METHODS FOR PROVIDING RURAL TELEMEDICINE WITH QUALITY VIDEO TRANSMISSION

A thesis submitted in fulfillment of the requirements for the degree of

Doctor of Technology (D. Tech): Electrical Engineering

in the Faculty of Engineering

of the Cape Peninsula University of Technology

by

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Supervisor: Professor MTE Kahn

February 2007

Declaration

I, Phumzile Malindi, hereby declare that the work

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on which this thesis is based is my original work and all the sources I have used or quoted have been indicated and acknowledged by means of complete references. Neither the whole work nor part of it has been, is being, or is to be submitted for another degree in any other institution.

Phumzile Malindi September 2006

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Key Words

Telemedicine, Rural Telemedicine, Quality Video, Multiservice Networks, Rural Area Networks (RANETs), Video communication, Quality of Service (QoS), IP-based Networking, H.264, DiffServ, MPLS, Traffic Engineering.

Abstract

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Telemedicine has been identified as a tool to distribute medical expertise to medically underserved rural community. However, due to the underdeveloped or non-existent telecommunication infrastructure, which is needed as the platform for telemedicine, the full benefits of telemedicine are yet to be realized in most parts of South Africa and Africa as a whole.

This study aims to explore ways on how to provide IP-based ICT system that can be used as a communication platform for telemedicine in rural areas. In order to emulate the on-site face-to-face consultation experience, the rural telemedicine system must be able to provide quality video transmission. Quality video is also important in order for the physician at the distant end to be able to make correct diagnosis. Hence the main focus of this study is on ways of providing quality video over IP-based multiservice network.

A conceptual model of a rural area network that can be used for rural telemedicine has been developed, and different access technologies that can be used for rural areas are presented. Techniques for compesating IP best effort datagram delivery are provided. Factors that can affect the quality of video transmission on an IP-based packet network are identified, and a holistic approach to mitigate them is proposed. That includes adopting coding techniques that will provide coding efficiency, high quality video that is

consistent at high and low bit rates, resilience to transmission errors, scalability, and network friendliness, which will result in perceived quality improvement, high-compression efficiency, and possibility of transportation over different networks. Secondly, it also includes mechanisms to compensate for packet networks idiosyncrasy, especially IP best-effort debilities, in order to meet the latency and jitter requirements of real-time video traffic.

For video coding, H.264 is proposed as it meets most of the encoding requirements listed above, and for prioritising and protecting video traffic from IP network's best-effort debilities a combination of differential services (DiffServ) and multi-protocol label switching (MPLS) have been adopted, where DiffServ is used for traffic classification and MPLS is used for traffic engineering and fast-rerouting in the event of route failure. To verify and validate the proposed solutions, modelling and simulation has been used, where the Network Simulator (NS-2.93) has been used to simulate network functions, and PSNR, VQM score and double stimulus impairment scale (DSIS) have been used for evaluating video quality.

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GLOSSARY

Nomenclature

ATM	Asynchronous Transfer Mode. A dedicated-connection switching technology that organises digital data into 53-byte cell units and transmits them over a physical medium using digital signal technology.
Busrtiness	Having a more or less variable bit within a stream
Communication	Sharing or exchange of information.
Communication network	A set of nodes which are interconnected to permit the exchange of information.
Convergence	In information technology, convergence is a term for the combining of personal computers, telecommunication, and television into a user experience that is accessible to everyone.
Converged networking	A type of networking where voice, video and multimedia traffic are converted to packets allowing them to be integrated with data traffic on a common shared IP transport architecture.
CSIR	Council for Scientific and Industrial Research. The body that was formed to foster industrial and scientific development so as to contribute to the improvement of the quality of life of the people of South Africa.
Group of Blocks	One or more rows of macroblocks in ITU-T video coding standards (H.261 and H.263)
ICT	Information and communications technology (or technologies). An umbrella term that includes any communication device or application, encompassing: radio, television, cellular phones, computer and network hardware

	and software, satellite systems.
IEEE802.11b	Institute of Electrical and Electronics Engineers standard for wireless LAN with data rates up to 11 Mbps.
IP	Internet Protocol. TCP/IP's primary network protocol, which provides addressing and routing information.
ISDN	Integrated Services Digital Network. A communication network that can transmit voice and data.
ISP	Internet service provider. A company that provides its clients with connectivity to Internet and other related services such as Web site building and virtual hosting.
ITU	International Telecommunications Union (formerly known as the CCITT). An international organisation within the United Nations System where governments and the private sector co-ordinate global telecom networks and services.
Jitter	The variation in the time between packets arriving, which is caused by network congestion, timing drift, or route changes.
LAN	Local Area Network. A collection of computers and other networked devices that fits within a scope of a single physical network.
Latency	The expression of how much time it takes for a packet of data to get from one designated point to another.
PABX	Private automatic branch exchange. An automatic telephone switching system within a private enterprise.
POTS	Plain old telephone service. A voice-oriented communication network, which is designed to offer telephone services.
PSTN	Public switched telephone network. The world's collection of interconnected voice-oriented public telephone networks, both commercial and government-owned. It is also referred to as the Plain Old Telephone Service (POTS).
RANET	Rural Area Network. A multi-service packet-based

	communication network, which can be used to provide an array of services to rural areas.
Remote area	An area that is situated far away from the main city.
Rural area	An area that is outside towns and cities, which is underdeveloped, underserved, and sometimes sparsely populated.
Simplex	A term used for transmission of signals in one direction only.
Telecommunication	Communication over a distance.
Telemedicine	The use of telecommunications and computer technologies with medical expertise to facilitate remote health care delivery.
Traffic engineering	Network function that controls the network response to traffic demands and other stimuli, such as network failures.
VQM	Video Quality Metric, which is an overall measure of video impairment
Wired	A term indicating that a network connection depends on access to a [copper or fiber] cable to carry the data transmissions from one networked device to another.
Wireless	A term indicating that a network connection depends on transmission at some kind of electromagnetic frequency through the atmosphere to carry the data transmissions from one networked device to another.

Abbreviations

ABR	Available Bit Rate
AF	Assured Forwarding
ATM	Asynchronous Transfer Mode
BA	Behaviour Aggregate
Bc	committed Burst

Be	excess Burst
BE	Best Effort
BGP	Border Gateway Protocol
CAC	Connection Admission Control
CAR	Committed Access Rate
CBR	Constant Bit Rate
CBQ	Class Based Queueing
CBS	Committed Burst Size
CBWFQ	Class Based Weighted Fair Queuing
CIF	Common Intermediate Format
CIR	Committed Information Rate
CME	Continuing medical education
CoPS	Common Open Policy Server
CoS	Classification on Flows
DiffServ	Differentiated Services
DS	Differentiated Services
DSCP	Differentiated Services Code Point
DSIS	Double Stimulus Impairment Scale
DTR	data terminal ready
EBS	Excessive Burst Size
EF	Expedited Forwarding
EGP	Exterior Gateway Protocol
EIGRP	Interior Gateway Routing Protocol
ERP	Enterprise Resource Planning
FIFO	First-In-First-Out
FRF	Frame-Relay Forum
FRTS	Frame Relay Traffic Shaping
FRTS	Frame-Relay Traffic Shaping

FTP	File Transfer Protocol
GOB	Group of Bits
GTS	Generic Traffic Shaping
HTTP	HyperText Transfer Protocol
IEFT	Internet Engineering Task Force
IGP	Interior Gateway Protocol
IntServ	Integrated Services
IP	Internet Protocol
IPDV	IP Packet Delay Variation
IS-IS	Intermediate System-to-Intermediate System
ITU-T	International Union for Telecommunications, Telecommunications
LAN	Local Area Network
LDP	Label Distribution Protocol
LLQ	Low Latency Queuing
MBS	Maximum Burst Size
MCR	Minimum Cell Rate
MF	Multi-Field
MIB	Management Information Base
MPEG	Motion Picture Experts Group
MPLS	Multi Protocol Label Switching
MQC	Modular QoS CLI
NS-2 (or NS2)	Network Simulator version 2
OSPF	Open Shortest Path First.
OWD	One Way Delay
PBS	Peak Burst Size
PCR	Peak Cell Rate
PDB	Per Domain Behavior

PHB	Per-Hop Behavior
PIR	Peak Information Rate
PSNR	Peak Signal-To-Noise Ratio
QCIF	Quarter Common Intermediate Format
QoS	Quality of Service
RAS	Rate Adaptive Shaper
RED	Random Early Detection
RFCs	Request for Comments
RMON	Remote Monitoring
RIO	Random Early Detection (RED) with IN/OUT
RIO-C	RIO Coupled
RSVP	Resource Reservation Protocol
SCFQ	Self-Clocked Fair Queueing
SFQ	Start-time Fair Queueing
SIF	Standard Image Format (also known as Standard Interchan[illegible] Input Format)
SLA	Service Level Agreement
SLS	Service-Level Specification
SMTP	Simple Mail Transfer Protocol
srTCM	Single-rate Three-Color Marker
SVC	Switched Virtual Circuit
TB	Token Bucket
TCA	Traffic Conditioning Agreement
TCP	Transfer Control Protocol
TCS	Traffic Conditioning Specification
TE	Traffic engineering
ToS	Type of Service
trTCM	Two-rate Three-Color Marker

UDP	User Datagram Protocol
VBR	Variable bit rate
VBR-rt	Variable Bit Rate, Real-Time
VoIP	Voice over IP
VQM	Video quality metric
WFQ	Weighted Fair Queueing
WRED	Weighted Randomly Early Detected

CHAPTER 1

1. NATURE AND THE SCOPE OF THE STUDY

This introductory chapter creates the context for this work and designs the research design and literature study.

1.1 BACKGROUND TO THE RESEARCH PROBLEM

About 70 to 80 percent of the population in Africa lives in remote areas (McClelland and Berendt, 1998). Most of these remote areas are rural and are having medical care service that is very poor. This is primary due to the fact that most physicians are reluctant to work in rural areas. So people in rural areas have to travel long distances in order to get medical care services.

The advent of telemedicine, which is the use of telecommunication to provide medical information and services (Perednia and Allen, 1995), more than fifty years ago has brought some hope to most of the medically under-served areas throughout the world. For Africa, which is regarded as the 'Dark Continent', the benefits of telemedicine are still yet to be realized. This is manly due to the lack of telecommunication infrastructure that is required as the platform for telemedicine in rural areas.

Providing rural telemedicine system with improved video quality can improve healthcare access and information service while reducing the isolation of healthcare providers and residents in rural areas. Rural telemedicine can also reduce the time and allay the costs of rural patient transportation significantly. Rural patients in health clinics who need special medical care can be able to have face-to-face consultation with a specialists that are situated in a hospital 50 kilometers away, using telecommunication system which allows interactive communication, including real-time full motion picture.

1.2 RESEARCH TOPIC

The title of this study is:

"METHODS FOR PROVIDING RURAL TELEMEDICINE WITH QUALITY VIDEO TRANSMISSION"

As it has already been mentioned in the background section, telemedicine needs telecommunication infrastructure to be in place before it can be implemented. This telecommunication infrastructure needs to be able to provide quality video transmission so that correct diagnosis can be made and the on-site face-to-face consultation experience can be emulated.

However, the advanced telecommunication infrastructure is usually unavailable or very expensive in rural areas. According to the Federal Communications Commission (FCC) telecommunication and health care advisory committee "... in most cases the telecommunication bandwidth that is available to urban health care providers and business is not available in rural areas." (FCC, 1997).

South Africa has a telecommunication infrastructure that resembles the mixture of first world and the third world. While in the big cities we are talking about scientific marvel of advanced modern telecommunication technologies with capacities that can support an array of telecommunication services, a number of rural communities lack rudimentary telecommunication services. Owing to historical reasons there is an imbalance in quality of telecommunication infrastructure between urban areas and rural areas. It is therefore not surprising to find that in some rural areas the infrastructure is nonexistent. This uneven distribution of modern telecommunications infrastructure has resulted in rural and remote areas having to settle for underdeveloped, unreliable, or non-existent telecommunication services.

Though in other parts of the world telemedicine has proved to be a cost effective solution for providing health care to medically under-served areas, the poor quality and lack of telecommunication infrastructure remains one of the major obstacles for introduction of telemedicine in most parts of South Africa and Africa as a whole. Even though in other places telemedicine is being used, the services offered are limited due to the lack of sufficient bandwidth, especially for quality video transmission.

From the preceding text it can be said that the main barriers to providing rural telemedicine are to a greatest extent technological, hence technological solutions are also required.

The research statement is: Providing rural telemedicine with quality video transmission.

Hypothesis:
IP-based multiservice rural area networks can provide the platform needed for rural telemedicine.

1.3 RESEARCH OBJECTIVES

The primary objective of this study is to investigate the methods for providing rural telemedicine with improved quality video transmission that can be used for most applications in telemedicine between the area hospital and referring clinics, in remote and rural areas, where the telecommunication infrastructure is either underdeveloped or nonexistent. In the process of realizing the primary objective, several secondary objectives are also identified. These are as follows:

- Development of a conceptual model of a heterogeneous rural area network (RANET) that will support a combination of audio, video, and data in both real-time and store-and-forward applications, which is required for rural telemedicine.

- Investigation of different methods that can be employed to ensure the improvement of quality of video transmission within the bandwidth confines of the rural telemedicine system.
- Development of algorithms and programs to model and simulate the network functions so as to validate the proposed system.

1.4 OUTLINE OF THE LITERATURE STUDY

In its report, the ITU-D Focus Group 7 have identified the information and communication technologies (ICTs) as a necessity to support rural economic development, health, distance education, emergency support, and disaster relief. This was followed by the design of specialized rural applications of ICTs, and the deployment of ICTs in rural areas in sectors such as business, education, agriculture, security, and health (ITU-D, 2001).

The use of ICT to deliver healthcare services over a distance is referred to as telemedicine or telehealth (Androuchko, 2003). Telemedicine is used to distribute or export the medical services and skills to medically underserved areas such as remote and rural areas so that there can be equitable access to quality healthcare services irrespective of the location. The services offered by telemedicine include the transfer of basic patient information, transfer of images such as radiographs, computer tomography (CT) scans, magnetic resonance imaging (MRI) pictures, ultrasound, pathology images, video images of endoscopes or other procedures, patient interviews and examinations, consultations with medical specialists and health care education services (Rao, 2001). The main driver of most telemedicine projects is the reduction of professional isolation among doctors and other healthcare staff located in remote and rural areas (Androuchko, 2003; Rao, 2001).

Acording to Rao (2001) the telemedicine must be able to provide high quality audio and video services so that internal and external examinations can be carried out and correct diagnosis can be made. In order to meet these quality requirements of audio and video telemedicine applications most developed countries have adopted ISDN and ATM as communication platforms for telemedicine. However, due to the fact that both ISDN and ATM services are expensive and are not available to the rural community, especially in the developing countries such as South Africa, ISDN based systems are commonly used in urban areas and most of the rural systems are IP-based.

Though IP has emerged as a de facto standard for data communication and new telemedicine systems, its default and traditional datatagram best effort delivery services make it not suitable for real time interactive traffic such as voice and video applications, which are sensitve to delays, delay variations and losses (Riley and Richardson, 1997, pp.91-99; Tobagi, 2005). So if IP-based networks are to be able to offer quality of service, IP's best effort delivery services need to be compensated with quality of service mechanisms.

So, in simplistic terms, the study aims to answer the question: How to provide a rural telemedicine with quality video transmission in rural areas where the telecommunication infrastructure is either non-existent or underdeveloped using IP-based networks?

The literature study follows a logical approach.
The starting point will be to investigate the requirements of a rural telemedicine system that incorporates video transmission.
The strategic review begins with investigating about the present rural telemedicine in South Africa and in the other parts of the world.
And progress through to the development of a model of a rural telemedicine and of an IP-based multiservice rural area network to support it.
The study continues by describing the different access technologies that can be implemented in rural environment for these rural area networks, which are used as platforms for telemedicine.
Mechanisms of providing quality of service using IP are explored in order to find the suitable ones.
Factors that affect the quality of video that is transmitted over packet networks will be identified.

Methods of mitigating some of the problems that degrade the quality of video when transmitted over packet networks will be discussed. These will include coding techniques for compression efficiency and error resilience, and also the prioritization and protection of video traffic.

The literature study will enable the analysis and the development of a video traffic model that can be used for simulation of video traffic.

The literature study concludes with the review of techniques used for video quality measurements.

1.5 THE SCOPE, ASSUMPTIONS AND LIMITATION OF THE STUDY

During the preliminary study it has become evident that the utilization of the convergent information and communication technologies (ICT) can be a solution to rural telemedicine. The ICT system with improved quality video transmission can help to provide the telecommunication platform that is required for telemedicine for medically under-served rural communities where the telecommunication infrastructure is either underdeveloped or nonexistent.

For this study it has been assumed that the clinic-to-hospital ratio in remote and rural areas is five-to-one and all the referring clinics are distributed evenly around the local hospitals, and are within a radius of 50 kilometers from the area hospital.

Though part of this study is to provide rural telemedicine, the emphasis of this study will be on improving the quality of video transmission within the rural telemedicine system that is based on IP network.

1.6 RESEARCH METHODOLOGY

Since the object of this study does not belong to empiria but is within the information segment of the computer science and engineering discipline, the research that is most relevant is of an engineering rather than a scientific orientation, and is concerned with technology, including artefacts, techniques and combination of both of them (Clarke, 2000).

Non-empirical and engineering research techniques are going to be used throughout this research. These will include review of exiting literature (or 'meta-analysis'), simulation, construction, and destruction. These different research methods will be used during the different phases of the research on the project. The phases of the research will include:

- The study of different network architectures in order to develop a conceptual model of a rural telemedicine.

- Study of different access technologies in order to determine the one with a potential to support an array of services for healthcare in rural areas.

- Having developed the conceptual model and identified the potential access network technology, different methods for ensuring quality of service over IP networks are going to be investigated.

- After investigating mechanisms for making IP suitable for interactive traffic, the focus of the study will shift to quality video transmission using IP. This part of the study will look at the challenges and possible solutions to quality video transmission over IP.

- Lastly, Modeling and Simulation is going to be done using network simulators, Matlab and other software packages to verify and validate the proposed system.

The literature study will provide background and insights to the research statement and permit an in-depth analysis. The literature study will involve:

- A study of the relevant journals, books, electronic publication, websites, and all other forms of published material;
- The examination of papers presented at conferences;
- Personal communications with local and international experts in the field of telemedicine and video.
- The use of keyword searches in the full text academic research databases, of Emerald, EBSCO Host and ScienceDirect; and
- The application of web searches such as Yahoo and Google.

The core field of study will include: telemedicine, rural telemedicine, rural communications, IP-based multiservice networks, video communication, IP video, quality video, quality of service, network simulators, perceived video quality measurement.

Further fields of study may be revealed by cross-references or citations. These will be followed up and studied for further relevance.

1.7 LAYOUT OF THE DISSERTATION

This thesis discusses methods of providing rural telemedicine with quality video transmission. In order to achieve the objective of this study the following chapters and content of this dissertation are structured as follows:

Chapter Two: Concepts.
Definitions and concepts used in this study are explained.

Chapter Three: Literature review
This chapter is about reviewing ways of providing

Chapter Four: Quality of Service in IP-based Multiservice Network
This chapter is about mechanisms that can be used to provide quality of service using IP-based multiservice rural area networks.

Chapter Five: Digital video and quality video transmission

This chapter introduces digital video, its transmission using packet network, how it is affected by the network, and solutions to mitigate some of the effects of the transmission system in order to render quality to the end user.

Chapter Six: Modeling and Simulation

This chapter is about performance evaluation in order to validating the proposed system. It is where the modeling and simulation of the network functions are going to be performed. This will include mathematical models and simulation.

Chapter Seven: Conclusion

This chapter will conclude this thesis, by showing that the goals of this research have been met, and also points to possible further research.

1.8 SUMMARY

This chapter introduces the background and context of the study. It covered the following:

- The discussion of the primary research objective
- Secondary objectives
- Research Methodology
- The outline of the dissertation

The chapter concluded with the introduction of the next chapter, which is about the definition and the explanation of the jargon, technical terminology that is used in this document so as to prevent misunderstanding.

CHAPTER 2

2. CONCEPTS

2.1 INTRODUCTION

In order to prevent the misunderstanding, a definitive understanding of terms used by the researcher and quoted in the context of this document is required. This chapter presents the definitions and explanation of concepts used in order to help differentiate between jargon and scientific meaning. However, not all the terms are covered here some are included under glossary in pages xvi through xxii.

2.2 TELEMEDICINE

Telemedicine has been defined as the use of telecommunication to provide medical information and services (Perednia and Allen, 1995). According to Tom Nesbitt, medical director for the UC- telemedicine program (USA), telemedicine can be seen as the way to distribute the medical expertise out to remote and rural health professionals who need consults to help them manage their patients (Anderson, 2000). According to Dean (2003, p.16), telemedicine brings patients and healthcare professionals together by exchanging voice, video, and data over distances when they cannot meet face-to-face.

Rapid development in computer technology and easiness to purchase has led to more amenability to computer-based telemedicine technology and the growing use of telemedicine. The use of ICT gave a new meaning to telemedicine and the way it is defined. Today definition of telemedicine is "the use of electronic information and communication technologies to provide and support health care when distance separates the participants" (Institute of Medicine (USA), 1996; Kienzle, Moses and McGowan, 2000; McGowan and Kienzle, 2000).

2.3 TELEHEALTH

Telehealth, often used interchangeable with telemedicine, refers to the use of information and communication technology (ICT) to deliver health services, expertise and information over distance. It includes internet or web-based e-health and video-based applications, and can be delivered real-time or through store-and-forward mode. Telehealth is unique in having the capability to cross-geographical, temporal, political, social and cultural barriers within the health sector.

2.4 PERIPHERAL DEVICES

Attachments to telemedicine systems that augment their communications or medical capabilities. Examples include: electronic stethoscopes, blood pressure monitors, oxygen saturation monitors, oto-/ophthalmoscope, dermascopes, graphic stands, cameras, scanners, etc.

2.5 REAL-TIME AND NEAR REAL-TIME

Real-time sends and receives data, video, or audio simultaneously without more than a fraction of a second delay. Near real-time refers to applications that are transmitted within few seconds.

2.6 STORE-AND-FORWARD

Captured audio clips, video clips, still images, or data that are transmitted or received at a later time. Sore-and-forward enables dissynchronous communication, with an advantage of not needing concurrent participant involved. For example, e-mail is a store-and forward system.

2.3 RURAL AREA

According to ITU-D Focus Group 7 (2001) the rural area is an area, usually remote, which exhibits one or more of the following characteristics:

- Scarcity or absence of public facilities such as reliable electricity supply, water, access roads and regular transport;
- Difficult topographical conditions, e.g. lakes, rivers, hills, mountains or deserts;
- Low level of economic activity mainly based on agriculture, fishing, handcrafts, etc.
- Low per capita income;
- Underdeveloped social infrastructures such as health, education, etc.;
- Underdeveloped telecommunications infrastructure
- Low population density
- Severe climatic conditions

2.4 RURAL AREA NETWORK

Rural area network (RANET) is a multi-service packet-based communication network, which can be used to provide an array of services to rural areas. Unlike other communication networks, RANET is designed for sparsely populated areas where nodes are geographically distributed over a radius of about 50 kilometers. The RANET concept and system architecture is almost similar to that of next generation networks (NGN).

2.5 QUALITY

From Wikipedia (the free encyclopedia) the definition of quality depends on the context in which it is used. For example, technically, quality refers to a specific characteristic of an object; practically, quality refers to the achievement or excellence of an object; artistically, quality refers to the essence of an object, whereas metaphysically, quality refers to excellence.

Philosophy tends to see quality as related either to subjective feelings or to objective facts. Hence when it comes to its assessment both subjective and objective techniques are used.

2.6 QUALITY OF SERVICE

Quality of service (QoS) refers to the ability of a network element to have some level of assurance that its traffic and service requirements can be satisfied. Enabling QoS requires the cooperation of all network layers, as well as every network element from end to end, for example, an application, host, and router must be able to meet the required QoS (Ma and Shi, 2000). According to ITU-T recommendation E.800, QoS can be defined as the collective effect of service performances, which determine the degree of satisfaction of a user of the service (Afullo, 2004).

2.7 VIDEO QUALITY AND QUALITY VIDEO

Video quality is a characteristic of video passed through a video processing system. It refers to the quality of video after being passed through the system, which can either be an encoder alone or encoder and transmission system.

2.8 CONVERGED NETWORK

A Converged network is an IP–based network that leverages a common infrastructure for voice, data, and video applications. It can also be defined as an IP network that has been enhanced with reliability and quality of service features to make it to meet the requirements of real-time, mission-critical business and communication applications.

2.9 MULTISERVICE NETWORK

A multiservice network is a packet network that can accommodate various applications such as data, streaming video, voice over IP (VoIP), interactive video, and others.

2.10 IN SUMMARY

This chapter introduced concepts referred to later in the dissertation. Some key definitions were discussed. The chapter concluded with the introduction of the next chapter, which is about literature review.

CHAPTER 3

3. LITERATURE REVIEW

3.1 INTRODUCTION

In this chapter literature study is done in order to conceptualize rural telemedicine, and rural area networks. A conceptual model of a rural area network that can be used as a communication platform for rural telemedicine is developed, and different access technologies that can be used for rural telemedicine are also reviewed in order to find one that is suitable for rural areas, which are characterized by mountainous terrain, rivers and trees.

3.2 TELEMEDICINE

The concept of telemedicine has been around for more than 40 years now. In 1959 the University of Nebraska School of Medicine began experimenting with a closed-circuit television link to provide psychiatric and other health services between the Nebraska Psychiatric Institute and Norfolk State Hospital (Benshoter, 1967; Preston, 1993). From then until the mid-1970s a number of rural and urban projects were developed that relied on satellites or dedicated video connections to supply services (Maxmen, 1978). However, in the late 1970s, the high cost of these systems, as well as problems with getting providers to integrate the technology into their practice patterns, forced almost all of these systems out of business (Bashur, 1997).

Telemedicine was revived in the 1980s, by the United States of America's military and correctional institutions (Brecht, 1997; Edwards, 1997). Many of these systems were designed to meet specific needs in environments where transporting patients to specialty providers would be both difficult and costly. Because of these challenges and the institutional missions of these organizations, even very expensive systems were relatively easy to justify (Kirby, Hardesty and Nickelson, 1998).

The 1990s witnessed the reemergence of telemedicine and its use world wide (Kirby, Hardesty and Nickelson, 1998; McGowan and Kienzle, 2000). This reemergence can be attributed to the developments in information and communication technology (ICT). Rapid development in computer technology and easiness to purchase has led to more amenability to computer-based telemedicine technology and the growing use of telemedicine. The use of ICT gave a new meaning to telemedicine and the way it is defined. Today definition of telemedicine is "the use of electronic information and communication technologies to provide and support health care when distance separates the participants" (Institute of Medicine (USA), 1996; Kienzle, Moses and McGowan, 2000; McGowan and Kienzle, 2000).

Early telemedicine systems, with the exception of University of Nebraska School of Medicine's case, involve mainly a simple telephone call from one health professional to another for advice and consultation. However, today telemedicine may be as simple as two health professionals discussing a case over the telephone, or as scphisticated as using a satellite technology to broadcast a consultation between providers at facilities in two countries using videoconferencing equipment (Brown, 2000). Whether simple or sophisticated, telemedicine needs telecommunication infrastructure to be in place to facilitate the linkage between the centers where the two health professionals are located. There are many telecommunication options available for telemedicine which include satellite, wireless, plain old telephone service (POTS), integrated service digital networks (ISDN), and asynchronous transfer mode (ATM), the choice depends on the program's current and projected needs, finances and developments in technology.

There are two modes used in most of the today's telemedicine applications. The first one is called store-and-forward or asynchronous mode and is used for non-emergent situations, where the diagnosis or consultation may be made within the next 24 – 48 hours. The application of store-and-forward includes teleradiology (the sending of x-rays, CT scans, or MRI), telepathology and dermatology.

The second mode is the interactive (real time) consultation or synchronous mode, which may involve two-way telephone conversation or two-way interactive videoconferencing that provides face-to-face consultation.

A key feature of telemedicine, which distinguish it from an ordinary telecommunication system, is the use of peripheral devices that include electronic versions of standard examination tools as well as other sense extending implements like cameras, document stands, dermascopes, and microscopes. These peripheral devices enable the clinician to better approximate an on-site physical examinations.

3.2.1 Rationale for Telemedicine

The reasoning behind telemedicine can be summarized as follows:

- Providing access to healthcare
- Enhancing the efficiency of the way the health is provided
- Bringing health care closer to people thereby shortening the time to treatment
- Enhancing professional collaboration between health providers and
- Cost savings

3.2.2 Telemedicine Applications

Telemedicine has already been defined as the use of information and communication technology to deliver medical services and information from one location to another. This includes applications in areas such as pathology and radiology, as well as consultations in specialties such as neurology, dermatology, cardiology, and general medicine. Telemedicine is also used for Continued Medical Education (CME), administration, research and development. Table 3.1 shows a summary of telemedicine application categories.

Table 3.1 A Summary of Telemedicine Application Categories

• Initial urgent evaluation of patients, triage decisions, and pre-transfer arrangements.
• Medical and surgical follow-up and medication checks.
• Supervision and consultation for primary care encounters in sites where a physician is not available.
• Routine consultation and second opinions based on history, physical exam findings, and available test data.
• Transmission of diagnostic images.
• Extended diagnostic work-ups or short-term management of self-limited conditions.
• Management of chronic diseases and conditions requiring a specialist not available locally.
• Transmission of medical data.
• Public health, preventive medicine, and patient education.

SOURCE: Grigsby et al., "Analysis of Expansion of Access to Care Through Telemedicine, Report 4, Study Summary and Recommendations for Further Research," Center for Health Policy Research, Denver, CO, December 1994, p.43

3.2.3 Rural Telemedicine

Most of the remote areas are rural and sparsely populated. The rural areas, which most of the time are underdeveloped, under-served and poor, constitute the greater part of South Africa and Africa as a whole. Most of the health professionals or healthcare providers in rural areas feel isolated from mentors, colleagues, and the information resources necessary to support them personally and professionally. Equipment may be less up to date and facilities less than adequate. These conditions have made it difficult attracting and retaining health professionals in rural areas resulting in geographic and socio-economic isolation that have disenfranchised millions of people from the health care services they require.

Telemedicine is a tool that may help address the problem of [healthcare] provider distribution by improving communication capabilities and providing convenient access to up-to-date information, consultations, and other forms of support (McCarthy, 1995).
Telemedicine is a solution for improving primary health care in rural areas. Dealing with patients at primary health care level relieves much of the strain on the infrastructure of secondary and tertiary institutions (MAS technology, 1998).
Telemedicine can be used to electronically transport the highest levels of medical care into most remote and rural, thus enabling rural people to access high quality medical expertise without leaving their communities.

3.2.4 Models for rural telemedicine

There are two possible models that can be used to provide rural telemedicine: local and open systems. These models are shown in Figure 3.1 and Figure 3.2, respectively.

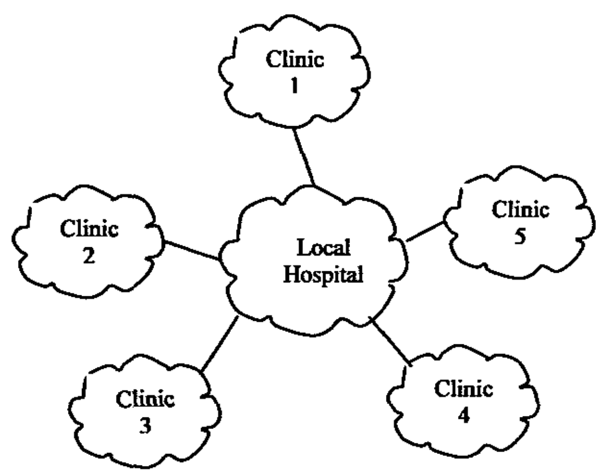


Figure 3.1 Local Rural Telemedicine System

The local system, which is a network that is constituted by the local hospital and referring clinics surrounding it and there is no direct link with other secondary and tertiary medical institutions. This model, though simple, tends to confine the clinicians to the limited expertise in their area. A system that is based on this model was launched beginning of

December 2001 in a deep-rural mountainous Qumbu district of the Eastern Cape. This project, which is a joint venture between CSIR and Tsilitwa community, uses ICT to link the Tsilitwa clinic to Sulenkama community hospital, which is about 20 kilometers away from the clinic. It is an autonomous system that is based on IEEE 802.11b wireless LAN, and it provides wireless connection between the clinic, school, police station using an IP-based PABX and a simplex wireless video link between the clinic and the hospital. The telephone and the video links between the clinic and the hospital enables the physician at the hospital to see what is happening to the patient at the clinic so that he can advise the nurse, diagnose and formulate a treatment plan for the patient (CSIR, 2001).

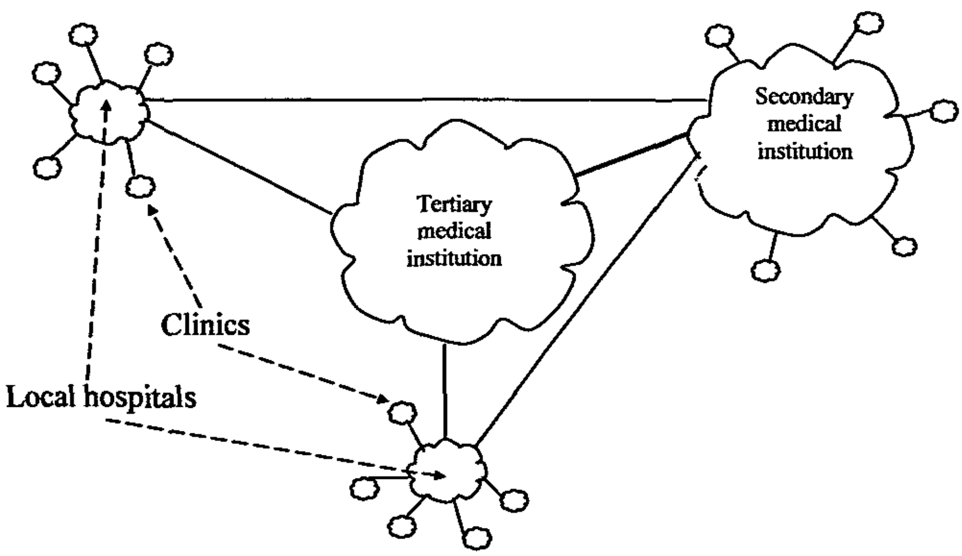


Figure 3.2 Open Rural Telemedicine System

The open system, on the other side, has direct links with other secondary and tertiary medical institutions to enable direct consultation between clinician and physician or specialist in any other medical institution, be it locally, or in the big city. The advantage of this model is that the rural health professional has a broader spectrum of expertise to his/her disposal to enable consults with physician or specialist in any other site.

There is presently a project that is based on open system model in South Africa and is called Healthlink. It is a project of the health system trust funded by the Henry J. Kaiser family foundation (USA). The main aim of the project was to develop a low-cost, off-line electronic communication network for use in South African health services systems. The project was piloted in three provinces: Eastern Cape, Free State and Northern provinces in 1994, but it has been extended to include five more provinces to make a total of eight provinces, out of nine, in South Africa. This network consists of 430 points connected to 10 Healthlink nodes operating in Pietermaritzburg, East London, Queenstown, Port Elizabeth, Umtata, Pietersburg, Bloemfontein, Mmabatho, Upington and Medical University of South Africa (MEDUNSA) in Pretoria. These nodes are connected to a central node in Durban, which has an access to internet via commercial ISP. The community hospitals and some few clinics are connected to one of the nodes. For example, fifty-three sites are now connected through the node in Umtata general hospital. The system enables the medical student when on training in rural hospitals to communicate with supervisors at University of Transkei (Banach, 1998).

3.3 RURAL AREA NETWORKS FOR RURAL TELEMEDICINE

The essence of telecommunication is breaking down the barriers like distance and time so that people can communicate even though they are not together. If the communication involves provision of healthcare at a distance, then it is no longer just a simple communication but it is referred to telemedicine.

So telemedicine can be defined as medicine practiced when the patient is at a site that is at some distance from the doctor or medical expert. The practice of telemedicine makes use of telecommunications and information technology (ICT), including digital video communications, to provide this healthcare at a distance (Riley and Richardson, 1997, p.9).

3.3.1 The Evolution of Telemedicine Systems

Except for the University of Nebraska case most of the early telemedicine systems were predominantly based on the use of telephone and fax machine as a telecommunications platform (Pattichis, Kyriacou, Voskarides, Istepanian and Schizas, 2002), where it was a physician at the hospital on one end and a nurse or any other health professional at a clinic on the other side, or physicians on both ends consulting each other during patient diagnosis.

The drawback of these first telemedicine systems was their inability to enable the distant physician to see the patient being examined. This was mainly due to their narrow bandwidth, which could not support the transmission of real-time video traffic, which is needed to provide the physician at the distant hospital with the ability to carry out remote examination of the patient with the aid of the clinician.

The rapid advances in telecommunication led to the development of integrated service digital network (ISDN) that can provide an array of services including video traffic to enable videoconferencing that can be used to provide face-to-face consultation between a patient at the clinic and a physician in hospital several kilometers away. However, the lack of ISDN connections and the high cost of services offered by ISDN technology have made the services of ISDN to be inaccessible to the most rural communities in developing countries, thus making videoconferencing to be out of reach to rural communities to enable face-to-face consultation using telemedicine.

As it has already been highlighted in chapter one that most of the rural areas have either underdeveloped or nonexistent telecommunication infrastructure, those which are lucky enough to have communication networks can only use them for voice applications, since data transfer over the telephone line is confined by the lack of bandwidth, flexibility, and quality of the transmission equipment. Using these pre-existing networks for new applications may lead to characteristic shortcomings, because these networks were not tailored to the needs of services that are provided by telemedicine.

3.3.2 Communication Networks for Rural Areas

Communication networks can be broadly defined as an arrangement of hardware and software that allow users to exchange information (Walrand, 1998, p.2). The information exchanged may be voice, sounds, graphics, pictures, video, text, or data, among users. Traditionally, there are two most common types of communication networks: the telephone network for voice transmission, and a computer network for data transmission between the computers and/or peripherals. However, the advancement in communication and information technologies has resulted in new communication networks merging the capabilities of both telephone networks and of computer networks into one network, thus enabling the simultaneous exchange of both voice and data over the same network. This has also paved way for development of high-speed networks that are able to transmit high quality, full-motion video.

A communication network consists of a set of nodes that are interconnected to permit the exchange of information. These nodes can be classified as either terminal nodes, or as communication nodes. The terminal node has terminal entities such as phone sets, computers, printers, file servers, or video monitors, whereas the communication node has communication devices such as switches, routers, or repeaters. The terminal nodes generate or use information transmitted over the network while the communication nodes transport the information but do not generate or use the information (Walrand, 1998, pp.2-3). The nodes can be few meters to many kilometers apart. To link the nodes a communication channel is used, which may be a transmission line, an optical fiber, a radio link, or a light wave in free space.

For rural areas the nodes can be a considerable distance apart; for example, a network that can enable the clinics and the local hospital to exchange information can have nodes that are 50 kilometers or more from one another. So the communication network that is designed for rural areas must be able to support nodes that are geographical distributed by distances of about 50 kilometers or more. Another challenge of rural communication networks is the terrain over which it has to be deployed. Most rural areas are characterized by mountainous terrain, forests and many rivers (ITU, 1996), which can make the use of other types of transmission media, such as transmission line and optical fiber, to be very difficult, if not impossible.

3.3.3 Rural Area Network (RANET)

The rural area network (RANET) is a multi-service packet-based communication network, which can be used to provide an array of services to rural areas. The system is autonomous and is tailored for providing the performance that is required by different applications that can be used.

3.3.4 Conceptual model

The RANET system architecture is designed to deliver integrated services to both wired and fixed-wireless networks. It is an open-system architecture for converged networking as shown in Figure 3.3, and it consists of the following four elements:

- Services and Applications network
- Core network
- Broadband access networks
- User terminals

The architecture assures that these functional elements are seamlessly interconnected and facilitates connecting additional networks, such as Internet, and traditional telephone service networks, to the core network. The architecture also enables, inter alia, the following set of services:

- *Packet services*: As a data transport, it must provide carriage for video, voice and data application with an appropriate quality of service for each. It must support the existing IP applications and new value-added services.

- *Real-time communication*: These are interactive real-time voice and video communication.
- *Multicasting services*: This allows for efficient delivery of a given information stream to multiple recipients at the same time, which is important for continued medical education (CME).
- *Network management*: These are supervisory functions, such as monitoring, and maintenance, to keep the network operating efficiently and effectively.
- *Unified messaging*: This involves the integration of voice, fax, and e-mail messages on a single IP-based message repository.

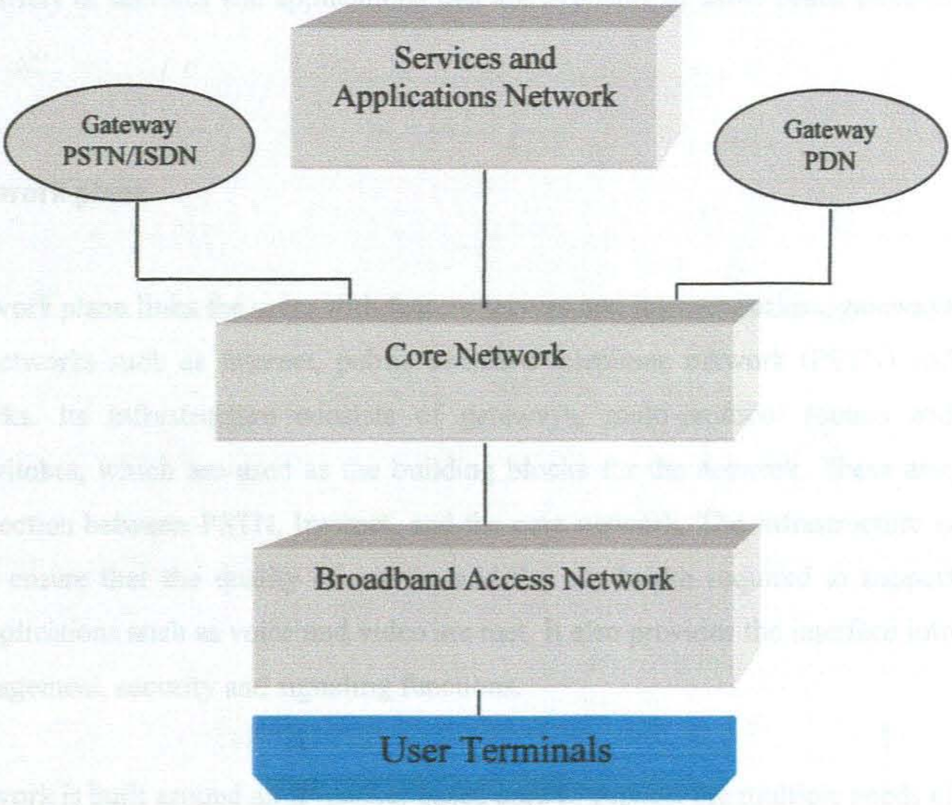


Figure 3.3 Conceptual RANET Model

Functional Components

The architecture defines the following four service planes that contain specific functional components:

Services and application plane

This is where all the capabilities of the network will reside. It consists of servers where all components that everyone needs to access the network reside. These end user servers will host a variety of services and applications that are available to users based on their profiles.

The core network plane

The core network plane links the users with feature servers and internetworking gateways to external networks such as Internet, public switched telephone network (PSTN) and other networks. Its infrastructure consists of gateways, multi-protocol routers and multilayer switches, which are used as the building blocks for the network. These also provide connection between PSTN, Internet, and the core network. The infrastructure is there to also ensure that the quality of service and the bandwidth required to support converged applications such as voice and video are met. It also provides the interface into network management, security and signaling functions.

The core network is built around an IP-packet based core to support the multiple needs of users, and to reduce the cost of network operation. It is designed to be independent of access technology. The Internet protocol (IP) provides data interoperability and emerging support for quality of service (QoS), and it works with other protocols to ensure reliable transmission of packets.

Broadband access plane

The access plane is the portion of the network that lies between the end user and the core network. Basically, it performs three functions: provision of interface to the core network, distribution of data, and provision of broadband access over cable (optical or transmission line), or air interface to the users. The broadband access technology may include the following:

(a) Copper line based systems, such as Digital subscriber line (xDSL) and EtherLoop, allow broadband signals to be carried over the already existing copper lines. They have data rates that are function of line length and wire gauge, that is, the longer the line the lower the rate and the thicker the wire the higher the rate.

(b) Cable modems may also be used to provide two-way broadband connections over coaxial cables.

(c) Fiber may also be installed in the place of copper to offer the bandwidth required for video communication.

(d) Hybrid fiber-coax can be used as an alternative for providing broadband connection since fiber connections can be costly. This can be done by using fiber for short distances like from the core network to a central location and then do the conversion from optical to electrical with delivery of high data-rate electrical signal via standard coaxial cable.

(e) Power Line Communication can be used for providing broadband connections over the existing power distribution cables.

(f) Fixed wireless provides another alternative way of broadband connection, where you have a transceiver at both ends to link the remote node to the core network. Most rural areas are characterized by mountainous terrain, forests and many rivers (ITU, 1996), which can make the use of other types of transmission media, such as transmission line and optical fiber, to be very difficult, if not impossible. Hence, wireless can be the most viable access technology for most RANETs.

User terminals plane

The user terminal plane is attached to the access plane and it will consist of telemedicine workstations for the users.

3.4 POSSIBLE NETWORKING MEDIA AND ACCESS TECHNOLOGIES FOR RURAL NETWORKING

In the previous section a conceptual model of a rural area network has been introduced. This section will discuss different technologies that can be used to connect different nodes of the network to enable the end users to access the services that are offered by the rural area network. These connections can be classified as network media and access technologies. Networking media is used to link the nodes within the network and the access technology is used to link the user terminals to the core (or backbone) network. These communication links or media can be wired, wireless, or hybrid of the two. The signals propagate through the media as electromagnetic waves, which may be guided by a transmission line, an optical fiber, or the waves may propagate in free space as radio waves or as optical waves

A wired link uses cable to carry the data transmissions from one networked device to another, whereas a wireless link depends on transmission at some kind of electromagnetic frequency through the atmosphere to carry the data transmissions from one networked device to another. Whether wired or wireless, each link is characterized by its bit rate, bit error rate, and distance.

3.4.1 Wired Links

Wired links need the communicating devices to be physically connected. There are two types of wired media: transmission lines and optical fiber.

3.4.1.1 Transmission lines

A transmission line is a metallic conductor (copper, galvanized steel, or copper-covered-steel) that can come in the form of open wire, twisted pair and coaxial cable.

Open wires

Open wires consist of two overhead conductors, which are not insulated. To prevent them from touching and shorting the conductors are placed about 20 cm apart.
They were the first, and for economical reasons in many developing countries, still are the main transmission medium in the access network and sometimes also used in distribution network, especially in rural areas. The relative modest investment for open wire line defers costs due to:

- The high vulnerability to damage by vandalism, theft, and natural calamities;
- Transmission quality, which is very much dependent upon whether conditions;
- The limited transmission capacity
- Significant crosstalk, radiation loss at high frequencies, and noise pick-up due to lack of shielding.

The above characteristics of the open wire make it impossible for it to be used for system with applications that need quality transmission and broad bandwidth.

Twisted pair (or cable)

Twisted pair (or cable) consists of a pair (or pairs) of insulated conductors that are twisted together. The information is transmitted as a difference voltage between the two conductors, thus resulting in a current that flows in opposite direction in each wire. Since these currents are equal and in close proximity, their magnetic fields cancel each other, and also cancel any magnetic interference caused by outside noise source. This makes the twisted pair to be self-shielding thus making it less prone to electromagnetic interference (EMI) and radio frequency interference (RFI) than open wires. The twists also minimize crosstalk between the pairs, and balance the mutual capacitance of the cable pair.

The twisted pair (or cable) can be divided into two groups: unshielded twisted pair (UTP) and shielded twisted pair (STP). The inclusion of the shielding in STP improves the cable's immunity to radiation loss at high frequencies, and noise-pickup. Unlike open wire that can be used external only, twisted pair (or cable) can be used both internal and external. When used external, the cable can be installed underground or overhead as an aerial cable. The conductors are relatively thin in diameter compared to open wire, which limits the transmission of unamplified transmission distance to be less than that of an open wire. In order to further extend the maximum distance a repeater is used. For example, in E1/T1 links (1.54/2.05 Mbps) regenerating repeaters, which are placed roughly every 1 to 3.5 km, are used for pairs of 1.2 mm diameter to cover a length up to 80 km (Huurdeman, 1997, p.144).

At higher frequencies or higher transmission rates, the power in the twisted pair or cable is lost due to skin effect and radiation loss. The radiated power has a potential to disturb the operation of the signal in the nearby pairs, resulting in crosstalk, which degrades the quality or halting the operation of the other pair altogether.

Digital subscriber line (xDSL) systems

To increase the capacity of each twisted pair, digital subscriber line (xDSL) systems can be used. This allows high-speed data, video, and multimedia to be transmitted on standard twisted copper pairs over the limited distance (Huurdeman, 1997, p.138). The digital subscriber line systems have data rates that are function of line length and wire gauge; that is, the longer the line the lower the rate and the thicker the wire the higher the rate. There are six popular forms of xDSL, namely:

(i) ***Asymmetric Digital Subscriber Line (ADSL)***, which operates at distances up to 6 km and provides downstream data rates of 1.544 to 8.448 Mbps and upstream data rates of 16 – 640 kbps over a single twisted pair of copper wires. This transmission of data in an asymmetric manner makes ADSL to be suitable for applications such as video-on-demand and video broadcast, which require high data rates for downstream and low data rates for upstream;

(ii) ***High-data-rate Digital Subscriber Line (HDSL)*** is a high-bit-rate digital subscriber line, which operates at distances up to 3.4 km and provides data rates of 2 Mbps in both directions over two twisted pairs of copper wires. It can be used to provide T1/E1 services without the need for repeater for up to 3.4 km. There is an advanced version of HDSL, which is called HDSL2. Unlike HDSL, which needs two twisted pairs of copper wire, HDSL2 uses one twisted pair of copper wire to offer the same T1/E1 service for distances up to 3.4 km;

(iii) ***Symmetric Digital Subscriber Line (SDSL)***, which operates at distances up to 4.3 km and provides data rates of 2 Mbps in both directions over a single twisted pair of copper wires. The transmission of data in a symmetric manner makes SDSL and HDSL to be suitable for applications such as videoconferencing, which require identical downstream and upstream transmissions;

(iv) ***Very High Speed Digital Subscriber Line (VDSL)***, which operates at distances up to few hundred meters, supporting both symmetrical and asymmetrical data transmission for data rates up to about 52 Mbps over a twisted pair of copper wires up to a distance of 1.4 km;

(v) ***Rate-adaptive Digital Subscriber Line (RADSL)*** is a variation of ADSL, which uses a single twisted pair to support 0.6 to 7 Mbps and upstream rates of 128 to 1000 kbps. Its bit rate can be adjusted to suite the bandwidth requirements of the application and to accommodate the link length and link conditions or line quality. Like ASDL, this transmission of data in an asymmetric manner makes RADSL to be suitable for applications such as video-on-demand and video broadcast, which require high data rates for downstream and low data rates for upstream up to a distance of 5.5 km;

(vi) ***Multirate Digital Subscriber Line (MDSL)*** is a variation of SDSL, which adapts its transmission rate to the line length and line quality, like RADSL. It can operate from 272kbps up to 232 Mbps. Like HDSL and SDSL the transmission of data in a symmetric manner makes MDSL to be suitable for applications such as videoconferencing, which require identical downstream and upstream transmissions up to a distance of 4.3 km;

The xDSL have the same basic configuration as shown in Figure 3.4 below. At the user end is an xDSL-based modem or a router and at the network side there are xDSL-based modems or multiplexer called DSL access multiplexer (DSLAM), which receiver traffic from multiple users and passes it to the respective networks.

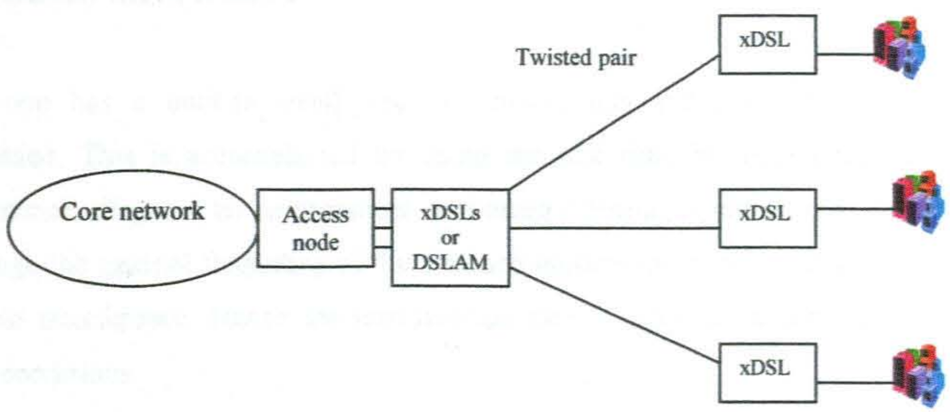


Figure 3.4 xDSL Network Architecture

EtherLoop

EtherLoop, also called next generation DSL or second generation DSL, is an access technology that uses Ethernet over twisted pair to allow the simultaneous voice and high speed data communications. It is a blending of DSL and Ethernet technologies; that is, it combines features of Digital Subscriber Line with features of Ethernet to provide both voice and high-speed data transmission (including Internet connection) over any ordinary phone line. EtherLoop uses a combination of QPSK and QAM digital modulation techniques to enable high transmission rates of up to 10 Mbps up to a distance of 6.4 km.

Like Ethernet and unlike DSL, EtherLoop transmits data packets in bursts using half-duplex communication model of Ethernet. This half-duplex and bust packet delivery capability of the EtherLoop technology makes EtherLoop to be less susceptible, than DSL, to certain kinds of interference from adjacent lines and devices such as bridge taps, loading coils, as well as wire gauge changes. EtherLoop can be deployed using the existing phone lines without special conditioning or replacement. Unlike Ethernet and like DSL, transmission is point-to-point, which means that connection speed is not decreased while multiple users are sharing the same path. Because transmission is point-to-point, at any given time one device can be designated as the server and the other the client. The server decides when the client can transmit, so collisions (which are a problem with Ethernet) can be avoided.

EtherLoop has a built-in intelligence to ensure that the signal quality is always maintained. This is accomplished by using the idle time between bursts to monitor performance, diagnose its environment, and using continuous rate adaptation techniques to change the internal frequency of the attached modem in order to reduce crosstalk and to avoid interference. Hence the transmission rate is adaptive, according to loop and traffic conditions.

EtherLoop may have an end-user plug-and play Ethernet device called EtherLoop modem (ELMo) on both ends or EtherLoop modem multiplexer on the core network end, which concentrate signals by multiplexing several loop circuits at the core network access node to concentrate modems into one single modem as shown in the Figure 3.5.

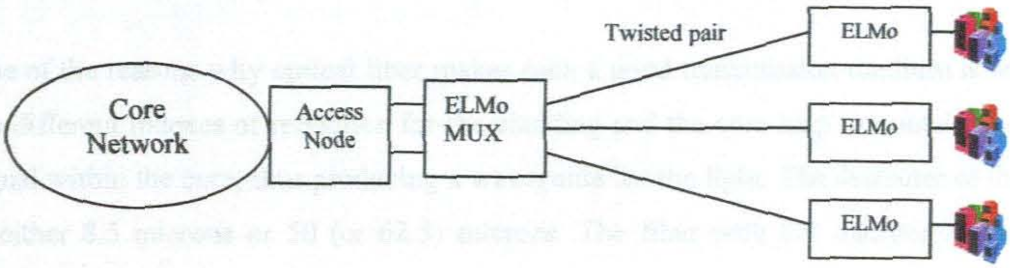


Figure 3.5 Ethernet Loop Architecture

Coaxial cable

At high frequencies attenuation in the above-mention types of transmission lines reaches prohibitive values mainly due to skin effect in the conductors, so the coaxial cable is used. The coaxial cable (or coax) consists of a center conductor surrounded by a concentric outer conductor, both made out of copper, with a dielectric material between them. The outer conductor acts as a shield to reduce radiation loss at high frequencies. It also protects the inner conductor, which is used as a carrier wire, from electromagnetic interference and radio frequency interference. The coaxial cable can be used, with repeaters, to provide long distance transmission of both analog and digital broadband signals (Huurdeman, 1997, p.146).

3.4.1.2 Optical fiber

Optical fiber (or fiber optic) refers to the medium and the technology associated with the transmission of information as light pulses along a glass or plastic fiber. Optical fiber carries much more information over long distances than conventional copper wire and is in general not subject to electromagnetic interference, crosstalk and radiation loss at high frequencies. The large bit rate and long distances are possible because of the small dispersion and low attenuation in the fiber.

One of the reasons why optical fiber makes such a good transmission medium is because the different indexes of refraction for the cladding and the core help to contain the optic signal within the core, thus producing a waveguide for the light. The diameter of the core is either 8.5 microns or 50 (or 62.5) microns. The fiber with 8.5 microns diameter is called single-mode fiber and the one with 50 or 62.5 microns is called a multimode fiber. Single-mode carries more data than multi-mode and has less attenuation than multi-mode fiber.

However, a major impediment to fiber is its cost of installation (Drewes, Hasholzner, and Hammerschmidt, 1997; Huurdeman, 1997, p.198; Goldman and Rawles, 2000, p.132; Dean, 2003, p.366), including the costs of right-of-way facilities, civil engineering, and its vulnerability to route damage over long service times.

3.4.1.3 Hybrid fiber-coaxial (HFC) and FTTx

Fiber connections are costly, and to minimize the costs an optical fiber can be used for short distances and then a transmission line can be used for the remaining part of the link. The type of the transmission line used can be either coaxial cable or a twisted pair of copper wire, thus giving two optical fiber/transmission line access technologies: hybrid fiber-coax (HFC) and FTTx.

3.4.1.4 Hybrid fiber-coax (HFC)

Hybrid fiber-coax can be used as an alternative for providing broadband connection since fiber connections can be costly. This can be done by using fiber for short distances, like from the core network to a central location and then do the conversion from optical to electrical with delivery of high data-rate electrical signal via standard coaxial cable. HFC combines high – bandwidth, low-loss fiber optic with low cost coaxial cables to provide more channels with better quality to the customers (Li and Liao, 1997). Figure 3.6 shows the HFC network architecture, where fiber trunks are used from head end to fiber nodes, which perform optoelectronic conversion, then the coaxial drops are used to cover the remaining distance to the end-user terminals.

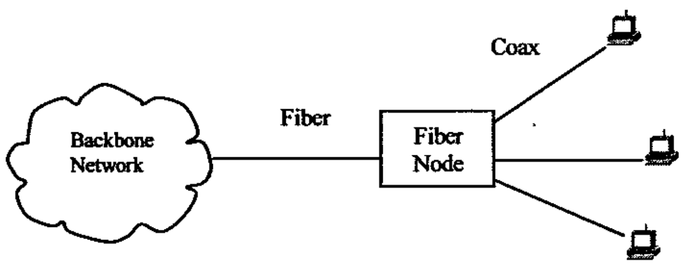


Figure 3.6 HFC Architecture

FTTx

FTTx is an abbreviation of fiber-to-the x, which is another hybrid of optical fiber and transmission line, but instead of using coax like in HFC, a twisted pair of copper wire is used. The "x" is used to describe the specific application of service or the location to which the optical fiber is deployed to in regards to the user's premises, which can be a business, home, or cabinet. Some of the FTTx access networks are listed below:

(i) ***Fiber-to-the-Cabinet (FTTCab):*** Fiber is used from the central point to the cabinet and then a conversion from optic to electrical is done to enable the signal to be transmitted over a twisted pair for the remaining part of the link or over xDSL if the distance is little bit longer.

(ii) ***Fiber-to-the-Curb (FTTC)***: Fiber is used from the central point to a point close to the user but not fully to the user's premises. Then conversion from optic to electrical is done to enable the signal to be transmitted over a twisted pair or a xDSL for the remaining part of the link.

(iii) ***Fiber-to-the-Business (FTTB)***: Fiber is used from the central point to a specific location within the business and then a conversion from optic to electrical is done to enable the signal to be transmitted over a twisted pair for the remaining part of the link within the business.

(iv) ***Fiber-to-the-Home (FTTH)***: Fiber is used from the central point to a specific location within the home and then a conversion from optic to electrical is done to enable the signal to be transmitted over a twisted pair for the remaining part of the link within the home of the user.

(v) ***Fiber-to-the-Building (FTTb)***: Fiber is used from the central point to a specific location within the building and then a conversion from optic to electrical is done to enable the signal to be transmitted over a twisted pair for the remaining part of the link within the building.

Unlike the first two, the latter three FTTx access networks: FTTB, FTTH, and FTTb, can also come completely comprising of fiber only and no copper.

3.4.1.5 Power Line Communication (PLC)

The wire-line (or wired) technologies described so far are solely designed for communication. Power line communication (PLC), on the other hand, is wire-line method of communication using the existing electric power transmission and electricity distribution lines.

Power lines were not designed for communication purposes as a result their properties as communication channels present a lot of challenges. Firstly, power lines are not shielded as a result they suffer radiation loss at high frequencies and they are more susceptible to noise, especially the RF interference. The radiated power can result in radio noise (or interference) in the radio frequency spectrum, especially in the ranges from very low frequency (VLF) to high frequency (HF) of the frequency spectrum, and the RF pickup can affect the quality of the channel on the power line. Secondly, impedance mismatches can result in reflections of the data being transmitted causing symbol error. Lastly, signal attenuation can occur due to the physical topology of the network, varying termination impedances, loads on the power lines, and the characteristics of the power line itself.

To meet some of the challenges listed above some techniques must be implemented at physical layer to provide reliable communication over power lines. These techniques must include modulation, and coding combined with retransmission strategies. Orthogonal frequency division multiplexing (OFDM) may be used to carry signals at high rates with few errors and forward error correction (FEC) may be used to redundantly encode the data to compensate for PLC's harsh channel characteristics. OFDM modulation spreads the signal over a very wide bandwidth, thus reducing the amount on power injected at a single frequency. This power spread decreases the transmitted power spectral density, which in turn reduces substantially the potential of interference to the radio users.

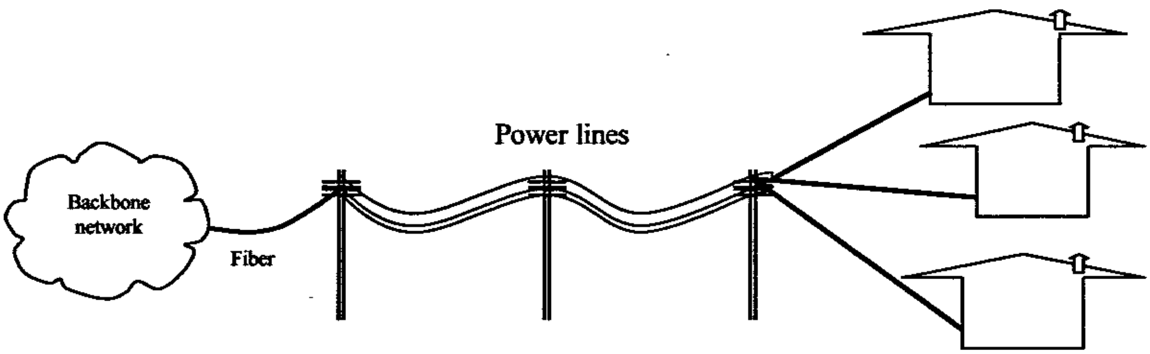


Figure 3.7 PLC Architecture

3.4.2 Wireless Links

To avoid the cost and inconvenience of setting up wired links or cabling, wireless links can be used to enable access into the network. In wireless links access to the network services and applications is no longer gained by conventional wired media but by wireless radio links. Wireless systems and technology can be classified as terrestrial or satellite (or extra-terrestrial) based, and as mobile or fixed.

3.4.2.1 Satellite links

Satellite communication use earth-orbiting satellites to relay information from the transmitting earth station to another station or stations. Satellite is commonly used for *access to remote locations, trans-ocean and intercontinental communication*. Satellite is capable of providing communication links to both fixed and mobile stations. However, the cost of satellite services makes the satellite not to be economically viable (Rampall and Mneney, 1997). Another disadvantage of satellite is propagation delay, which is due to the large distance between earth station and satellite. The approximation propagation delay equals 275 ms and two-way delay is 550 ms, which is not good for quality real-time communication.

3.4.2.2 Terrestrial radio

The bringing of broadband wireless communications information and the provision of wireless networking includes methods such as point-to-point, point-to-multipoint and multipoint-to multipoint. The literature reveals that there are many wireless systems in several bands, which are competing for broadband access. However, when it comes to range others fall short. Some of the broadband wireless technologies are discussed below:

3.4.2.3 Wireless local area networks (WLAN)

Wireless local area networks (WLAN) eliminate the need for wiring, thus enabling the nodes to move. They are cellular packet networks, which were originally devised to replicate in a wireless fashion the structures of the wired local area network (LAN) and to support mobility. There are two major standards of wireless LANs: IEEE802.11, being defined by the Institute of Electrical and Electronics Engineers (IEEE), and High-Performance Radio LAN (HiperLan), being standardized by the European Telecommunications Standards Institute (ETSI).

IEEE802.11 WLAN

The IEEE802.11 wireless LAN specification was ratified in 1997. It specifies the features of the medium access control (MAC) and of physical layer. During its inception, IEEE802.11 *provided for 1 Mbps and 2 Mbps, using spread spectrum technology:* frequency hopping spread spectrum (FHSS) and direct sequence spread spectrum (DSSS), or infrared at 2.4 GHz ISM band. To increase the throughput of IEEE802.11 to a higher rate of up to 11 Mbps, IEEE ratified the 802.11b standard, using direct sequence spread spectrum at 2.4 GHz ISM band. This was followed by the ratification of 802.11a, using orthogonal frequency division multiplexing (OFDM) at 5 GHz unlicensed national information infrastructure (UNII) band to offer data rates of 54 Mbps. Another IEEE wireless standard is 802.11g, which uses orthogonal frequency division multiplexing (OFDM) at 2.4 GHz ISM band to support data rates of up to 55Mbps. IEEE802.11g is backward compatible with 802.11b.

Broadband Radio Access Network (BRAN)

In Europe, the ETSI workgroup Broadband Radio Access Network (BRAN) have standardized the broadband radio LANs. They started by ratifying HiperLan 1 in 1997, which specifies an IEEE802.11-like radio LAN operating at 5GHz license-exempt band with data rate of up to 19 Mbps. The HiperLan 1 was followed by HiperLan 2, HiperAccess, and HiperLink, which support access to IP, and ATM core networks.

Like Ethernet and 802.11, HiperLAN 1 uses a carrier sense multiple access protocol to give wireless LAN access to end-user devices. CSMA regulates traffic on the LAN by having devices listen for other transmissions and transmit data only when no other device is transmitting. HiperLAN 1 also provides quality-of-service (QoS) support for the varying needs of data, video, voice, and images.

HiperLan 2, like 802.11a, operates at 5 GHz license-exempt band, using OFDM. It offers high-speed wireless connectivity of up to 54 Mbps and seamless connectivity with corporate LAN, 3G cellular systems, mobility and QoS for future applications such as multimedia, voice over Internet protocol (VoIP) and real-time video.

Network Structure

Wireless LAN standards are meant to accommodate one or both of the two different types of network structures: infrastructure-based networks and ad hoc networks.

Infrastructure mode has at least one access point connected to the wired network to serve wireless end (or client) stations as illustrated in Figure 3.8. The access point uses a point control protocol, which grants every station a chunk of airtime in an orderly, sequential manner, much like a token ring. At the end of each contention free period, a configurable length of time is left open to allow stations to authenticate to the network or transmit high priority data out of turn. The infrastructure is sometimes referred to as client/sever network configuration and the access point is referred to as Basic Service Set (BSS) or simply base station.

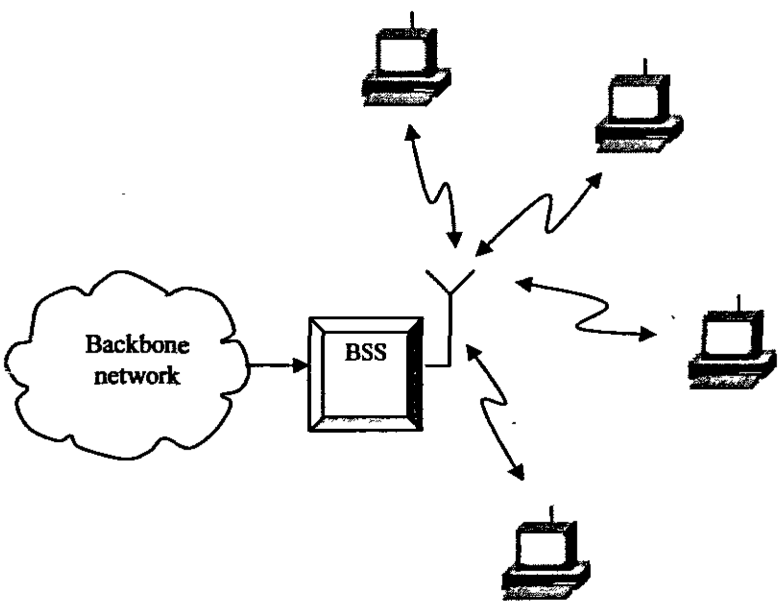


Figure 3.8 Infrastructure Network

In ad hoc mode, shown in Figure 3.9, several stations communicate with each other without the intervention of the access point. This mode is also referred to as peer-to-peer mode or independent basic service set (IBSS). Ad hoc network uses spokesman election algorithm (SEA) to elect one machine as the base station (master) of the network with others being slaves. Another algorithm in ad hoc network uses a broadcast and flooding method to all the other nodes to establish who's who.

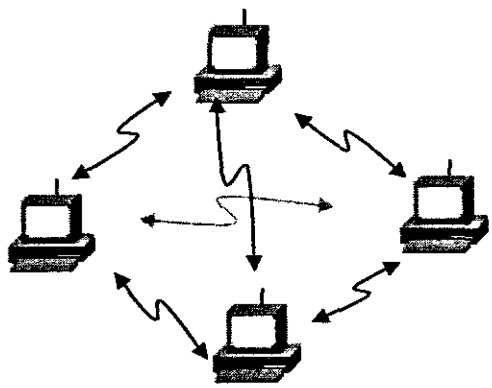


Figure 3.9 Ad hoc Network

3.4.2.4 Wireless personal area networks (WPAN)

Wireless Personal Area Networks (WPAN) are short range wireless networks, which are standardized by IEEE 802.15 Working Group for WPAN. WPAN can be used to exchange information in the range from couple of centimeter to a couple of meters. The standard caters for low data rates of about 200 kbps or less, medium data rates, and higher data rates of 20 Mbps or greater. It includes Bluetooth, Spike, IrDA, and HomeRF technologies (Cordeiro, Gossain, Ashok, and Agrawal, 2003). Table 3.2 gives a summary of some of the WPAN technologies.

Table 3.2 Wireless Personal Area Network (WPAN) as standardized by IEEE802.15

TECHNOLOGY	DATA RATE	FREQUENCY	RANGE
Bluetooth	721 kbps	2.4 GHz (ISM)	10 m to 100 m
IrDA	16 Mbps	2.4 GHz (ISM)	300 cm to 1.2 m
HomeRF	1.6 to 10 Mbps	2.4 GHz (ISM)	50 m

Though all the technologies are using wireless, not all of them are using radio waves, for example, IrDa transmit data via infrared light waves, instead of radio waves, while Bluetooth, and HomeRF are using radio waves to transmit data.

3.4.2.5 Fixed broadband wireless systems

Fixed Broadband Wireless Systems are wireless links that have data rates that are typically in excess of 2 Mbps, which can be used to provide the wireless connection between the users and backbone network, thus enabling the users to have radio access to broadband services like video applications. Fixed wireless loop has an advantage that it can maintain a much high signal-to-interference ratio than mobile wireless station and therefore will be robust to fading, traffic, multipath interference, etc. (Cox, Haskell, Lecun, Shanraray, and Rabiner, 1998).

High-Performance Radio Link (HiperLink)

HiperLink is an ETSI-BRAN's high-speed radio links for static interconnections. It can provide data rates up to 155 Mbps in the 17 GHz band for distance up to 150 meters. HiperLink is used most of the time to provide a high-rate interconnection between HiperAccess and HiperLan 2 networks (Chevillat and Schott, 2003).

High-Performance Radio Access (HiperAccess)

HiperAccess is an ETSI-BRAN's fixed broadband point-to-multipoint wireless system. It is optimized for the 26 GHz, and 40.5 to 43.5 GHz band, and has a typical data rate of 25

Mbps and can cover a range of up to 5 kilometers. HiperAccess provides remote access to an IP or ATM backbone network. HiperAccess is depicted in Figure 3.10.

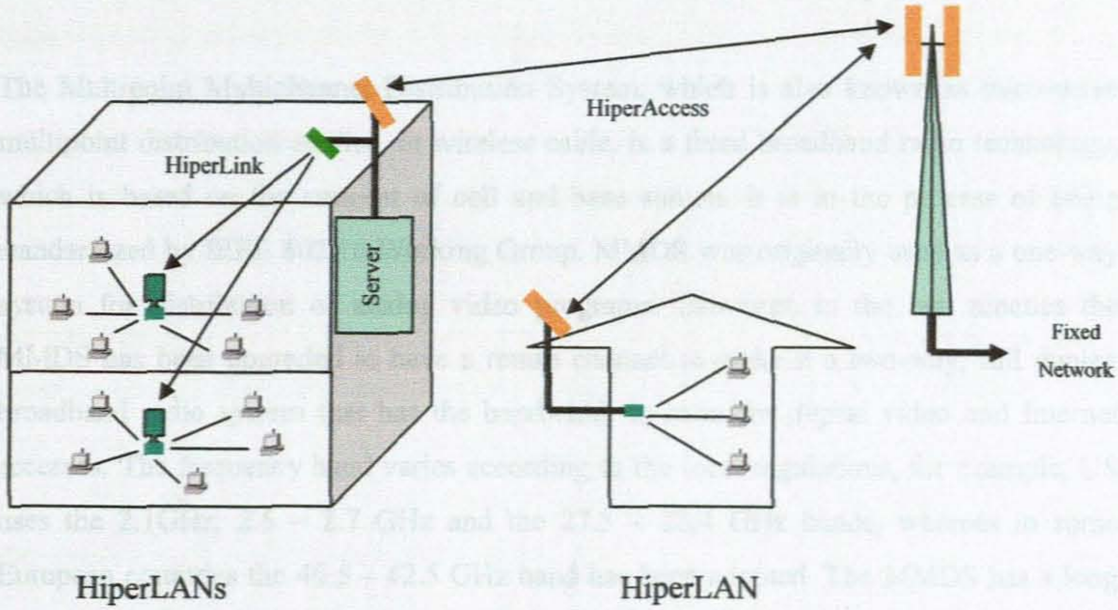


Figure 3.10 ETSI Broadband Radio Access

Local Multipoint Distribution System (LMDS)

Local Multipoint Distribution System (LMDS) is a fixed broadband radio technology that is defined by the IEEE 802.16 standard, which is used to deliver voice, data, Internet access, and video services at carrier frequencies equal or higher than 24 GHz (typically within specified bands in the 24 GHz to 40 GHz range). It provides asymmetrical two-way transmission operation at up to 1.5 Gbps downstream and 200 Mbps upstream. It is a line-of-sight, point-to-multipoint microwave system, which propagates communication signals with relatively short RF range of up to 5 kilometers to multiple end-users. It is based on the concept of cell and base station. The base station, which is attached to the core network, transmits signals in a point-to-multipoint method to the end-users, and the return paths from end-user to the base station is accomplished by point-to-point links.

Multipoint Multichannel Distribution System (MMDS)

The Multipoint Multichannel Distribution System, which is also known as microwave multipoint distribution service, or wireless cable, is a fixed broadband radio technology, which is based on the concept of cell and base station. It is in the process of being standardized by IEEE 802.16 Working Group. MMDS was originally used as a one-way system for distribution of analog video programs. However, in the late nineties the MMDS has been upgraded to have a return channel to make it a two-way, full duplex broadband radio system that has the bandwidth to cater for digital video and Internet accesses. The frequency band varies according to the local regulations, for example, US uses the 2.1GHz, 2.5 – 2.7 GHz and the 27.5 – 28.4 GHz bands, whereas in some European countries the 40.5 – 42.5 GHz band has been adopted. The MMDS has a long reach of up to 50 kilometers from the base station and a throughput of 128 kbps to 10 Mbps, which makes it a match for rural applications (Dean, 2003, p.436).

3.4.2.6 Medium Choice

For rural area network (RANET) the nodes can be a considerable distance apart; for example, a network that can enable the clinics and the local hospital to exchange information can have nodes that are 50 kilometers or more from one another. So RANET must be able to support nodes that are geographical distributed by distances of about 50 kilometers or more. The other challenge of RANET is the terrain over which it has to be deployed. Most rural areas are characterized by mountainous terrain, forests and many rivers (ITU, 1996), which can make the use of other types of transmission media, such as transmission line and optical fiber, to be very difficult, if not impossible.

The choice of the medium is determined by the required transmission speed, security needs, as well as the physical characteristics of the environment in which the medium is to be deployed (Dean, 2003, p.365). Additionally, to the factors referred to by Goldman and Rawles, distance and cost of connection can also have a big influence on choosing the network media.

The first two types of media, transmission lines and optical fiber, need the communicating devices to be physically connected. Though physical connection is sufficient for many applications, especially for short distances, for rural scenario, the deployment of wired links can be impractical due to vast distances to subscribers, copper theft, high labor and maintenance costs (Rampall and Mneney, 1997).

For rural areas, which are characterized by mountainous terrain and many rivers, radio transmission solutions are the best (ITU, 1996), and wired links such as twisted pair, coax, optical fiber, and hybrid fiber-coax can only be used for short distances. Wireless LANs can also be used for short distances, especially in cases where it is either impractical or expensive to used wired links.

For nodes that fall beyond the footprint of DSL, wireless LAN, LMDS, or HiperAccess, MMDS can be implemented in a super-cell scenario to provide broadband links of up to 50 kilometers. And to avoid the effects of heavy rain and fog, which are common in mountainous topography, the low frequency bands (2.1GHz and 2.5 to 2.69 GHz) can be used, and to mitigate the multipath distortion due to forests and mountains, which characterize the rural areas, OFDM must be used.

Another possibility for rural networks can be the hybrid of wired and wireless media. Where a [fiber or copper] cable can be used for a short distance, like from the core network to a central location, and then change to wireless. Figure 3.11 shows a hybrid of wired-wireless system, where a wired technology like transmission line or fiber optic is used as a feeder from the core network to a line termination (LT) unit, then wireless technology takes over to cover the remaining part.

Alternatively, since the government has embarked on the program of electrifying rural areas, resulting in a presence of a vast infrastructure being deployed for power distribution, this penetration of the service in rural areas can be exploited to provide broadband access to remote and rural areas. Where power lines have been used, the

WPAN can be used to provide the last 50 m connections, so as to avoid the signal attenuation, which is common in the buildings due to physical topology of the house wiring, varying network impedances, loads on the power lines, and the characteristics of the power line itself.

To cut out the civil engineering costs in optical fiber, fiber optic cable can be attached to existing power lines to offer the last mile connectivity (Ernberg, 1999).

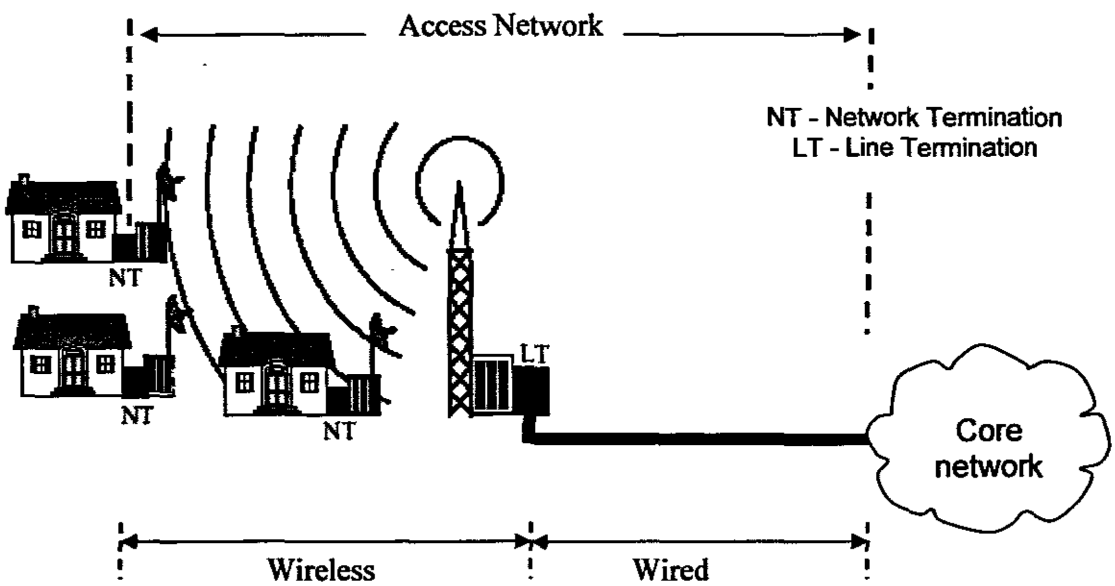


Figure 3.11 Hybrid Wired-Wireless System

3.5 IN SUMMARY

This chapter covered the concept of telemedicine, rural telemedicine and rural area networks that can be used as a communication platform for rural telemedicine. The chapter also provides a review of access technologies that can be used in rural area networks in order to provide rural telemedicine. To conclude the chapter, the next chapter is introduced, which is about the mechanism that can be used in order to provide quality of service in the proposed multiservice IP-based rural area networks.

CHAPTER 4

4. IP-BASED MULTISERVICE NETWORK AND QUALITY OF SERVICE

4.1 INTRODUCTION

This chapter introduces the concept of IP-based multiservice networks, and it also highlights the advantages of using IP over ATM. The chapter also discusses some mechanisms that can be used to compensate for IP's best effort debilities so as to make it suitable for real time traffic such as voice and video applications, which are having stringent network requirements.

4.2 MULTISERVICE NETWORK

Rural area networks that are used as telecommunication platforms for rural telemedicine must be able to support an array of services like voice, data and video applications. Traditionally, disparate networks are used to provide these services, for example; within one enterprise there will be a separate network for data, voice and video applications. These have been deployed autonomously and operated in isolation, and in many instances video is not even considered at all because of its prohibitive costs.

Convergence is derived by the coming together of voice, data, and video contents and by the integration of data and voice networks. This is accomplished by the merging of packet switching technology with telephony signaling and call-processing intelligence, allowing carriers to consolidate typically separate voice and data overlay networks and provide new multiservice network, which is capable of providing differentiated integrated communication services. Instead of having distinct and separate network over different media being implemented for data, voice, and video services within one enterprise, these multiservice networks are capable of providing voice, data and video communications leveraging a common packet-based platform. These multiservice networks are referred to as converged networks and they are utilizing Internet Protocol (IP) as the ubiquitous transport to carry traffic from dissimilar applications and networks on a single unified communication infrastructure.

The choice of IP over ATM is that IP has proven to be extremely powerful because it was designed to provide internetworking capabilities over a variety of underlying network technologies and is also ubiquitous (Leon-Garcia and Widjaja, 2004, p.705). However, IP is designed to provide best effort service to every user regardless of its requirements. This best effort service often makes congestion to be common in the IP-based networks resulting in serious degradation for applications that need a minimum bandwidth to operate satisfactory and those that require real-time response. The ATM, on the other hand, was designed at the outset to provide end-to-end quality of service to support integrated traffic, which includes real-time traffic. So for IP to meet all the requirements of converged networking such as scalability, reliability, and support for quality of service, which is essential for real time traffic, it needs to be enhanced.

Rationale for Convergence utilizing IP

- Leveraging a single infrastructure for all kinds of traffic.
- More efficient than the use of disparate facilities for each application transport.
- Lowered network operation, administration, and management costs.
- Easy to add and deploy new functionalities and technologies.
- IP routers are now as fast as ATM switches or even faster
- Communication is possible in both real-time and non-real-time.
- Enhanced communication capabilities brought about by unified messaging, video conferencing, video streaming, IP-telephony, etc.
- Lowered overall bandwidth requirements and improved bandwidth efficiency
- Ability to provide internetworking across any layer 2 technology

4.3 IP AND REAL-TIME APPLICATIONS

The default and traditional datagram delivery service used in IP-based networks is called best-effort service. This type of service delivers the packets as fast as possible, using, in most cases, first-in-first-out (FIFO) scheduling mechanism, which delivers the datagram to the output port in the same order as they came from the input port without giving preference to any type of traffic. For slightly loaded network this service can be sufficient, but if the network is congested queuing losses and queuing delays can make the network not to be suitable for real-time traffic.

Another problem with IP network is the link-state routing protocols that are used to distribute IP routing information throughout a single autonomous system (AS) in an IP network; that is, Interior Gateway Protocols (IGPs). These IGPs like Intermediate System-Intermediate System (IS-IS) protocol, and Open Shortest Path First (OSPF) protocol create a shortest path matrix for each router and then forward traffic along this shortest path. Unlike ATM's PNNI, IP routing protocols do not have a Connection Admission Control (CAC) mechanism to balance network traffic along multiple paths. The result is under-utilization of certain links, creation of hot spots in the network and congestion along the shortest path route (Sayeed, 2000). Congestion can cause long queues, which will result in transit delays and packet losses.

Real-time traffic can be divided into two categories: that which is tolerant to variation in delay (or jitter) and that, which is not. For an example, a unidirectional video streaming can tolerate jitter if at the destination the received data is first buffered and then played back at some offset delay (RFC1633, 1994). However, for applications that involve interactive real-time communication, jitter and end-to-end delays (or latency) can degrade the quality of the received signal. In addition to minimum delay requirements, real-time traffic also needs enough bandwidth from the network.

4.4 QUALITY OF SERVICE (QOS)

Quality of service (QoS) refers to the ability of a network element to have some level of assurance that its traffic and service requirements can be satisfied. Enabling QoS requires the cooperation of all network layers, as well as every network element from end to end, for example, an application, host, and router must be able to meet the required QoS (Ma and Shi, 2000). According to ITU-T Recommendation E.800, QoS can be defined as the collective effect of service performances, which determine the degree of satisfaction of a user of the service (Afullo, 2004).

Parameters such as bandwidth, delays (both latency and jitter) and packet loss rate characterize the nature of packet delivery or quality of service (QoS) that can be provided by the network (RFC2216). Different applications have varying needs for latency, delay variation (jitter), bandwidth, packet loss, and availability. These parameters form the basis of QoS. If the network meets all the delay and the bandwidth requirements for a certain application, the network is said to be able to provide a quality of service for that application. However, if the requirements are not met the network is said to be not able to provide the quality of service required by that application.

From the explanation of the best-effort nature of packet delivery offered by traditional IP networks, it can be said that in its original form IP does not guarantee that the user will be able to get a service whose quality is sufficiently predictable. That is, there is no guarantee that the application can operate in acceptable way over duration of time determined by the user. Another shortfall of traditional IP networks is that they always use the shortest paths to forward traffic to its destination. This may cause congestion on some links while leaving other links underutilized.

In order to manage all the various applications that can be accommodated in a multiservice network such as streaming video, voice over IP, interactive video, and others, a converged network should be designed to provide the requisite QoS to applications in addition to best-effort service. This is necessary so that the different network performance requirements of different applications can be met. In addition these converged IP networks need to include traffic engineering to control the paths of the packets.

To facilitate true end-to-end QoS on an IP-network, the Internet Engineering Task Force (IETF) has defined two models: Integrated Services (IntServ) and Differentiated Services (DiffServ). IntServ follows the signaled-QoS model, where the end-hosts signal their QoS need to the network and the network reserves the resources for each flow. DiffServ, on the other hand, works on the provisioned-QoS model where the traffic is divided into different classes and network elements are set up to service these multiple classes of traffic, with varying QoS requirements. Integrated service was the first quality of service model to be defined, however, it was found that it did not scale to large networks, which led to the definition of differentiated service model (Sayeed, 2000).

Traffic engineering can help to ease congestion at a network level by creating alternate paths for traffic to be routed between the same ingress-egress pair. For traffic engineering to work in IP environment, either routing metrics must be manipulated so that traffic has multiple paths or some form of connection orientation mechanism must be introduced to create equal bandwidth paths between the same ingress-egress router pair. While the former is unscalable to large networks, the latter, if engineered well, can scale to very large networks and achieve better results. IETF proposes multi-protocol label switching (MPLS) as a means to provide the connection orientation mechanism to IP traffic without sacrificing bandwidth overheads like ATM (Sayeed, 2000). MPLS can be used to provide connections with label switched paths (LSPs) in a connectionless IP networks (Xiao, Irpan, Hannan, Tsay, and Ni, 1999). During the creation of these virtual connections (LSPs), some QoS requirements can be introduced in order to make these LSPs QoS compliant.

4.5 PROVIDING QOS IN MULTISERVICE IP-BASED NETWORKS

Real-time traffic such as audio and video needs a network that has a low latency, low jitter and low packet loss rate in order to ensure quality communication. Traditional IP networks are designed to provide a best-effort service, which at times of congestion has variable queuing delays and packet losses that can produce very low quality video and audio (Riley and Richardson, 1997, pp.91-99). To make IP networks to be suitable for real-time traffic, enhancement of the packet delivery method is necessary in order to provide real-time audio and video traffic with a guaranteed QoS, in the form of bandwidth, packet loss, and end-to-end delay guarantees.

In this chapter, the two IETF defined service models for providing QoS in IP-based networks, IntServ and DiffServ, will be introduced. The strength and the shortcomings of each model will be discussed, and then the best model will be proposed, which will consist of either the one or the combination of the two QoS models.

4.5.1 Integrated Services Model

As a first step towards providing quality of service in IP-based networks, the IETF developed the integrated services (IntServ) model. In IntServ model resources such as bandwidth and buffers are reserved, using the reservation scheme called Resource Reservation Protocol (RSVP), for a given data flow to ensure that the application receives the requested quality of service. The IntServ model provides three classes of services (CoS): guaranteed services, controlled load services, and the traditional best-effort services.

Guaranteed service is designed to provide a firmly bounded end-to-end delay and bandwidth with no-loss guarantees, which is essential for applications that have stringent real-time delivery requirements. Controlled load services is a class of service that is designed to provide approximately the same quality of service under heavy loads as it does under light loads, and it is intended for applications that can tolerate a certain amount of loss or delay.

The IntServ model consists of four components: the packet classifier, the packet scheduler, the admission control, and the reservation setup protocol as shown in Figure 4.1. Packet classifier identifies flows that are to receive a certain level of service. Packet schedulers handle the forwarding of different packet flows in a manner that ensures that the QoS commitments are met. The Admission control determines whether the router has necessary resources to accept the new flow. The resource reservation protocol (RSVP) is used as a signaling protocol for application to reserve appropriate resources along the path traversed by a new flow requesting a QoS service.

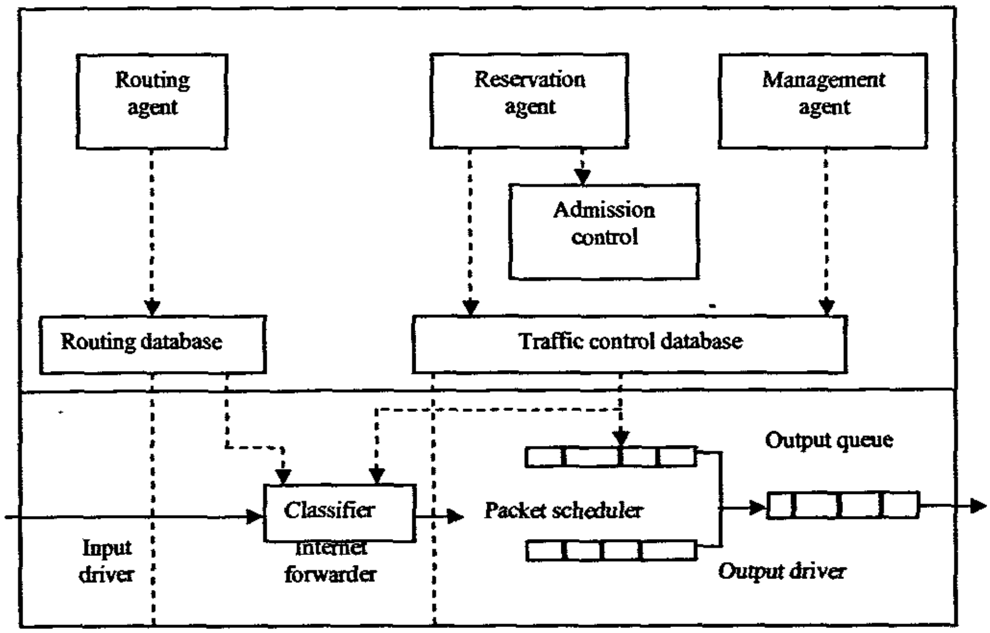


Figure 4.1 Router model in Integrated Service IP

The signaling process is shown in Figure 4.2. To startup a RSVP session, the host (or sender) sends a RSVP path message to the receiver specifying the characteristics of the traffic. Upon receiving this path message the receiver responds with an appropriate reservation-request (RESV) message informing each router of the requested QoS, and if the flow is found admissible, each router in turn adjust its packet classifier and packet scheduler to handle the given packet flow. The same RESV massage is used to notify the sender that the requested network performance requirement has been allocated so that it can start sending packets. However, if one of the routers does not have the resources to support the new flow, the request is rejected and the router in question will send an error message to the receiver, and the signaling process will terminate (Ibe, 2002, pp.224 -225; Aidarous and Plevyak, 2003, pp.96-110; Leon-Garcia and Widjaja, 2004, pp.706-716).

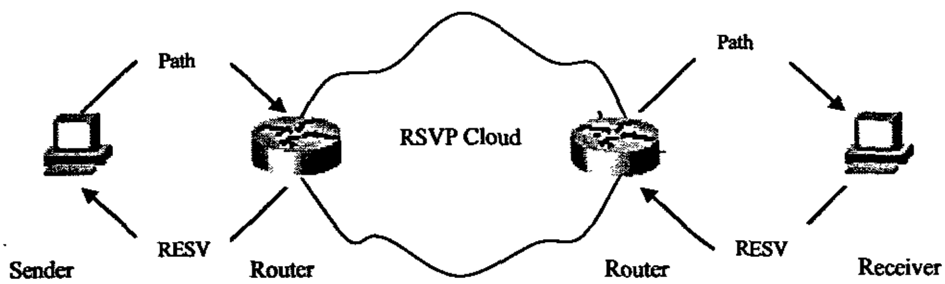


Figure 4.2 RSVP Signaling process

IntServ model, however, has some shortcomings, which are as follows:

- Every node along a packet's path, including the end systems like servers and PCs, need to be fully aware of RSVP and be capable of signaling the required QoS.
- Reservations in each device along the path are "soft," which means they need to be refreshed periodically, thereby adding to the traffic on the network and increasing the chance that the reservation may time out if refresh packets are lost. Though some mechanisms alleviate this problem, it adds to the complexity of the RSVP solution.
- Maintaining soft-states in each router, combined with admission control at each hop adds to the complexity of each network node along the path, along with increased memory requirements, to support large number of reservations.
- Since state information for each reservation needs to be maintained at every router along the path, scalability with hundreds of thousands of flows through a network core becomes an issue.

4.5.2 Differentiated Services

Due to scalability and complexity issues associated with IntServ model and RSVP, IETF introduced another service model called the differentiated services (DiffServ) model, which is intended to be simpler and more scalable. DiffServ is a coarse-grained approach that provides QoS to aggregated traffic and it allows all types of digitally represented information such as conventional data, voice, video, text, graphics, etc. to be transported on a single and unique IP network infrastructure, while their QoS requirements are being supported (Aidarous and Plevyak, 2003, p.111; Leon-Garcia and Widjaja, 2004, p.717).

DiffServ is generally deployed at the network edge by access devices and propagated through the backbone network by DiffServ-capable routers (Ibe, 2002, p.225). This enable the traffic entering the network to be categorized into different classes, called class of service (CoS), and applying QoS parameters to those classes.
To accomplish this, packets are first divided into classes by marking the packets according to their required QoS in the IP header field at network boundaries (RFC 2474). Packets that have the same marking or DSCP are assigned for the same DiffServ class and will receive a similar predefined service or processing in network core nodes.

Once packets are marked and classified at the edge of the network, specific forwarding treatments, formally called Per-Hop Behaviors (PHBs), are applied on each network element, providing the packet the appropriate delay-bound, jitter-bound, bandwidth, etc. among competing traffic streams.

4.5.2.1 DiffServ architecture

The basic DiffServ network or domain consists of a set of network nodes, typically routers, which are working together to provide predefined services to packet streams that are classified into classes. According to RFC 2475, there are two types of nodes in DiffServ domain: boundary and interior node. Boundary nodes connect the DiffServ domain cloud to other domains. Interior nodes are connected to other nodes (interior and boundary) within the same DiffServ domain. Boundary nodes can be either ingress or egress nodes, depending on the packet flow direction. Where the ingress node is the one at the entry of the DiffServ domain, while egress is the one at the outer edge of the DiffServ domain. Figure 4.3 illustrates at typical DiffServ network.

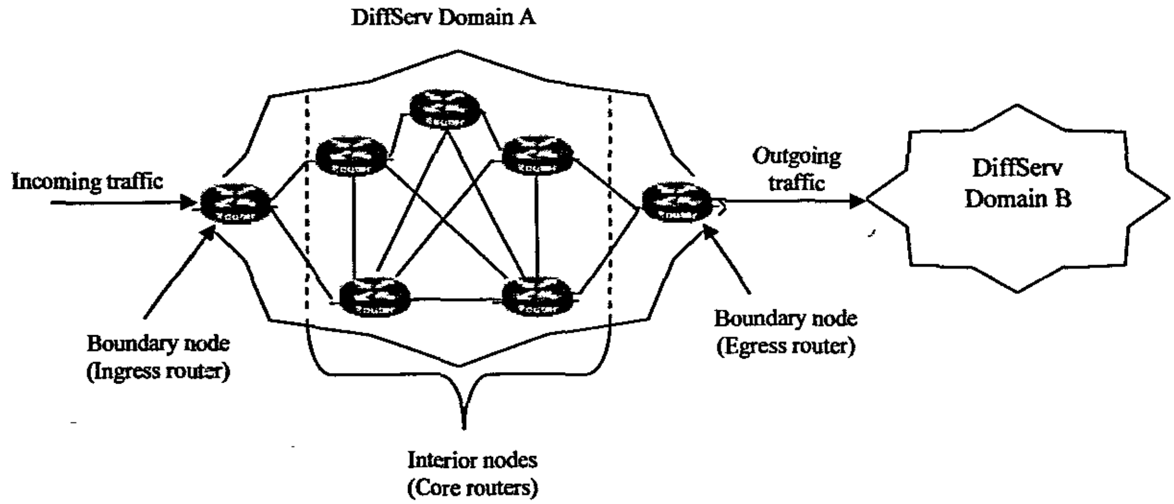


Figure 4.3 Basic DiffServ Network Architecture

Both the boundary and interior network nodes are used to implement the functional elements that compose the DiffServ architecture.

Boundary Nodes

Boundary nodes in a DiffServ domain act both as an ingress and as an egress node for different directions of traffic.

Ingress routers are responsible for packet classification, stream monitoring, marking, and conditioning. This is done to the ingress traffic to ensure that packets, which transit the domain, are appropriately marked to select a PHB from one of the PHB groups supported within the domain.

Egress node performs traffic conditioning functions on traffic forwarded to a directly connected peering domain, depending on the details of the TCA between the two domains.

Interior Nodes

Internal nodes are core routers, which are mainly used for the optimized IP processing for the marked packets. They perform the task of packet forwarding processing by identifying the correspondent packet behavior and scheduling it accordingly. According to RFC 2475, the interior nodes may also perform limited traffic conditioning functions such as DSCP remarking.

4.5.2.2 Traffic marking

Both Ipv4 and Ipv6 headers have 8-bit fields called Type of Service (ToS) and Traffic Class bytes, respectively. DiffServ model redefines these 8-bit IPv4 ToS Octet and IPv6 Traffic Class Octet fields as Differentiated Services (DS) fields, and replaces them with a 6-bit bit-pattern called the Differentiated Services Code Point (DSCP) and a two-bit currently unused (CU) field (RFC 2474). The DSCP replaces the six most significant bits of the octet while CU replaces the two least significant bits of the octet (RFC 3260), as shown in Figure 4.4. This marking can be either done by the host or by the edge routers of the DS domain.

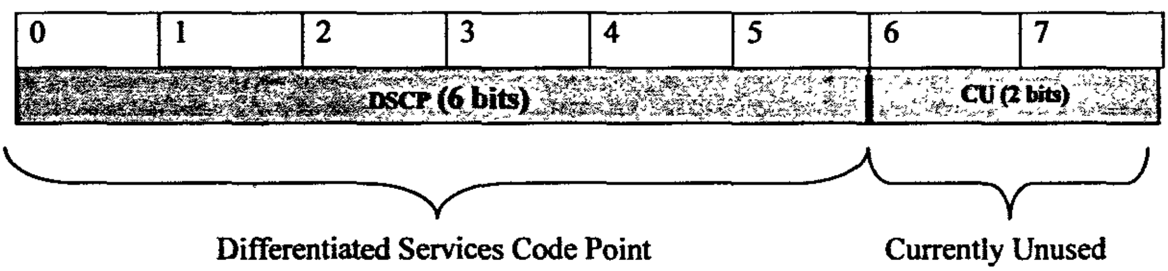


Figure 4.4 Differentiated Services (DS) field

The DS field supersedes the Ipv4 ToS or Ipv6 Traffic Class octet definitions and. is used to specify how the packet forwarding must be treated (RFC 2474), while the DSCP is used to select the per-hop behavior at each node, and all classification and QoS revolves around the DSCP in the DiffServ model. With DSCP up to 2^6 or 64 different aggregates or classes can be supported in any given node (Cisco Systems, 2001). Packets from multiple applications or sources could have the same DSCP value, and the collection of packets that have the same DSCP value in them, and crossing in a particular direction is called a behavior aggregate (BA).

4.5.2.3 Traffic Classification and Conditioning

Packet classification involves the identification of traffic that may receive a differentiated service. Packets are selected in a traffic stream based on content of some portion of their header using packet classifier. There are two types of classifiers: behavior aggregate (BA), and the multi-field (MF) classifier. The BA classifier classifies packets based on the DSCP only, whereas the MF classifier selects packets based on the value of a combination of one or more header fields, such as source address, destination address, DS field, protocol ID, source port and destination port numbers.

Once packet classification is done, packets are fed to a traffic conditioner, which performs metering, shaping, policing and/or remarking to ensure that the traffic entering the DS domain conforms to the rules for services specified in Traffic Conditioning Agreement (TCA) in accordance with the domains service provisioning policy. Figure 4.5 shows the block diagram of typical ingress router functionalities.

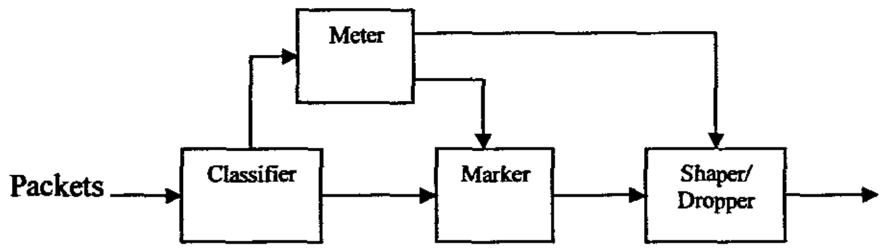


Figure 4.5 DiffServ Packet classifier and Traffic Conditioner

Meters

Traffic meters checks compliance to traffic parameters; that is, they measure the temporal properties of the stream of packets selected by a classifier against a traffic profile specified in a TCA and passes the results to the marker and shaper/dropper to trigger action for in/out-of-profile packet.

Markers

Packet markers write/rewrite the DSCP values; that is, they set the DSCP in packets based on defined rules. The rewrite or remark is done to ensure conformance and only done when necessary.

Shapers

Shapers delay some or all the packets in a stream in order to bring them into compliance with a traffic profile.

Dropper

Droppers perform the policing of the packet streams; that is, they discard some or all the packets in a stream that are not in compliance with a traffic profile.

4.5.2.4 Per-Hop Behaviors (PHBs)

Per-Hop Behaviors (PHBs) refers to specific forwarding treatments of traffic aggregates or packets in a DiffServ network. This includes packet scheduling, queuing, policing or shaping behavior of a node on any given packet belonging to a BA. To date there are four standard PHBs that have been defined: the assured forwarding (AF) PHB, expedited forwarding (EF) PHB, class-selector PHB, and default PHB (Cisco, 2001).

Assured Forwarding (AF) PHB

Assured Forwarding Per-Hop Behavior (AF PHB) is designed to give a reliable service even in times of congestion. That is, there is a high probability of traffic being delivered to the destination as long as the aggregate traffic does not exceed the traffic profile. However, AF PHB service does not define any specific restriction on bandwidth or delay for packets.

AF PHB is designed to give different forwarding assurances. For example, traffic may be divided into gold, silver, and bronze, with each having its own drop preference. AF PHB defines four priority classes, and within each class, it is possible to specify 3 drop precedence values. Table 4.1 illustrates the DSCP values for each class and the drop precedence, where bits 0, 1, and 2 are used to define the class; bits 3 and 4 specify the drop precedence; and bit 5 is always 0 (RFC 2597).

Table 4.1 Assured Forwarding Codepoint Values

Drop Precedence	Class 1	Class	Class 3	Class 4
Low Drop	001010	010010	011010	100010
Medium Drop	001100	010100	011100	100100
High Drop	001110	010110	011110	100110

Drop precedence is intended for deciding on packet priority during congestion; that is, once the network becomes congested, the packet with high drop precedence will be dropped first before the one with medium drop precedence, before the one with low drop precedence.

Expedited Forwarding (EF) PHB

Expedited Forwarding Per-Hop Behavior (EF PHB) is designed for traffic that is required to be guaranteed enough resources to ensure that it receives its minimum guaranteed rate (Ibe, 2002, p.227), as a result it provides a low-loss, low-latency, low-jitter, assured-bandwidth, end-to-end service through DS domains. The end-to-end service received by using EF PHB is often called a "premium service" and is equivalent to a "virtual leased line", which is essentially equivalent to CBR service in ATM (Leon-Garcia and Widjaja, 2004, p.719). This type of service is designed for real-time traffic from applications such as voice over IP and interactive video.

In order to ensure premium service, which is characterized by very low latency and assured bandwidth, the aggregate arrival rate of packets with EF PHB at every router should be less than the aggregate allowed departure rate. EF PHB is assigned the DSCP 101110 (Cisco, 2001; Aidarous and Plevyak, 2003, p.119; Leon-Garcia and Widjaja, 2004, p.719; RFC 2474), as shown in Table 4.2.

Table 4.2 Expedited Forwarding PHB format

0	1	2	3	4	5	6	7
1	0	1	1	1	0	Currently Unused	

Class-Selector PHB

Class-Selector PHBs are designed to preserve backward compatibility with the IP-Precedence scheme defined in [RFC1812]; that is, they retain almost the same forwarding behavior as nodes that implement IP-Precedence based classification and forwarding. They ensure that DS-compliant nodes can co-exist with IP-Precedence aware nodes, with the exception of the DTR or type of service bits of the Ipv4 ToS Octet, as defined in [RFC791] and [RFC1349] (Cisco, 2001; RFC2474; RFC3260). Class-Selector PHBs are having DSCP values of the form xxx000, where x is either 0 or 1.

Default PHB

The standard best-effort treatment that nodes (routers) perform when forwarding traffic is known as the default Per-Hop Behavior (DE PHB). DE PHB is assigned the DSCP 000000 (Cisco, 2001; Ibe, 2002, p.228; Aidarous and Plevyak, 2003, p.119; Accellent, 2003; Leon-Garcia and Widjaja, 2004, p.719; RFC 2474), as shown in Table 4.3 below.

Table 4.3 Default PHB format

0	1	2	3	4	5	6	7
0	0	0	0	0	0	Currently Unused	

4.5.2.5 Providing QoS using DiffServ

For true QoS using DiffServ model, the entire path that is traveled by the packet must be DiffServ enabled. There must be service level agreement (SLA) between the service provider and the customer to specify a traffic conditioning agreement (TCA), which defines classification rules as well as metering, marking, shaping and policing rules. For assured service, SLAs are usually static; that is, customers can start transmission without signaling their service providers, however, for premium service, the SLAs can be either static or dynamic. Dynamic SLAs allow customers to request by signaling their service provider for premium service on demand without subscribing to it.

Once SLA is in place, the incoming packets need to be classified, and from these classification results, traffic profiles and corresponding policing, marking and shaping can be performed on the traffic based on the SLA. For customers with assured service, policing should be implemented with a token bucket, so that some kind of burst is allowed. For customers with premium service, policing should be implemented with a leaky bucket so as to avoid the introduction of burst. When the packet arrives and there are tokens in the bucket the packet is considered to be in profile or conformant and the DSCP of the packet is set 101000 for assured service or 111000 for premium service. However, if a packet arrives and there is no token in the bucket, the packet is considered as out-of-profile or non-conformant, and the DSCP of the packet is set 100000 for assured service. For premium service, the non-conformant packets may be dropped immediately or may be transmitted, depending on the SLA, and if they are transmitted, their DSCPs are also set to 111000.

All assured service packets that are not dropped are put into the same queue, called assured queue (AQ) to avoid out-of-order delivery. Similarly, all premium service packets that are not dropped are put into the same queue, called premium queue (PQ).

Once a packet is marked its treatment is determined by its DSCP. For assured and best-effort traffic, the random early detection (RED) algorithm is first applied to determine whether to drop a particular packet or to queue it. RED is a buffer management scheme, which avoids queue overflow at the router by dropping packets randomly based on their DSCPs. By default, RED only deals with in-packets, however, RED with in and out (RIO) maintains two RED algorithms, one for in-packets and another for out-packets, thus resulting in a more advanced buffer management scheme than simply RED. The RIO has two thresholds for each queue, when the average queue length is below the first threshold, no packets are dropped, when the average queue is between the two thresholds only the out-of-profile packets are randomly dropped. However, when the average queue length exceeds the second threshold, all the out-of-profile packets are dropped, and some of the in-profile may be randomly dropped, in order to avoid network congestion. If assured service is replaced by gold, silver and bronze services for more service classes, the handling of the packets will be exactly the same as assured service, with the exception that instead of having one AQ there will be three queues: gold queue (GQ) for gold packets, silver queue (SQ) for silver packets and bronze queue (BQ) for bronze packets, and there will be decrease in service quality from gold to bronze service (Xiao and Ni, 1999).

For premium service, which most of the time is using UDP, neither RED nor RIO is applied, so all the packets are put into the premium queue. Thus in premium service, no in-profile packet is dropped, so to ensure that the PQ does not overflow, the output rate of the PQ is configured to be significantly higher than the input rate. This will cause PQ to always be short or empty most of the time, thus resulting in very low delay or jitter being experienced by premium packets.

4.5.2.6 Traffic scheduling

The traffic from the three services needs to be scheduled so that the bandwidth-hungry applications of the premium service do not starve the assured and best-effort traffic. Weight fair queuing (WFQ), or a variant of it, can be used to schedule the traffic from the default queue (DQ) of the best-effort service, PQ of the premium service, and AQ of the assured service (Xiao and Ni, 1999; Accellent, 2003).

Basically, there are two methods that can be used to allocate and control bandwidth within a DiffServ domain. One approach is to have users individually decide which service to use. Another approach is to have an agent for each domain to control the resources. This agent or resource controller, which can be a host, a router or a software process on the exit router, is called a bandwidth broker, and it manages the traffic for all hosts within that domain. To accomplish this, the bandwidth broker tracks the current allocation of the traffic to various services and handles new request for service according to organizational policies and the current state of traffic allocation (Xiao and Ni, 1999; Leon-Garcia and Widjaja, 2004, p.721).

Besides service allocation, bandwidth brokers are responsible for setting up of packet classifiers and meters in the boundary or edge routers. The bandwidth broker is also responsible for maintaining bilateral agreements with bandwidth brokers in the neighboring domains so that they can also do resource allocation in their own domains to cater for flows that request service to a destination in a different domain. This bilateral agreement is a viable means of reaching agreement on the exchange of DiffServ traffic across multiple DiffServ domains.

4.6 TRAFFIC ENGINEERING

Uneven network traffic distribution can cause high utilization or congestion in some part of a network, even when total capacity of the network is greater than total demand (Xaio, Telkamp, Fineberg, Chen and Ni, 2002). Uneven distribution may be due to the fact that packets tend to follow the shortest path of route, thus resulting in the shortest path being highly utilized and congested, while a longer path remains under-utilized. This uneven traffic distribution in the core network can result in QoS problems such as packet loss, latency, and jitter increase as the average load on a router rises, even when the network utilization is low. The IntServ and DiffServ QoS schemes, discussed above, do not eliminate traffic congestion, but provides differentiated performance degradation for different traffic during network congestion. Additional to congestion, another problem that can be encountered is route failure, which can result in packet loss and quality of service degradation if not attended to quickly.

Traffic engineering (TE) is the process of controlling how traffic flows through one's network as to optimize resource utilization and network performance (Xiao, Hannan, Bailey and Ni, 2000); that is, the traffic flows are moved from a congested shortest path to a less congested non-shortest path that would not normally be used between a given source-destination pair (Ibe, 2002, p.234). Thus, traffic engineering forms the integral part of providing QoS in IP networks. According to [RFC3272], traffic engineering can be either intra-domain (that is, traffic engineering within a given autonomous system) or inter-domain (that is, originating in one domain and terminating in another).

In this study, the focus will be on aspects pertaining to intra-domain traffic engineering. This will include network capacity management and traffic management; that is, managing network resources such as bandwidth, and controlling routing and traffic in order to minimize congestion and provide fast re-route in the event of route failures so that traffic oriented performances, such as latency, jitter, and packet loss, can be improved. In order to facilitate efficient and reliable network operations while simultaneously optimizing network resource utilization and traffic performance, the IETF has introduced multi-protocol label switching (MPLS), constraint-based routing (CR) and an enhanced link state IGP.

4.6.1 MPLS

MPLS is an IETF specified framework that provides for efficient designation, routing, forwarding, and switching of traffic flows through the network (IEC). It approaches the QoS by attempting to address the shortcomings of the IP routing, which include speed, scalability, QoS management and traffic engineering. In traditional IP networks each router makes an independent forwarding decision on each packet as it traverses the network. This decision usually involves a complex manipulation of a large routing table to determine the next hop of a packet. MPLS also uses IP addresses either Ipv4 or Ipv6, to identify end points and intermediate switches and routers. This makes MPLS networks IP-compatible and easily integrated with traditional IP networks. However, unlike traditional IP, MPLS flows are connection-oriented and packets are routed along pre-configured Label Switched Paths (LSPs). MPLS also slightly modifies the IP packet format to include a new Shim Label header, called the MPLS header, which contains a label field. MPLS works by tagging each packet with an identifier to distinguish the LSPs. This identifier is contained within the label field and it consists of fixed-length value, called label, which is used as index into table, which specify the next hop, and new label (RFC3031). When a packet is received, the MPLS router uses this label, instead of the traditional IP destination address, to deliver the packet along the selected path or LSP, and then looks up the LSP in its own forwarding table to determine the best link over which to forward the packet, and the label to use on this next hop.

The 32-bit MPLS header sits between the layer 2 (data link layer) header and the layer 3 (network layer) header as shown in Figure 4.6.

Layer 2 header (Link layer header)	MPLS header	Layer 3 header (Network layer header)	Packet Data

Label 20 bits	CoS or Exp 3 bits	S 1 bit	TTL 8 bits

Figure 4.6 32-bit MPLS Packet Header

The MPLS header contains four fields: a 20-bit label field, a 3-bit Class of Service (CoS) or Experimental (EXP) field, a 1-bit Stack (S) field, and an 8-bit Time-to-Live (TTL) field. The label field carries the actual value of the MPLS label, which is the index into a smaller table that specifies the next hop for the packet. CoS field is used to define different packet scheduling schemes that can be applied to different packets in the network, and each field permits up to eight different packets with the same label. The Stack field is used to indicate whether or not multiple headers are stacked; that is, it is set to a value 1 when the current label is the last in the stack; otherwise it is set to 0. The TTL field provides the same functionality as the TTL field in a conventional IP packet; that is, the time the packet is allowed to spend in the network (Ibe, 2002, p.230).

4.6.1.1 MPLS Architecture

The basic MPLS domain consists of a set of MPLS capable network nodes, typically routers and switches (Aidarous and Plevyak, 2003, p.127), which are working together to provide the ability to forward packets over arbitrary non-shortest paths, and emulate high-speed tunnels between non-label-switched domains (Armitage, 2000). According to RFC 3031, a MPLS node is a node that is aware of MPLS control protocols, operate one or more layer 3 protocols, and is capable of forwarding packets based on labels. A router, which supports MPLS, is known as a Label Switched Router (LSR). These LSR can either be at the boundary or at the core of a MPLS domain. Boundary node, which is called MPLS edge node, connects the MPLS domain cloud with a node, which is outside of the domain, either because it is not running MPLS, or it is in a different domain. Interior nodes are connected to other nodes (interior and boundary) within the same MPLS domain.

However, LSR is usually reserved for MPLS core or backbone node, and the MPLS edge node or edge LSR is often referred to as or Label Edge Router (LER), instead of LSR. LER can be either an ingress or egress node, depending on the packet flow direction. Where the ingress is the one at the entry of the MPLS domain, while egress is the one at the outer edge, or exit, of the MPLS domain. Figure 4.7 illustrates at typical MPLS network.

Both the boundary and interior network MPLS nodes are used to implement the functional elements that compose the MPLS architecture, such as executing MPLS signaling protocols, and performing routing and forwarding functions. The role of the ingress LSR is to handle the traffic as it enters the MPLS domain, while the role of the egress LSR is to handle the traffic that leaves the MPLS domain.

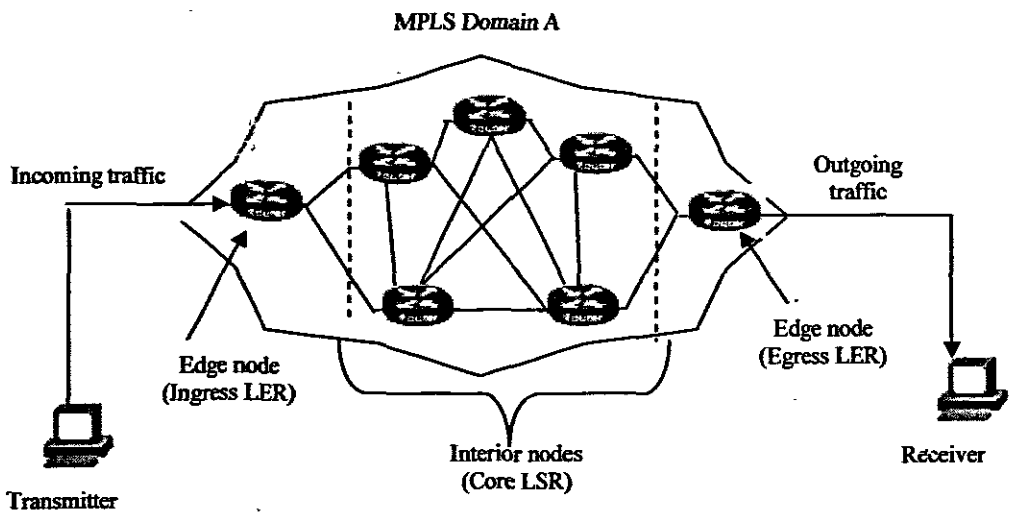


Figure 4.7 Basic MPLS Network Architecture

The ingress LER operates at the edge of the access network and MPLS network. Its basic function is to assign FEC, and label to the incoming packets as they arrive. It also helps in establishment of label switched paths (LSP) for packets within the MPLS domain.
The egress LER's function is to strip labels, which have been allocated to packets and used within the MPLS domain, from the packets that are leaving the MPLS domain.

Core LSRs are optimized for forwarding processing, and are used to efficiently forward labeled packets in order to achieve the best possible performance. They also participate in the establishment of LSPs for packets within the MPLS domain, in conjunction with other LSRs or LERs. As an MPLS-labeled packet arrives at a LSR, the label field inside the MPLS header is used to determine the next hop of each packet by looking up the forwarding table resided in each LSR. Before switching the packet to the next hop, LSR can replace the incoming label by another label (i.e. label swap) or stack a new MPLS header on top of the incoming MPLS header, depending on the forwarding table in each LSR.

4.6.1.2 Forward equivalence class and labels

As stated earlier, the ingress LER labels packets before they enter the core network. These labels are used to partition a group of packets into forward equivalence class (FEC) as they are arriving at the MPLS domain edge. According to EL-Gendy et al., FEC refers to a group of packets that meet the same forwarding behavior over the same path, and an aggregation of flows with the same service class, or FEC, that can be put into a LSP is called a traffic trunk. Packets that have been assigned to the same FEC will receive the same transport treatment at any MPLS node inside the domain. Several criteria can be used to define FECs, including source IP address, destination IP address, and port address.

4.6.1.3 Label Distribution Protocol

In the MPLS, the LSRs must agree on the meaning of labels used to forward traffic between and through them. Label Distribution Protocol (LDP) is a protocol that defines a set of procedures and messages by which one LSR informs another of the label bindings it has made. The LSR uses this protocol to establish label switched paths through the network by mapping network routing information directly to data-link layer switched paths (Aidarous and Plevyak, 2003, p.136). LSRs, which use LDP to exchange label-mapping information is called LDP peers and they have an LDP session between them. The LDP peers exchange the following categories of messages:

- Discovery messages – used to announce and maintain the presence of an LSR in a network
- Session messages – Used to establish, maintain, and terminate LDP sessions between LDP peers
- Advertisement messages – Used to create, change, and delete label mappings for FECs
- Notification messages – Used to provide advisory information and signal error information.

These LDP messages are required to be delivered reliable. Thus, except for discovery messages, all other message types use TCP transport; discovery messages use UDP transport. LDP supports two label distribution methods: downstream unsolicited distribution and downstream on demand distribution. Downstream-unsolicited distribution permits an LSR to advertise label/FEC mappings to its peers when it is ready to forward packets in the FEC, while the downstream on demand distribution permits an LSR to provide label/FEC mappings to a peer in response to a request from a peer for a label for the FEC (Ibe, 2002, p.223). On receipt of label bindings each LSR creates entries in the label information base (LIB), which is a table that represents Label-to-FEC mapping.

4.6.1.4 Label Switched Paths

Label Switched Path (LSP) is a path through the MPLS domain from the ingress LER to the egress LER. Each FEC need an LSP route or path through which it is going to traverse the MPLS domain. These LSPs may have dedicated resources associated with them and can either be explicitly determined or implicitly created. Once LSPs have been established, packets received by MPLS nodes are mapped onto the paths according to the following:

- Traffic requirements
- Information carried in their IP headers, and/or
- Local routing information

4.6.1.5 MPLS Protocol Stack Architecture

According to IEC the core components that constitute MPLS architecture can be broken down into the following parts:

- Network layer (IP) routing protocols
- Edge of network layer forwarding
- Core network label-based switching
- Label semantics and granularity
- Signaling protocol for label distribution
- Traffic engineering
- Compatibility with various layer-2 forwarding paradigms

The following figure depicts the protocols that can be used for MPLS operations.

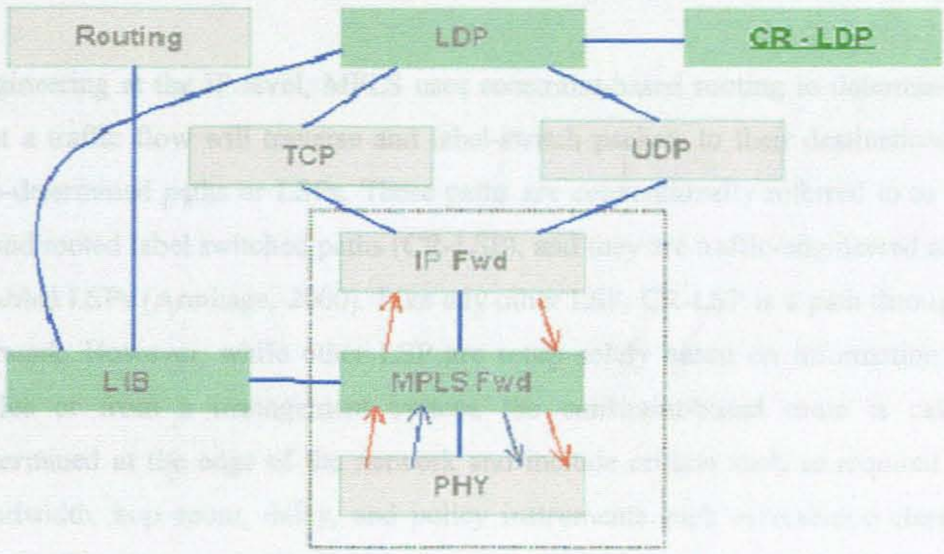


Figure 4.8. Protocols used for MPLS operations

***Next Hop Label Forwarding Entry (NHLFE)*:** The NHLFE is used when forwarding a labeled packet. It contains the outgoing interface (next hop), the data link encapsulation used for the transmitted packets, the outgoing label and the operation (add, replace, or remove) to perform on the label stack.

***Incoming Label Map (ILM)*:** The ILM is a mapping from incoming labels to NHLFEs. It is used when forwarding packets that arrive as labeled packets.

***FEC-to-NHLFE Map (FTN)*:** The FTN is a mapping from FECs to NHLFEs. It is used when forwarding packets that arrive unlabeled, but which are to be labeled before forwarding.

4.6.2 Constraint-based Routing

According to RFC3212, constraint-based routing (CR) is a routing mechanism that supports Traffic Engineering requirements. Label switching alone does not make MPLS to satisfy the traffic engineering requirements. So in order to achieve true traffic engineering at the IP level, MPLS uses constraint-based routing to determine the paths that a traffic flow will traverse and label-switch packets to their destinations using the pre-determined paths or LSPs. These paths are conventionally referred to as constraint-based routed label switched paths (CR-LSP), and they are traffic-engineered and/or QoS-enabled LSPs (Armitage, 2000). Like any other LSP, CR-LSP is a path through a MPLS network. However, while other LSP are setup solely based on information in routing tables or from a management system, the constraint-based route is calculated or determined at the edge of the network and include criteria such as required values for bandwidth, hop count, delay, and policy instruments such as resource class attributes (RFC3272). That is, CR-LSPs are established according to particular criteria or constraints in order to meet specific traffic requirements. The intention is that this functionality shall give desired special characteristics to the LSP in order to better support the traffic sent over the LSP (RFC3212).

MPLS uses either constraint-based routed LDP (CR-LDP) or a Resource Reservation Protocol enhanced to support MPLS traffic engineering (RSVP-TE) as the label distribution protocol for traffic engineering signaling (Ibe, 2002, p.235). CR-LDP is an extension of LDP that is designed to promote constraint-based routing of LSP, while RSVP and extension of RSVP. Both CR-LDP and RSVP-TE enable ingress LER to:

- Trigger and control the establishment of CR-LSP between itself and a remote (egress) LER,
- Strict or loose specify the route to be taken by the established CR-LSP, and
- Specify QoS parameters to be associated with this CR-LSP, leading to specific queuing and scheduling behavior at every hop

The constraints that are used to establish theses CR-LSP in IP networks can be divided into three categories: explicit, implicit, and mixed routing.

4.6.2.1 Explicit routing

Explicit routing (ER) is a method of pre-selecting and specifying the exact path for traffic to flow through the network. The explicit routing information contains a list of nodes or group of nodes along the constraint-based path or tunnel. The benefit of this method is that the exact flow of the traffic through the network can be determined rather than relying on current interior gateway protocols, such as Routing information Protocol (RIP), Intermediate System-Intermediate System (IS-IS) and Open Shortest Path First (OSPF), which tend to send all the traffic across shortest common links while leaving other links under-utilized. CR explicit paths can be strict or loose. Strict paths specify an exact physical path, including every physical node, while loose paths include hops that have local flexibility. The ingress LER does constraint- based route computation in order to automatically compute explicit routes used for traffic originating from that LER. However, explicit routing tends to be static in nature.

4.6.2.2 Implicit routing

Implicit routing is a routing method that route traffic through the MPLS domain based on the forwarding requirements of the traffic. Unlike the explicit routing, the implicit routing process does not specify paths through the network, but it selects a path through the network that is capable of satisfying the specific forwarding requirements of that particular traffic and dedicates the resources along the selected path for that traffic. Implicit routing offers a dynamic method to traffic engineering a network by calculating the paths using automated process.

4.6.2.3 Mixed routing

Alternatively, a combination of both explicit and implicit routing can be used to establish CR-LSP. For, example, the LSR that traffic should not flow through may be explicitly specified, and then rely on the implicit routing to determine the best LSP that satisfies the constraints (Yip, 2002).

4.6.3 Enhanced Link State Interior Gateway Protocols

In order to calculate CR-LSPs, the routing protocol is required to distribute network topology, and QoS information through the network. An interior gateway protocol, such as OSPF or IS-IS, is normally used. However, these routing protocols only distribute network topology. So to cater for traffic engineering and establishment of LSPs with guaranteed QoS characteristics (CR-LSPs) and backup LSPs that avoid any single point of failure, Enhanced link state IGPs, which has the traffic engineering extensions to these protocols are used. These extensions distribute QoS and Shared Risk Link Groups (SRLGs) information to enable the router calculator to determine routes through the network with guaranteed QoS parameters and backup LSPs that transverse different links from the primary path (Data Connection, 2004).

According to RFC3272, these enhanced link state IGPs are used to propagate link attributes so that constraint-based routing can be able to recognize 'reachability' and compute CR-LSPs. Unlike ordinary IGP, the enhanced link state IGP floods information more frequently. This is done to track any changes in the link attributes, such as topology, bandwidth and link affinity

4.6.4 Network Robustness and Resiliency

Links and/or routers failures cause route failures that can result in packet loss, which can compromises the quality of service. So as part of traffic engineering the network must be able to dynamically adapt to new conditions in order to keep network utilization optimized and make network resilient. One possible solution for a fault-tolerant system is to invest in redundancy, so that MPLS constraint-based routing can be able to establish a 'protection' LSP around the 'working' or operational LSP, which can be used as an alternative route in the event of any link/s or router/s along the working LSP fail. Redirecting traffic onto these alternative paths during route failures can either be pre-planned or done automatically in the time of the failure condition. The first method, preplanning, is usually referred to as protection while the second method is referred to as restoration. Protection is proactive whereas restoration is reactive. The problem with reactive method is the interruption time, while IGPs are still determining an alternative optimal route, can be unacceptable to real-time traffic.

According to RFC3272, MPLS has four types of protection against route failure, which can be categorized as link protection, node protection, path protection, and segment protection.

Link protection: It protects an LSP from a given link failure by disjointing the path of the protection or backup LSP from the path of operational LSP at a particular link over which protection is required. Then when the protected link fails, traffic on the working LSP is switched over to the protection LSP at the head-end of the failed link.

Node protection: It protects an LSP from given node failure by disjointing the path of the protection LSP at a particular node to be protected. The secondary path is also disjoint from the primary path at all links associated with the node to be protected. When the node fails, traffic on the working LSP is switched over to the protection LSP at upstream LSR directly connected to the failed node.

Path protection: It protects an LSP from failure at any point along its routed path by completely disjointing the path of the protection LSP from the working path; that is, creating a backup LSP between the same ingress-egress LER pair. The advantage of path protection is that it protects the working LSP from all possible link and node failures within the core or backbone network.

Segment protection: This type of protection involves the partitioning of MPLS domain into multiple protection domains whereby a failure in a protection domain is rectified within that domain. In the case where an LSP transverses multiple protection domains, a protection mechanism within a domain only needs to protect the segment of the LSP that lies within the domain.

Alternatively, fast reroute can be used. Fast reroute is to temporarily repair the affected path so that it can continue to carry traffic before a more optimal alternative path is computed. This is to avoid large amounts of data being buffered for a long time or being dropped while IGPs are propagating information and re-computing alternative optimal paths.

4.7 IN SUMMARY

In this chapter, the concept of converged networking using IP was reviewed. The challenges of using IP for real time traffic were highlighted and possible solutions. As part of the solution the two models that can be used for providing quality of service in IP-based networks were presented. These two models are integrated services (IntServ) and differentiated services (DiffServ), where the IntServ provides per flow QoS while DiffServ provides QoS to aggregated traffic. The advantage of providing QoS to aggregate traffic instead of individual flows is that the amount of signaling required in the network is minimized. This together with the moving of complex processing from the core to the edge makes DiffServ to be more scalable than IntServ. DiffServ provides differentiated services for different classes by using mechanisms that include packet classification, policing, class-based queuing and scheduling, and Random Early Detection (RED).

To complement the two models and provide a holistic approach to provision of QoS in converged IP based networks, traffic engineering using MPLS was also presented. Other mechanisms that are used in collaboration with MPLS to provide traffic engineering were also presented. These include constraint-based routing, LSP path signaling and enhanced link state IGPs. MPLS together with constraint-based routing, LSP signaling, and enhanced link state IGPs work together to provide traffic engineering, which ensures network optimization and congestion avoidance. Lastly, this chapter also addresses the question of network robustness and resiliency that is needed in order to avoid packet loss, which can compromise QoS during link and/or router failure.

To conclude the chapter, the next chapter is introduced, which is about digital video and how it can be transmitted using IP-based multiservice network without compromising its quality.

CHAPTER 5

5. QUALITY VIDEO TRANSMISSION USING IP-BASED MULTISERVICE NETWORK

5.1 INTRODUCTION

In the previous chapter it was highlighted that with converged or multiservice networks it is possible to transmit video together with voice and data within one network. This chapter is about quality video transmission over IP-based network. Firstly, the author introduces video applications using IP-based networks and their use in telemedicine. Secondly, the concept of quality video is introduced. Thirdly, digitised video and its transmission is discussed. This chapter will also look at how the QoS and traffic engineering mechanisms introduced in the previous chapter can be utilized to provide solutions for quality video transmission.

The application, which involves transmission of video over the IP-based network, is called video over IP. This video over IP includes the following services:

- Video on demand (VoD). The VoD is an interactive service, which involves the storage of digital information in large databases. This information includes centrally stored films, programs, archival, and educational learning video material. The user can access the stored information and play back the video with options to stop, start, fast-forward, or rewind (Siemon, 2005).

- Video conferencing (VC). VC is a full duplex, real-time video and audio communication, which enables people in different locations to have a face-to-face like conversation. The VC consists of cameras on both endpoints to capture the video, microphones for capturing sound, monitor or screen for displaying the picture, and speakers for playing sound on both endpoints. VC can be one user to one user (point-to-point), or multiple-users participating in the same session (multipoint). VC can also include an electronic whiteboard in the conference to allow users to write notes on the same board and/or view each other's presentations and notes while speaking (Siemon, 2005).

- Video broadcast. Video broadcast over IP involves a simplex transmission of a video file content. The broadcast video signal can either be replicated by the server for each user (i.e. Unicast), or the server can send the same signal to multiple users at the same time (i.e. Multicast) (Siemon, 2005).

All of the above video applications can be integrated into healthcare system to provide services such as tele-consultation, tele-examination, training, e-learning, continued medical education (CME), etc. Video on demand can allow the health professional to access computer-based information relevant to the problem of the patient being attended. Video conferencing can be used for tele-consultation, and tele-examination. Video broadcast can be used as means to distribute training, presentations, meeting minutes, and it can also be utilized for continual medical education.

5.2 QUALITY VIDEO TRANSMISSION

Video quality depends on four variants: coding, transmission, recording and display. Each of these variants has some limitations, which makes it impossible to achieve ideal quality transmission of a video signal. However, in most cases a compromise must be reached between the ideal and the practical quality video transmission.

If we accept that a practical system provides video at a particular level of quality, then this quality level should be ideally constant from frame to frame. Secondly, the quality of the displayed video sequence should be independent of the visual content of the sequence, or the current state and the utilisation of the network. Thirdly, the decoded video sequence should not show any quality degradation due to errors or packet losses in the communication system.

5.3 DIGITISED VIDEO

Digitised video enables the storage, processing and the transmission of video information in a digital form. Digital video presentation has a number of advantages over traditional analog video, which include compression to minimize its size, integration into computer applications and system to make it possible to create interactive applications, where the user is no longer a passive observer but has the opportunity to interact with the video information. Video over IP is one of these applications, which integrates video and computer. It consists of four stages: capturing, digitising, streaming, and managing over IP networks of video signals (Jesshope and Liu, 2001; Siemon, 2005). Digitised video has an inherently high bandwidth; that is, it requires a very high data rate for transmission (Riley and Richardson, 1997, p.15). To minimize the bandwidth requirements, the digitised video data needs to be compressed by encoding it into a smaller number of bits (Jesshope and Liu, 2001).

5.3.1 Video Compression and Coding

Digitised video clips or signals require too many bits to represent them, which means that to store or transmit them a large amount of space or a large bandwidth is also required. For example, a video signal of a small image can require a bit rate of 60 Mbps to transmit while high definition television (HDTV) requires about 8000 Mbps (Walrand, 1998, p.258), and most of the systems do not have such a capacity. Video compression and video coding technology is utilized in video application to compress the digitised video signal so that its bandwidth consumption is reduced. Most compression algorithms exploit the fact that there is a lot of redundancy within digital video information in order to achieve a substantial reduction in bit rate. A still image, or a single frame within a video sequence, contains a significant amount of spatial redundancy, which makes it possible to represent or encode the information in a more compact form that eliminates some form of redundancy. Similarly, a moving video sequence contains temporal redundancy; that is, successive frames of video are usually similar, which makes it possible to achieve further compression by sending only the parts of the image that have changed from the previous frame (Riley and Richardson, 1997, p.16).

There are various standards that are available for encoding video signals so as to make it possible to store, transmit, and manipulate video signals. The main standards for still image and video coding include the Joint Photographic Experts Group (JPEG) for compressing still images or individual frames of video, H.261 for motion video coding for videoconferencing and video telephony applications at bit rates that are provided by ISDN, H.263 for video coding and compression for very low bit rate applications (less than 64 kbps), and ISO/IEC Motion Picture Experts Group (MPEG) standards for video coding for entertainment, broadcast and integrated video communication. Table 5.1 shows some of the video compression algorithms or standards together with their bit rates and their applications. The evolution of the video coding standards of both ITU-T and ISO's MPEG is shown in the chart of Figure 5.1.

Table 0-1Table 5.1 Video compression algorithms or standards

Algorithm	Bit Rate	Application
H.261	p×64 kbps	**ISDN Video Conferencing and Video telephony**
H.263	≤ 64 kbps	Video Conferencing
MPEG 1	1.5 Mbps	Direct Broadcast Satellite (DBS), Community Antenna Television (CATV)
MPEG 2/ H.262	10 Mbps and more	DBS, CATV, HDTV, Digital Video Cassette (DVC), and Digital Video Disc (DVD).
MPEG 4	64 kbps	Integrated Video Communication or Multimedia
H.264/MPEG- 4/AVC	≥32 kbps	HDTV, Portable game console, mobile video services, video on solid-state camcorders, instant video messaging on cell phones, video conferencing

The Digital Imaging and Communication in Medicine (DICOM) committee has adopted various JPEG variants, such as lossy and lossless JPEG, JPEG-LS, and JPEG 2000. Since most of the still images in telemedicine are medical images, which are having a diagnostic use, medical image compression techniques have primarily focused on lossless methods, where the signal can be reconstructed exactly from its uncompressed format (Pattichis, Kyriacou, Voskarides, Pattichis, Istepanian and Schizas, 2002).

For digital video, the DICOM committee has adopted MPEG 2 for diagnosis video. However, due to its high bandwidth requirements and frame synchronization problem, MPEG 4 standard can be used for real-time diagnosis video and for non-diagnosis video applications such as video conferencing, H.263 may be acceptable (Pattichis, Kyriacou, Voskarides, Pattichis, Istepanian and Schizas, 2002).

H.264 can also be another possible candidate since it can be used for video conferencing, video storage and video broadcast or streaming. The evolution of these video coding standards is as follows:

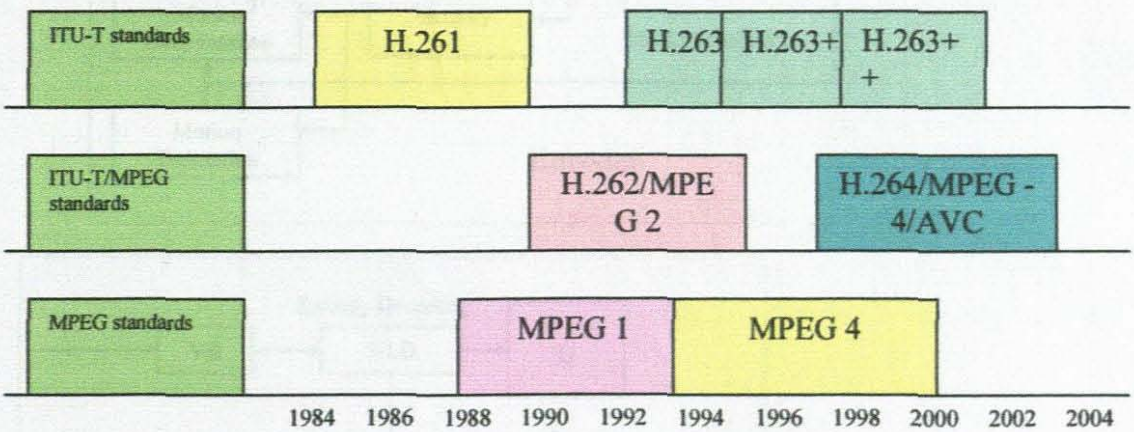
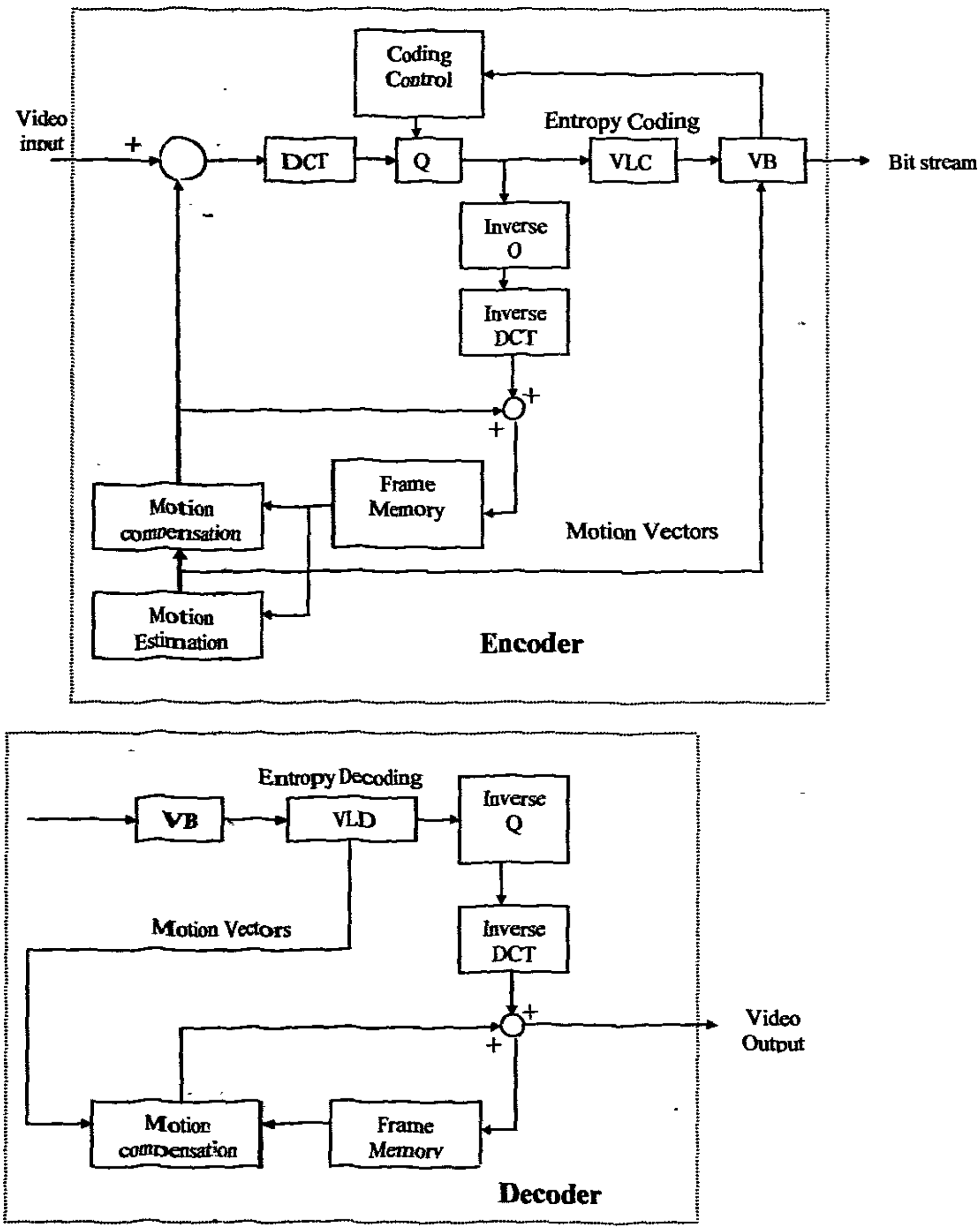


Figure 5.1 Evolution of the video coding standards of both ITU-T and ISO's MPEG

Today's MPEG and ITU's H.26x video coding algorithms implement a hybrid block-based motion estimation and compensation/discrete cosine transform (MC/DCT) coding strategies. MC/DCT technology provides a significant compression gain compared to pure INTRA frame DCT coding (JPEG) for video compression. The DCT is applied in these standards strictly as block-based approach usually on blocks of sizes 8 x 8 pixels (Cox *et al.*, 1998; Girod and Färber, 1999; Sikora, 2005). The block diagram of a basic hybrid MC/DCT encoder and decoder are depicted in Figure 5.2, and following that is the summary of the some of these video coding standards, which are designed for low bit rates.



Where DCT: Discrete Cosine Transform, Q – Quantization, VLC – Variable Length Coding, VB – Video Buffer, VLD – Variable length Decoding

Figure 5.2. Block diagram of a basic hybrid MC/DCT encoder and decoder

5.3.1.1 H.263

H.263 is a low bit rate video coding standard for videoconferencing, which has been adopted in several network transport standard such as ITU-T H.324 (PSTN), H.320 (ISDN), H.310 (BISDN), H.323 (packet-based networks), and 3GPP. It was developed by ITU-T Video Coding Experts Group (VCEG) as an evolutionary improvement based on experience from H.261, MPEG 1 and MPEG 2 standards. Version 2 of the H.263 standard (H.263+) was introduced to provide additional features that improve video quality, increase robustness and functionality of the target video system, and version 3 of the H.263 standard (H.263++) was introduced to add more features such as enhanced coding efficiency for low delay applications, enhanced error resilience for wireless video communications and support for interlaced video sources (UB Video, 2002).
All versions of H.263 supports five resolutions or standardized picture formats: sub-QCIF (SQCIF) (88x72), QCIF (176x144), CIF (352x288), 4CIF (704x576), and 16CIF (1408x1152), where SQCIF is approximately half the resolution of QCIF, and 4CIF and 16 CIF are 4 and 16 times the resolution of CIF respectively. Each picture in the input video sequence is divided into macro blocks, consisting of four luminance blocks of 8 pixels x 8 lines followed by one chrominance C_b block and one chrominance C_r block, each consisting of 8 pixels x 8 lines. A group of blocks is defined as an integer number of macro block rows, a number that is independent on picture resolution.

H.263 provides a compact representation of the information in the video frame by removing spatial redundancies that exist within the frames, and also temporal redundancies that exist between successive frames. Discrete Cosine Transform (DCT) is used to remove spatial redundancies, and motion estimation and compensation are used to remove temporal redundancies.

5.3.1.2 MPEG 4

MPEG 4 is an ISO/IEC standard, which provides a broad framework for the joint description, compression, storage, and transmission of natural and synthetic (computer generated) audio-visual data. It defines improved compression algorithms for audio and video signals and efficient object-based representation of audio-video scenes (Liu, Wei, and Zarki, 2001).

MPEG 4 absorbs many of the features of MPEG 1, MPEG 2, H.263 and other related standards. Like H.263 it is designed for low bit rate, however, it adds advanced error detection and correction services on top of H.263, which makes it robust to errors. In fact it is designed to deliver DVD (MPEG 2) quality video at lower data rates and smaller file sizes, and it scales to transport media at any data rate; that is, from media suitable for delivery over dial-up modems to high-bandwidth networks. MPEG 4 allows the hybrid coding of pixel-based images and video together with synthetic scenes. MPEG 4 Visual supports the following formats and bit rates:

- Bit rates: typically between 5 kbps and more than 1 Gbps
- Formats: progressive as well as interlaced video
- Resolutions: typically from sub-QCIF to 'Studio' resolutions (4k x 4k pixels)

The basic coding structure used in MPEG 4 involves shape coding (for arbitrarily shaped video objects) and motion compensation as well as DCT-based texture coding using standard 8x8 DCT or shape adaptive DCT (Koenen, 2002)

5.3.1.3 H.264

H.264 was developed by Joint Video Team (JVT), which is a joint venture between ISO's MPEG and ITU-T's VCEG. The official title of the H.264 standard is Advanced Video Coding (AVC); however, it is widely known as by its old working title, H.26L, and by its ITU document number, H.264 or as ISO MPEG 4 Part 10, or just MPEG AVC.

The main objective of H.264 standard is to provide means to achieve substantially higher video quality as compared to what could be achieved using any of the existing video coding standard. It has a number of advantages that distinguish it from existing standards, while at the same time, sharing common features other existing standards. These advantages include up to 50% in bit rate saving compared to H.263 or MPEG 4, high quality video that is consistent at high and low bit rates, error resilience, and network friendliness that makes it possible to transport H.264 bit streams over different networks. These advantages make H.264 an ideal standard for several video applications such as video conferencing, storage and broadcast video.

The underlying approach is similar to that adopted in H.263 and MPEG 4; that is, dividing each video frame into blocks of pixels and providing a compact representation of the information in the video frame by removing spatial redundancies that exist within the frames, and also temporal redundancies that exist between successive frames.

5.4 DIGITAL VIDEO COMMUNICATIONS SYSTEM AND FACTORS AFFECTING QUALITY TRANSMISSION

Figure 5.3 shows a functional block diagram for a video communications system.

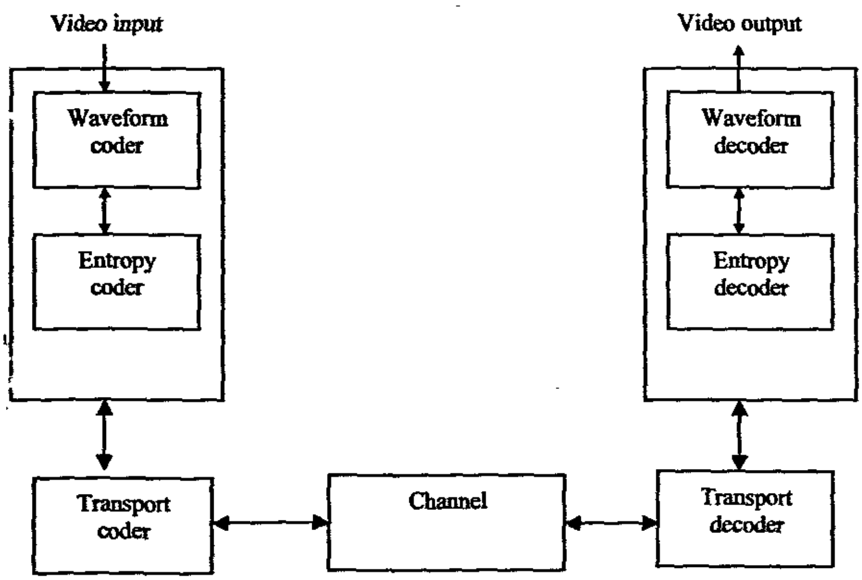


Figure 5.3 A functional block diagram of a video communications system

In order to transmit video over packet networks, the input video must be compressed to save bandwidth using the source encoder, channel coded, packetized and/or modulated;

that is, converted into data units suitable for transmission using the transport coder. At the receiver side, the inverse operations are performed in order to reconstruct and display the video signal (Wang and Zhu, 1998).

Digital video transmission has a number of demanding requirements such as high transmission bandwidth, minimum delay, minimum jitter, and low error rate. For other applications not all of these requirements are critical, however, for delivery of real-time video, data transmission rate, transmission delay, transmission delay variations, bit errors, and packet losses have all significant effects on the quality of the video transmission (Wu, Hou and Zhang, 2001).

As mentioned before in the previous chapter, these requirements are parameters that determine the quality of service. The following discussion will consider the definition of these parameters, the factors in a network that influence them, and the possibilities for managing and controlling them.

5.4.1 Transmission Bandwidth

Bandwidth represents the amount of capacity the network has to guarantee for the application on an end-to-end basis; that is, it expresses the data rate that the network is suppose to guarantee for the application (Aidarous and Plevyak, 2003, p.68). According to Wu, Hou and Zhang a minimum bandwidth of 28 kbps is required in order to achieve acceptable presentation quality for transmitted real-time video (Wu, Hou and Zhang, 2000; 2001). For networks that cover a wide area, bandwidth requirements could be a point of concern, due to the fact that such types of networks use routers, and during periods of high demand, these routers may become a bottleneck, since the available bandwidth for their interconnections may not be necessary dimensioned to guarantee the flow of packets from all applications simultaneously at all times. In order to guarantee the minimum bandwidth required, the processing in the routers must be able to reserve a certain bandwidth along the route for video traffic, and failure to provide this minimum bandwidth for real-time video traffic can result in an impaired video quality.

5.4.2 Delay

Delay (or latency) is the time that elapses from the instant a packet is transmitted at the source until it is received at the destination (Ibe, 2002, p.159). This includes delays introduced by network equipment (hubs, switches, and routers); delays introduced by service providers (carriers), propagation delays in the communication medium (wireless, transmission line, and optical fiber), and the network stack processing delay at the hosts (Aidarous and Plevyak, 2003, p.69). The sum of all these delays is called end-to-end delay, and for real-time video this end-to-end delay must be bounded (Wu, Hou, and Zhang, 2000).

5.4.3 Delay Variation (or Jitter)

Jitter is the variation in the interval times at the destination of all packets belonging to the same data stream (Ibe, 2002, p.161; Aidarous and Plevyak, 2003, p.73). Factors that influence variability of delay include variable processing time at intermediate equipment, congestion, and other factors related to network operation. Jitter may also cause packets to be received out of order. To correct the problem of packets received out of order in real-time applications, which most of the time are UDP-based, a high level protocol like real-time protocol, or procedure must be used. Additional to controlling the jitter, another mechanism to guarantee that packets will be received in a precise time windows for applications needs to be implemented. This is accomplished by creating a buffer, called jitter buffer, which will keep incoming packets for processing in such a way that the receiver will always have an incoming packet for processing (Aidarous and Plevyak, 2003, p.73). This process is called dejitter buffering (Ibe, 2002, p.161). Figure 5.4 depicts the effects of jitter in the transmission of packets through the network.

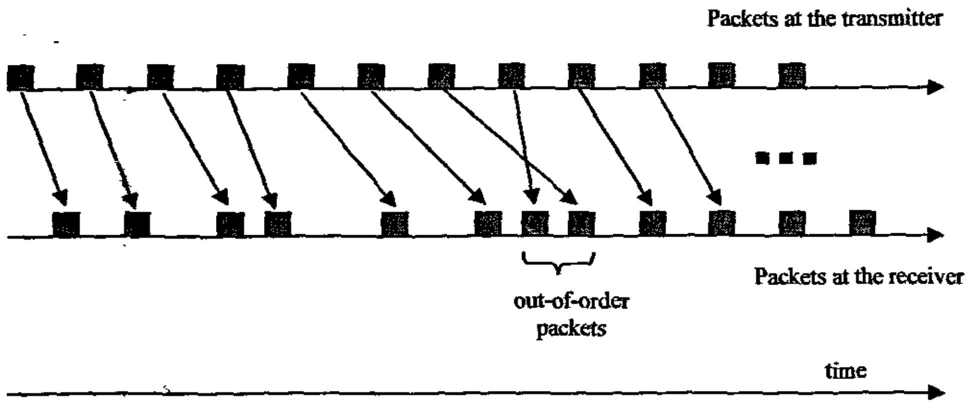


Figure 5.4 Jitter effect on transmitted packets

5.4.4 Errors and Losses

Practical communication networks introduce errors and losses into transmitted data. Errors are mostly due to noise in the communication network, and bit error rate or bit error ratio (BER) describes the mean rate at which bit errors occur. BER depends mostly on the characteristics of the transmission or communication channel. For data, a BER of about 10^{-5} is generally considered to be acceptable for data, however, coded video data is highly sensitive to transmission errors and requires a much lower probability of error than 10^{-5} for acceptable video quality at the receiver (Riley and Richardson, 1997, p.108).

Packet loss occurs in networks mainly due to packet dropping. If a video packet is delayed too long such that its time slot for decoding and display is past, it is treated as a lost packet. Packet loss can also occur if the capacity of a node (especially the router) within the network is exceeded due to congestion or limited resources (Riley and Richardson, 1997, p.109; Aidarous and Plevyak, 2003, p.74). Loss of packet can potentially make the presentation displeasing to human eyes, or, in some cases, make the presentation impossible. This implies that, in order to achieve acceptable visual quality, the packet loss is required to be kept low (Wu, Hou, and Zhang, 2000).

Compressed video is very sensitive to errors because of the use of predictive coding and variable length coding (VLC) by the source coder. Due to the use of spatio-temporal prediction, a single erroneously recovered sample can lead to errors in the following samples in the same and the following frames; that is, error propagation. Likewise, because of the use of VLC, a single bit error can cause the decoder to lose synchronization, so that even correctly received following bit become useless (Wang, Wenger, Wen and Katsaggelos, 2000). To make sure that transmission errors and packet losses do not spoil the quality of the transmitted video, mechanisms must be employed to make the bit stream to be resilient to transmission errors, and to make sure that network congestion that causes packet loss is under control.

5.5 APPROACHES FOR QUALITY VIDEO TRANSMISSION

The factors discussed above make the packet-based network to be an unpredictable communication system with parameters that are variable according to network usage. In order to overcome the challenges posed by such a system, which exhibit wide variability, and be able to provide quality video transmission, the following are suggested:

- Video encoding scheme that will produce a low bit rate without compromising much on quality
- Video delivery or distribution method that will utilize the bandwidth efficiently
- Error control and concealment
- Scalable coding to address heterogeneity both at network and receiver level
- Make the video source to be able to adapt to network conditions to address bandwidth fluctuations and congestion
- Prioritise and protect video traffic

5.5.1 Video Encoding for Low Bit Rate

In section 5.4 various video encoding schemes were mentioned and out of these schemes it was discussed that MPEG 4, H.263, and H.264 have an ability to produce low bit rates;

that is, they have minimum bandwidth requirements. It was also mentioned that, out of the three standards, H.264 has better performance, since. H.264 has up to 50% in bit rate saving compared to H.263 or MPEG 4, high quality video that is consistent at high and low bit rates, error resilience, and network friendliness that makes it possible to transport H.264 bit streams over different networks. According to Sikora (2005), H.264 is presently the most advanced standard in terms of compression efficiency. These advantages make H.264 an ideal standard for several video applications such as video conferencing, storage and broadcast video.

5.5.2 Video Delivery or Distribution

To deliver or distribute video there are two methods that are commonly used: unicast and multicast. The unicast delivery uses point-to-point transmission, where only one sender and one receiver is involved. Figure 5.5 shows unicast video distribution to multiple receivers using multiple point-to-point connections.

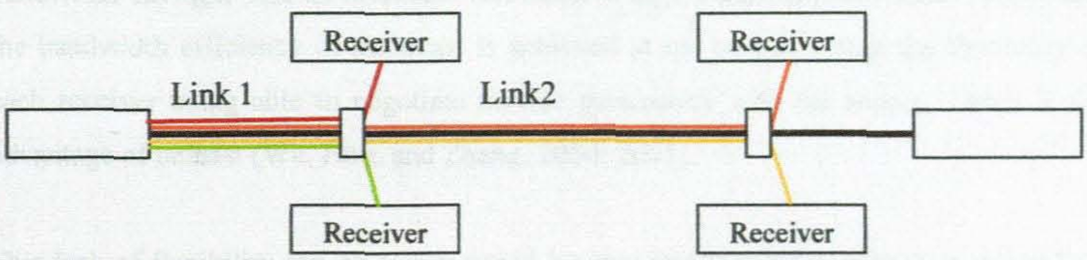


Figure 5.5 Unicast Video Distribution (1 sender to 5 receivers)

From Figure 5.5, it can be seen that, for unicast, five copies of the same video content must be sent out to five different receivers; that is, there must be five point-to-point connections carrying the same content. Since some of the flow has to traverse the same links; for example, in Figure 5.5, all five copies of the same video content flow across Link 1 and three copies flow across Link 2, there is more bandwidth that required for Link 2, which is five times the bandwidth required for each video content.

Multicast delivery method, shown in Figure 5.6, on the other hand, uses point-to-multipoint transmission, where one sender and multiple receivers are involved. This enables sharing some of the links that are common, thus removing replication.

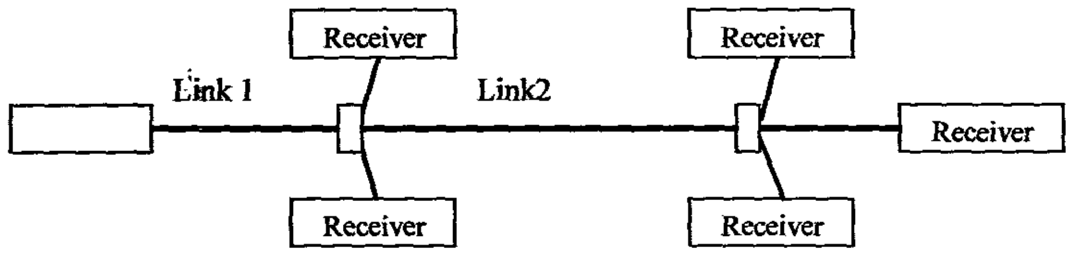


Figure 5.6 Multicast Video Distribution (1 sender to 5 receivers)

From Figure 5.6, it can be seen that there is only one copy of the video content traversing any link, which means the bandwidth required at Link 1 and Link 2 to transmit this video content has been reduced to one-fifth and one-third, respectively, compared to that required for unicast in Figure 5.5. The sharing of links in multicast results in a substantial bandwidth savings; that is, multicast can achieve high bandwidth efficiency. However, the bandwidth efficiency of multicast is achieved at the cost of losing the flexibility of each receiver being able to negotiate service parameters with the source, which is the advantage of unicast (Wu, Hou, and Zhang, 2000; 2001).

This lack of flexibility can be compensated by ensuring that heterogeneity is minimised both at network and receiver level. Removing network heterogeneity and receiver heterogeneity will ensure that all the users have the same requirements and there would be no need for individual receiver to individually negotiate service parameters with the source. Network heterogeneity can be minimized or removed by making sure that the network nodes such as routers and switches in the network can provide quality of service support to guarantee bandwidth, delay, jitter, and packet loss for video applications.

5.5.3 Error Control and Concealment

One of the inherent problems with any communication channel is the introduction of errors to the data being transmitted through it. According to Wang and Zhu (1998) transmission errors can be roughly classified into two categories: random bit errors and erasure errors. Random bit errors are caused by the imperfections of physical channels, which result in bit inversion, bit insertion, and bit deletion. Erasure errors, on the other hand, can be caused by packet loss, or system failures for a short time (Wang and Zhu, 1998). These errors of transmission channel can be effectively corrected by channel coding methods such as forward error correction (FEC) and automatic repeat request (ARQ) or a mixture of them (Sullivan and Wiegand, 2005). Both FEC and ARQ channel coding methods improves the reliability of the information being transferred by adding additional bits to the transmitted data stream, which of course increases the amount of data being sent (Bateman, 1999, pp.173-175). These two traditional methods do not provide the bounded delay which is required for real-time traffic as a result other types of error-control mechanisms such as error resilience and error concealment have be proposed in addition to ARQ and FEC schemes (Wu, Hou, and Zhang, 2000; Stanković, Hamzaoui and Xiong, 2003). Most of these schemes add an amount of redundancy at the waveform-, entropy-, or transport-coder level (of Figure 5.3), which is referred to as concealment redundancy (Wang and Zhu, 1998). FEC, ARQ and error resilience are performed at both the source and the receiver side, while error concealment is carried out only at the receiver side. The location of each error control scheme is depicted in Figure 5.7.

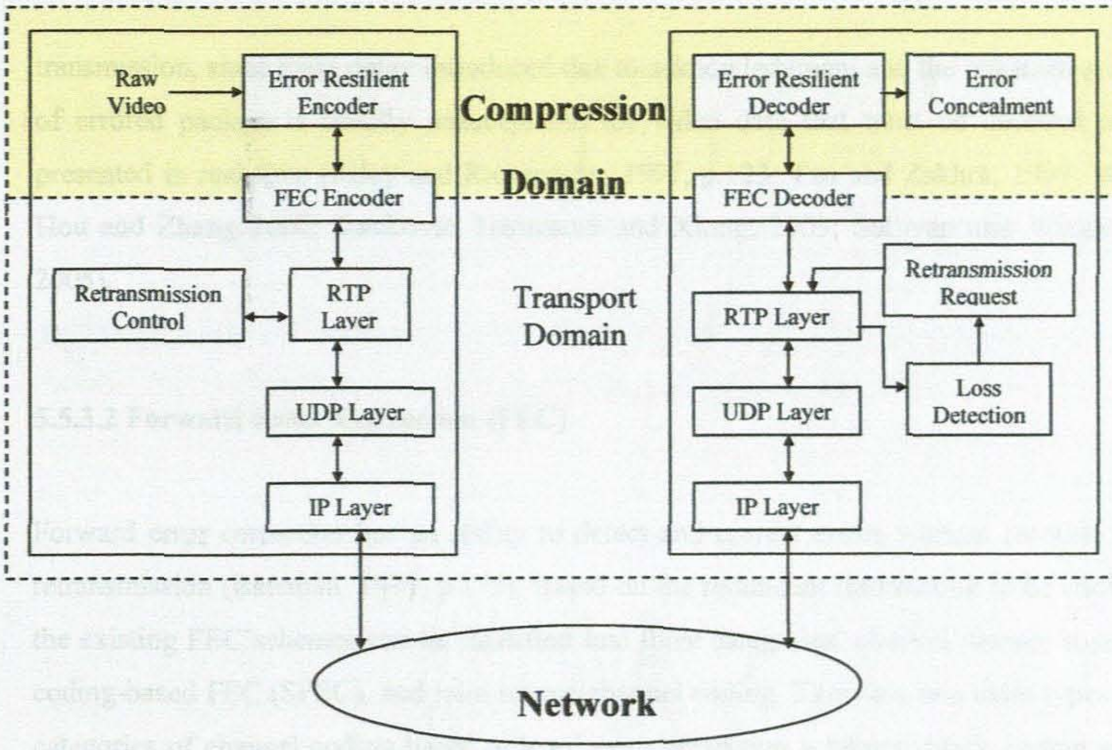


Figure 5.7 Architecture for Error Control Mechanisms

Figure 5.7 shows that ARQ, which is also referred to as retransmission, deals with error *control from transport perspective; error resilience and error concealment approach error* control from compression perspective; and FEC falls in both transport and compression domains (Wu, Hou, and Zhang, 2000).

5.5.3.1 Automatic Repeat Request (ARQ) or Retransmission

Automatic repeat request, or retransmission, is the most elementary form of error correction, which transmit additional bits to the data flow and use those bits at the receiver to determine whether an error has occurred or not. In the event of the error being detected, a request for repeat transmission is sent to the transmitter. However, the feedback error control method used in ARQ is not suitable for real-time video transmission, since extra delay introduced due to acknowledgment and the retransmission of errored packets is usually unacceptable for video data that must be decoded and presented in real time (Riley and Richardson, 1997, p.123; Tan and Zakhor, 1999; Wu, Hou and Zhang,2000; Stanković, Hamzaoui and Xiong, 2003; Sullivan and Wiegand, 2005).

5.5.3.2 Forward Error Correction (FEC)

Forward error correction has an ability to detect and correct errors without recourse to retransmission (Bateman, 1999, p.173). Based on the redundant information to be added, the existing FEC schemes can be classified into three categories: channel coding; source coding-based FEC (SFEC), and joint source/channel coding. There are two main types or categories of channel coding-based forward error correction schemes: block coding and convolutional coding (Riley and Richardson, 1997, p.132; Bateman, 1999, p.175). Both types of these FEC schemes append additional redundant parity data to the original data prior to transmission. If some of the traffic is lost in transit, an FEC scheme allows the receiver to use the additional redundant parity data to detect and recover the lost data without retransmissions (Riley and Richardson, 1997, p.132).

FEC has been successfully used in some applications like broadcasting to control the error rate. This reduction in error rate comes at the expense of increased bandwidth. Another shortcoming of FEC is that it can be ineffective when bursty packet losses or errors, which exceed the FEC codes recovery capability, occur (Wu, Hou and Zhang 2000; Kumar, Xu, Mandal and Pachanathan, 2006).

5.5.3.3 Error Resilience

Error resilient schemes deal with packet loss on the compression layer by considering the semantic meaning of the compression layer and attempt to limit the scope of damage caused by the packet loss on the compression layer. The standard error resilient schemes are resynchronization, data partitioning, and data recovery (Arankalle, 2003; Wu, Hou and Zhang 2000).

Resynchronization

According to Talluri (1998), if the received bitstream is having some errors, a video decoder that is decoding that corrupted bitstream will lose synchronization with the encoder; that is, it will be unable to identify the precise location in the image where the current data belongs. This, if no remedial measures are taken, will cause the quality of the decoded video to rapidly degrade and become totally unusable (Talluri, 1998).

In order to counter this, the encoder must introduce resynchronization markers in the video bitstream at various locations (e.g. beginning of the frame in MPEG –1/2, H.261/263, or periodically in MPEG 4) so that when the decoder detects an error, it can hunt for this resynchronization marker to regain synchronization with the encoder, thereby preventing error propagation across different segments of the video bitstream, which are separated by the markers. These markers are designed such that they can be easily distinguished from all other codewords and a small perturbation of these codewords (Talluri, 1998; Villasenor, Zhang, and Wen, 1999; Wang, Wenger, Wen and Katsaggelos, 2000; Du, Maeder and Moody, 2003). This is shown in Figure 5.8, below.

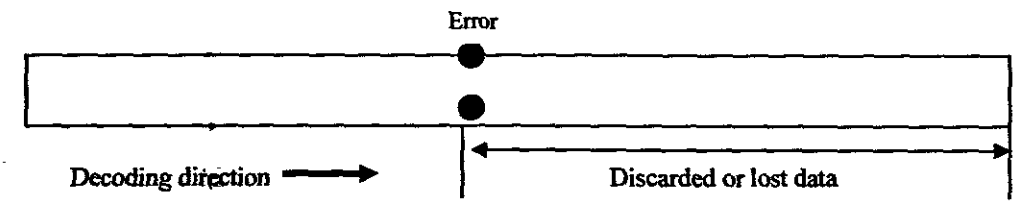


Figure 5.8(a) Decoding video bit stream without resynchronization markers

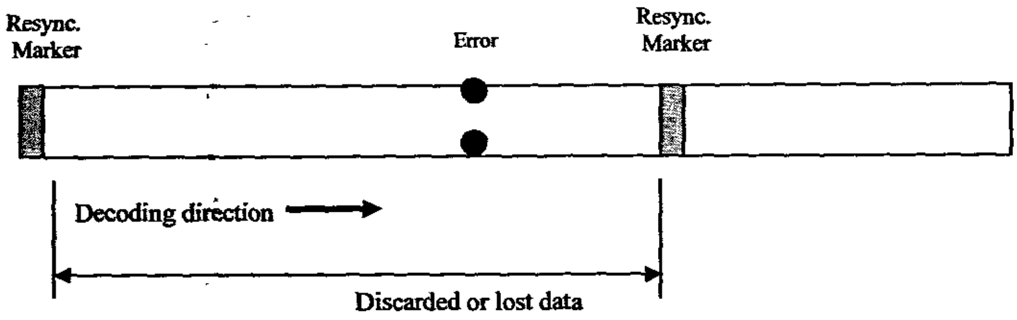


Figure 5.8(b) Decoding video bit stream with resynchronization markers

Data Partitioning

Some bits in the encoded video bitstream are more important than others within the same bitstream. For example, DCT coefficients are less important than GOB headers, motion vectors, and Macroblock information bits. Hence an error occurred in the received DCT information has less impact on the decoded quality than bit errors received in group of block (GOB) headers, motion vectors, and Macroblock information bits. Because of this unequal data importance within the encoded video bitstream, it is possible to separate the bitstream into two layers of importance to allow each to be protected according to its level of importance; that is, unequal error protection (UEP). This technique, which involves segmenting the bit stream into segments, is called data partitioning (Katsaggelos, Ishtiaq, Kondi, Hong, Banham and Brailean, 1998). It helps during the decoding to prevent the errors occurring in one segment from affecting other segment.

Data Recovery Using Reversible Variable Length Coding (RVLC)

One of the shortcomings of using resynchronization markers alone is that, after detecting an error in a video bit stream and resynchronizing to the next resynchronization marker, the video decoder discards all the data between the two resysnchronization markers. This is due to the fact that the decoder cannot be certain of the exact location, between the two resynchronization markers, where the error has occurred.

Reversible variable-length codes (RVLCs) alleviates this problem and enable the decoder to better isolate the error location by enabling data recovery in the presence of errors. Unlike ordinary VLCs, RVLCs are VLCs that are wrapped in an error correcting code; that is, they are having the prefix property when decoding them in both the forward and reverse directions. This enables RVLCs to be uniquely decoded in both forward and reverse direction. So when a decoder detects an error while decoding a bit stream in a forward direction, it jumps to the next resynchronization marker and decodes the bit stream in the backward direction until it encounters an error, and based on the locations of these two errors, the decoder can recover some of the data between the two consecutive resynchronization markers that would otherwise have been discarded (Talluri, 1998). This is illustrated in Figure 5.9 below.

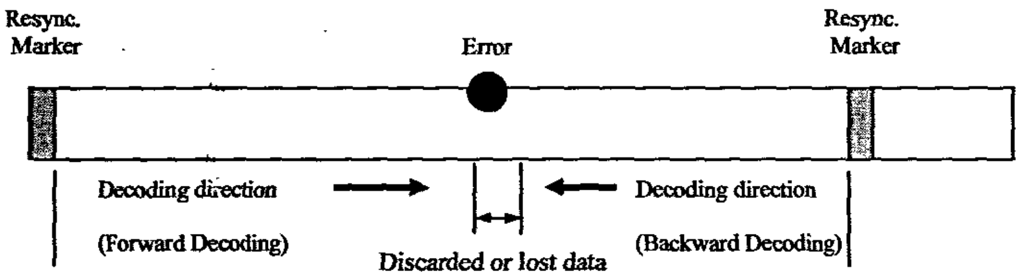


Figure 5.9 RVLC Decoding

According to Li, Kittitornkun, Hu, Park and Villasenor (2000), RVLC can be used in conjunction with data partitioning, where data partitioning is used to separate the less important information bits from the most important information bits and RVLC is used in coding the most important information bits so that the decoder is able to recover data within the erroneous segments (Villasenor, Zhang, and Wen, 1999; Li, Kittitornkun, Hu, Park and Villasenor, 2000). RVLC has been adopted in both MPEG-4 and H.263, in conjunction with insertion of synchronized markers (Wang, Wenger, Wen and Katsaggelos, 2000).

5.5.3.4 Joint source-channel coding (JSCC)

Joint source-channel coding (JSCC) is a hybrid of error control schemes, which uses source coding and channel coding. The goal of JSCC is to achieve an optimum distribution of source and channel bits between source and channel coders in order to minimize the effects of errors. For video applications, JSCC consists of three tasks: finding an optimal bit allocation between source coding and channel coding for a given channel loss characteristics; designing the source coding to achieve the target source rate; and designing the channel coding to achieve the required robustness (Wu, Hou and Zhang 2000; Zhai, Eisenberg, Pappas, Berry and Katsaggelos, 2004).

In order to achieve minimum distortion, JSCC uses a feedback mechanism by which the source coder chooses an optimum quantizer to achieve target bit rate while the channel coder adapts to the loss characteristics by using an appropriate number or redundant bits (Arankalle, 2003). The architecture for JSCC is shown in Figure 5.10.

The receiver monitors the quality of service and infers the channel loss characteristics to the transmitting side through the feedback control protocol. The joint source/channel optimiser at the transmitter uses this feedback information to make an optimal rate allocation between the source and channel coding. The allocated optimal rate is conveyed to both the source and the channel coders. Once the source encoder receives the allocated rate from the joint source/channel optimiser, it chooses an appropriate quantizer to achieve the allocated rate. Similarly, when the channel encoder receives the allocated rate from the joint source/channel optimiser, it chooses a suitable channel code to match the channel loss characteristics (Wu, Hou and Zhang 2000).

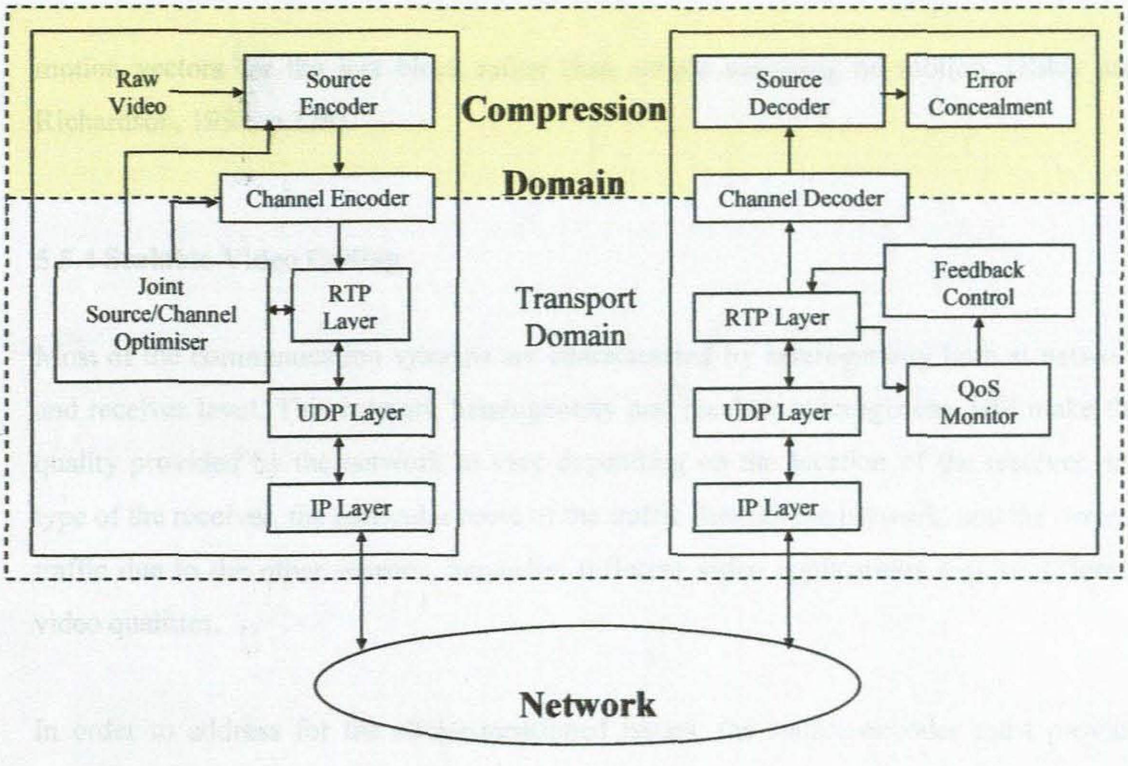


Figure 5.10 Architecture for JSCC

5.5.3.5 Error Concealment

Error concealment is implemented at the receiver by the decoder, usually after data recovery, in order to hide the effects of errors once they have occurred. It improves the quality of the decoded sequence in the presence of errors when the error rate is not that high. There are three types of concealment: temporal concealment, spatial concealment, and motion-compensated concealment. Temporal concealment replaces a corrupted area with a pixel values in the same location in the previous decoded frame, and it is only effective if there is a little change from the previous frame. Spatial concealment replaces the distorted block by a spatial interpolation between adjacent error-free blocks. It is used for those situations where temporal concealment is not effective, for example, where there is high motion or scene change. Motion-compensated concealment estimates the motion vectors for the lost block rather than simple assuming no motion. (Riley and Richardson, 1997, p.124).

5.5.4 Scalable Video Coding

Most of the communication systems are characterized by heterogeneity both at network and receiver level. This network heterogeneity and receiver heterogeneity will make the quality provided by the network to vary depending on the location of the receiver, the type of the receiver, the particular route of the traffic through the network, and the current traffic due to the other sources. Secondly, different video applications require different video qualities.

In order to address for the above-mentioned issues, the source encoder must provide coded video for a range of different visual qualities and a range of available network qualities of service. This is possible through scalable or layered coding, where the video is encoded as a range of layers that contain different components of video information and at the receiver the decoder can choose to decode a particular subset of these layers in order to scale the received video to a particular quality. This is illustrated in Figure 5.12, where the video sequence is encoded at a low quality to form a base layer and the difference between the coarse quality of the base layer and the full quality of the coded sequence is encoded as one or more enhancement layers.

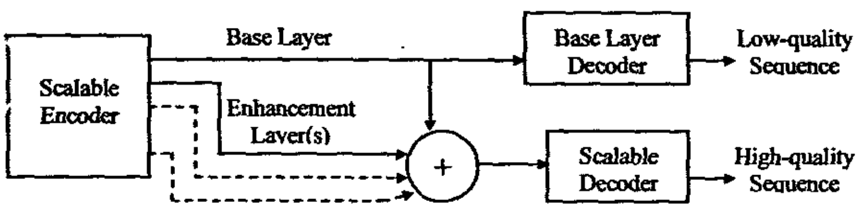


Figure 5.11 Architecture of a scalable or layered coding system

The base layer sub-stream can be independently decoded to provide coarse visual quality, and the other enhancement layer(s) sub-stream can only be decoded together with the base sub-stream. In order to get a quality that is better than the low-quality level, the base layer must be decoded together with the enhancement layer(s). This layered coding improves the flexibility of the video communication system to provide video at a range of spatial and temporal resolutions or visual qualities to compensate for heterogeneity (Riley and Richardson, 1997, p. 158; Wu, Hou, and Zhang, 2001; Hua, Tantaoui, and Tavanapong, 2004; Ohm, 2005). This is illustrated in Figure 5.12, where the raw video sequence is layered coded into three layers: a base layer (BL), enhancement layer (EL) 1, and enhancement layer (EL) 2.

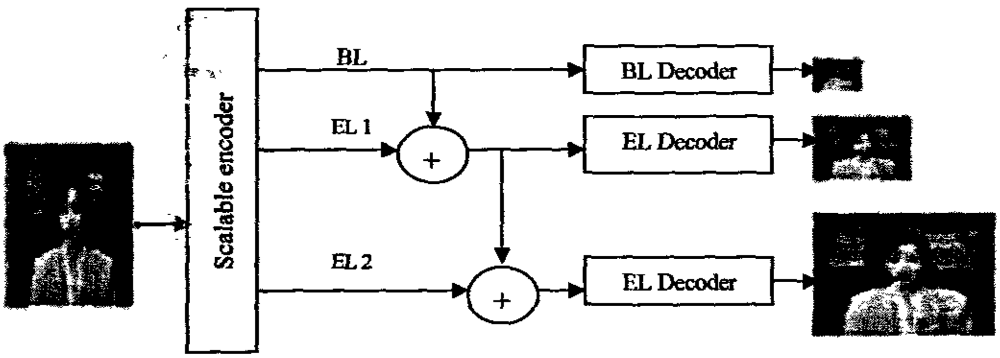


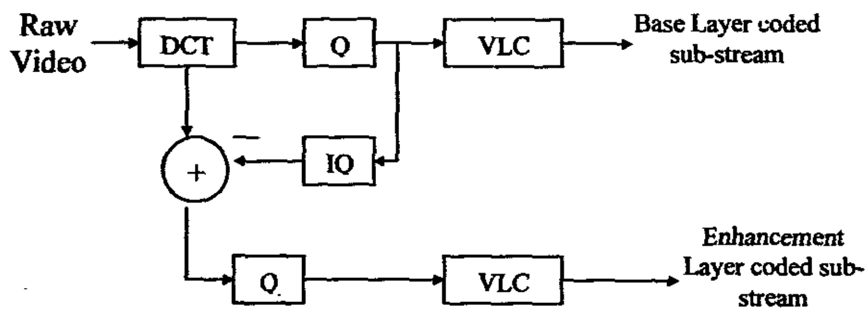
Figure 5.12 Layered video coding for different resolutions

Layered coding can also be used to improve the performance in the presence of transmission errors (Riley and Richardson, 1997, p.158; Ohm, 2005). However, to serve as an error resilient tool, layered coding must be paired with unequal error protection in the transport system, so that the base layer is protected more strongly than the enhancement layers (Wang, Wenger, Wen and Katsaggelos, 2000). Scalability can also withstand bandwidth variations (Wu, Hou, and Zhang, 2001).

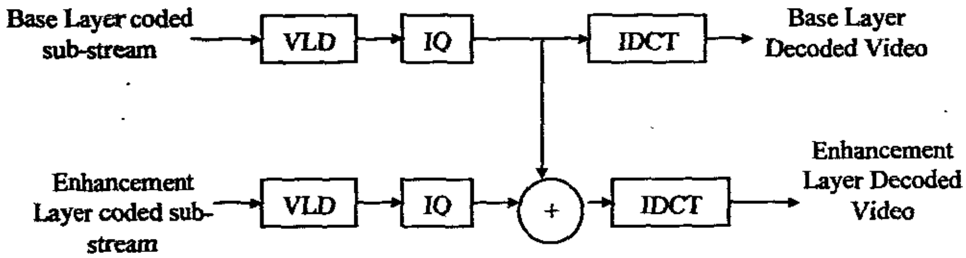
Basically, video scalability consists of signal-to-noise (SNR) scalability, spatial scalability, and temporal scalability. Each video representation has different significance and bandwidth requirements. For example, the base layer is more important than the enhancement layers, however, the enhancement layer requires more transmission bandwidth due to its finer quality while the base layer needs less bandwidth due to its coarser quality. As a result, SNR, spatial and temporal scalability makes it possible to transmit the same video in different representation, or at different rates (that is, bandwidth scalability) (Mrak, Šprljan, and Grgić, 2001; Wu, Hou, and Zhang, 2001). Except for the three basic scalabilities mentioned above, there can be other scalabilities that can be derived by their combinations, such as spatio-temporal scalability. Other scalability mechanisms include frequency scalability and object-based scalability, and fine-granular scalability. The difference between the three basic types of scalability coding is discussed in the following text.

5.5.4.1 SNR Scalability

SNR scalability is defined as the representation of the same video in different SNR or perceptual quality. SNR scalability coding quantizes the DCT coefficient to different levels of accuracy by using different quantization parameters, thus resulting in streams with different quality levels (Wu, Hou, and Zhang, 2001). Figure 5.13 depicts an SNR-scalable encoder and a decoder with two-level scalability.



(a)



(b)

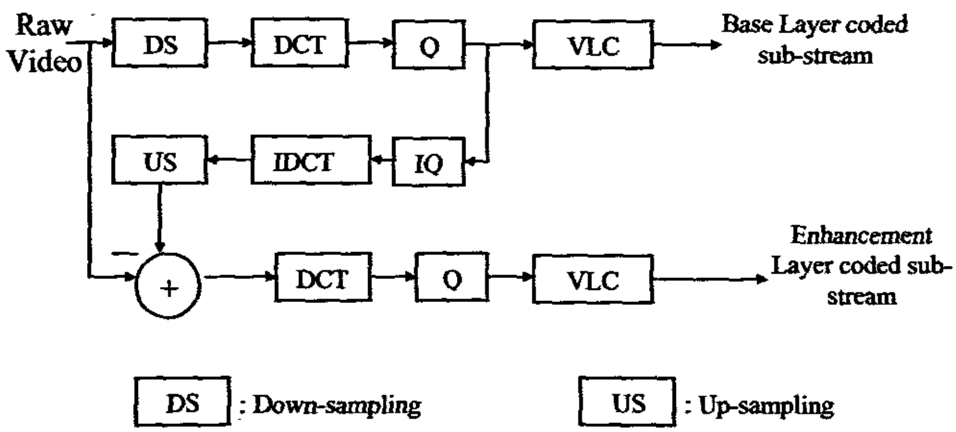
Figure 5.13 SNR-scalable (a) encoder, (b) decoder

5.5.4.2 Spatial Scalability

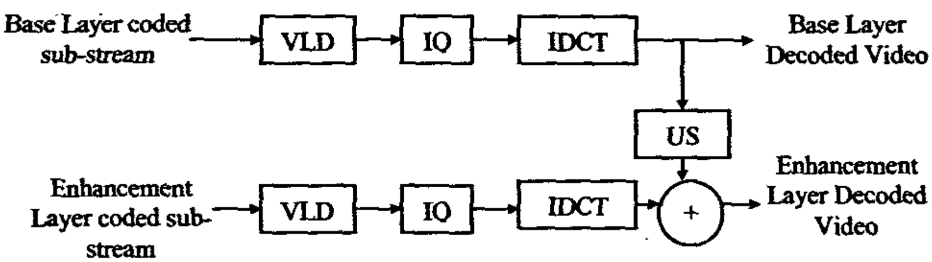
Spatial scalability is defined as representing the same video in different spatial resolutions or sizes. In spatial scalability spatially up-sampled pictures from a lower layer are used as a prediction in a higher layer. For base layer, the raw video is first spatially down sampled; that is, selecting few pixels from a couple of pixels and discarding them, then DCT transformed, quantized, and VLC coded the non-discarded pixels. For enhancement layers, the raw video is spatially down sampled, DCT transformed, and quantized at the base layer. The base–layer image is reconstructed by inverse quantization and inverse DCT. The base layer image is spatially up-sampled (i.e. making copies of each pixel and transmit together with the original pixels to the next stage) and then subtracted from the original image to get the residual, which is then DCT transformed, quantized by a quantization parameter that is smaller than that of the based layer and coded by VLC. The use of the smaller quantization parameter makes the enhancement layer to achieve finer quality than the base layer (Wu, Hou, and Zhang, 2001).

5.5.4.2 Temporal Scalability

Temporal scalability is defined as representing the same video in different temporal resolutions or frame rates. The process of temporal scalable encoding is almost the same as that of spatial scalable coding. The only difference, however, is that in temporal scalable codecs, temporal down-sampling and temporal up sampling are used instead of spatial down sampling and spatial up sampling. That is, instead of selecting and discarding pixels like in spatial down-sampling, the temporal down-sampling, used in temporal scalability, uses frame skipping, where few frames are selected from a couple of frames and the selected frames are discarded (Wu, Hou, and Zhang, 2001). For temporal up-sampling, the codecs make copies of each frame and transmit them together with the original frames to the next stage, while in spatial up-sampling it is the copy of pixels of each pixels that are made and transmitted together with the original pixels to the following stage. Figure 5.14 depicts a spatially/temporally-scalable encoder and a decoder with two-level scalability.



(a)



(b)

Figure 5.14 Spatially/temporally-scalable (a) encoder, (b) decoder

5.5.5 Making Video Source Adapt to Network Conditions

Although TCP ensures reliable packet delivery, in general it is not suited for real time traffic due to delays that are introduced by its retransmission, hence most of the multimedia traffic, especially real-time traffic, is UDP traffic, which does not provide error handling or congestion control (Arankalle, 2003; Wu, Hou and Zhang 2000). Since it has already explained that network congestion can cause busty loss and excessive delay, which can have devastating effects on video presentation quality, some mechanisms for congestion control need to be implemented in order to reduce packet loss and delay (Wu, Hou, and Zhang, 2000). One mechanism to minimize the effects of network congestion is by making the source be able to adapt to the network parameters (Katsaggelos, Eisenberg, Zhai, Berry, and Pappas, 2005). Wu, Hou, and Zhang proposed three mechanisms for controlling congestion: rate control, rate-adaptive video encoding, and rate shaping. The architecture of the congestion control scheme is depicted in Figure 5.15, where a quality of service (QoS) monitor is maintained at the receiver side of the network to monitor the network congestion status based on the behaviour of the arriving packets, e.g., packet loss and delay. Such information is used in the feedback control protocol, which sends information back to the source. Based on this feedback information the rate control module estimates the available bandwidth and regulates the video output rate of the video stream according to the estimated network bandwidth (Wu, Hou, and Zhang, 2000; Mrak, Šprljan, and Grgić, 2001).

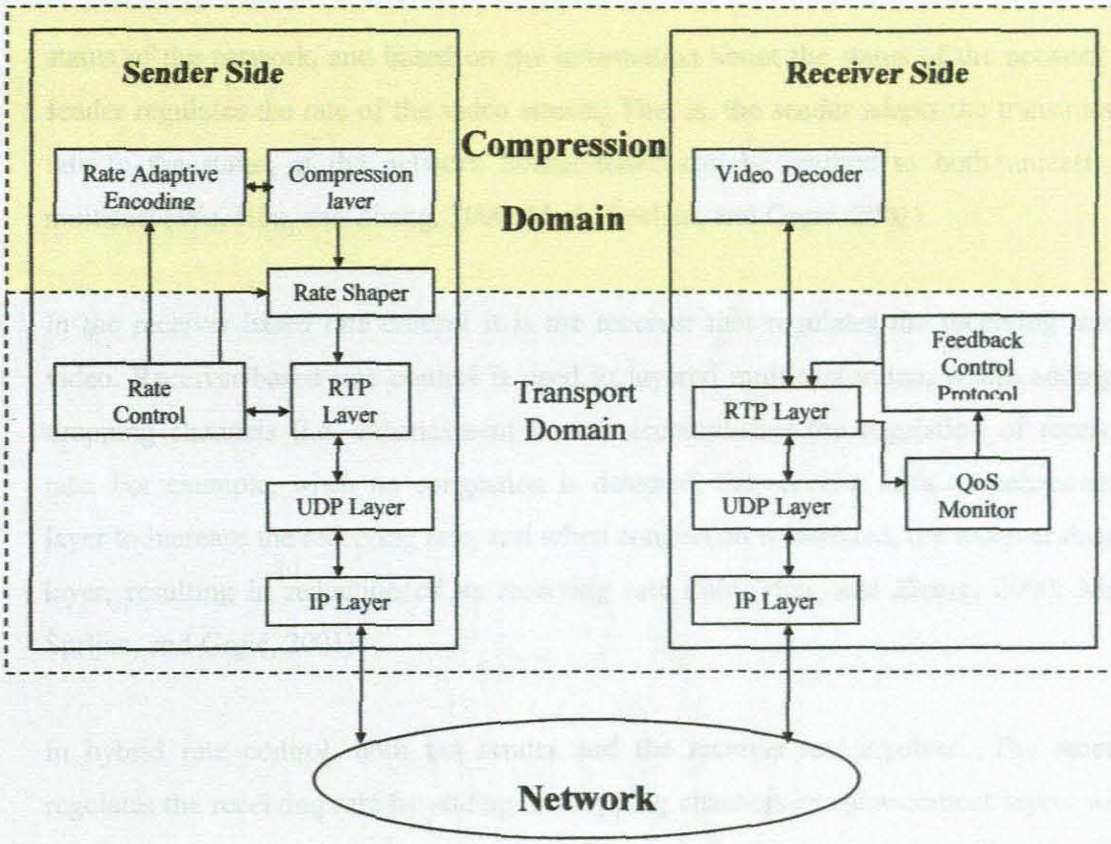


Figure 5.15 Architecture for congestion control

5.5.5.1 Rate Control

Rate control minimizes congestion and packet loss by making the source to send the video stream at a rate dictated by the available network resources so that no packets will be discarded due to the traffic exceeding the available bandwidth. There are three categories of rate control schemes for real-time video: source-based, receiver-based and hybrid rate control (Wu, Hou, and Zhang, 2000; Mrak, Šprljan, and Grgić, 2001).

In the source-based rate control the transmitter is responsible for matching the rate of the video stream to available network bandwidth. A feedback is employed to convey the status of the network, and based on the information about the status of the network the sender regulates the rate of the video stream. That is, the sender adapts the transmission rate to the status of the network Source-based can be applied to both unicast and multicast (Wu, Hou, and Zhang, 2000; Mrak, Šprljan, and Grgic, 2001).

In the receiver-based rate control it is the receiver that regulates the receiving rate of video. Receiver-based rate control is used in layered multicast video, where adding or dropping channels (i.e. enhancement layers) accomplishes the regulation of receiving rate. For example, when no congestion is detected, the receiver adds an enhancement layer to increase the receiving rate, and when congestion is detected, the receiver drops a layer, resulting in reduction of its receiving rate (Wu, Hou, and Zhang, 2000; Mrak, Šprljan, and Grgić, 2001).

In hybrid rate control, both the sender and the receiver are involved. The receiver regulates the receiving rate by adding or dropping channels or enhancement layers while the sender adjusts the transmission rate of each layer based on information from the receivers. Like receiver-based rate control, hybrid rate control is targeted at multicast video (Wu, Hou, and Zhang, 2000; Mrak, Šprljan, and Grgić, 2001).

5.5.5.2 Rate-Adaptive Video Encoding

The objective of the rate-adaptive video encoding is to maximize the perceptual video quality under a given encoding rate, which can be either fixed or dynamically changing based on the network congestion status. This adaptation can be achieved by the alteration of the encoder's Quantization parameter and/or the alteration of the video frame rate. Alteration of the frame rate is achieved by skipping a frame at the encoder so that the time that was going to be used by the skipped frame can be used by coded bits of the previous frame, thereby reducing the buffer level (Wu, Hou, and Zhang, 2000).

5.5.5.3 Rate Shaping

According to Eleftheriadis and Batra, rate shaping can be defined as an operation, which, given an input compressed video bit stream and a set of rate constraints, produces another compressed video bit stream that complies with these constraints (Eleftheriadis and Batra, 2003). Rate shaping plays an important role in transmitting compressed video on a network with dynamic bandwidth. According to Wu, Hou, and Zhang (2000), rate shaping is a technique used to adapt the rate of compressed video bit streams to the target rate constraint. Figure 5.15 depicts a rate shaper as an interface, or bitstream filter, between the video encoder and the network, through which the encoder's output can be matched to the available bandwidth. Rate shaping can either be implemented at compression domain or transport domain as illustrated in Figure 5.15.

One typical example of implementing rate shaping at compression domain is dynamic rate shaping. The concept of dynamic rate shaping (DRS) was introduced in early 1995 as a technique to adapt the rate of compressed video bit streams to dynamically varying bit rate constraints. DRS bridges the gap between constant and variable bit rate video, allowing a continuum of possibilities between the two. It provides bit stream filtering to perfectly match the encoder's output to the network's quality of service characteristics (Eleftheriadis and Batra, 2003). Figure 5.16 illustrates the dynamic rate shaping operation, where B and B^* are the bit rate of the encoded video bit stream before and after transcoding, respectively, and B_T is the constraint rate of the network over which the output bit stream is transmitted.

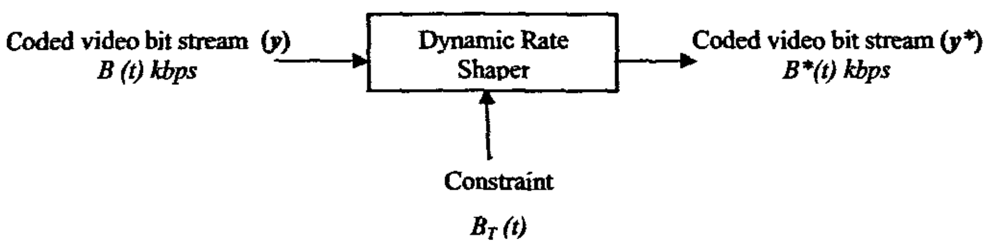


Figure 5.16. Dynamic rate shaping operation

The objective of a rate shaping algorithm is to minimize the conversion distortion; that is,

$$\min_{B^*(t)\leq B_T(t)}\{\|y-y^*\|\}\tag{5.1}$$

where $\|\,.\,\|$ denotes the squared error criterion, y is the bit stream before transcoding and y^* is the bit stream after transcoding.

There are two ways to reduce the rate for motion-compensated block-based transform coding techniques:

- Modifying the quantized transform coefficients by employing coarser quantization (i.e. requantization)
- Eliminating transform coefficients, which is selective discarding of coefficients

Both schemes can be used to perform rate shaping; requantization, however, leads to recording-like algorithms, which are not amenable to very fast implementation (Eleftheriadis, and Anastassiou, 1995) that is required in real-time applications. Eliminating transform coefficient scheme, which is also known as selective-transmission, can be categorized into two: truncation, and arbitrary selection. In truncation case, a set of DCT coefficients at the end of the block is eliminated. This approach is referred to as constrained DRS (CDRS). In arbitrary selection case, DCT coefficients are arbitrary selected for elimination from bitstream. This approach is referred to as general or unconstrained DRS (GDRS) (Eleftheriadis, and Anastassiou, 1995; Eleftheriadis and Batra, 2003).

5.5.6 Prioritise and Protecting Video Traffic

In a multiservice network or converged network, video applications must battle for bandwidth with other applications, causing latency, delay variations, packet loss, and other problems that can jeopardize the quality of video over IP services. In order to provide quality transmission to coded digital video the communication network is required to monitor, control, and provide network resources so that the video traffic can be protected and controlled.

In the previous chapter, the best effort debilities of IP-based networks were highlighted, and some mechanisms to make IP suitable for real-time applications were also discussed. These mechanisms or technologies, which included IntServ, DiffServ, and MPLS, are used to provide QoS to real-time traffic. All the three mechanisms are able to guarantee the appropriate QoS treatment to real-time traffic concerning its special delay, jitter, packet loss and bandwidth requirements. However, it was also highlighted in the previous chapter that IntServ suffers from scalability problems. Hence in this study the combination of DiffServ and MPLS will be considered for the core network. Where DiffServ is used to divide traffic into a small number of classes with each class being identified by a mark called DSCP, which is carried in the six-bit differentiated field of the IP header, and MPLS is used for traffic engineering and for offering efficient IP QoS services over diverse layer-two backbone networks (Patrikakis, Despotopoulos, Rompotis, Boukouvalas, Pediaditis and Lambiris, 2003).

The reason for suggesting the adoption of a combination of DiffServ and MPLS, is that, though MPLS offers traffic engineering and great savings in terms of switching speed, it still lacks an effort to distinguish among the flows of different delay characteristics, and drop precedence in order to provide real quality of service guarantees (Saad, Yang, Yu, Makrakis, Groza, and Petru, 2001). Secondly, using DiffServ alone cannot guarantee QoS since DiffServ does not influence a packet path, and therefore during a congestion or failure, even high-priority packets would not get guaranteed bandwidth. Pairing MPLS with DiffServ can compensate these downsides, so that DiffServ can be responsible for classifying the traffic according to different flow characteristics, while MPLS is responsible for providing a connection-oriented environment that enables traffic engineering.

DiffServ defines Classes of Service (CoS), called aggregates, and QoS resource management functions with node-based, or per-hop, operation. The CoS definitions include a behaviour aggregate (BA) which has a specific requirements for scheduling and packet discarding, and an ordered aggregate (OA) which performs classification based on scheduling requirements only, and may include several drop precedence values as mentioned in the previous chapter. Thus, OA is a coarser classification than a BA and may include several BAs. DiffServ offers per hop behaviour (PHB) to BA, which includes scheduling and packet discarding, and PHB scheduling class (PSC) to OA, which only concerns scheduling. The value of the DSCP field is used to specify a BA (or class), which is used by DiffServ-compliant nodes for choosing the appropriate PHB (or a queue servicing treatment). Fourteen PHBs have been defined: one for expedited forwarding (EF), twelve for assured forwarding (AF), and one for Default (or Best Effort) PHB. The twelve AF PHBs are divided into four PSCs, and each of the PSCs consists of three sub-behaviours, which are related to different packet discarding treatment.

As discussed in the previous chapter, MPLS provides fast packet switching and an opportunity for traffic engineering, resulting in better utilization of network resources such as link capacity as well as the ability to adapt to node and link failures. To deploy MPLS in an IP-based network, the MPLS Shim header is inserted between the link layer and network layer headers as shown in Figure 4.6 (previous chapter). It was also mentioned in chapter four that this Shim header contains four fields: Label field, CoS or EXP field, S field, and TTL field. Label field is used for identifying the FEC, which is a group of packets that are forwarded to the same next hop, specified by the Next Hop Label Forwarding Entry (NHLFE). The CoS or EXP field is used to select the appropriate PHB for the packet.

5.5.6.1 MPLS with DiffServ

According to Fineberg [2003], there are two conditions that are necessary for QoS: guaranteed bandwidth, and class-related scheduling and packet discarding treatment. DiffServ only satisfies the second condition, but not the first. In order to satisfy the first condition we need to supplement DiffServ with MPLS. MPLS allows the implementation of traffic engineering in networks to achieve bandwidth assurance, diverse routing, load balancing, path redundancy, and other services that lead to QoS (Fineberg, 2003). Thus, the combination of DiffServ-based classification and per-hop behaviours with MPLS-based traffic engineering will lead to true QoS in packet backbones.

However, there are two basic challenges when combining MPLS and DiffServ, which need to be addressed in order to fully support DiffServ in an MPLS network. First, the DiffServ's DSCP is carried in the IP header, but the MPLS routers (LSRs) only examine the label header. Second, the DSCP uses 6 bits to identify a PHB associated with a packet, while MPLS's CoS or EXP field has only 3 bits to specify the PHB for the packet (Law and Raghavan, 2001; Minei, 2004). To overcome these challenges, RFC 3270 defines two types of LSPs for use in MPLS-DiffServ network: EXP-Inferred-PSC LSPs (E-LSPs) and Label-Only-Inferred-PSC LSPs (L-LSPs).

Mapping Between DiffServ IP Headers and MPLS Shim Headers

In E-LSP, a label is used as the indication of the FEC destination, and the 3-bit CoS or Exp field of the MPLS Shim header is used by the LSR to determine the PHB to be applied to the packet, including both scheduling and the drop precedence (or priority). E-LSP can carry packets with up to eight distinct PHB in a single LSP (Law and Raghavan, 2001; Le Faucheur *et al.*, 2002; Fineberg, 2003; Minei, 2004). Figure 5.17 shows the mapping between IP header with DiffServ and MPLS shim header for E-LSP

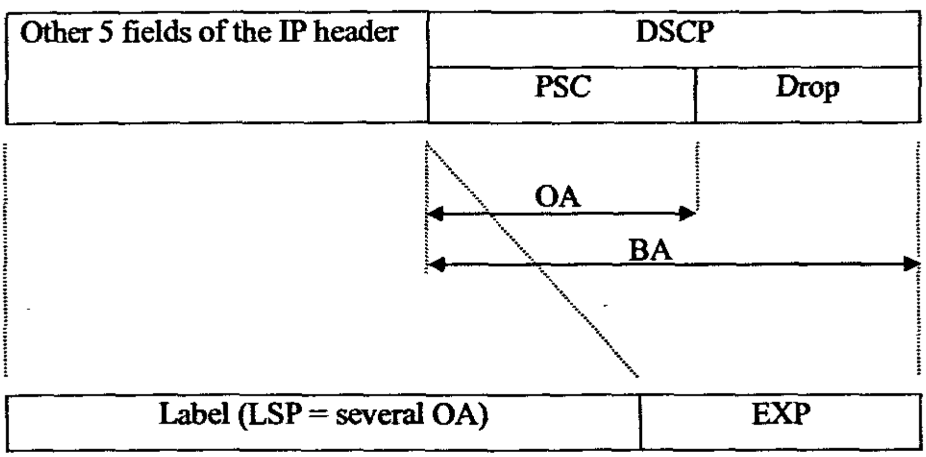


Figure 5.17. Mapping between IP header and MPLS shim header for E-LSP

L-LSP is designed for networks that support more than eight PHBs, where the three bits of the Exp field alone are not enough to carry all the necessary information for distinguishing between the PHBs. The only other field in the MPLS shim header that can be used for distinguishing between the PHBs is the label field itself, hence in an L-LSP, the label is used as an indication of both the FEC destination and its scheduling priority, while the Exp field is used only for the indication of the drop precedence. Thus, in L-LSP, the PHB is determined from both the label and Exp bits (Law and Raghavan, 2001; Le Faucheur *et al.*, 2002; Fineberg, 2003; Minei, 2004). Figure 5.18 shows the mapping between IP header with DiffServ and MPLS shim header for L-LSP.

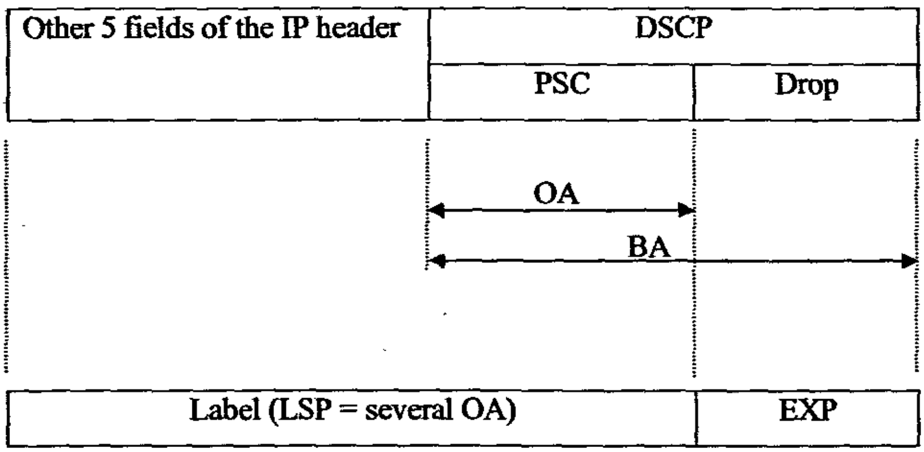


Figure 5.18. Mapping between IP header and MPLS shim header for L-LSP

The following table, Table 5.1, summarizes the differences between E-LSPs and L-LSPs.

Table 5.1 Differences between E-LSPs and L-LSPs.

E-LSP	L-LSP
PHB is determined by the Exp bits	PHB is determined by the label and Exp bits together
Can carry traffic with up to 8 distinct PHBs in a single LSP	A single PHB per LSP or several PHBs with the same scheduling regimen and different drop priorities
No signalling required to convey the PHB information	The PHB information needs to be signalled when the LSP is established
Up to 8 PHBs can be supported in the network when only E-LSPs are used. However, E-LSPs can be used in conjunction with L-LSPs when more	Any number of PHBs can be supported in the network

PHBs are required	

DiffServ-Aware MPLS Traffic Engineering (DS-TE)

MPLS traffic engineering (MPLS-TE) is used in networks to achieve performance objectives such as bandwidth assurance, diverse routing, load balancing, path redundancy, and other services that lead to QoS. If the LSRs in the MPLS network are DiffServ-enabled, it is possible to engineer the network with traffic engineering that is applied on a per-class basis. This type of traffic engineering is referred to as DiffServ-aware MPLS traffic engineering (DS-TE). The essential goal of DS-TE is to guarantee network resources separately for each type of traffic in order to improve and optimise its compliance with QoS requirements (Minei, 2004). This makes it possible to establish separate LSPs for different classes, taking into consideration resources available to each class. For example, a separate LSP can be established for each type of real-time or premium traffic, and these LSPs can be given higher priority than other LSPs (Goode, 2002).

DS-TE enables the introduction of the concept of LSP priorities, where some LSPs are marked as more important than others, and it also allows the more important LSPs to be able to confiscate resources from the less important LSPs; that is, pre-empting the less important LSPs. This is done to ensure that important LSP always establishes along the optimal path regardless of the existing reservations, and when LSP needs to reroute in the event of node or link failure, important LSPs have a better chance of finding an alterative path. It is also done to ensure that when high-priority-bandwidth is not needed for real-time traffic, it can be used for lower priority class of traffic that is carried on less important LSPs (Goode, 2002; Minei, 2004). Each LSP has two priorities associated with it: a setup priority and a hold priority. The setup priority controls access to the resources for an LSP at the time of LSP establishment, and the hold priority control access to the resources for an LSP that is already established.

The fundamental requirement for DS-TE is to be able to enforce different limits on the percentage of a link's bandwidth for different sets of aggregation of traffic flows of the same class, which are placed inside an LSP; that is, DS-TE must enforce different bandwidth constraints (BCs) to different sets of traffic trunks (TT). This implies keeping track of how much bandwidth is available for each type of traffic at a given time on all the routers throughout the MPLS network. To address this, Le Faucheur and Lai introduced the concept of a class type (CT), which is the set of traffic trunks crossing a link, that are governed by a specific set of bandwidth constraints. This CT is used for the purpose of link bandwidth allocation, constraint based routing and admission control. Once a traffic trunk has been assigned to a CT it will remain an element of that same CT on all links. Each CT may carry traffic from more than one class of service (RFC 3564, 2003).

According to RFC 3564, the DS-TE solution must support up to 8 CTs. Those 8 CTs are referred to as CTc, $0 = c = MaxCT\text{-}1 = 7$; that is, CT0 through CT7. The DS-TE solution must enforce a different set of bandwidth constraint for each CT, and a DS-TE implementation must support at least 2 CTs. LSPs that are traffic engineered to guarantee bandwidth from a particular CT are referred to as DS-TE LSPs, and in current IETF model, a DS-TE LSP can only carry traffic from on CT. By convention the best effort traffic is mapped to CT0.

5.5.6.2 Traffic Protection

Once the high-priority traffic has been classified and the important LSPs have been established, the high-priority or premium traffic can be mapped on these important LSP to all it to utilize the resources available for the premium class. To make sure that this premium traffic is less affected by route failures, the important LSP will have Fast Reroute enabled so that when there is a route failure the premium traffic can be switched to the protection LSP, which is established around the 'working' or operational LSP as an alternative route in the event of any link/s or router/s along the working LSP fail. This traffic protection and fast rerouting is essential for applications that cannot tolerate packet loss in order to make sure that premium traffic is less affected by route failures.

5.6 IN SUMMARY

This chapter was about quality video transmission using IP-based network; that is video over IP. Firstly, the author introduces video applications using IP-based networks and how they can be integrated into healthcare system to provide telemedicine services such as tele-consultation, tele-diagnosis, training, CME, etc. These services include video conferencing, video-on-demand and video broadcast. Secondly, the issue of quality video is introduced. Thirdly, digitised video including compression and encoding for low bit rates was introduced. Under video coding, three standards that can produce low bit rates were discussed, and these include H.263, MPEG-4 and H.264. Fourthly, factors that are affecting the quality of video transmission were discussed. These included transmission bandwidth, latency, jitter, errors and losses. Lastly, approaches to ensure quality video transmission were discussed. These approaches included video encoding scheme that will produce a low bit rate without compromising much on quality in order to enable the video transmission over small bandwidth; video delivery or distribution method that will utilise the bandwidth efficiently; error control and concealment to address losses and errors introduced by the communication network; scalable coding to address heterogeneity both at network and receiver level; making the video source to be able to adapt to network conditions to address bandwidth fluctuations and congestion, and prioritising and protecting video traffic using the QoS and traffic engineering mechanisms introduced in the previous chapter to provide solutions for quality video transmission.

To conclude the chapter, the next chapter is introduced, which is about modeling and simulation in order to validate the proposed solutions to quality video transmission.

CHAPTER 6

6. MODELLING AND SIMULATION

6.1 INTRODUCTION

When designing a new communication network, it is desirable to be able to predict its level of performance before implementing it. This chapter briefly reviews some of the methods used to predict performance. After reviewing the methods, implementation of these methods will be done in order to evaluate and validate the proposed system.
These methods include modelling and simulations (Walrad, 1998, p.270), which are the most commonly used by performance analysts to represent constraints and optimise performance. In this study the following are going to be used for performance analysis:

(i) Mathematical analysis for analyzing mathematically the simplified models of communication systems,

(ii) Matlab and Simulink for modelling and analyzing communication system

(iii) Network Simulator (ns) to simulate the networking environment

6.1.1 Models as Tools

In this study the term model will be used for something which represents a system, or part of the system, in the form which allows the making of prediction about the behaviour of the complete or part of the system. That is, a model will be used as an abstraction and approximation of the system, which is an attempt to distil from the mass details of the system those aspects that are more essential to the system's behaviour. The accuracy of these predictions will depend on the detail invested in the model; that is, a very crude model will suffice where only an approximate or relative measure of the behaviour of the system is analysed, whereas a more detailed model is needed for crucial analysis. Thus, the development of the model and the representation of the system ranges from the very abstract, in which the behaviour of the system can be captured by a single equation, to the

6.1.2 Performance Evaluation

Performance evaluation is concerned with the description, analysis and optimisation of the dynamic behaviour of a communication system. It involves the investigation of the flow of data and control information within and between components of the system. The aim is to understand the behaviour of the system and identify the aspects of the system, which are sensitive from the performance perspective. Performance evaluation can be used to evaluate the design and its alternatives, to determine the optimal value of a parameter or parameters, to find a performance bottleneck, to characterise the load of the system, to determine the number and the size of the components of the system, and to predict the performance at future loads (Hillston, 2002).

There are a variety of measures or indices to characterise the behaviour of the system, however, in this study the emphasis is going to be on those that influence quality of service of video transmission over IP. These include throughput (usually specified as bandwidth), delay (latency), delay variations (jitter), loss, and reliability.

The modelling approach that is going to be used in this study is divide-and-conquer; that is, each component of the system is going to be modelled separately, and then the resulting model is the combination of these components. Since the main aim of modelling is for performance, the elements that will be modelled are those that affect the performance of the system, which include the source models and the network elements. The principles that will be used in modelling in this study will be as follows:

- Use the big picture to find the most important parts of the system, and areas of interest
- Break the system into smaller parts and construct small basic models
- Decide what results and performance is required
- Include in these small basic model only the detail that effects the results of modelling
- Integrate the small basic models into a more detailed model for credibility

6.2 VIDEO COMMUNICATION SYSTEM

In chapter five it was mentioned that, in order to transmit video over packet networks, the input video must be compressed to save bandwidth using the source encoder, channel coded, packetized and/or modulated, and at the receiver side, the inverse operations are performed in order to reconstruct and display the video signal (Wang and Zhu, 1998). A functional block diagram for a video communications system was also shown in Figure 5.3.

According to Riley and Richardson (1997, p.198), the overall quality of communications is influenced at a number of stages of the video communication system: encoding, protocol processing, transmission, and decoding. This means that in order to fully model the video communication system all these components of the communication systems must be considered.

6.3 MODELLING TRAFFIC SOURCES

According to Ghanbari, Hughes, Sinclair and Eade (1997, p.240), packet networks are variable bit-rate (VBR) systems that offer bandwidth on demand; that is, the resources of the network are engaged by the source only when packets are to be transmitted. So to estimate the performance of the network by analysis and/or simulation, it is therefore necessary to specify both the statistics of the sources generating the information to be transmitted and the rules for assembling the information (Ghanbari, Hughes, Sinclair and Eade, 1997, p.240).

The goal of source traffic modelling is to find a traffic model, defined by a small set of statistics, that can still capture the significant statistical information of the real modelled traffic, so that an accurate queueing performance evaluation can be obtained if the model is used as input into a queueing system instead of the real traffic (Tatipamula and Khasnabish, 1998, p.133). Source traffic modelling involves characterizing and building mathematical models that mimic the source traffic characteristics. These traffic models can be used for performance analysis in simulation. In this section the characterizing and modelling of traffic sources that is foreseen to be on the network is going to be done. Though all the three foreseen traffic sources (voice, data, and video) are going to be covered, the emphasis placed on video type traffic and the others are going to be covered briefly. Modelling video source depends on the coding technique used, and on the video application itself. For example, it was mentioned in the previous chapter that H.264 will generate low bit-rates than most of the existing coding techniques, and interactive video application will be different from entertaining video programs, which include action packed sports and movies. In this study the emphasis is going to be on H.264 (MPEG-4 AVC) video encoding standard.

6.3.1 Video Characterization

Video, like voice, is natural bursty but can be manipulated to give a constant output bit rate. Due to its bandwidth requirements, video is first compressed by a video encoder before it is transmitted in order to save bandwidth. The output of the video encoder will change as the information content of the picture changes. For example, only the initial scene's full encoded information needs to be transmitted in full and thereafter only the small updating changes are necessary to reproduce the succeeding frames. This results in a bit-rate that varies widely. The variable bit rate (VBR) output from the video encoder is also a function of how quick the movements are occurring in the picture. For example, sudden movements in the picture or scene changes will generate high bit rate than a slow moving picture, a 'talking head' or a caption, while small changes in the picture or scene will result in low bit rates (Murphy and Teahen, 1994; Ghanbari, Hughes, Sinclair and Eade, 1997, p.242).

Generally, there are three different methods to characterize encoded video for the purpose of network performance evaluation: video bit stream, video traffic trace, and video traffic model (Seeling, Reisslein and Kulapala, 2004). Each of these methods has its strengths and weaknesses, as discussed below.

6.3.1.1 Video bit stream

The video bit stream is generated using a video encoder, and it contains the complete video information. The traffic characterization such as frame size or number of bits per frame can be obtained by measuring or by parsing the bit stream. The advantage of the video bit stream is that it allows for network experiments where the quality of the transmitted video, after suffering losses in the network, can be evaluated. The video bit stream can be either compressed or uncompressed. However, uncompressed bit stream are very large in size, hence the video bit streams that are transmitted through the network are compressed in order to save bandwidth. The major problem with video bit stream is that you need video encoders to generate them, which are usually proprietary and/or protected by copyright. This limits the access of other researchers to their availability (Seeling, Reisslein and Kulapala, 2004).

6.3.1.2 Video traffic trace

Unlike video bit streams, which consists of the actual bits carrying the video information, the video traffic traces only give the number of bits used for the encoding of the individual video frames, hence are sometimes referred to as frame size traces (Reisslein, Lassetter, Ratnam, Lotfallah, Fitzek and Panchanathan, 2002). Video traces can be used as alternative to video bit streams, thus eliminating the copyright issues.

A general video trace structure gives some, or all, the following quantities:

- Frame number, n
- Cumulative display time T_n
- Frame type (I, P, or B)
- Frame size X_n
- Luminance quality Q_n^Y (in dB)
- Hue quality Q_n^U (in dB)
- Saturation quality Q_n^V (in dB)

These quantities are given in ASCII format with one video frame per line, and the last three quantities denote the quality in terms of the Peak Signal to Noise Ratio (PSNR) of the colour components: luminance (Y) and two chrominance components hue (U) and saturation (V) of the encoded video frame n in decibels. Table 6.1 shows an excerpt of a H.264 trace file of 'talking head' encoding.

Table 6.1 Excerpt of a H.264 trace file of 'talking head' encoding.

Frame number	type	offset	time	size
0	I	0x0000000000000178	00:00.0	3384
1	P	0x0000000000000eb0	00:00.0	376
2	P	0x0000000000001028	00:00.1	188
3	P	0x00000000000010e4	00:00.1	188
4	P	0x00000000000011a0	00:00.2	188
5	P	0x000000000000125c	00:00.2	188
6	P	0x0000000000001318	00:00.2	188
7	P	0x00000000000013d4	00:00.3	188
8	P	0x0000000000001490	00:00.3	188
9	P	0x000000000000154c	00:00.4	752
10	P	0x000000000000183c	00:00.4	188
11	P	0x00000000000018f8	00:00.4	188
12	P	0x00000000000019b4	00:00.5	376

6.3.1.3 Video traffic model

A traffic model is derived or developed from video traffic traces. This development is based on the statistical properties of a set of video trace samples of the real video traffic. For example, if the number of frames in a given trace is N, then the mean, coefficient of variation and autocorrelation are as follows (Fitzek and Reissslein 2000; Reisslein, Lassetter, Ratnam, Lotfallah, Fitzek and Panchanathan, 2002):

The sample mean $\overline{X}$ of a frame size trace is

$$\overline{X}=\frac{1}{N}\sum_{n=0}^{N-1}X_n \tag{6.1}$$

The sample variance S_X^2 of the frame size trace is

$$S_X^2=\frac{1}{N-1}\sum_{n=0}^{N-1}\left(X_n-\overline{X}\right)^2 \tag{6.2}$$

Substituting Equation (6.1) for $\overline{X}$, equation (6.2) becomes

$$S_X^2=\frac{1}{N-1}\left[\sum_{n=0}^{N-1}X_n^2-\frac{1}{N}\left(\sum_{n=0}^{N-1}X_n\right)^2\right] \tag{6.3}$$

The coefficient of variation CoV_X

$$CoV_X=\frac{S_X}{\overline{X}} \tag{6.4}$$

The maximum frame size X_{max} is

$$X_{\max}=\max_{0\le n\le N-1}X_n \tag{6.5}$$

The autocorrelation coefficient *?x(k)* for lag *k*, *k* = 0, 1, 2, ..., *N*-1, is

$$\rho x(k)=\frac{1}{N-k}\sum_{n=0}^{N-k-1}\frac{\left(X_n-\overline{X}\right)\left(X_{n+k}-\overline{X}\right)}{S_X^2} \tag{6.6}$$

The goal of the video traffic model is to capture the essential properties of the real traffic in an accurate, computationally efficient, and preferable mathematically tractable description that should also be parsimonious; that is, require only a small number of parameters. The developed model is verified by comparing the traffic it generates with the video traces, and if the model is deemed sufficiently accurate, it can be used for mathematical analysis of networks, for model driven simulation and also for generating so called virtual or synthetic video traces (Seeling, Reisslein and Kulapala, 2004).

6.3.1.4 Generation of Video Traffic for Performance Analysis

Two videos are captured: one is 'talking head', which is a person sitting in front of a webcam, and another one is a movie called 'Mr. Bones' by Leon Schustler.

(a) Talking head

(b) Mr. Bones

Figure 6.1 Captured Videos

'Talking head' is captured using a camera and the captured video frames are stored on a disk as Common Intermediate Format (CIF) with 352 pels in the horizontal direction and 288 pels in the vertical direction (i.e. 352x288 pels) and as well as Quarter Common Intermediate Format (QCIF) with 176 pels in the horizontal direction and 144 pels in the vertical direction (i.e. 176x144 pels). 'Mr. Bones' is captured from a DVD player and the captured video frames are also stored on the disk as Standard Image Format (also known as Standard Interchange Format or Source Input Format) (SIF) with 384 pels in the horizontal direction and 288 pels in the vertical direction (i.e. 384x288 pels). In all the three frame formats, each video frame is divided into three components, which are the luminance component (Y), and two chrominance components: hue (U) and intensity (or saturation) (V). This YUV information is grabbed at a certain rate, which is called the frame rate.

One of the advantages of using YUV is that the chroma channels can have a lower sampling rate than the Y channel without a dramatic degradation of the perceptual quality. A notation called the *A:B:C* notation is used to describe how often U and V are sampled relative to Y, for example:

- 4:4:4 means no down-sampling of the chroma channels.
- 4:2:2 means 2:1 horizontal down-sampling, with no vertical down-sampling. Every scan line contains four Y samples for every two U or V samples.
- 4:2:0 means 2:1 horizontal down-sampling, with 2:1 vertical down-sampling.
- 4:1:1 means 4:1 horizontal down-sampling, with no vertical down-sampling. Every scan line contains four Y samples for every U or V sample.

In this study 4:2:2 sampling, as defined in ITU-R Recommendation BT.601, is used for the 'talking head' video, and for 'Mr. bones' video, a 4:1:1 sampling is used. Figure 6.2 shows the sampling grids defined by these standards, where luma samples are represented by a cross, and chroma samples are represented by a circle.

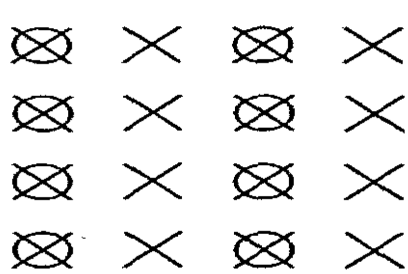


Figure 6.2 YUV 4:2:2 sample positions

The videos were captured as uncompressed YUV information using Asymetrix DVP Capture 4.0 software. However, due to its inefficiency (dropping of about 50 % of frames while capturing), there was a change to other software packages: Ulead VideoStudio 8 software and Pure Motion Capture software, which proved to be more efficient than the Asymetrix DVP capture software, because when they were used there was no dropping of even a single frame during capturing. The Ulead VideoStudio is used for capturing the 'talking head' video from the webcam, whereas the Pure Motion Capture software together with Brooktree Fusion Video card were used to capture 'Mr. Bones' movie from a DVD player. The resulting YUV information is grabbed at a frame rate of 25 frames/second and stored on the disk. For CIF the video runs for 180 seconds (3 minutes), for QCIF the video runs for 300 seconds (5 minutes), and for SIF the video runs for 181 seconds (3 minutes and 1 second).

Both QCIF and CIF formats use 4:2:2 chrominance sampling and quantization into 16 bits. This means that each macro-pixel is two pixels encoded as four consecutive bytes. This results in horizontal down-sampling of the chroma by a factor of two, thus producing the following number of pels per frame

$$No.\text{ of pels/frame for } 4:2:2 = (W \times H) + 2\left(\frac{W}{2} \times H\right) \tag{6.7}$$

For CIF: $= 352 \times 288 + 2 \times \frac{352}{2} \times 288$

$= 202752$ pels per frame

For QCIF: $= 176 \times 144 + 2 \times \frac{176}{2} \times 144$

$= 50688$ pels per frame

SIF format use 4:1:1 chrominance sampling and quantization into 16 bits. This will produce the following number of pels per frame

$$No.\text{ of pels/frame for } 4:1:1 = (W \times H) + 2\left(\frac{W}{2} \times \frac{H}{2}\right) \qquad (6.8)$$

$$= 384 \times 288 + 2 \times \frac{384}{2} \times \frac{288}{2}$$

$$= 165888 \text{ pels per frame}$$

These YUV frame sequences are used as inputs for the H.264 encoders: one with an average rate of 64 kbps and another with an average rate of 600 kbps. However, after decoding and replaying the videos it was found that the quality for decoded 'Mr. Bones' (which was encoded at 64kbps) is very poor, hence the low bit rate was changed from 64 kbps to 150 kbps for 'Mr. Bones'. For each encoded output, a corresponding trace file is generated. From these trace files video traffic metrics are retrieved. These retrieved metrics are then used for the statistical analysis of the recorded video signal. This whole process is depicted in Figure 6.3.

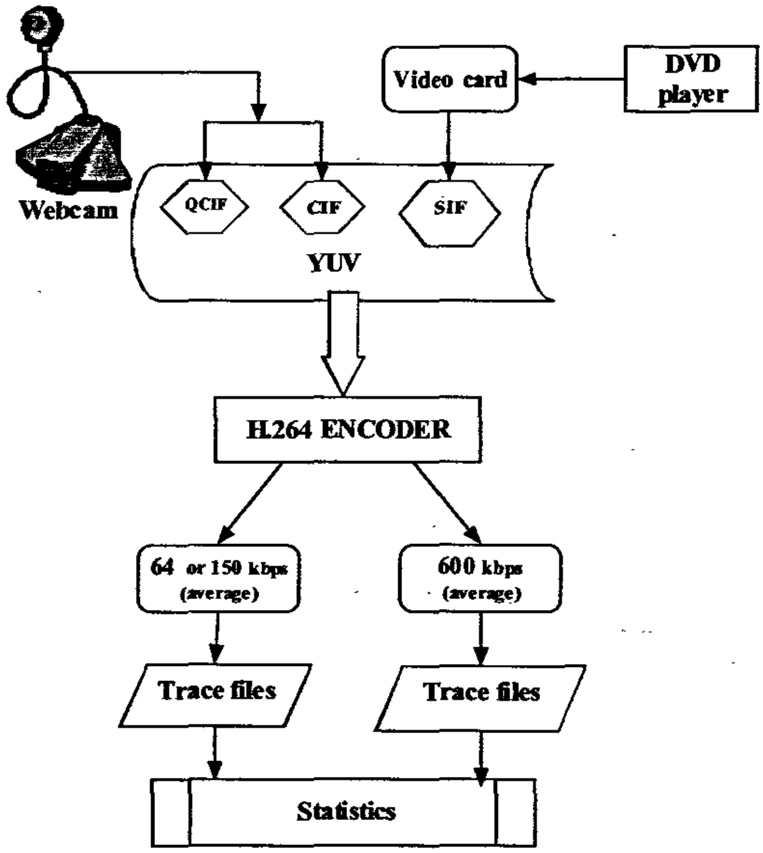


Figure 6.3 Video traffic generation

6.3.1.5 Encoding approach

The three YUV frame sequences are encoded into H.264 bit streams using MainConcept H.264/AVC encoder v2 software at two target bit rates: 64 kbps (or 150 for 'Mr Bones') (low) and 600 kbps (high). The video type is set to H.264 Baseline, stream type to program video, video mode to NTSC for 'talking head' and to PAL for 'Mr. Bones', video width to 176 for QCIF, 352 for CIF format ('the talking head') and 384 for 'Mr. Bones', video height to 144 for QCIF, 288 for CIF format and 'Mr. Bones', frame rate to 25 fps, frame/field encoding to progressive, bit rate mode to variable bit rate, and quantization is set to best. For low bit rate, bit rate is set to 64 kbps (or 150 for 'Mr Bones'), average bit rate to 64 kbps (or 150 for 'Mr Bones'), and maximum bit rate to

120 kbps (or 250 for 'Mr. Bones') and for high bit rate the bit rate is set to 600 kbps, average bit rate to 600 kbps and maximum bit rate to 1200 kbps.

6.3.1.6 Extraction of the properties of uncompressed YUV information

The summary of the details of the properties of the captured videos are given in Table 6.2 below.

Table 6.3a Statistical properties of uncompressed videos

Talking head CIF format	Talking head QCIF format	Mr. Bones SIF format
File Size : 892.905 Mbytes [Movie] Valid : Yes [AVI] Duration : 00:03:00 Movie complete : Yes [Video] Resolution : 352x288 Codec : MS YUV 4:2:2 FPS : 25.00 Video Sample Size: 16 bits	File Size : 366.656 Mbytes [Movie] Valid : Yes [AVI] Duration : 00:05:00 Movie complete : Yes [Video] Resolution : 176x144 Codec : MS YUV 4:2:2 FPS : 25.00 Video Sample Size: 16 bits	File Size : 754.294 Mbytes [Movie] Valid : Yes [AVI] Duration : 00:03:01 Movie complete : Yes [Video] Resolution : 384x288 Codec : BT YUV 4:1:1 FPS : 25.00 Video Sample Size: 12 bits

6.3.1.7 Extraction of the parameters of compressed video

The six encoded H.264 bit streams are analyzed using Elecard StreamEye tool, and the summary of the statistical properties of the encoded video is given in Table 6.3.

Table 6.3a Statistical properties of 'talking head' compressed video

	Encoded CIF (600 kbps avg)	Encoded CIF (64 kbps avg)	Encoded QCIF (600 kbps avg)	Encoded QCIF (64 kbps avg)
Video Stream Type	AVC/H.264	AVC/H.264	AVC/H.264	AVC/H.264
Resolution	352x288	352x288	176x144	176x144
Profile: Level	Baseline:31	Baseline:31	Baseline:31	Baseline:31
Aspect Ratio	12:11	12:11	12:11	12:11
Frames Count	4 501	4 501	7 502	7 502
Max Frame Size	24 576	5 264	11 092	4 324
Min Frame Size	0	188	188	188
File Size	13 740 036	4 016 244	47 567 572	6 717 428
Frame Rate Declared	25.00	25.00	25.00	25,00
Real	25.00	25.00	25.00	25.00

Table 6.3b Statistical properties of 'Mr. Bones' compressed video

	Encoded SIF (600 kbps avg)	Encoded SIF (150 kbps avg)
Video Stream Type	AVC/H.264	AVC/H.264
Resolution	384x288	384x288
Profile: Level	Baseline:31	Baseline:31
Aspect Ratio	1:1	1:1
Frames Count	4 532	4 532
Max Frame Size	43 008	14 336
Min Frame Size	0	0
File Size	13 824 004	3 483 652
Frame Rate Declared	25.00	25.00
Real	25.00	25.00

6.3.1.8 Statistical Analysis

Now that the videos has been captured and encoded, it is time to do statistical analysis of the captured uncompressed videos (YUV), encoded videos and their traces.

6.3.1.9 Statistical Analysis of YUV Information

With a frame rate of 25 frames/second, 16 bits quantization for QCIF and CIF, and 12 bits quantization for SIF, the bit rate of the YUV information is equal to

$$No.\text{ of pels/frame} \times No.\text{ of bits/pels} \times \text{frame rate} \qquad (6.9)$$

Therefore, for QCIF bit rate is

$$= 50688 \times 16 \times 25$$
$$= 20.275 Mbps$$

For CIF the bit rate is

$$= 202752 \times 16 \times 25$$
$$= 81.101 Mbps$$

For SIF the bit rate is

$$= 165888 \times 12 \times 25$$
$$= 49.766 Mbps$$

and the size of the captured video is equal to

$$No.\text{ of frames} \times size \text{ of 1 frame} \qquad (6.10)$$

where total number of frames captured will be the product of the duration of the video and the frame rate. That is,

$$No.\text{ of frames} = duration\ of\ \text{video} \times frame\ rate \qquad (6.11)$$

For CIF: Number of frames $= 180s \times 25 frames/s$

$= 4500 frames$

For QCIF: Number of frames $= 300s \times 25 frames/s$

$= 7500 frames$

For SIF: Number of frames $= 181s \times 25 frames/s$

$$= 4525\,frames$$

Re-arranging equation 6.10 and making size of the frame the subject of the formula, it is possible to determine the size of the frame for each format as follows:

$$size\ of\ the\ frame = \frac{Size\ of\ uncompressed\ YUV\ information\ in\ bytes}{Total\ number\ of\ frames}$$

For QCIF frame size is:

$$\frac{366.656 Mbytes}{7500}$$

$$= 48.887 kbytes$$

For CIF frame size is:

$$\frac{892.905 Mbytes}{4500}$$

$$= 198.423 kbytes$$

For SIF frame size is:

$$\frac{754.294 Mbytes}{4525}$$

$$= 166.695 kbytes$$

6.3.1.10 Statistical Analysis of H.264 and its Traces

After the YUV sequence is encoded the size of the QCIF video is reduced from 366.656 MB (YUV) to 6.717 MB (for 64 kbps average bit rate) and 47.567 MB (for 600 kbps average bit rate). For CIF video is reduced from 892.904 MB (YUV) to 4.016 MB (for 64 kbps average bit rate) and 13.740 MB (for 600 kbps average bit rate). Therefore, the compression ratio, which is defined as the ratio of the size of the entire uncompressed YUV video sequence to the size of the entire H.264 compressed video, is

$$YUV : H.264 = \frac{Size\ of\ uncompressed\ YUV}{Size\ of\ compressed\ H.264} \tag{6.12}$$

For 64 kbps (or 150 kbps for 'Mr Bones') the average video compression is:

$$\mathrm{YUV : H.264_{QCIF}} = \frac{366.656}{6.717} \text{ for QCIF}$$
$$= 54.586$$

$$\mathrm{YUV : H.264_{CIF}} = \frac{892.904}{4.016} \text{ for CIF}$$
$$= 222.337$$

$$\mathrm{YUV : H.264_{SIF}} = \frac{754.294}{3.484} \text{ for SIF}$$
$$= 216.502$$

For 600 kbps average video compression:

$$\mathrm{YUV : H.264_{QCIF}} = \frac{366.656}{47.567} \text{ for QCIF}$$
$$= 7.708$$

$$\mathrm{YUV : H.264_{CIF}} = \frac{892.904}{13.740} \text{ for CIF}$$
$$= 64.986$$

$$\mathrm{YUV : H.264_{SIF}} = \frac{754.294}{13.824} \text{ for SIF}$$
$$= 54.564$$

The sample mean $\overline{X}$ of a frame size trace is

$$\overline{X} = \frac{1}{N}\sum_{n=0}^{N-1} X_n$$

where N is the number of frames of the encoded video. Using data analysis tool (within Excel) to perform analysis of the generated traces parameters gives the following mean values:

For QCIF (600 kbps): $\overline{X} = 6340.602 bytes$

For QCIF (64 kbps): $\overline{X} = 895.368 bytes$

For CIF (600 kbps): $\overline{X} = 3052.205 bytes$

For CIF (64 kbps): $\overline{X} = 892.217 bytes$

For SIF (600 kbps): $\overline{X} = 3049.855 bytes$

For SIF (150 kbps): $\overline{X} = 768.224 bytes$

The sample variance S_X^2 of the frame size trace is

$$S_X^2 = \frac{1}{N-1}\left[\sum_{n=0}^{N-1} X_n^2 - \frac{1}{N}\left(\sum_{n=0}^{N-1} X_n\right)^2\right]$$

Using data analysis tool (within Excel) to perform analysis of the generated traces parameters gives the following simple variance values:

For QCIF (600 kbps): $S_X^2 = 212112.388$

For QCIF (64 kbps): $S_X^2 = 297303.233$

For CIF (600 kbps): $S_X^2 = 4944454.935$

For CIF (64 kbps): $S_X^2 = 576886.609$

For SIF (600 kbps): $S_X^2 = 6535797.102$

For SIF (150 kbps): $S_X^2 = 1825736.376$

The coefficient of variation CoV_X is given by

$$CoV_X = \frac{S_X}{\overline{X}}$$

Using the simple variance and mean values above equation produces the following coefficients of variation:

For QCIF (600 kbps): $CoV_X = 0.073$

For QCIF (64 kbps): $CoV_X = 0.609$

For CIF (600 kbps): $CoV_X = 0.729$

For CIF (64 kbps): $CoV_X = 0.851$

For SIF (600 kbps): $CoV_X = 0.838$

For SIF (150 kbps): $CoV_X = 1.759$

The maximum frame size X_{max} is

$$X_{\max} = \max_{0 \le n \le N-1} X_n$$

For QCIF (600 kbps): $X_{\max} = 11.092 kbytes$

For QCIF (64 kbps): $X_{\max} = 4.324 kbytes$

For CIF (600 kbps): $X_{\max} = 24.576 kbytes$

For CIF (64 kbps): $X_{\max} = 5.264 kbytes$

For SIF (600 kbps): $X_{\max} = 43.008 kbytes$

For SIF (150 kbps): $X_{\max} = 14.336 kbytes$

The display time or frame period of a given frame is given by the ratio of the captured video runtime to the number of video frames in a given trace; that is,

$$t = \frac{Captured\ video\ runtime}{No.\ of\ \ frames\ in\ a\ given\ trace} \tag{6.13}$$

From Table 6.2 the video runtime is 5 minutes (300 seconds) for QCIF, 3 minutes (180 seconds) for CIF and 3 minutes, 1second (181 seconds) for SIF, and from Table 6.3 the number of frames is 4501 for both of the CIF traces, 7502 for both of the QCIF traces and

4532 for both of SIF traces. Therefore substituting these values in Equation 6.13 will produce the following frame periods:

For QCIF (600 kbps): $t = 39.989\ ms$

For QCIF (64 kbps): $t = 39.989\ ms$

For CIF (600 kbps): $t = 39.991\ ms$

For CIF (64 kbps): $t = 39.991\ ms$

For SIF (600 kbps): $t = 39.938\ ms$

For SIF (64 kbps): $t = 39.938\ ms$

The above statistical properties of the generated traces are summarized in Table 6.4 below, and illustration is provided by plots in Figure 6.4 and 6.5.

Table 6.4 Overview of the statistical summary of the encoded video frame sizes

Video Format	Average Bit Rate	Compression Ratio YUV:H.264	Mean ($\overline{X}$)	CoV_X $S_X/\overline{X}$	Peak/ Mean $X_{max}/\overline{X}$	Mean Bit Rate $\overline{X}/t$ [kbps]	Peak Bit Rate X_{max}/t [kbps]
QCIF	600	7.708	6340.602	0.073	1.749	158.559	277.376
QCIF	64	54.586	895.368	0.609	4.829	22.390	108.130
CIF	600	64.986	3052.205	0.729	8.052	76.322	614.538

CIF	64	222.337	892.217	0.851	5.900	22.310	131.630
SIF	600	54.564	3049.855	0.838	14.102	76.364	1076.863
SIF	150	216.502	768.224	1.759	18.661	19.235	358.956

Another observation from all the generated four video traces is that the encoded video has two types of frame formats: the intra-coded (I), and the forward motion predicted (P). These I and P frames are arranged in a fixed deterministic pattern IPPPPPPPPPPPP, which is called Group of Pictures (GoP), as shown in Table 6.5 below.

Table 6.5 Excerpt of a H.264 trace file of 'talking head' encoding (QCIF 600 kbps)

Frame number	type	offset	time	Frame size
0	I	0x0000000000000178	00:00.0	11092
1	P	0x0000000000002ccc	00:00.0	1880
2	P	0x0000000000003424	00:00.1	564
3	P	0x0000000000003658	00:00.1	188
4	P	0x0000000000003714	00:00.2	376
5	P	0x000000000000388c	00:00.2	376
6	P	0x0000000000003a04	00:00.2	564
7	P	0x0000000000003c38	00:00.3	376
8	P	0x0000000000003db0	00:00.3	376
9	P	0x0000000000003f28	00:00.4	564
10	P	0x000000000000415c	00:00.4	564
11	P	0x0000000000004390	00:00.4	564
12	P	0x00000000000045c4	00:00.5	564

Comparing the 600 kbps target rate encodings for both formats with 64 kbps target rate encodings it can be observed in Table 3 above that lower target rate encodings have higher compression ratios than their corresponding higher target rate encodings. However, the coefficient of variation (CoV_X) values in Table 6.4 and plots in Figure 6.3 reveal that this efficiency comes at the expense of larger variability of frame sizes; that is, the higher the compression rate, the larger the variability of the frame sizes. This is also reflected by the histogram plots in Figure 6.4. It can also be seen from the plots in Figure

6.3 that the first frame in all the plots contains a large number of bits. This is because there is no previous frame to predict from and each macroblock is intracoded without any prediction.

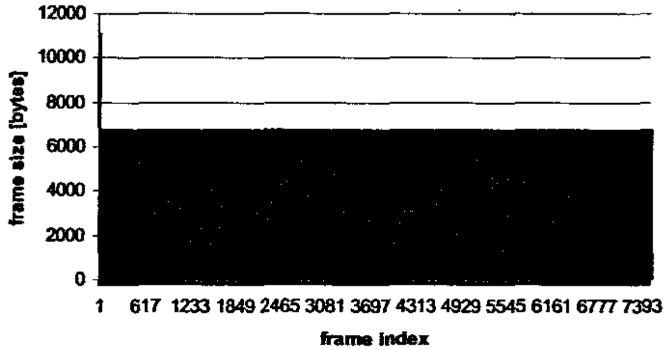


Figure 6.4(a) Frame sizes of encoded 'Talking head' (QCIF 600 kbps)

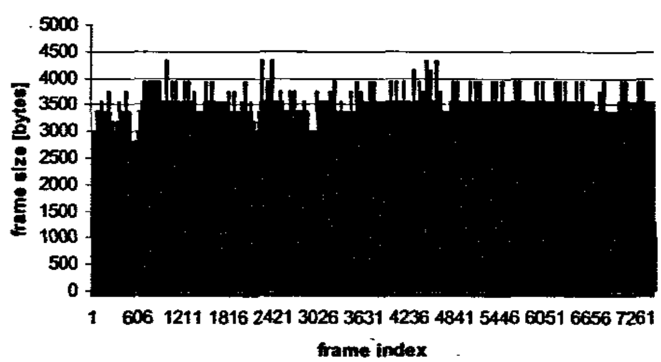


Figure 6.4(b) Frame sizes of encoded 'Talking head' (QCIF 64 kbps)

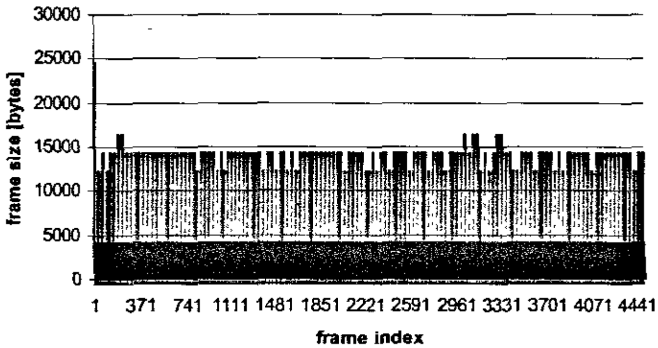


Figure 6.4(c) Frame sizes of encoded 'Talking head' (CIF 600 kbps)

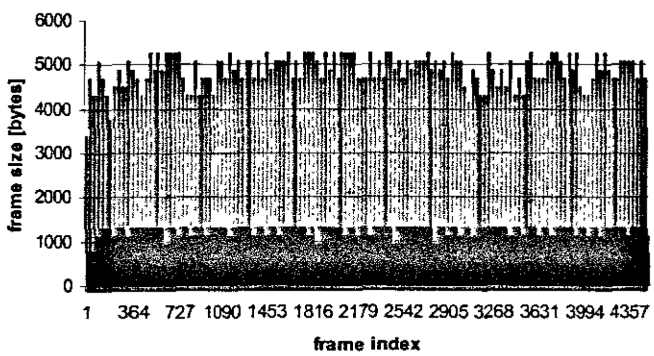


Figure 6.4(d) Frame sizes of encoded 'Talking head' (CIF 64 kbps)

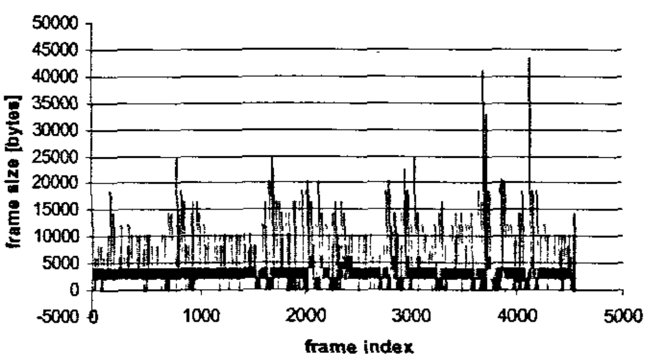


Figure 6.4(e) Frame sizes of encoded 'Mr. Bones' (SIF 600 kbps)

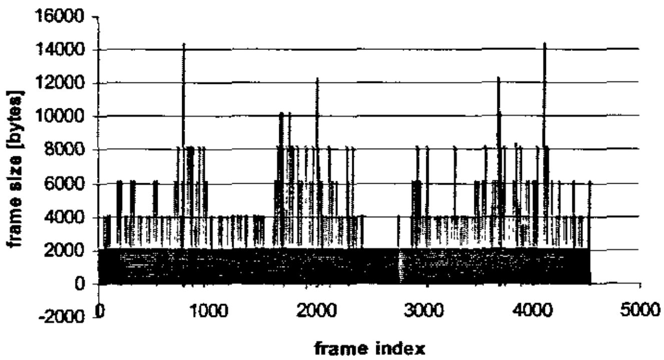


Figure 6.4(f) Frame sizes of encoded 'Mr. Bones' (SIF 150 kbps)

Figure 6.4 Frame size traces of the encoded 'Talking head' and 'Mr. Bones'

Figure 6.4 above gives the plots of the frame size X_n (in bytes) as a function of the frame number n. It can be observed from the plots that QCIF encoding at high target rate (600 kbps) is relatively smooth for "talking head', while the encoding at low target rate (64 kbps), on the other side, exhibits extreme changes in the frame sizes.

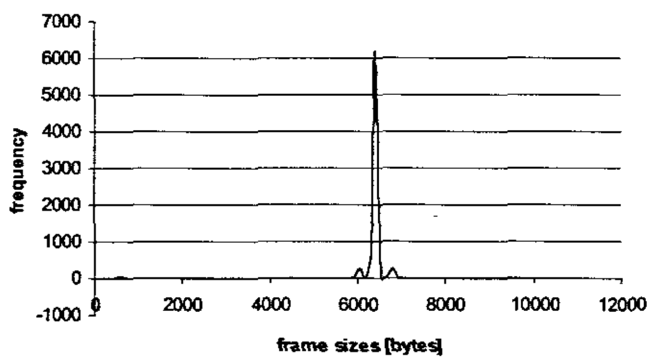


Figure 6.5(a) Frame size histogram of encoded 'Talking head' (QCIF 600 kbps)

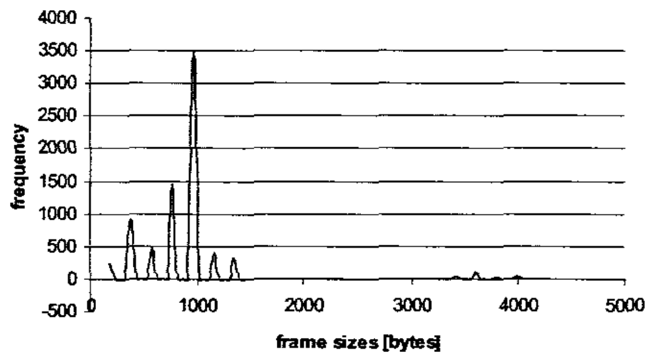


Figure 6.5(b) Frame size histogram of encoded 'Talking head' (QCIF 64 kbps)

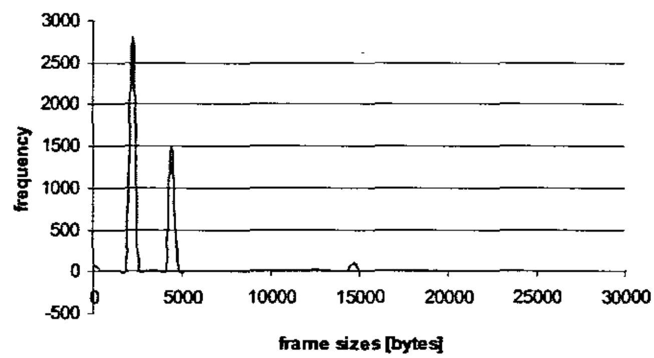


Figure 6.5(c) Frame size histogram of encoded 'Talking head' (CIF 600 kbps)

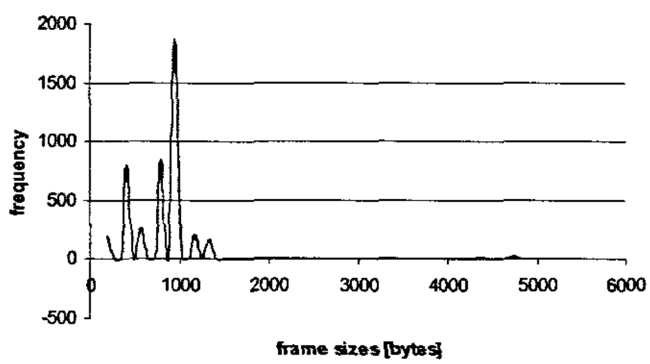


Figure 6.5(d) Frame size histogram of encoded 'Talking head' (CIF 64 kbps)

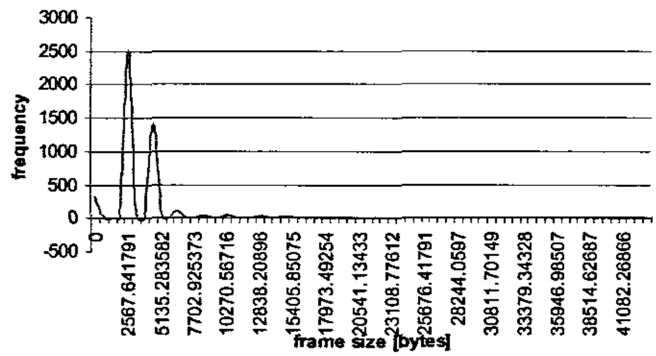


Figure 6.5(e) Frame size histogram of encoded 'Mr. Bones' (SIF 600 kbps)

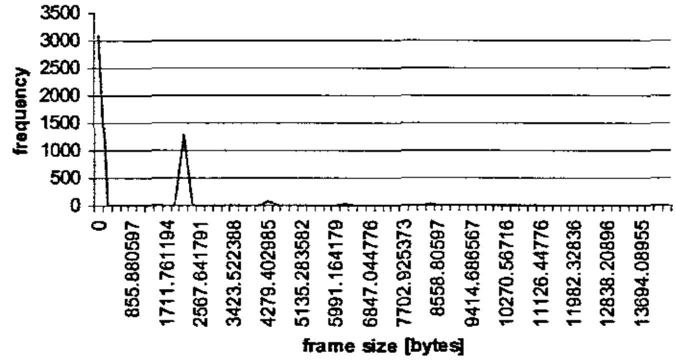


Figure 6.5(f) Frame size histogram of encoded 'Mr. Bones' (SIF 150 kbps)

Figure 6.5 Frame size histograms of the encoded 'Talking head' and 'Mr. Bones'

The histogram plots above reflect again that the high compression ratios result in more variability of the encoded video stream.

Figure 6.6 gives the autocorrelation coefficient $?_x(k)$ of the frame size sequence X_n, $n = 1, 2, 3, \ldots, N$, as a function of the lag k in frames.

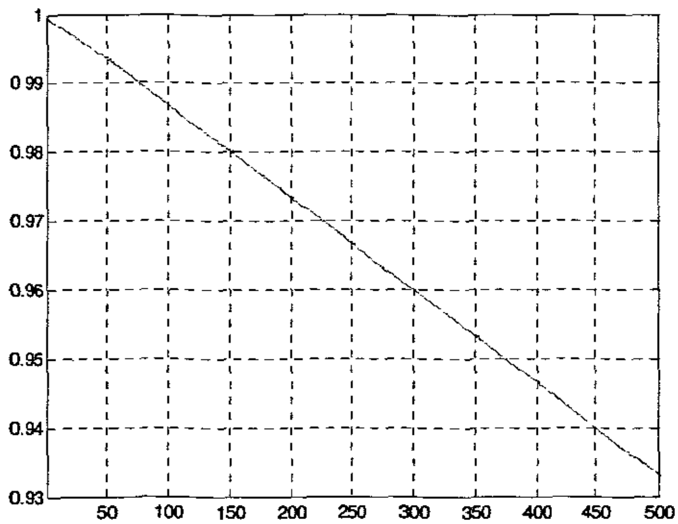


Figure 6.6(a) Autocorrelation of frame size trace (QCIF 600 kbps)

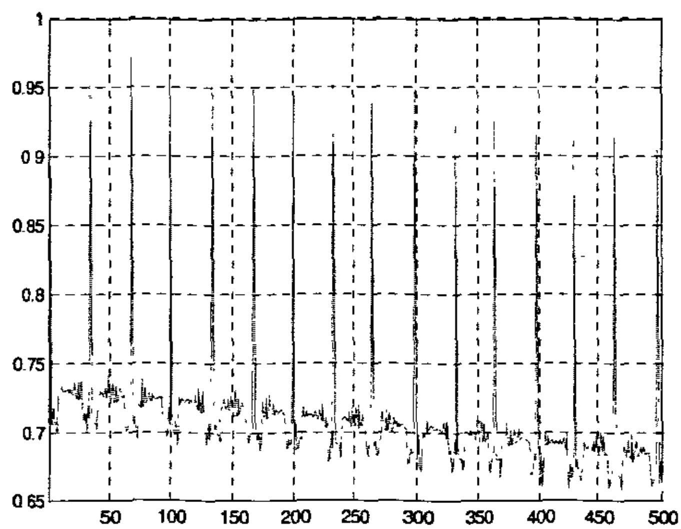


Figure 6.6(b) Autocorrelation of frame size trace (QCIF 64 kbps)

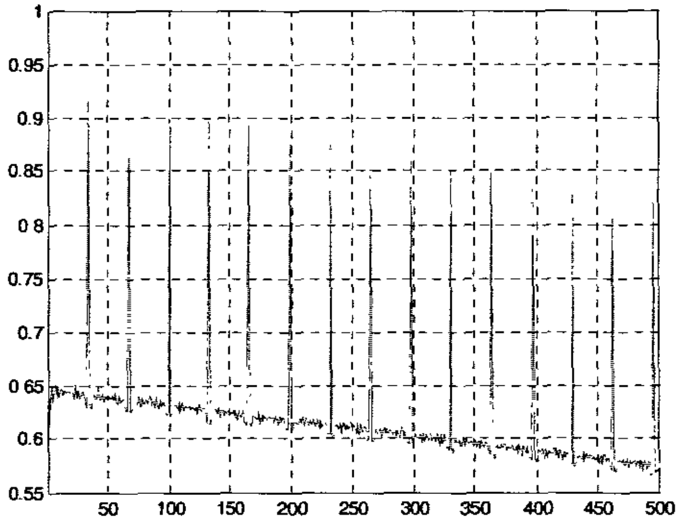


Figure 6.6(c) Autocorrelation of frame size trace (CIF 600 kbps)

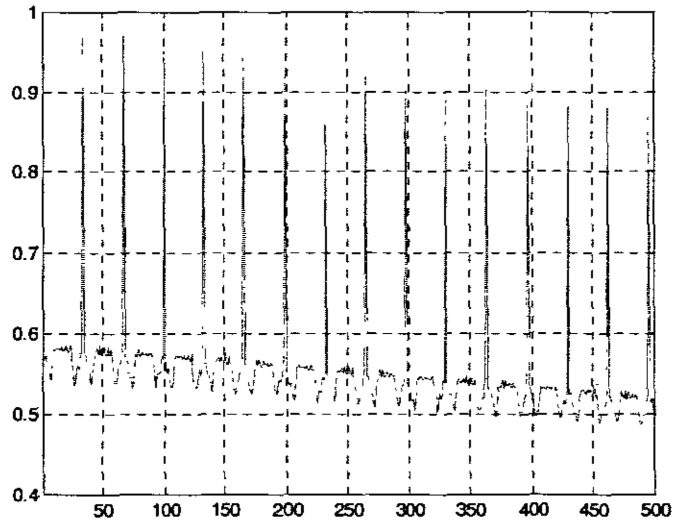


Figure 6.6(d) Autocorrelation of frame size trace (CIF 64 kbps)

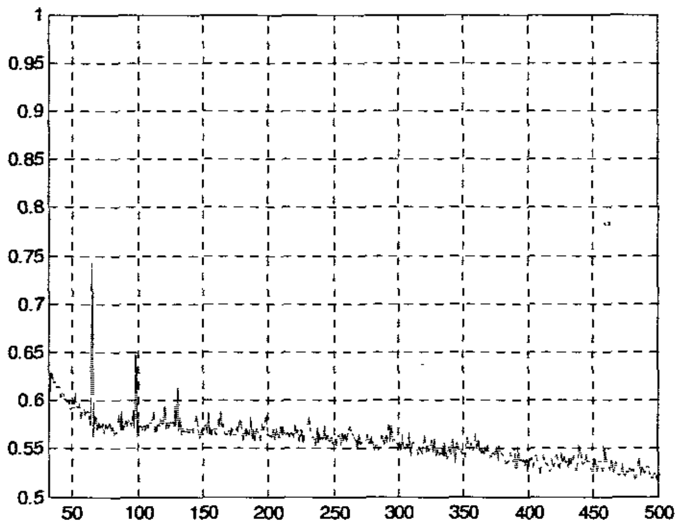


Figure 6.6(e) Autocorrelation of frame size trace (SIF 600 kbps)

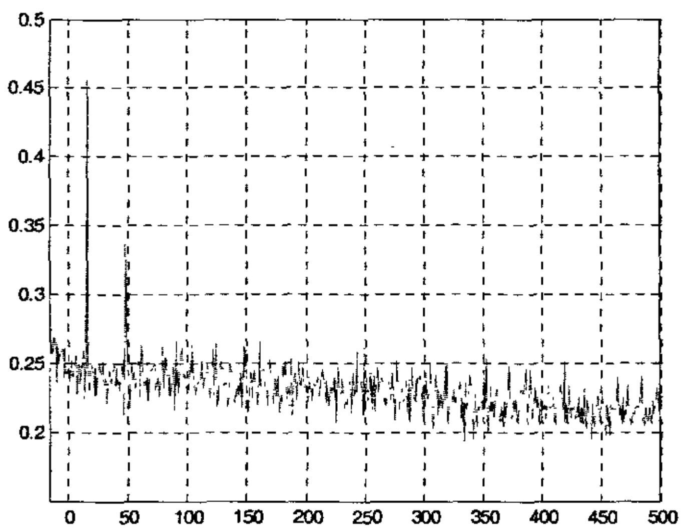


Figure 6.6(f) Autocorrelation of frame size trace (SIF 150 kbps)

Figure 6.6 Autocorrelation of traces of the encoded 'Talking head' and 'Mr. Bones'

From the autocorrelation function plots of the frame sizes, in Figure 6.5, it is observed that the autocorrelation of the frame size series $\{X_0, X_1, X_2, ..., X_{N-1}\}$ consist of periodic spikes pattern, which is superimposed on a decaying curve. The periodic spike pattern reflects the repetitive GoP pattern, where the large spikes represent I frames and the subsequent small spikes represent P frames. The decaying slope is a characteristic of long term correlations in the encoded video (Fitzek and Reisslein, 2001). It can also be observed from the autocorrelation plots that the correlation structure is significant for relatively small lags, which is a sign of short-range dependence (Adas, 1997).

To assess the long-range dependence properties of the encoded video sequences, the Hurst parameter must be determined (Fitzek and Reisslein 2001; Tatipamula and Khasnababish 1998). There are several methods for determining the Hurst parameter, however, the most commonly used ones includes the R/S statistic plot, Variance-time plot, Periodogram plot (Seeling, Reisslein and Kulapala 2004; Fitzek and Reisslein 2001; Droz and LeBoudec 1996). According to Fitzek and Reisslein (2001), Hlavacs, Kotsis and Steinkellner (1999), Tatipamula and Khasnababish (1998, p.136), and Droz and LeBoudec (1996), and Lan and Heidemann (2002), the Hurst parameters between 0.5 and 1 indicate long-range dependence or self-similarity. The traffic which exhibits this long-range dependence or self-similarity is usually referred to as fractal traffic (Tatipamula and Khasnababish 1998, p.135; Leland, 1994)

For this study the variance-time plot method, which is a heuristic graphical method to estimate the Hurst parameter, has been used. This method relies on the fact that a self-similar process has slowly decaying variances. The variance-time plot is obtained by plotting the normalized variance of the aggregated trace $S_X^{2(a)}/S_X^2$ as a function of the aggregated level a in a log-log plot as shown in Figure 6.6. Where the variance of the aggregated trace is given by

$$S_X^{2(a)} = \frac{1}{(N/a)-1} \sum_{n=0}^{(N/a)-1} \left(X_n^{(a)} - \overline{X}\right)^2 \tag{6.14}$$

and the aggregated frame size series $X_n^{(a)}$ is derived from the original trace X_n by averaging it over non-overlapping blocks of length a as follows

$$X_n^{(a)} = \frac{1}{a} \sum_{j=na}^{(n+1)a-1} X_j \text{, for } n = 0, 1, 2, \ldots (N/a)-1 \qquad (6.15)$$

Once the plot is obtained, the slope of the linear part of the variance-time plot is estimated using a least squares fit, and thereafter, the Hurst parameter is then estimated as

$$H = 1 - \frac{|\beta|}{2} \qquad (6.16)$$

where H is the Hurst parameter and β is the slope.

For all of the variance time plots in Figure 6.6, the initial value of a is 12 and then each iteration is multiplied by 2; that is, $a = \{12, 24, 48, 96, \ldots, 6144\}$. The plots are generated using Matlab and they are shown in Figure 6.7, the straight line is used to estimate the slope of the variance plot, and the values of the Hurst parameter are given in Table 6.6

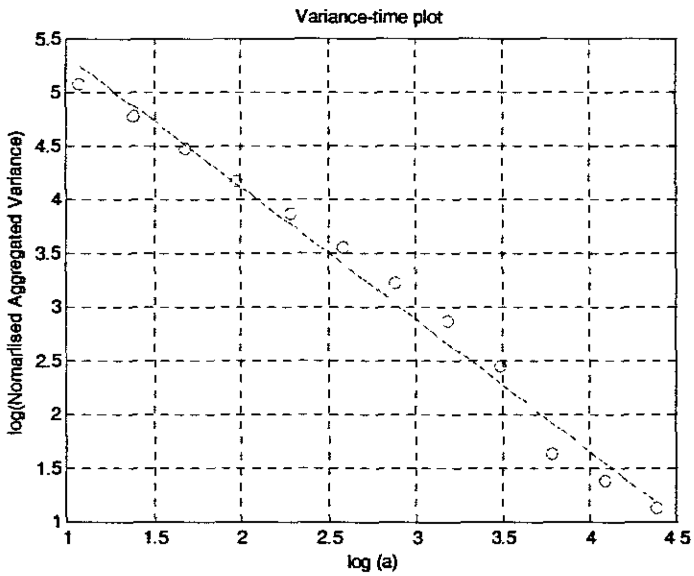


Figure 6.7(a) Variance-time plot of 'talking head' (QCIF 600 kbps)

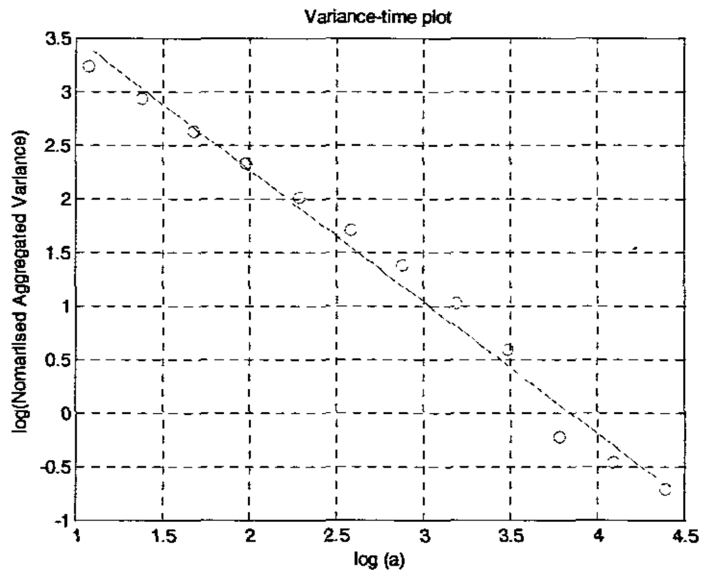


Figure 6.7(b) Variance-time plot of 'talking head' (QCIF 64 kbps)

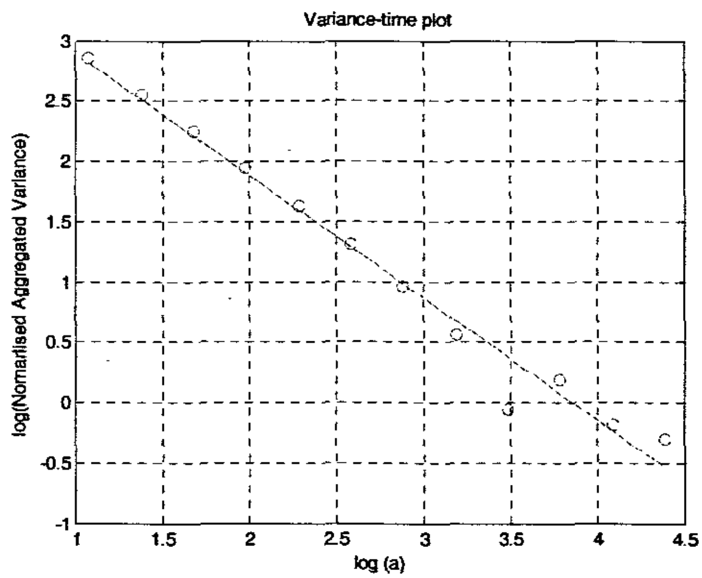


Figure 6.7(c) Variance-time plot of 'talking head' (CIF 600 kbps)

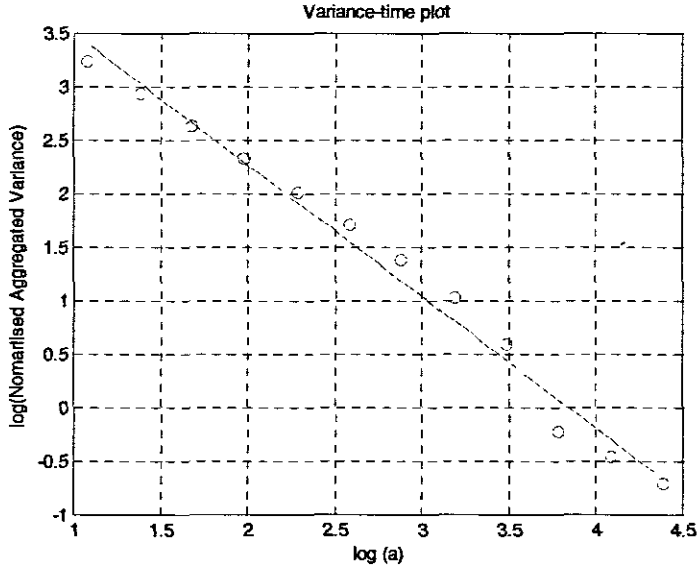


Figure 6.7(d) Variance-time plot of 'talking head' (CIF 64 kbps)

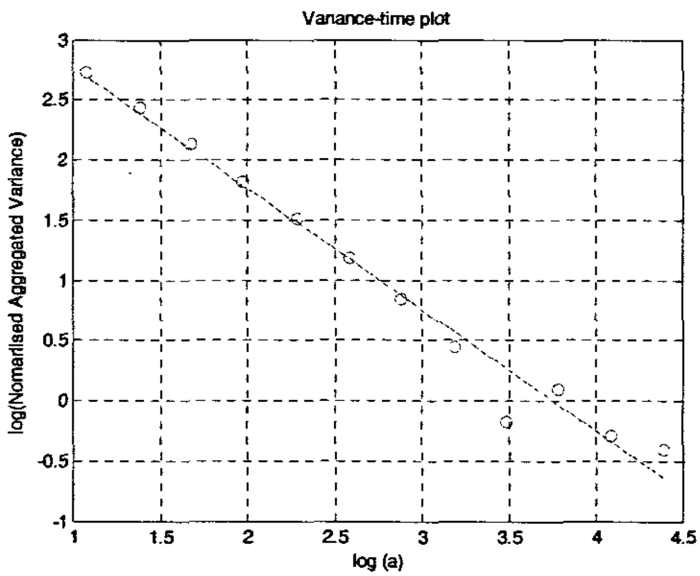


Figure 6.7(e) Variance-time plot of 'Mr. Bones' (SIF 600 kbps)

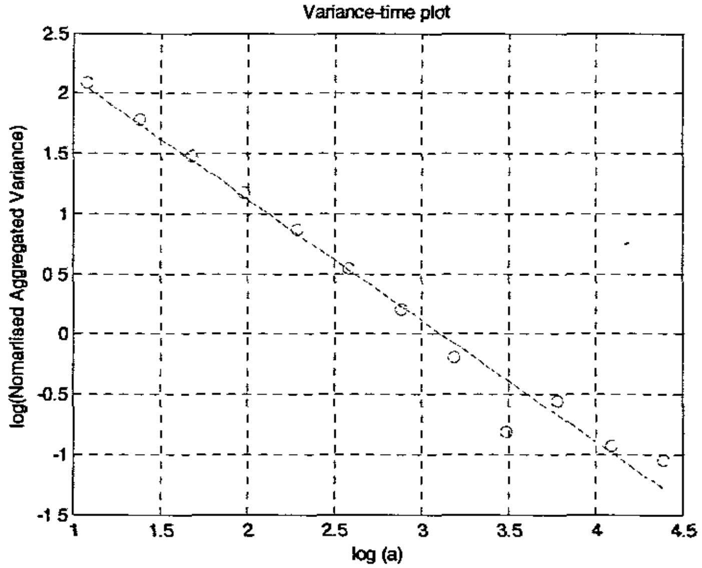


Figure 6.7(f) Variance-time plot of 'Mr. Bones' (SIF 150 kbps)

Figure 6.7 Aggregated variance-time plots for 'Talking head' and 'Mr. Bones'

Table 6.6 Slope and Hurst parameters

Video format	Average Bit rate	Slope (ß)	Hurst parameter (H)
QCIF	600	-0.763	0.619
QCIF	64	-0.763	0.619
CIF	600	-0.838	0.581
CIF	64	-0.763	0.619
SIF	600	-0.841	0.579
SIF	150	-0.841	0.579

All the values of the slope are greater than -1, which according to Hlavacs, Kotsis and Steinkellner (1999), Marot and Kotsis (1999), Lan and Heidemann (2002), and Gospodinov and Gospodinova (2005) indicates self-similarity when using a variance-time plot method. The values of the estimates of Hurst parameter are between 0.5 and 1, which also confirms the self-similarity or 'fractality' of the encoded video sequences.

6.3.1.11 Building a Video Traffic Model

Now that the captured encoded video is analyzed and characterized, its traffic model can be built. Many studies have been conducted in the area of video source traffic modeling and many models have been developed for VBR video sequences. Traditional models based on Markovian structures have been used to statistically approximate VBR video traffic (Huang, Devetsikiotis, Lambadaris, and Kaye, 1995). However, most video source traffic modeling research has fallen behind the rapid advances in video coding techniques, especially H.264 (Dai and Loguinov, 2005). So in this study the focus is going to be on developing the model for H.264 video sequences.

The traffic models that are used for VBR video sequence can be divided into three categories: Markov Chains, autoregressive processes, and self-similar models (Maglaris et al., 1988, Beran et al., 1995, Blondia and Casals, 1992; Frost and Melamed, 1994). Markov process can be used to model processes, where the observation depends on only the previous observed value. For example, user behavior, system/network state change/failure, and network traffic if the observed traffic shows no correlation.
Autoregressive models define the next random variable in a sequence X_n as an explicit function of previous ones within a time window stretching from present to past. Autoregressive models are suitable for modeling short-range dependencies, but fail to model long-range dependencies, which is common in VBR-coded video, web and Ethernet traffic (Hlavacs, Kotsis and Steinkellner, 1999).

In Figure 6.5, 6.6 and 6.7 plots of the distribution of frame sizes, autocorrelation, and variance-time were presented, respectively. The autocorrelation captured the dependencies between the frame sizes within each video sequence. Variance plots estimate the Hurst parameter, which is measure of long-range dependence or self-similarity. From these plots of real traffic streams, it has been observed that VBR video traffic streams exhibit the phenomenon referred to as short-range dependence (SRD) as well as long-range dependence (LRD) or self-similarity, which is in agreement with the findings by Garret and Willinger (1994), Beran *et al.* (1995), Ma and Ji (2001), and Dai and Loguinov (2005). The traffic, which behaves in this way, is usually referred to as fractal traffic, and according to Huang, Devetsikiotis, Lambadaris, and Kaye (1995), fractal traffic cannot be sufficiently represented by traditional models, but instead can be more accurately matched by self-similar or fractal models.

6.3.1.12 Self-similar models

The empirical data from the encoded video statistics shows that the real video traffic has a variable bit rate and exhibit self-similar characteristics. Hence the model that will be suitable for VBR video must be the one which exhibits the self-similar properties. Self-similar traffic source models include Fractional Gaussian Noise (FGN), Fractional Brownian Motion (FBM), Linear Fractional Stable Motion (LFSM), and asymptotically self-similar Fractional Autoregressive Integrated Moving-Average (F-ARIMA, sometimes called ARFIMA). While the first three models are used mainly to generate pure self-similar signals (Garret and Willinger 1994; Casilari, Reyes, Díaz-Estrella, and Sandoval, 1998), the F-ARIMA models' advantage is that, unlike other models which only capture the LRD and not SRD, they can model both long time dependence and short time dependence; that is, LRD and SRD at the same time (Huang, Devetsikiotis, Lambadaris, and Kaye, 1995; Harmantzis and Hatzinakos, 2005). Since it was also found from the empirical data that the VBR video also possesses significant SRD, thus, using either an LRD or SRD model alone does not provide satisfactory results (Dai and Loguinov, 2005). Thus, a more suitable model will be the one that is able to capture both SRD and LRD properties of VBR video such as F-ARIMA model.

Fractional ARIMA or F-ARIMA came as a natural extension of the standard Autoregressive Integrated Moving-Average (ARIMA) model (Leland, Taqqu, Willinger and Wilson, 1993), which was popularized by Box and Jenkins (1994). F-ARIMA is built on classical ARIMA model by allowing the degree of differencing d in ARIMA(p, d, q) to take non-integer values (Harmantzis and Hatzinakos, 2005). According to Harmantzis and Hatzinakos (2005), the F-ARIMA model is defined by

$$X_t = \Phi(B)^{-1}\Theta(B)\Delta^{-d} \in_t \qquad (6.17a)$$

Rearranging, equation 6.17a can be rewritten as follows

$$\Phi(B)X_t = \Theta(B)\Delta^{-d} \in_t \qquad (6.17b)$$

where d is the fractional difference parameter, which can be derived from the Hurst parameter (Casilari, Ríos, and Sandoval, 2000; Harmantzis and Hatzinakos, 2005) as follows

$$d = H - \frac{1}{2} \qquad (6.18)$$

F (B) and *T (B)* are the p^{th} and q^{th} degree polynomials

$$\Phi(B) = 1 - \sum_{j=1}^{p} \phi_j B^j \qquad (6.19a)$$

and

$$\Theta(B) = 1 + \sum_{j=1}^{q} \phi_j B^j \qquad (6.19b)$$

representing autoregressive (AR) and moving-average (MA) components respectively, ϕ_j are the coefficients of the filter, and B is the backward operator defined by

$$BX_t = X_{t-1}, B^2 X_{t-2} = X_{t-2}, \ldots, \qquad (6.20)$$

B^j denotes the backward operator iterated j times, and ? is the difference operator that is defined by

$$\Delta X_t = X_t - X_{t-1} = (1 - B)X_t \qquad (6.21)$$

$?^d$ is ? iterated d times, and is equal to $(I - B)^d$.

$$\Delta^{-d} = (I - B)^{-d} = \sum_{j=0}^{\infty} b_j(-d)B^j \tag{6.22}$$

where b_j are moving average coefficients with $b_0(-d) = 1$ and

$$b_j(-d) = \prod_{k=1}^{j} \frac{k+d-1}{k}$$

$$= \frac{\Gamma(j+d)}{\Gamma(d)\Gamma(j+1)}, \ j = 1, 2, \ldots, \tag{6.23}$$

Graphically the F-ARIMA(p, d, q) process is illustrated in Figure 6.8 below.

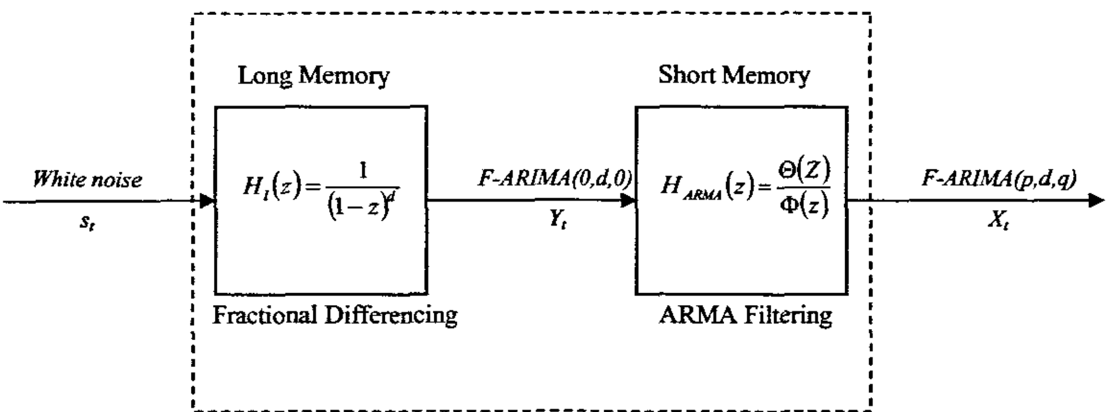


Figure 6.8 Linear model for the F-ARIMA(p, d, q) process

6.3.2 Voice Traffic Source

For voice to be transmitted over a digital communication system it needs to be digitally encoded into PCM or any of the low bit rate technique. At source level, voice is intermittent; that is, it is characterized by periods of activity (ON), called talk spurts, and periods of inactive (OFF), called silence. During the ON period, the source sends packets at regular intervals of length T (packetization time). The duration of talk spurt and silence periods is generally estimated by independent and exponential distributed parameters a and ß. The traffic generated by voice applications is usually characterised by an ON/OFF Process (Staehle, Leibnitz and Tran-Gia, 2001; Ji and Asano, 2000) and can be modeled as a two state Markovian ON-OFF model as shown in Figure 6.9 below (Ghanbari, Hughes, Sinclair and Eade, 1997, p. 240; Hassan, Garcia and Brun, 2005) with the following parameters:

- The mean duration of the ON period:

$$T_{ON} = \frac{1}{\alpha} \tag{6.24}$$

- The mean duration of the OFF period:

$$T_{OFF} = \frac{1}{\beta} \tag{6.25}$$

- The constant packet transmission rate during the on period:

$$\lambda = \frac{1}{T} \tag{6.26}$$

Where a is the parameter of exponential law of the activity period ON, and β is the parameter of exponential law of the inactivity period OFF.

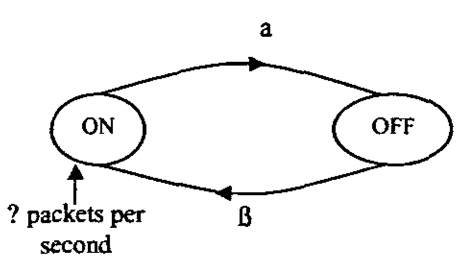


Figure 6.9 Two-state Markov diagram to model speech or VoIP

According to Hassan, Garcia and Brun (2005), the mean rate of the source (in packets per second) on all states is given by

$$\begin{aligned}\bar{\lambda} &= \frac{T_{ON}}{(T_{ON}+T_{OFF})T} \\ &= \frac{\beta}{(\alpha+\beta)T}\end{aligned} \tag{6.27}$$

Packet sizes and inter-arrivals depend on the codec used for voice encapsulation. For example, Table 6.7 below shows some of the values for three different codecs: G711, G726 and G729, respectively, with durations of ON and OFF periods taken from measures by Deng (1995).

Table 6.7 Speech model parameters by codec

CODEC	Packet Size (Bytes)	Packet Int-arrival (ms)	T_{ON} (sec)	T_{OFF} (sec)
G711	136	12	0.352	0.65
G726	104	16	0.352	0.65
G729	70	30	0.352	0.65

6.3.3 Data Traffic Source

Traditionally traffic source models have been based on a well-known Poisson process, which introduces to the use of exponential distribution that is of the form $f(x) = \lambda e^{-\lambda x}$. However, according to Staehle, Leibnitz and Tran-Gia (2001), and Leppänen, Prokkola and Bräysy (2003), models based on Poisson process do not fully represent the modern data traffic, which is highly variable and exhibits burstiness over a wide range of time scales; that is, having a fractal like behavior. Hence long- and heavy-tailed distributions, such as Pareto-distribution, are used in modeling modern data traffic (Staehle, Leibnitz and Tran-Gia, 2001; Leppänen, Prokkola and Bräysy, 2003; Prokkola, Leppänen and Bräysy, 2003; Hassan, Garcia and Brun, 2005).

According to Staehle, Leibnitz and Tran-Gia (2001), IEEE Working Group 802.20 and Shankaranayanan (2003) TCP data traffic may include web-browsing (HTTP), E-mail, and FTP as shown in Figure 6.10. However, nowadays FTP application constitute less than 10 % of the total TCP traffic because file transfers are now increasingly performed via HTTP (Klemm, Lindermann and Lohmann, 2001).

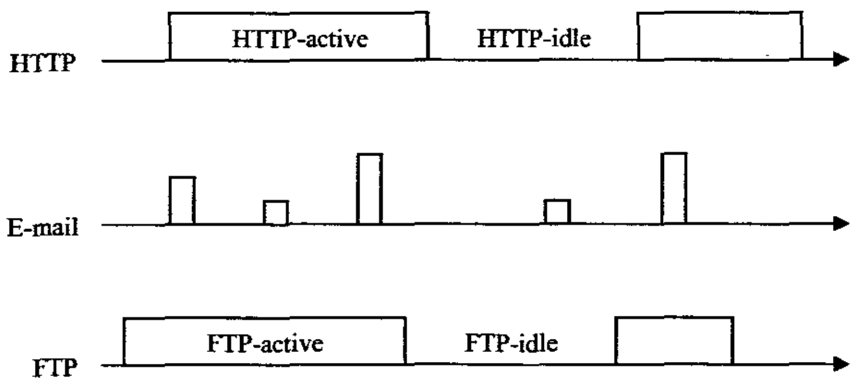


Figure 6.10 Speech model parameters by codec

As it can seen on the figure above, most of the data applications activity phases consists of a sequence of ON- (active) and OFF- (idle) phases with some application specific length or data volume distribution, respectively. Where the ON-state is when the user is using the application, and OFF is during the idle period. Thus, it can be said that the data traffic source can be assumed to have an ON-OFF behaviour, with each file transmission time Pareto distributed, and the thinking time between the transmissions is Lognormal. A user may run different applications that may be concurrently active, for example, web-browsing while downloading a file.

6.4 MODELING NETWORK FUNCTIONS

In chapter two a model of rural area network was proposed, and one of the main features of this network is being multiservice so that it can be able to support multimedia traffic. In this section the network requirements of multimedia traffic is presented together with tools for providing QoS. Multimedia traffic includes voice, video and interactive data applications, with each application having its own requirements in terms of data rates, variability, and limits on latency, jitter and packet loss.

Voice encoders produce data at relatively low bit rate, which are ranging from 5kbps to 64 kbps, depending on the encoding scheme. Table 6.8 is a summary of the different ITU-T recommendations, their nominal bit rates, one-way codec delay, and mean opinion scores (MOS) that specify the speech quality on a 5-point scale (where 1 is bad, 2 is poor, 3 is fair, 4 is good, and 5 is excellent) (Ibe, 2002, p.125).

Table 6.8 ITU-T Recommendation for Voice Compression

Standard	Bit rate (kbps)	Delay (ms)	Quality (MOS)
G.711	64	0.125	4.3
G.722	48/56/64	0.125	4.1
G.723.1	5.3/6.3	67-100	4.1
G.726	16/24/32/40	0.125	2.0-4.3
G.728	16	2	4.1
G.729	8	25-35	4.1
G.729A	8	25-35	3.4
GSM6.10	13	20	3.7

For the network to provide quality voice service, packet delay and loss must meet stringent requirements, such as maximum round trip time of 200 to 300 ms. That is, the one-way delay incurred in voice encoding, packetization, network transit time, dejittering and decoding must be kept below 100 to 150 ms. Jitter must also be limited to about 50 ms in order to ensure smooth playback at the receiver (Gruber and Strawczynski, 1985; ITU-T Recommendation G.114, 1996 and 1998; Sze, Liew, Lee and Yip, 2002; Cisco, 2002; Tobagi, 2005). Loss should be no more than 1%; guaranteed priority bandwidth per call must be 21 to 106 kbps (depending on the sampling rate, codec and Layer 2 overhead), and a guaranteed bandwidth of 150bps (plus Layer 2 overhead) per phone is required for voice control traffic. The total bandwidth consumed by voice over IP streams is calculated by adding the packet payload and all the headers (in bits), then multiplying by the packet rate per second (default of 50 packets per second), resulting in a bandwidth ranging from 21 kbps to 106 kbps, depending on codec, the sampling rate and the Layer 2 media used (Cisco, 2002).

Video encoders produce a video traffic that is stream-oriented with data rates ranging from tens of kbps to tens of Mbps. The characteristics of the encoded video vary tremendously according to the content, the video compression scheme and video encoding control scheme. For example, more complex scenes and more frequent scene changes produce large bit rates and more variability than video streams of talking heads. Different video compression schemes, such as H.261, H.263, MPEG 1, MPEG 2, MPEG 4, and H.264 are designed to meet different objectives and therefore have different bit rates and stream characteristics. Video characteristics are also affected by the video encoding control scheme used. For example, for a given content and compression scheme, the control scheme can be either constant bit rate (CBR) or variable bit rate (VBR). CBR has almost a constant bit rate, while VBR has a peak rate that can be many times the average rate. Latency requirements of video depend on the video application. There are two main types of video applications: video streaming, which may be either be on-demand (video-on-demand) or multicast (video broadcast or video content distribution), and interactive video such as video conferencing. Video-on-demand needs latency that is not more than 5 seconds, video broadcast is latency and jitter insensitive, and interactive video communication requires low delay of 200 to 300 ms round-trip and an average jitter that is not more than 30 ms (Cisco, 2002; Tobagi, 2005; Hassan, Garcia and Brun, 2005). Additional to delay requirements, the packet loss in the network must be kept small, since there is dependency in the encoded video bit stream (Boyce and Gaglianello, 1998).

Interactive data application includes mission-critical applications that directly contribute to the core operation of the enterprise. These applications are highly-interactive and are therefore sensitive to loss and delay like voice and interactive video.

6.4.1 QoS Requirements for Multimedia Traffic

Now that the characteristics and the requirements of the multimedia traffic have been obtained it is time to find out how these different traffic types can be supported in a single network, such that the user-perceived performance is maximized. In chapter four it has been discussed that the key enabling technology in the IP-based multiservice network is QoS. This is because voice, video, and mission-critical data have stringent service requirements from network infrastructure, which supersede the requirements of generic data traffic, and if they are not given priority service they need from network devices, the quality of these applications would quickly degrade to a point of being unusable.

In chapter four it was also highlighted that the provision of QoS includes QoS packet conditioning, and management, as shown in Figure 6.11.

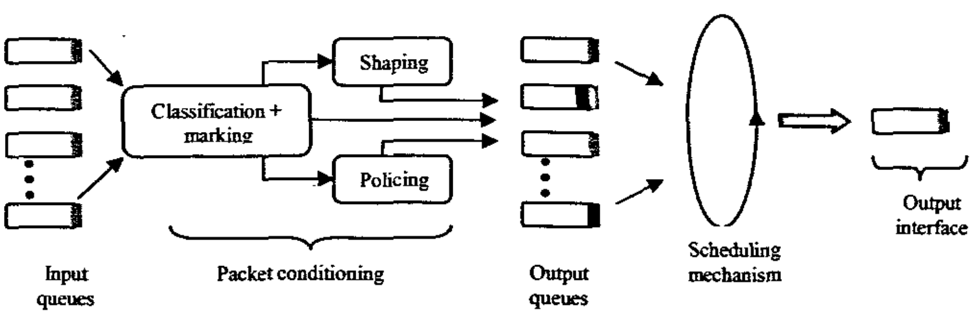


Figure 6.11 Packet conditioning and management

It was also mentioned in chapter four that in order to provide a holistic approach to QoS, a combination of differential services and multi-protocol label switching must be used, where DiffServ will be for classification and MPLS will be for traffic engineering.

6.4.1.1 Packet conditioning

Packet conditioning corresponds to a set of actions carried out on packets in an attempt to establish and preserve their relative importance or priority in relation to any other packet in the network. Packet conditioning can be accomplished through packet prioritization, packet classification and markup, traffic shaping and policing (Aidarous and Plevyak, 2003, p.81).

The first step in providing differentiated services is enabling network nodes to identify the class of service of each packet they receive through a special marking carried by the packet. In chapter four it was mentioned that DiffServ architecture uses the byte in the IP header, previously allocated to TOS and renamed it DS Field, as a priority code.

The contents of the DS field are checked by the classifier upon reception of a packet to determine the queue in which the packet is to be placed. However, before a packet is enqueued, it goes through traffic conditioners, which perform functions as metering and policing to ensure that the traffic entering a queue does not exceed the limits determined by the queue's allocation of the link resources. This functionality is particularly needed for queues that are serviced with high priority in order to avoid starvation of lower priority traffic (Tobagi, 2005). In DiffServ, packets from applications with similar QoS requirements are assigned the same service class at the edge of the DiffServ network and aggregated in the core network. As it has already been mentioned in chapter four, DiffServ model meet the QoS requirements of different service classes by providing per hop behaviours (PHB), which include EF PHB (for low-loss, low-latency, low-jitter and assured bandwidth), AF PHB (for providing forwarding assurances), and BE PHB (for best-effort services).

Network traffic can be categorized into three broad classes: important, best-effort and less-than-best-effort. Important traffic can further be subdivided into multiple categories. For example, mission-critical data and real-time [voice and video] traffic can be assigned to the highest class of data application, which is gold. While applications that are viewed as secondary in importance to business operation or are highly asynchronous in nature can be assigned to the second highest class of data application, which is silver. These applications include video-on-demand, groupware, and intranet browsing.

Applications that play an indirect role in the normal enterprise operations, such as e-mail, generic Internet browsing, and video broadcast can be assigned bronze, which is the best-effort or default class.

6.4.1.2 Traffic management

Traffic management is concerned with delivery of QoS to the end user and with the efficient use of network resources. It involves packet queueing and packet scheduling at switches, routers and multiplexers in order to provide differentiated treatment for packets belonging to different QoS classes. Scheduling manages bandwidth allocation for different flows, management tools are used for queue management in the buffers. To meet the QoS requirements of multiple services, a queueing system must implement strategies for controlling the transmission bit rates that are provided to various information flows (called queue scheduling), and strategies for managing how packets are placed in the queueing system (called queue management) (Leon-Garcia and Widjaja, 2004, p.539).

Queue scheduling

With traffic classified and put into separate queues, a scheduler is required to service them. Scheduling deals with how a frame/packet entering the node exits that node. The scheduler is what decides which queue to process and in what order. The scheduler's service discipline needs to be carefully designed in order to provide the appropriate delay through the node for each traffic type and to avoid congestion. Queue scheduling schemes involves first-in first out (FIFO), priority queueing (PQ), round robin (RR), weighted round robin (WRR), fair queueing (FQ), weighted fair queueing (WFQ), and class-based queueing (CBQ) (Ibe, 2002, p.223; Aidarous and Plevyak, 2003, p.84).

FIFO transmits the packets in the order of their arrival, and it treats all the packets in the same manner, hence it is not possible to provide different information flows with different qualities of service (Leon-Garcia and Widjaja, 2004, p.540).

PQ is a scheduling approach that involves defining a number of priority classes with each class having its own buffer, and packets in high priority queues or buffers are serviced first. However, though PQ does provide different levels of services to the different classes, it does not discriminate among users of the same priority, and it does not allow for providing some guaranteed access to transmission bandwidth to the lower priority classes, which can result in low-priority queues starve for resources (Ibe, 2002, p.223; Aidarous and Plevyak, 2003, p.85; Leon-Garcia and Widjaja, 2004, p.541; Gospodinov, 2004).

In RR queue scheduling, classified packets are sent to *m* queues and these queues are serviced in order 0 to (*m -1*), one packet at a time. RR solves the starvation problem. However, RR does not consider packet size, so small critical packets may have to wait for long periods in queues while large non-critical packets are being served (Aidarous and Plevyak, 2003, p.86).

WRR is a variant of RR where more than one packet can be serviced in turn for a particular queue. However, like RR, WRR does not consider packet size, and allocate resources accordingly (Aidarous and Plevyak, 2003, p.87).

FQ attempts to provide equitable access to transmission bandwidth. Each user has its own logical buffer and the transmission bandwidth is divided equally among all the buffers that have packets to transmit; that is, among nonempty buffers, and the different user buffers are serviced one packet at a time in a round robin fashion (Leon-Garcia and Widjaja, 2004, p.542).

WFQ, also known as packet-by-packet generalized processor sharing (PGPS), is a variant of FQ which is designed to address the situation in which different users have different requirements. As in FQ, each user flow has its own buffer, but each user flow also has weight that determines its relative share of the bandwidth. For example, if buffer 1 has weight 1 and buffer two has weight 2, then when both buffers are nonempty, buffer 1 will receive ¼ of bandwidth and buffer 2 will receive ¾ of the bandwidth. The objective of WFQ is to share the bandwidth proportional to the weights so as to ensure that no one traffic class uses up the entire bandwidth, thereby shutting other classes out (Ibe, 2002, p.223; Aidarous and Plevyak, 2003, p.87; Leon-Garcia and Widjaja, 2004, p.545; Gospodinov, 2004).

CBQ partitions the traffic into different classes such that each class has its own queue and is assigned a proportion of the total link capacity. No traffic is allowed to exceed its assigned capacity, even if there is unused capacity belonging to other classes (Ibe, 2002, pp.223-224).

Cisco has proposed two other scheduling algorithms to support DiffServ PHBs: Modified Deficit RR (MDRR) and Low Latency Queueing (LLQ). Their structures are similar as shown in Figure 6.12, and under both schedulers, EF is implemented using priority queueing, while AF and BE are provisioned by fair queueing. The major difference is that LLQ uses weighted fair queueing and MDRR uses Deficit round robin to implement fair queueing (Zeng, Lung and Huang, 2004).

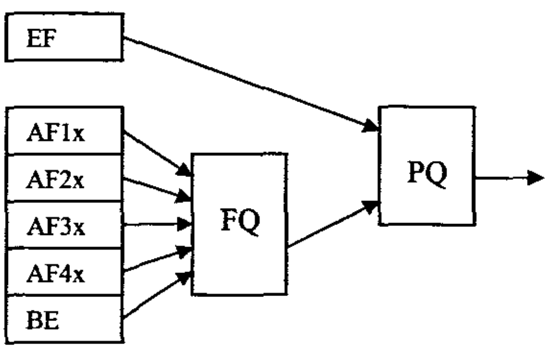


Figure 6.12 Cisco MDRR and LLQ scheduler

The MDRR and LLQ scheduler structure in Figure 6.11 shows that both MDRR and LLQ are hierarchical schedulers that have a PQ scheduler at their higher level. As it has been alluded to before, relying on PQ as the final scheduling algorithm can starve the low priority AF and BE classes, especially if high-priority traffic is excessive. Furthermore, lower priority traffic can experience an extensive packet drops and high queueing delays due to starvation problem (Gospodinov, 2004).

FQ schedulers are capable of providing service isolation among service classes. However, they need proper engineering of bandwidth so that each class can be guaranteed its QoS independently to other service classes. Instead of assigning bandwidth statically for different service classes, Wang, Shen and Shin proposed a dynamic weighted fair queueing (DWFQ) scheduler in order to achieve a higher link utilization (Wang, Shen and Shin, 2001). However, though DWFQ scheduler is able to dynamically allocate the resources proportionally to all the classes, without over-provisioning to EF, monitoring QoS and changing bandwidth allocations too often in a high-speed core routers may lead to dramatic computational overhead. Furthermore, connection admission control can also become difficult as the bandwidth allocations changes with time.

In order to compensate for the debilities of the above scheduling algorithms Zeng, Lung and Huang proposed a solution based on FQ, called weighted fair queueing scheduler with priority (WFQ-P), which is a hierarchical scheduler that has a FQ scheduler at its higher level as shown in Figure 6.13 below. Where each AF class has its own bandwidth reservation so that hard QoS can be provided while EF and BE classes share the same bandwidth with EF having the priority to use the allocated bandwidth and BE packets only get served if there are no EF packets (Zeng, Lung and Huang, 2004).

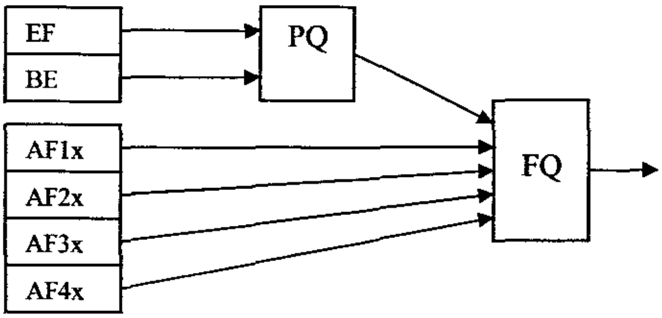


Figure 6.13 WFQ-P scheduler

Another solution is based on the combination of CBQ and WRR scheduling algorithms, where CBQ is used as a scheduling mechanism to provide link sharing between traffic classes, and on a higher level between agencies that are using the same physical link while WRR scheduling is used to distribute time between the classes. CBQ link sharing enables any excess bandwidth resulting from an agency that is not fully utilizing its share to be distributed to other agencies (Rakocevic, 2003).

Queue management

If the rate at which the packets enter the node is faster than the rate at which they can exit, then bottleneck or congestion occur. Congestion may influence basic QoS parameters, such as packet loss, latency and jitter. During congestion, packets are take time to be serviced and some are dropped out, causing increase in delay and packet loss. Whenever congestion is experienced the queueing management algorithms are activated, and in most cases are deactivated when the congestion clears.

Queue management, most of the time, is accomplished by dropping packets whenever necessary or appropriate. This dropping can either be done as packets are arriving or selectively before the buffer is full. There are different buffer management techniques, namely, tail-drop, random early detection (RED), and explicit congestion notification (ECN).

In tail-drop, incoming packets are dropped as they are arriving if the output queue for the packets is full. RED technique discards packets with a certain probability in order to prevent congestion. There are variations of RED in which the thresholds are chosen according to traffic priority, or discard probability is adjusted to guarantee fairness, among other possibilities. Examples of these variations include weighted RED (WRED), distributed weighted RED (DWRED), fair RED (FRED), stabilized RED (SRED), and balanced RED (BRED) (Christiannsen *et al.*, 2001). Another variation of RED is RED with IN/OUT (RIO) (Kim, Lim and Montgomery, 2002). RIO maintains two RED algorithms, one for in-packets and another for out-packets. It has two thresholds for each queue: first and second threshold. When the average queue length is below the first threshold, no packets are dropped, when the average queue is between the two thresholds only the out-of-profile packets are randomly dropped. However, when the average queue length exceeds the second threshold, all the out-of-profile packets are dropped, and some of the in-profile may be randomly dropped, in order to avoid network congestion.
ECN technique allow routers to set a congestion signal bit in packets from ECN-capable transport protocols in order to signal congestion explicitly, instead of signaling congestion by dropping packets. That is, ECN is a congestion avoidance technique, and its advantage is that congestion is avoided thereby reducing packet loss (Aidarous and Plevyak, 2003, p.93).

6.4.1.3 MPLS - DiffServ synergies and implementation issues

In a differentiated Service domain, all the IP packets crossing a link and requiring the same DiffServ behavior are said to constitute a behavior aggregate (BA). At the ingress node of the DiffServ domain, the packets are classified and marked with a DSCP, which correspond to their BA. At each transit node, the DSCP is used to select the PHB that determines the queue and scheduling treatment to be used, and in some cases, drop probability for each packet. In an MPLS domain, LSP is established, using signaling protocols, where packets will transverse. A packet is assigned a label to identify its FEC. The packet is assigned to FEC only once, when it enters the network at the ingress edge

LSR, and at each LSR along the LSP, only the label is used to forward the packet to the next hop.

From the above brief descriptions, one can see the similarities between MPLS and DiffServ: an MPLS LSP or FEC is similar to a DiffServ BA or PHB, and the MPLS label is the similar to the DiffServ code point in some way. The only difference is that MPLS simplifies the routing or switching process used in IP networks while DiffServ is rather about queueing, scheduling and dropping in order to provide differentiated QoS according to the user's demand (Dreilinger, 2002; Kim, Lim and Montgomery, 2002). Because of this, MPLS and DiffServ can be used at the same time in a single network to provide QoS.

In a DiffServ enabled MPLS system, when packets marked with DiffServ code points arrive at an MPLS network, the information provided by the code points must be transferred onto MPLS labels so that MPLS can make decisions that respect the differentiated service requirements. This can be done by either using the 3-bit long EXP field of the MPLS header (for up to eight service classes) or by mapping DSCP to a <Label, EXP> pair (for more than eight service classes).

6.4.2 Simulation Tool for Performance Analysis

According to Shannon (1975), simulation can be defined as the process of designing a model of a real system and conducting experiments with this model for the purpose either of understanding the behaviour of the system or evaluating various strategies (within the limits imposed by a criterion or a set of criteria for the operation of a system (Shannon, 1975). It enables the analysis of the sensitivities of the different network parameters and their best setting-up without deploying several scenarios in a real network.

This section presents simulation, which is performed with the Network Simulator version 2 (NS-2), which is an object-oriented, discrete event-driven network simulation package developed at UC Berkeley. Like other simulation packages, NS network simulator provides the user with a convenient interface and a large number of facilities for setting up the model of the network to be simulated and for analyzing the results of the simulation. It consists of a rich set of protocols such as TCP and UDP and various types of applications. These applications include FTP, Telnet and HTTP, which use TCP as the underlying transport protocol, and applications requiring a constant bit rate (CBR) traffic pattern, which use UDP as an underlying protocol. It also supports various queueing and scheduling policies. NS also allows for the definition of arbitrary network topologies, composed of routers, links and shared media (Andreozzi, 2001).

The main objective of this simulation is to model and investigate the effects as well as the potential and the performance of DiffServ with MPLS in multi-service rural networks, especially when it comes to providing QoS required by the real-time interactive traffic, using both wired and fixed wireless communication media. In the following subsections the simulation settings are discussed, and there after the simulation results for different scenarios are going to be presented.

6.4.2.1 Topology definition

Figure 6.14 shows a basic topology that is gong to be used, which involves a simple core network with 6 attached end nodes, representing the hospital and 5 referring clinics, respectively.

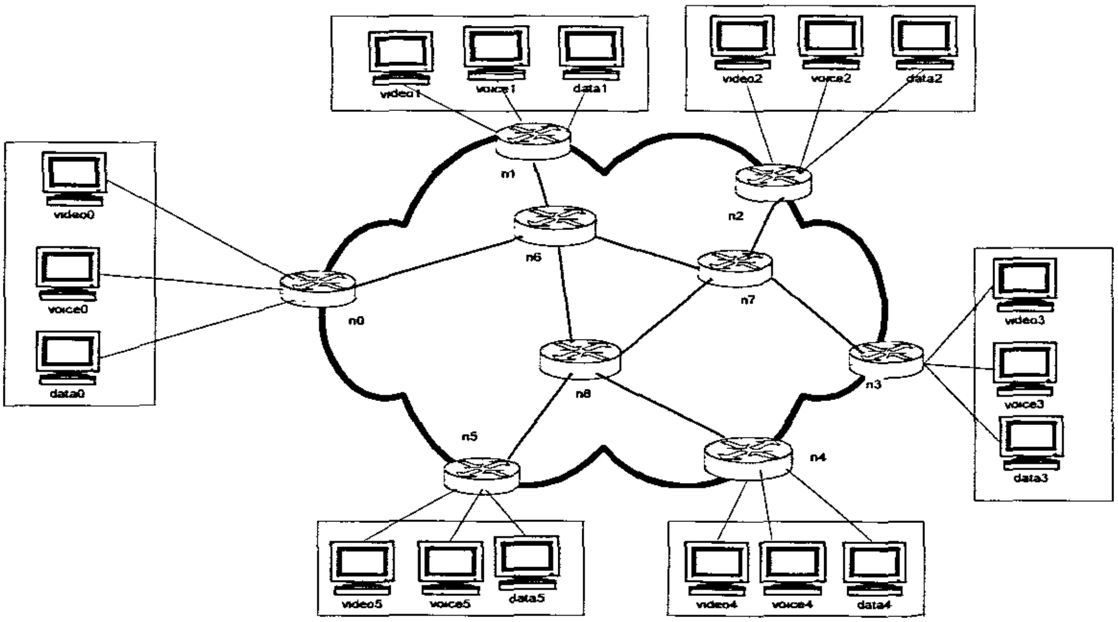


Figure 6.14 Topology setup

Each site has video, voice and data source/sink; that is, within this network, every site is capable of communicating using video, voice or data, and these applications can be activated in parallel by any user within the network. Within the core the links are represented by bidirectional E1 (2.048 Mbps) while at the edges the end-stations are attached using 4 Mbps and more. The reason for selecting higher bandwidth links for access network is to prevent any bottlenecks in the access points that could impact the results of end-to-end application performance. Since it has been mentioned in chapter two that the envisaged communication media for rural networks will be a hybrid of wireless and wired media, some of the links are going to be wireless and others will be wired. For most of the simulation purposes the setup in Figure 6.13 is simplified to be as shown in Figure 6.15.

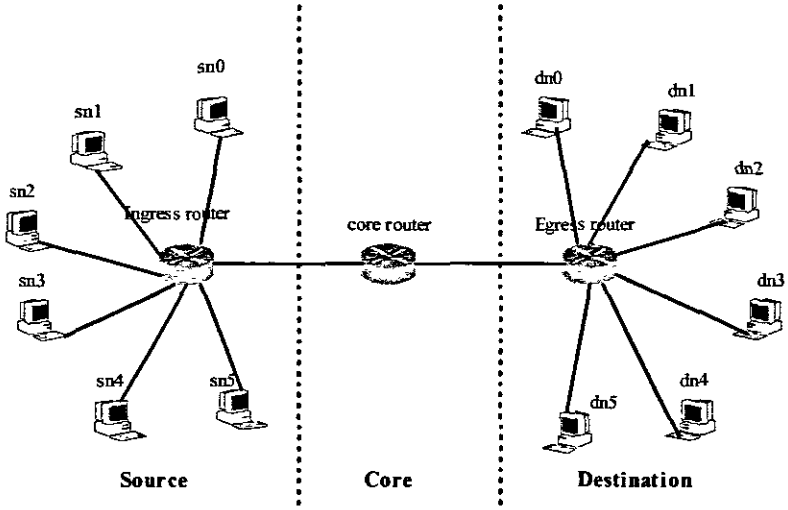


Figure 6.15 Simulation setup

6.4.2.2 Traffic source definition

Multi-service traffic involves data, voice and video traffic. Data traffic is usually transmitted using Transport Control Protocol (TCP) while real-time traffic such as voice and interactive video are transmitted using User Datagram Protocol (UDP). To simulate data applications, Telnet and FTP traffic sources are going to be considered. To simulate speech traffic, which is characterized by alternating spurts and periods of silence, an ON/OFF process is going to be used. Since the ON (or OFF) periods do not have a heavy-tailed distribution, a non fractal version of the ON/OFF model with an exponential distribution of on and off times will be suitable (Tatipamula and Khasnabish, 1998, p.137). VoIP traffic is simulated by the G.711 audio codec, which transmit data over RTP. The G.711 codec outputs data at a constant rate of 64 kbps with a frame size of 240 bytes, and in the case of IP-based networks, these frames are encapsulated by the IP/UDP/RTP protocols that augment the basic frame size with their headers resulting in packet sizes of 280 bytes for IPv4 and 300 bytes for IPv6, and transmission rates of 74.667 kbps for IPv4 and 80 kbps for IPv6. The parameters for speech traffic ON/OFF source are given in Table 6.9.

Table 6.9 Simulation parameters for the voice

Traffic type	Mean ON duration (seconds)	Mean OFF duration (seconds)	Peak rate (Packets/second)
Voice	0.35	0.65	80k

These ON/OFF parameters for VoIP are widely used in literature (Rakocevic, Stewart and Flynn, 2003) are taken from measurements by Deng (1995). For data sources, the packet size is 1500 bytes for each type.

The real-time video will be simulated using a trace driven source. This is due to the fact that from the statistical analysis of the different video traces it was found that encoded video traffic is dependent on the content, the encoding standard, and the encoder settings, hence no independent video model can be developed to cover all video traffic sources. So to enable simulation that evaluates video sequences, video traces are used as video traffic. Figure 6.16 depicts an overview of video traffic modelling versus the video trace approach.

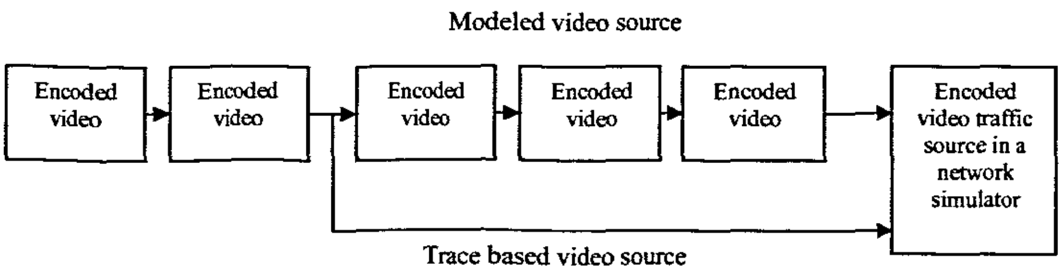


Figure 6.16 Overview of video traffic modeling versus the video trace approach.

It is evident from the figure above that using video traces instead of modelled video sources is beneficial in that you have a vast amount of video models, each with a specific point of view. According to Fitzek, Seeling and Reisslein (2003), direct utilization of video traces in network simulators also facilitates the fastest method to incorporate video sources into existing network models. However, for utilization of the trace files in the network simulator as video sources, some interface has to be used. For Network

Simulator ns-2, an interface that was written by Michael Savoric from the Technical University of Berlin (Germany) is used (Fitzek, Seeling and Reisslein, 2003). The function of the interface is to convert the trace files into binary format (Altman and Jiménez, 2003).

6.4.3 Simulation Results

During simulation, it was found that the native NS-2 DiffServ module, which was developed by Nortel Networks and added to NS on November 2000, does not support some of the functions of the DiffServ router, such as traffic marking on a per-transport-protocol and per-packet-type basis among several source-destination nodes; that is, it is not possible to define a meter for aggregate traffic. Secondly it was also found that the schedulers suitable for differentiated services are also limited. To address this shortfall the native Nortel Networks developed DiffServ module was replaced by the one that was developed by Andreozzi at Lappeenranta University of Technology (Finland). This redesigned DiffServ module has the following features:

- Marking – possibility to define mark rules on per-packet basis that are needed in order to provide differentiated services and the marking is decoupled from metering.
- New schedulers – WFQ, WF2Q+, SCFQ, SFQ and LLQ.
- New Policy – possibility to define a DSCP based rate limiter to enable a drop out-of-profile traffic capability on drop precedence level basis
- New Monitoring possibilities for both UDP- and TCP-based traffic, such as, average, instantaneous, minimum and frequency-distributed one-way delay (OWD) and IP packet delay variations (IPDV) for UDP-based traffic, and TCP Goodput, Round-Trip Time, Window Size, queue length, maximum burstiness, departure rate and received, transmitted and dropped packets monitoring for TCP-based traffic. OWD and IPDV will enable the computation of end-to-end one-way delay and delay variation for UDP based traffic, respectively (Andreozzi, 2001).

6.4.3.1 Scenario 1: Scheduler comparison for real-time traffic

The first simulation is to determine the most appropriate scheduler that can be used for real-time traffic, which is delay and delay variation sensitive, in a differentiated services environment. Nine schedulers: Priority Queueing (PQ), Self-Clocked Fair Queueing (SCFQ), Start-time Fair Queueing (SFQ), Weighted Fair Queueing (WFQ), Weighted Fair Queueing Plus (WF2Q+), Low-Latency Queueing (LLQ), Weighted Interleaved Round Robin (WIRR), Round Robin (RR), and Weighted Round Robin (WRR), are compared using the three QoS parameters: packet loss, [one way] delay, and jitter. Two traffic types are used for simulation: video trace (from Mr. Bones) for video traffic source and constant bit rate (CBR) traffic for other traffic source. Two queues are configured to support both Expedited Forwarding (EF) and Best Effort (BE) aggregates. Video is transmitted as premium traffic using Expedited Forwarding services, while CBR is transmitted using Best Effort services. To avoid synchronization problems, background traffic is generated with several CBR sources whose rate is chosen from a uniform random distribution in the range [10 kbps, 100 kbps], while the starting time is chosen from a uniform random distribution in the range [0 s, 5 s]. The simulation is done using packet sizes ranging from 64 to 1600 bytes.

Scenario-1 Results

Figure A.1 presents the IP Delay Variation (IPDV) plots for the different schedulers, which can be used for jitter performance analysis. WFQ have the lowest average jitter values which are less than 1.55 ms up to packet size of 1 600 bytes, followed by LLQ with average values which are less than 1.65 ms for packet sizes up to 1 600 bytes, PQ and RR have average values which are less than 1.7 ms for packet sizes up to 1600 bytes, SCFQ has average values which are less than 1.8 ms for packet sizes up to 1600 bytes, and other schedulers are having values that are greater than 1.8 ms.

Figure A.2 presents the One Way Delay (OWD) plots for the different schedulers, which can be used for latency performance analysis. PQ and LLQ have the lowest average one-way latency values which are less than 17.5 ms up to packet size of 1 600 bytes, followed by RR with average values of less than 19 ms for packet sizes up to 1 600 bytes, then WRR with an average values less than 31.5 ms for packet sizes up to 1 600 bytes, WIRR and SFQ with an average values less than 32.5 ms for packet sizes up to 1 600 bytes, and other schedulers are having values that are greater than 35 ms.

Figure A.3 presents the packet loss plots for all transmitted packets using different schedulers. These plots reveal that the performance of almost all the schedulers is the same, with the percentage of transmitted packets of about 90 %, with the exception of WIRR, which has a little bit more loss that drops the percentage of transmitted to about 85 % compared to other schedulers

6.4.3.2 Scenario-2: Complete service model with DiffServ only

In this simulation, the setup is as in scenario-1; however, the number of applications and classes is increased. For premium services, voice traffic is also transmitted. Assured forwarding (AF) service is also added to accommodate additional classes of service, which include gold services for Telnet and FTP.

Test 1

The service model is tested using only those schedulers that had a jitter less than 2 ms in the previous simulation; that is, RR, LLQ, WFQ, PQ, and SCFQ. These schedulers are again compared using the two QoS parameters: [one way] delay, and delay variations or jitter to check their performance when there are more classes of service and applications.

Scenario-2 Test1 results

Figure B.1 in appendix B presents the plots for jitter and latency (one-way) for RR, LLQ, WFQ, PQ, and SCFQ schedulers. From the IPDV plots PQ has the lowest jitter values, which is less than 4.5 ms, followed by LLQ with values less than 7 ms, other schedulers have values which are more than 40 ms. From the OWD plots LLQ has the lowest one way latency values of less than 28 ms followed by PQ with values less than 45 ms, and other schedulers have one way latency values which are more than 210 ms. However looking at service rate plots in Figure B.2 it can be seen that though PQ has the lowest jitter compared to other scheduler, it starves the low priority BE classes, while LLQ gives a fair service to all the classes.

From the above discussion it can be concluded that though PQ has the lowest jitter values than any of the schedulers used, LLQ has a better performance when it comes to one way latency and servicing of all the classes. Hence from now onwards the scheduler that is going to be used in this study is going to be the LLQ scheduler.

Test 2

The second test is done using the setup in test 1 but with more video traffic; that is, instead of using a single video source, the number of video traffic sources is increased to two and then to three video. The LLQ scheduler is used and the performance of the network when there are more classes of service and applications is analyzed using the two QoS parameters: [one way] delay, and delay variations.

Scenario-2 Test 2 results

Figure B.3 in appendix B presents the plots for jitter and latency (one-way). From these plots it can be seen that as the number of video streams increases to three, both jitter and latency increase to about 26 ms and 270 ms, respectively. Jitter is still within limits; that is, it is still below the maximum acceptable value for the average jitter, which is 30 ms, however the average one way delay exceeds 150 ms (which is half of 300 ms round-trip delay allowed) (Cisco, 2002; Tobagi, 2005; Hassan, Garcia and Brun, 2005).

Test 3

The third test is done to test the ability of DiffServ to perform load balancing in the case where the shortest route is congested, and rerouting in the event of node or route failure. In order to perform this test, the simulation setup in Figure 6.15 is modified to have two core routers, thus offering two paths between the ingress and egress router as shown in Figure 1.17 below.

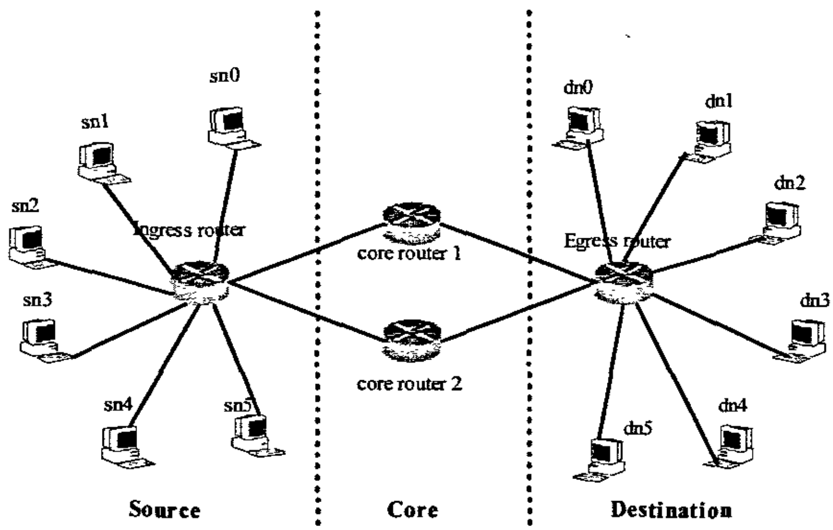
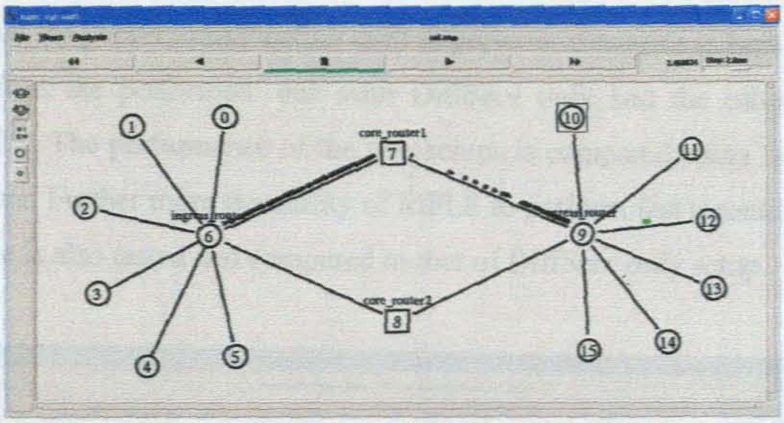


Figure 6.17 Simulation setup for Test 3

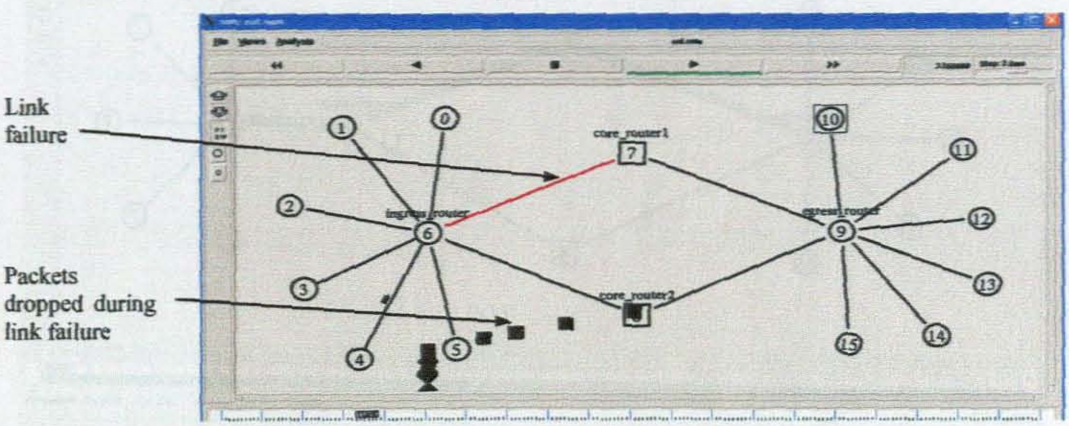
Six video streams are sent from the source to destination together with 3 VoIP traffic and data so that the network is fully loaded. Then during the course of simulation a route failure is simulated between the ingress and core router 1, to check if the traffic will be rerouted to the other path that goes via core router 2, after a minute the route is restored.

Scenario-2 Test 3 results

The NAM screenshots of the simulation are shown in Figure 6.18 (a) and (b), where the dotted line represents the movement of packets from the source nodes 0 through ingress router and core router 1 to the destination nodes 10. Nodes 7 and 8 are core routers, node 6 and node 9 are ingress and egress routers, respectively.



(a) Traffic Distribution



(b) Route failure (indicated by the red line between Node 6 and 7)

Figure 6.18 NAM screenshots for Scenario 2 Test3 simulation

From these NAM screenshots it can be observed from Figure 6.18 (a) that though an alternative path has been provided, no load sharing is done. It can also be observed in Figure 6.18(b) that no rerouting was done when there is a route failure, instead the packet are dropped and the flow just stops even though there is an alternative path between the ingress and egress router, which goes via core router 2.

6.4.3.3 Complete service model with DiffServ enabled MPLS

In this scenario, a complete service model with DiffServ enabled MPLS is simulated for real-time video and voice application, and data applications. The simulation setup is as shown in the NAM screenshot of Figure 6.19. The DiffServ router (node 5) is included to make sure that the packet classification used in previous scenarios is kept the same. Then, two simulations are performed: one with DiffServ only and the other with DiffServ enabled MPLS. The performance of the two setups is compared using two QoS metrics: delay and jitter. Further more the ability of MPLS to perform fast rerouting when there is a route failure is also tested and compared to that of DiffServ only setup.

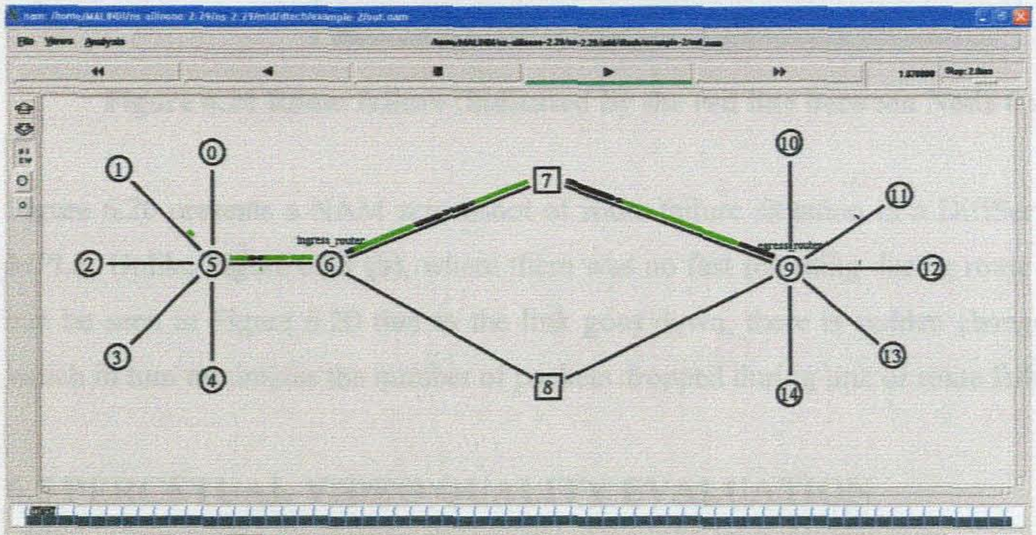


Figure 6.19 Simulation Setup for Scenario 3

Scenario 3 results

The IPDV and OWD plots for both DiffServ and DiffServ enabled setups are shown in Figure C.1 and C.2 (Appendix C), respectively. Comparing the plots, it can be seen that the use of MPLS has improved both jitter (IPDV) and one way delay to acceptable values, which is less than or equal to 30 ms for jitter, and less than or equal to 150 ms for one way delay (which is half of 300 ms round-trip delay allowed) (Cisco, 2002; Tobagi, 2005; Hassan, Garcia and Brun, 2005).

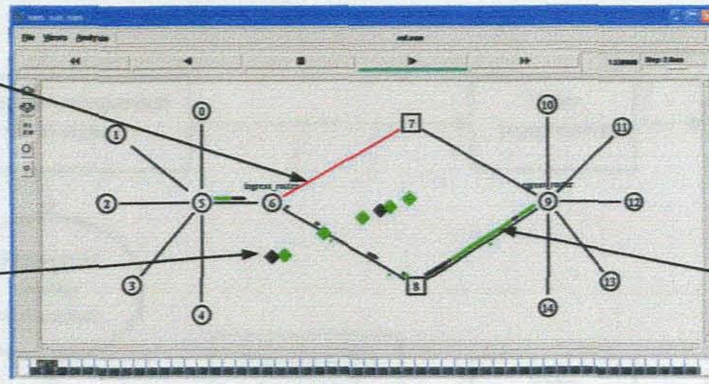


Figure 6.20 Route failure (indicated by the red line between Node 6 and 7)

Figure 6.20 presents a NAM screenshot of route failure situation in a DiffServ enabled MPLS. Unlike Figure 6.18 (b), where there was no fast rerouting during route failure, it can be seen in Figure 6.20 that as the link goes down, there is sudden change of path, which in turn minimizes the number of packets dropped during link or route failure.

6.5 PERCETUAL VIDEO QUALITY EVALUATION

According to Klaue, Rathke and Wolisz (2003), the network performance metrics such as packet loss, jitter and latency, which have already been evaluated in the preceding section may be insufficient to adequately rate the evaluated perceived quality by the end user. So in order to evaluate the perceptual quality of the received video, the reconstructed or decoded video must be compared with the original video using a perceptual video quality technique, as shown in Figure 6.21. There are many widely accepted techniques to measure video quality. They normally fall into two main categories: Subjective and Objective evaluation assessments (Klaue, Rathke and Wolisz, 2003; Shanmugham, 2006, pp. 2 and 24; Orozco and Ros, 2004).

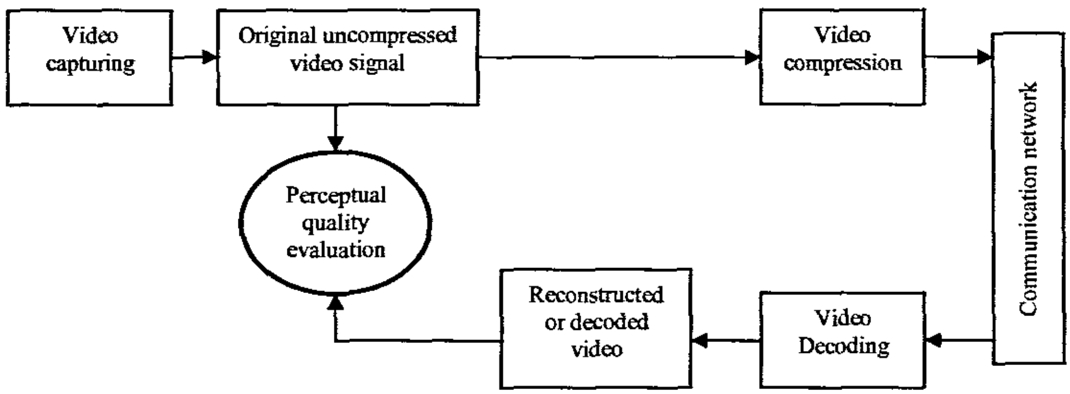


Figure 6.21Perceptual video evaluation process

The subjective evaluation assessment was proposed by ITU and the Video Quality Experts Group (VQEG) is concerned with how the reconstructed video is perceived by the viewer or the user. Many of the subjective measurements are described in ITU-R Recommendation BT.500 and ITU-T Recommendation P.910, P.920 and P.930. The idea is almost the same as that of Mean Opinion Score (MOS) used for audio: the reconstructed (decoded) video is viewed by an audience who in turn designate their opinions on that particular video sequence, and their opinion is averaged to evaluate the video quality of the video sequence. However, there are many parameters, such as room illumination, display type, brightness, contrast, resolution, viewing distance, age, education level, which can influence the subjective video quality assessment.

According to Nakajima's (2003) explanation of the subjective assessment as used in telemedicine, which is based on the ITU-R recommendation 500-9, it can be agreed with Koumaras, Martakos and Kourtis (2005), Koumaras (2006), Orozco and Ros (2004), and Klaue, Rathke and Wolisz (2003) that subjective quality evaluations are complex and time-consuming, both in their preparation and execution. Hence, other algorithms that provide an objective measure of quality difference between the original and the decoded video sequences have been developed (Fitzek, Seeling, and Reisslein, 2002). Evaluating using these algorithms is known as objective evaluation assessment. This technique exploits mathematical models or algorithms for emulating the results of subjective evaluation procedures. There are two approaches to objective evaluation technique: psychophysical approach and the engineering approach. In the psychophysical approach the metric design is based on models of human visual system, while in engineering approach the metrics make assumptions about the artifacts that are introduced by compression process or transmission link. Objective techniques are further classified according to the availability of the original uncompressed video signal, which is considered to be in high quality. Therefore, they can be classified as Full Reference Methods (frame-by-frame comparison between the original and reconstructed video), Reduced Reference Methods (extract a number of features from the reference and reconstructed video), and Non-Reference Methods (no reference information needed). The advantages of the objective evaluation technique are that it is economically affordable, no audience is required, no statistical analysis is needed to process the scores of the audience and thus it is faster than subjective procedures (Koumaras, 2006; Shanmugham, 2006, p.27).

6.5.1 Video Quality Measurement Metrics

6.5.1.1 Peak Signal-to-Noise Ratio (PSNR) and MSE

The most traditional ways of evaluating the quality of digital video processing system are counting on the signal-to-noise ratio (SNR) and peak signal-to-noise ratio (PSNR) (Wang, Lu and Bovik, 2004; Koumaras, Martakos and Kourtis, 2005) between the original and reconstructed signal, where SNR of an image is usually defined as the ratio of the mean pixel value to the standard deviation of the pixel values, and PSNR is a term used for the ratio between the maximum possible power of a signal and the power of corrupting noise that affects the fidelity of its representation. In quality video evaluation, PSNR is used as a measure of quality of reconstructed video, and is most easily defined via the mean squared error (MSE). If the original image is given by $f(i, j)$ and the decoded or reconstructed image is $F(i, j)$ and both images contain $m \times n$ pixels, then error metrics, then the MSE is defined as

$$MSE = \frac{1}{mn}\sum_{i=0}^{m-1}\sum_{j=0}^{n-1}\left\| f(i,j) - F(i,j) \right\|^2 \qquad (6.28)$$

And the PSNR in decibels (dB) is defined as

$$PSNR = 10\log\left(\frac{MAX_I^2}{MSE}\right) = 20\log\left(\frac{MAX_I}{\sqrt{MSE}}\right) \qquad (6.29)$$

Where MAX_I is the maximum pixel value of the image, which is defined by

$$MAX_I = 2^b - 1 \qquad (6.30)$$

Where b is number of bits per sample or pixel. The typical PSNR values range between 20 and 40 dB. According to Klaue, Rathke and Wolisz (2003) the conversion from PSNR to MOS can be done as follows:

Table 6.10 PSNR to MOS conversion

PSNR [dB]	MOS
> 37	5 (Excellent)
$31 - 37$	4 (Good)
$25 - 31$	3 (Fair)
$20 - 25$	2 (Poor)
< 20	1 (Bad)

6.5.1.2 Video Quality Measurement (VQM) and double stimulus impairment scale (DSIS)

Video quality measurement (VQM), which was developed at the Institute for Telecommunication Sciences in America (ANSI), is a framework of objective parameters that can be used to measure the quality of digital video system. VQM objective metrics are derived from gradients that represent instantaneous changes in pixel values over space and time. The VQM score measures visual quality degradation (or impairment) in the range that goes from 0 to 1, where 0 means no distortion (or no perceived impairment) and 1 is maximum distortion (or maximum perceived impairment) (Orozco and Ros, 2004; Wolf and Pinson, 2002, p. 64).

VQM also provides for double stimulus impairment scale (DSIS) measurements. DSIS as defined by ITU-R Recommendation 500-11 measures the global amount of degradation perceived which is due to transmission path impairments, and it provides the subjective ratings on a scale ranging from 1 to 5, where 1 is "impairments are very annoying" and 5 is "impairments are imperceptible", as shown in the table below (ITU-R, 2002; Winkler and Campos, 2003; Orozco, 2005, pp.27-28).

Table 6.11 DSIS Impairment scale

Score	Description
5	Imperceptible
4	Perceptible but not annoying
3	Slightly annoying
2	Annoying
1	Very annoying

6.5.1.3 Blurring and Block Distortion (or Blockiness) measure

Video encoding can introduce a number of different impairments, which are often related to the bit rate used for encoding and to the characteristics of the video sequence. Common compression artifacts include blockiness (or block distortion), and blur.

Block distortion or blockiness is a block like pattern in the processed video that occurs because typical video codecs process images in small blocks, and quantization of transform coefficient values can lead to discontinuities between the blocks. This can be exacerbated by the Mach Band effect in which the human eye tends to sharpen edges. Block distortion measure metric was created to measure the percentage of block distortion.

Blur is the loss in the detail that occurs around edges, typically due to the quantization of transform coefficients representing higher frequencies. The higher the blurring the lower will be the quality of the signal. In order to determine the level of image blurring, the blurring measure metric is used. This metric allows for the measure of the percentage of blurring of the processed video.

Scenario 4

This scenario is designed for perceptual video quality assessment. Since most of the quality assessment or measurement tools available are using raw YUV formats instead of AVI, a new raw YUV digital video sequence, called "Mother & Daughter", was downloaded in CIF format from the site http://www.tkn.tu-berlin.de/research/evalvid/cif.html, and processed by JM1.7 codec to generate a H.264 bitstream at 213.95 kbps. After encoding the trace files are generated, then followed by all the perceptual video evaluation process steps as depicted in Figure 6.21. Error module is also included in order to simulate link-level errors and losses. Then the quality of video is evaluated before and after transmission using PSNR, and visual quality metric (VQM). The first test is done without any Intra-coded MBs, and the second test is done with Intra-coded MBs, which are used in order to prevent error propagation.

Scenario 4 Results

The PSNR of the three videos (i.e. decoded video before transmission, decoded video after transmission without any Intra-coded MBs and decoded video after transmission with Intra-coded MBs) were determined and their values are plotted in Figures 6.22 through 6.25, VQM and DSIS scores are depicted in Figures 6.27(a) and (b) through 6.28(a) and (b), and the images of the original and the processed videos are shown in Figures 6.26, 6.27(c) through 6.28(c). These measurements were performed with the help of Evalvid tools, the general model of video quality measurement (VQM) PC tools, and Matlab.

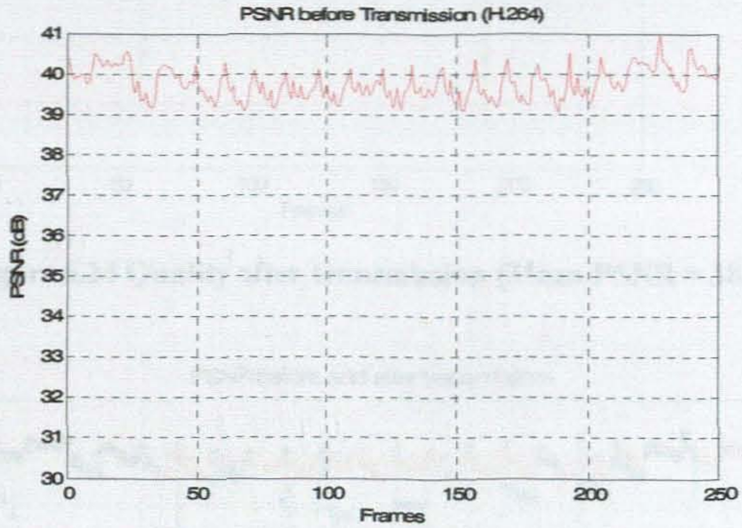


Figure 6.22 Quality before transmission (Mean PSNR = 39.77 dB)

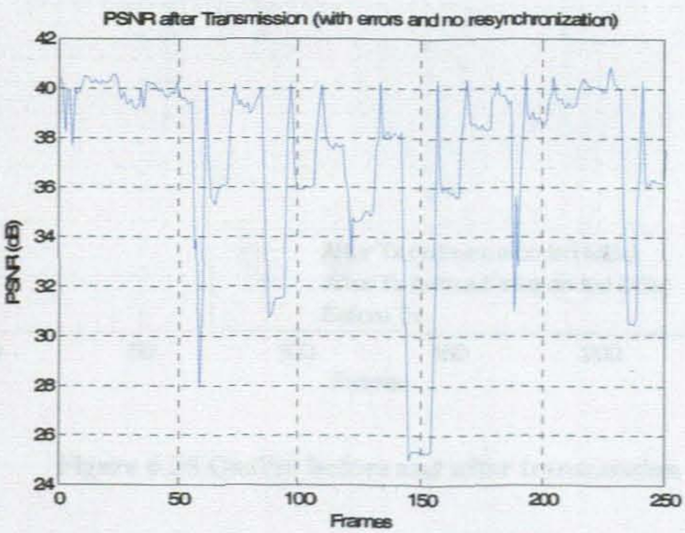


Figure 6.23 Quality after transmission (Mean PSNR = 37.28 dB)

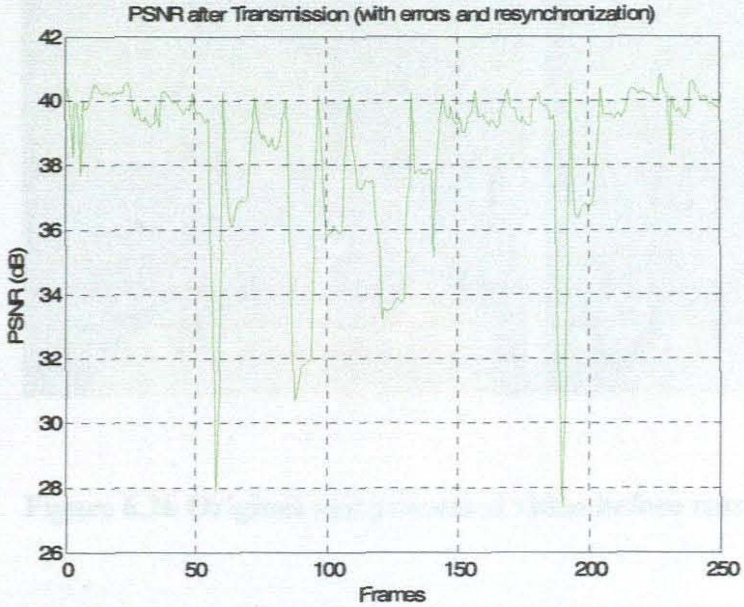


Figure 6.24 Quality after transmission (Mean PSNR = 38.35 dB)

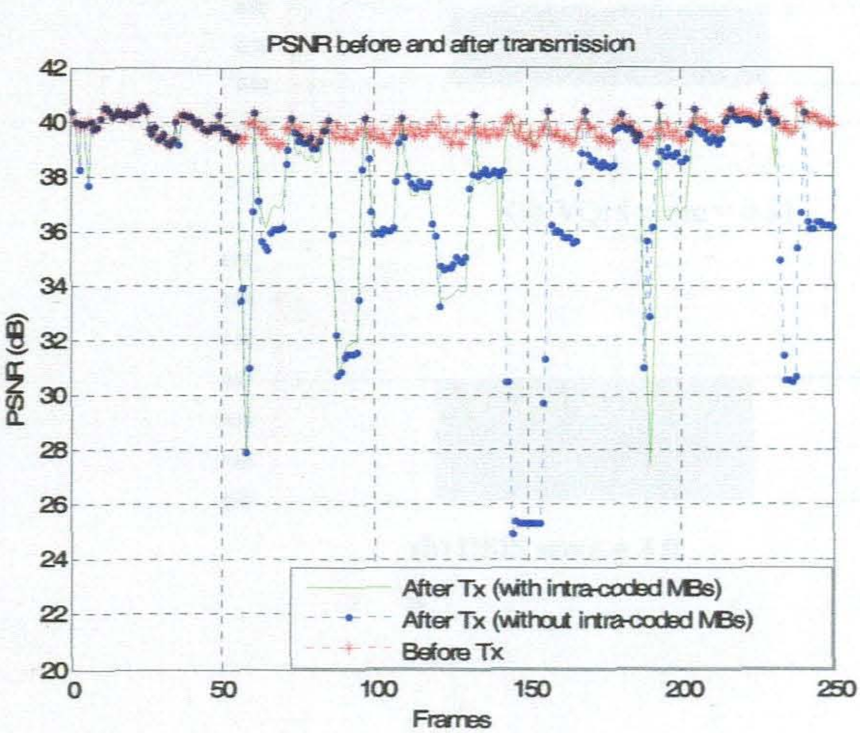
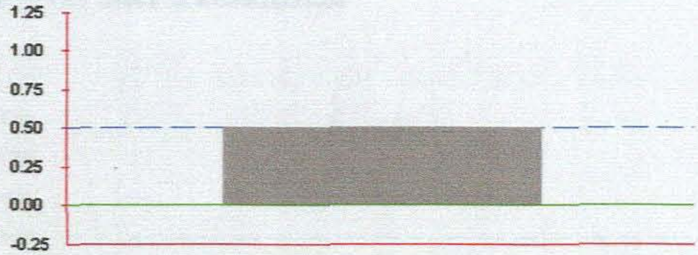
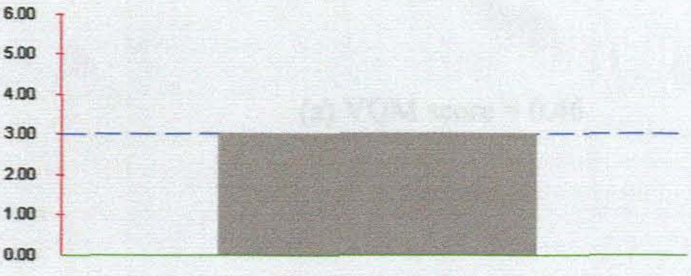


Figure 6.25 Quality before and after transmission

Figure 6.26 Original and processed video before transmission



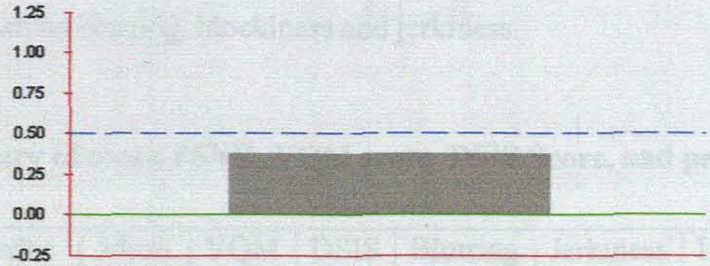
(a) VQM score = 0.51



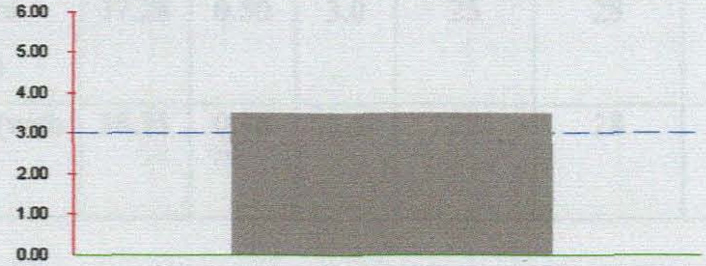
(b) DSIS score = 3.0

(c)

Figure 6.27 (a) VQM score (b) DSIS score and (c) Original and processed video (without Intra- coded MBs) after transmission



(a) VQM score = 0.46



(b) DSIS score = 3.5

(c)

Figure 6.28 (a) VQM score (b) DSIS score and (c) Original and processed video (with Intra- coded MBs) after transmission

Table 6.12 gives a summary of the values of the mean PSNR, VQM and DSIS scores, and impairments such as blurring, blockiness and jerkiness.

Table 6.12 Summary of mean PSNR, VQM score, DSIS Score, and perceptual effects

Processed video under measurement	Mean PSNR (dB)	VQM score	DSIS score	Blurring %	Jerkiness %	Blockiness %
Before transmission	39.77			21	28	26
After transmission (no intra-MBs)	37.28	0.50	3.0	25	28	51
After transmission (with Intra-MBs)	38.35	0.38	3.5	22	28	37

From the PSNR plots and PSNR mean, VQM and block distortion values in Table 6.12 it can be seen that the errors introduced by the channel can have a profound effect on the quality of the transmitted video. However, using Intra-coded MBs, as proposed in chapter five, during coding can help minimize the effects of the errors. This is evident in PSNR plots in Figures 6.23-25, and the VQM plots in Figures 6.27(a)-28(a). This can also be seen in the images of the transmitted processed videos in Figures 6.27(b)-28(b). For example, in Figure 6.27(b) the fingers of the left hand, picture on the wall and the tip of the head of the mother is distorted due to the errors introduce by the channel, whereas in Figure 6.28(b), which is the video that is encoded with Intra-MBs, only the hand of the mother has been affected by errors.

6.6 IN SUMMARY

This chapter was about validating the proposed solutions on how to provide a rural telemedicine with quality video transmission using IP based multiservice rural area network. Firstly, the author introduced the need for modeling and simulation in communication systems. Secondly, traffic source modeling was done, with emphasis being on video traffic, where video characterization was done and video model was developed. Having characterized and developed the traffic models, simulation was done using the network simulator NS-2 version 2.93.3 in order to verify whether the proposed quality of service mechanisms will be able to meet the network requirements for quality video transmission and to determine the appropriate scheduling. Parameters used included one-way delay (OWD) or latency, delay variation (IPDV) or jitter, and packet loss. In addition to the performance evaluation mentioned above, perceptual video evaluation was done using mainly the objective techniques such as PSNR and VQM scores. Additional to PSNR and VQM, DSIS was also used for perceptual video quality measurement. The outcome of simulations and perceptual evaluation is as follows.

From the simulation performed, it was found that DiffServ-enabled MPLS, which is using a LLQ scheduler, provides a performance that meets the network requirements for quality video transmission. During perceptual video evaluation it was found that the erasure effect that is caused by the packet loss and errors in the channel can be minimized by using resynchronization (or Intra-coded MBs), which prevents error propagation.

The next chapter provides a final, concluding overview of this dissertation.

CHAPTER 7

7. CONCLUSION AND RECOMMENDATIONS

7.1 INTRODUCTION

From the literature consulted, it is evident that telemedicine system must be able to bring patients and healthcare professional together by exchanging voice, video, and data over distances when they cannot meet face-to-face. In order to emulate the face-to-face consultations and to make correct diagnosis the telemedicine system must provide quality audio and video delivery. The literature reveals that in many rural telemedicine ATM has been used to meet most of these telemedicine requirements, however, most of these systems are in the developed countries. For the developing countries, like South Africa and other countries in Africa, the rural telemedicine is based on IP and/or PSTN (i.e. telephone and fax).

IP has emerged as 'de facto' standard for data communication. However, its best-effort debilities make it to be insufficient for most of innovative and demanding applications such as e-commerce, e-health and others that hit the market today, hence some quality of service mechanisms such as IntServ and DiffServ have been proposed in order to compensate for these IP's best-effort debilities. However, from the literature consulted it has been argued that IntServ has scalability problems, thus making DiffServ to be the most preferred technique for providing quality of service for IP-based network.

7.2 NEW QOS METHODS FOR RURAL TELEMEDICINE

The primary objective of this study is to provide new methods for providing rural telemedicine with improved quality video transmission to be used for most of the applications in telemedicine between the area hospital and referring clinics, in remote and rural areas, where the telecommunication infrastructure is either underdeveloped or non-existed.

As a first step towards meeting this objective a conceptual model of a rural area network (RANET) that can be used in rural areas as a communication platform for rural telemedicine in areas where telecommunication infrastructure is either underdeveloped or nonexistent was developed. In order to cater for all the telemedicine applications, which include data, voice and video, RANET is a multiservice IP-based network. Different access technologies that can be used with RANET were also investigated and it was found that due to the topography of the rural areas, which is characterized by trees, rivers, and mountainous terrain, a combination of wired and fixed wireless and a hybrid of wired-wireless access technologies will be suitable.

Having developed the RANET system, the second step was to compensate for IP's best-effort datagram delivery service debilities, which make IP not to be able to meet the QoS requirements of real time traffic. To facilitate quality of service within IP DiffServ has been adopted in this study so as to provide differentiated services for different classes by using mechanisms that include packet classification, policing, class-based queuing and scheduling, and random early detection. To complement the DiffServ and provide a holistic approach to the provision of QoS, traffic engineering using MPLS was incorporated. MPLS provides connection-oriented paths in a connectionless IP environment. It also provides fast reroute during link or route failure, thus minimizing packet loss that can cause quality degradation and unavailability. The two technologies: DiffServ and MPLS complment each other to minimize delay, jitter and packet loss.

However, for video, literature study revealed that, for a holistic approach to quality video transmission, bandwidth, heterogeneity, and errors need to be also considered in addition to delay, jitter and packet losses mentioned above. In order to address all these factors and ensure a holistic approach to quality video transmission the following were proposed:

- Adopting video coding technique that will produce a low bit rate without compromising much on quality in order to enable the video transmission over small bandwidth;

- Multicast video delivery or distribution method that will utilise the bandwidth efficiently for those video applications that involve video distribution of the same content to many users;
- Error control and concealment to address losses and errors introduced by the communication network;
- Scalable coding to address heterogeneity both at network and receiver level;
- Making the video source to be able to adapt to network conditions to address bandwidth fluctuations and congestion, and
- Prioritising and protecting video traffic using the QoS and traffic engineering mechanisms to provide solutions for quality video transmission.

For video coding, H.264 is proposed because it provides high quality video that is consistent at high and low bit rates, resilience to transmission errors, scalability, high-compression efficiency, and possibility of transportation over different networks. For prioritising and protecting video traffic from IP network's best-effort debilities a combination of DiffServ and MPLS has been adopted and to minimizing the effects of errors on the transmitted video resynchronization is used so as to prevent error propagation.

7.3 VALIDATION

In order to verify and validate the proposed system and the proposed techniques modeling and simulation was carried out. First two video were captured, encoded analyzed and characterized in order to develop a video traffic source model that was used during simulation. From these analysis it was found that the encoded captured videos possessed both short range dependence (SRD) and long range dependence (LRD) and the only model that can capture both SRD and LRD was found to be a fractional autoregressive integrated moving-average (FARIMA) model. Additional to video traffic, other traffic types such as voice traffic and data were also defined.

This was followed by simulations using network simulator (NS-2.93). The main objective of the simulation was to model and investigate the effects as well as the potential and the performance of DiffServ with MPLS in multi-service rural networks, especially when it comes to providing QoS required by the real-time interactive voice and video applications. The simulation started by determining the suitable scheduler that can be used with DiffServ to provide low latency, low jitter and low loss. Nine schedulers: PQ, SCFQ, SFQ, WFQ WF2Q+, LLQ, WIRR, RR and WRR were compared. From the results obtained it was found that LLQ provides the best overall performance than other schedulers, hence it was used for the remainder of the NS-2 simulations.

The second scenario was to evaluate the performance of DiffServ under light and heavy load conditions. The results obtained reflected that under light load DiffServ meets the video latency and jitter requirements, however, as the number of video traffic sources was increased, the performance of the DiffServ dropped to unacceptable level, where latency and jitter exceeded the limits set for quality video. Test were also performed to check the network response to link failure and the results revealed that traditional IP and DiffServ are not capable of performing fast rerouting in the event of link or route failure.

MPLS was introduced and the network performance was evaluated again. Comparing the results with those of DiffServ only scenario it was found that the inclusion of MPLS yielded better performance than when it was DiffServ only. For example, latency and jitter decreased tremendously and during link failure there was a fast reroute to an alternative path.

7.4 PERCEPTUAL VIDEO QUALITY EVALUATION

The literature review read revealed that the network performance metrics such as packet loss, jitter, and latency that are have already been evaluated may be insufficient to adequately rate the evaluated perceived video quality by the user. So in order to evaluate the perceptual quality the received video must be decoded and compared to the original video using perceptual video quality evaluation methods such as subjective and objective evaluation techniques.

In this study PSNR, VQM score and DSIS were used to evaluate the perceived video quality. Errors were induced in the transmitted video and in the receiver the perceived video quality was measured, then Intra-coded MBs were introduced during encoding and again error were induced to the transmitted video to check the effectiveness of Intra-coded MBs in containing error propagation. The results revealed that using Intra-coded MBs can help reduce the effects of channel errors, thus improving the quality of the transmitted video.

7.5 FINAL COMMENTS AND RECOMMENDATION FOR FURTHER STUDY

This thesis proposes a framework for proving rural telemedicine with quality video transmission using an IP-based multiservice rural area network. From simulation results it can be concluded that DiffServ-enabled MPLS can compensate for IP best-effort debilities and make IP suitable for multiservice network where there are various applications with different network requirements. From perceived video quality measurements it can also be concluded that including Intra-coded MBs during encoding can help minimize the erasure effects of the channel by containing errors and preventing error propagation.

From simulation conducted it can be concluded that H.264 provides coding efficiency, high quality video that is consistent at high and low bit rates, resilience to transmission errors, scalability, and network friendliness, which will result in perceived quality improvement, hence it is the one that is used in this study.

This study was limited to the communication between the clinic and hospital where all the stations involved are fixed to one point. This study can be expanded to also incorporate quality video transmission to mobile stations in order to accommodate those health professionals who are not always in one place, like paramedics who most of the time are in ambulances on the road with patients, so that the physician can be able to start diagnosing the patient while still on the way to the hospital.

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APPENDICES

APPENDIX A

SCENARIO-1 NS SCRIPTS AND SIMULATION PLOTS

```
#========================================================================
#
# Scenario 1: Comparison of Different schedulers for Real-time traffic
# within the DiffServ environment. Adapted from Simulation-1 by Sergio
# Andreozzi, which is a file that is part of DiffServ4NS
# However, instead of using CBR traffic source for both EF and BE,Video
# trace is used as a traffic sourcefor EF,the number of nodes has been
# increased to 6 for both the source and destination and the number of
# schedulers is also increased to 9 to include PQ, SCFQ, SFQ, WFQ,
# WF2Qp, LLQ, WIRR, RR, and WRR
#
# TOPOLOGY
#
#  ----                                                          ----
# | s0 |-------------|                        |-------------| d0 |
#  ----  100Mb/s     |                        |     100Mb/s  ----
#          1ms       |                        |       1ms
#  ----              |                        |              ----
# | s1 |--------     |                        |     --------| d1 |
#  ---- 100Mb/s \    |                        |   / 100Mb/s  ----
#         1ms    \   |                        |  /    1ms
#  ----           \  |                        | /            ----
# | s2 |--------   | |                        | |   --------| d2 |
#  ---- 100Mb/s \  | |                        | |  / 100Mb/s ----
#         1ms    \ | |                        | | /   1ms
#                 \----        ----        ----/
#                 | e1 |------|core|------| e2 |
#                 / ----  5ms  ----  3ms  ----\
#  ----          /  | |                        | | \           ----
# | s3 |--------    | |                        | |  \--------| d3 |
#  ---- 100Mb/s     | |                        | |   100Mb/s  ----
#         1ms      /  |                        |  \   1ms
#  ----           /   |                        |   \          ----
# | s4 |-------- /    |                        |    \--------| d4 |
#  ---- 100Mb/s       |                        |     100Mb/s  ----
#         1ms         |                        |      1ms
#  ----               |                        |              ----
# | s5 |--------------|                        |-------------| d5 |
#  ---- 100Mb/s                                    100Mb/s   ----
#        1ms                                        1ms
#
#========================================================================


set alg          [lindex $argv 0]
set testTime     [lindex $argv 1]
set EFPacketSize [lindex $argv 2]
```

```
set quiet false

set ns [new Simulator]

set cirEF       300000                   ; # parameters for Token Bucket
Policer
set cbsEF             [expr $EFPacketSize+1]
set EFRate          300000b

#set cbsEF            [expr $EFPacketSize]        ; #

set BGRate          100000b
set BGPacketSize    64

set rndStartTime [new RNG]
$rndStartTime seed 0                          ; # seeds the RNG
heuristically
set rndSourceNode [new RNG]
$rndSourceNode seed 0

set txTime          [expr $EFPacketSize*8.0/2000]; # in millisec

set sumEFQueueLen     0
set sumBEQueueLen     0
set samplesNum        0

#tracing
if {($quiet == "false")} {

set ServiceRate  [open ServiceRate.tr w]

set EFQueueLen  [open EFQueueLen.tr  w]
set BEQueueLen  [open BEQueueLen.tr  w]

set OWD  [open OWD.tr w]
set IPDV [open IPDV.tr w]
}

#set f [open test1.out w]
#$ns trace-all $f

#set nf [open test1.nam w]
#$ns namtrace-all $nf

# Set up the network topology shown at the top of this file:
set s(0) [$ns node]
set s(1) [$ns node]
set s(2) [$ns node]
set s(3) [$ns node]
set s(4) [$ns node]
set s(5) [$ns node]

set e1 [$ns node]
set core [$ns node]
set e2 [$ns node]

set dest(0) [$ns node]
```

```
set dest(1) [$ns node]
set dest(2) [$ns node]
set dest(3) [$ns node]
set dest(4) [$ns node]
set dest(5) [$ns node]

$ns duplex-link $s(0) $e1 100Mb 1ms DropTail
$ns duplex-link $s(1) $e1 100Mb 1ms DropTail
$ns duplex-link $s(2) $e1 100Mb 1ms DropTail
$ns duplex-link $s(3) $e1 100Mb 1ms DropTail
$ns duplex-link $s(4) $e1 100Mb 1ms DropTail
$ns duplex-link $s(5) $e1 100Mb 1ms DropTail

$ns simplex-link $e1 $core 2Mb 5ms dsRED/edge
$ns simplex-link $core $e1 2Mb 5ms dsRED/core

$ns duplex-link $core $e2 5Mb 3ms DropTail

$ns duplex-link $e2 $dest(0) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(1) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(2) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(3) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(4) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(5) 100Mb 1ms DropTail

set qE1C [[$ns link $e1 $core] queue]
set qCE1 [[$ns link $core $e1] queue]

# Set DS RED parameters from Edge1 to Core:
$qE1C set numQueues_ 2
$qE1C setNumPrec 0 2; # queue 0, two levels of precedence
$qE1C setNumPrec 1 1; # queue 1, one level of precedence

if {($alg=="RR")} {
$qE1C setSchedularMode RR
}

if {($alg=="PQ")} {
$qE1C setSchedularMode PRI
}

if {($alg=="WFQ")} {
$qE1C setSchedularMode WFQ
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1     17
}

if {($alg=="SCFQ")} {
$qE1C setSchedularMode SCFQ
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1     17
}

if {($alg=="WRR")} {
$qE1C setSchedularMode WRR
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1     17
```

```
}

if {($alg=="WIRR")} {
$qE1C setSchedularMode WIRR
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1     17
}

if {($alg=="SFQ")} {
$qE1C setSchedularMode SFQ
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1    17
}

if {($alg=="WF2Qp")} {
$qE1C setSchedularMode WF2Qp
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1     17
}

if {($alg=="LLQ")} {
$qE1C setSchedularMode LLQ 1700000
$qE1C addQueueWeight 1    17
}

if {($alg=="LLQ_SFQ")} {
$qE1C setSchedularMode LLQ SFQ 1700000
$qE1C addQueueWeight 1    17
}

$qE1C setQSize 0 30
$qE1C setQSize 1 50

$qE1C setMREDMode DROP ;# could be also specified for each queue: e.g.
$qE1C setMREDMode DROP 0

$qE1C addMarkRule 46 -1 [$dest(0) id] any any ;# EF   recommend
codepoint
$qE1C addMarkRule  0 -1 [$dest(1) id] any any ;# BE   recommend
codepoint

$qE1C addPolicyEntry  46 TokenBucket $cirEF $cbsEF    ;# depending on
SLS
$qE1C addPolicyEntry  48 Dumb
$qE1C addPolicyEntry   0 Dumb

$qE1C addPolicerEntry TokenBucket 46 48
$qE1C addPolicerEntry Dumb  0 0

$qE1C addPHBEntry 46 0 0
$qE1C addPHBEntry 48 0 1
$qE1C addPHBEntry  0 1 0

$qE1C configQ 0 0 30
$qE1C configQ 0 1 -1
$qE1C configQ 1 0 50
```

```
# Set DS RED parameters from Core to Edge1:
$qCE1 setMREDMode DROP
$qCE1 set numQueues_    1
$qCE1 setNumPrec  1
$qCE1 addPHBEntry  10 0 0
$qCE1 configQ      0 1 50

# VBR AND CBR TRAFFIC ACTIVATION
source traffic-generator.tcl

# EF
set startTime [$rndStartTime uniform 0 5]
set sourceNodeID [$rndSourceNode integer 5]
video_connection 0 0 $s($sourceNodeID) $dest(0) 0 $startTime $testTime
if {($quiet == "false")} {
      puts "EF : s($sourceNodeID)->d(0) - Traffic: VIDEO FROM TRACEFILE
- Start $startTime"
}

$Sink_(0) FrequencyDistribution 0.00001 0.00001 $EFPacketSize\_FD.tr

# BE

for {set i 0} {$i < 20} {incr i} {
        set startTime [$rndStartTime uniform 0 5]
        set sourceNodeID [$rndSourceNode integer 5]
      cbr_connection [expr $i+40] 1 $s($sourceNodeID) $dest(1) 1
$BGPacketSize $BGRate $startTime $testTime
      puts "BE: s($sourceNodeID)->d(1) - Traffic: CBR - PktSize:
$BGPacketSize - Rate: $BGRate - Start $startTime"
      set BGPacketSize [expr $BGPacketSize+64]
}

proc record_departure_rate {} {
          global qE1C ServiceRate
          #Get an instance of the simulator
          set ns [Simulator instance]

          #Get the current time
          set now [$ns now]

          #Set the time after which the procedure should be called
again
          set time 1

        set EFRate          [expr [$qE1C getDepartureRate 0  ]/1000 ]
        set BERate         [expr [$qE1C getDepartureRate 1  ]/1000 ]
        puts $ServiceRate "$now $EFRate $BERate"

              #Re-schedule the procedure
          $ns at [expr $now+$time] "record_departure_rate"
}

proc record_delay {} {
          global Sink_ OWD IPDV txTime
          #Get an instance of the simulator
          set ns [Simulator instance]
```

```
        #Set the time after which the procedure should be called
again
        set time 0.5

      set sumOWD  [$Sink_(0) set sumOwd_]
      set sumIPDV [$Sink_(0) set sumIpdv_]
      set pkts    [$Sink_(0) set npktsFlowid_]

        #Get the current time
        set now [$ns now]

        if {($pkts<2)} { puts $OWD "$now 0" } else {
        puts $OWD "$now [expr $sumOWD*1000/$pkts-$txTime]" }
        puts $IPDV "$now [expr $sumIPDV*1000/($pkts-1)]"

            #Re-schedule the procedure
        $ns at [expr $now+$time] "record_delay"
  }


proc record_queue_len {} {
        global EFQueueLen BEQueueLen qE1C samplesNum sumEFQueueLen
sumBEQueueLen quiet

        #Get an instance of the simulator
        set ns [Simulator instance]
        #Set the time after which the procedure should be called
again
        set time 0.5

      set queue0Len [$qE1C getQueueLen 0]
      set queue1Len [$qE1C getQueueLen 1]

      set sumEFQueueLen  [expr $sumEFQueueLen +$queue0Len ]
      set sumBEQueueLen  [expr $sumBEQueueLen +$queue1Len ]
      set samplesNum     [expr $samplesNum+1]

        #Get the current time
        set now [$ns now]

      if {($quiet == "false")} {
          puts $EFQueueLen  "$now $queue0Len"
          puts $BEQueueLen  "$now $queue1Len"
        }

            #Re-schedule the procedure
        $ns at [expr $now+$time] "record_queue_len"
}


proc finish {} {
    global ns  Sink_ OWD IPDV ServiceRate quiet EFPacketSize  alg
EFQueueLen BEQueueLen sumEFQueueLen samplesNum qE1C

    $ns flush-trace
    $Sink_(0) flushFD
```

```
    if {($quiet == "false")} {

      close $OWD
        close $IPDV
        close $ServiceRate
      close $EFQueueLen
      close $BEQueueLen

      source gnuplot-x.tcl
      exec   gnuplot bw_x.p
      exec   gnuplot owd_x.p
      exec   gnuplot ipdv_x.p
      exec   gnuplot queue_x.p
      exec   gnuplot owdFD_x.p
      exec   gnuplot ipdvFD_x.p
    }

    set gIPDV       [open "$alg\_IPDV.tr" a]
    set gOWD        [open "$alg\_OWD.tr"  a]
    set gQueueLen   [open "$alg\_QueueLen.tr"  a]
    set gPktLoss    [open "$alg\_PktLoss.tr"  a]
    set gEFPktLoss  [open "$alg\_EFPktLoss.tr"  a]
    set gEF_MBS     [open "$alg\_EF_MBS.tr"  a]

    set sumOWD  [$Sink_(0) set sumOwd_]
    set sumIPDV [$Sink_(0) set sumIpdv_]
    set npkts   [$Sink_(0) set npktsFlowid_]
    puts $gOWD       "$EFPacketSize [expr $sumOWD*1000/$npkts]"
    puts $gIPDV      "$EFPacketSize [expr $sumIPDV*1000/($npkts-1)]"
    puts $gQueueLen "$EFPacketSize [expr $sumEFQueueLen/$samplesNum]"

    set  EF_MBS     [$qE1C set MBS0_ ]
    puts $gEF_MBS   "$EFPacketSize $EF_MBS"

    set  Pkt          [$qE1C getStat pkts    ]
    set  dropPkt    [$qE1C getStat drops   ]
    set  edropPkt   [$qE1C getStat edrops ]
    puts $gPktLoss   "$EFPacketSize [expr ($Pkt-$dropPkt-
$edropPkt)*100.0/$Pkt]  [expr $dropPkt*100.0/$Pkt] [expr
$edropPkt*100.0/$Pkt]"

    set  Pkt       [$qE1C getStat pkts   46 ]
    set  dropPkt   [$qE1C getStat drops  46 ]
    set  edropPkt  [$qE1C getStat edrops 46 ]
    puts $gEFPktLoss  "$EFPacketSize [expr ($Pkt-$dropPkt-
$edropPkt)*100.0/$Pkt]  [expr $dropPkt*100.0/$Pkt] [expr
$edropPkt*100.0/$Pkt]"

    close $gOWD
    close $gIPDV
    close $gQueueLen
    close $gPktLoss
    close $gEFPktLoss
    close $gEF_MBS

   exit 0
}
```

```
        puts "EF packet size: $EFPacketSize"
        $qE1C printPolicyTable
        $qE1C printPolicerTable
        $ns at 0.0 "record_departure_rate"
        $ns at 6   "record_delay"
        $ns at [expr $testTime/2]       "$qE1C printStats"
        $ns at [expr $testTime - 0.1] "$qE1C printStats"


$ns at 6 "record_queue_len"
$ns at $testTime "finish"

$ns run
```

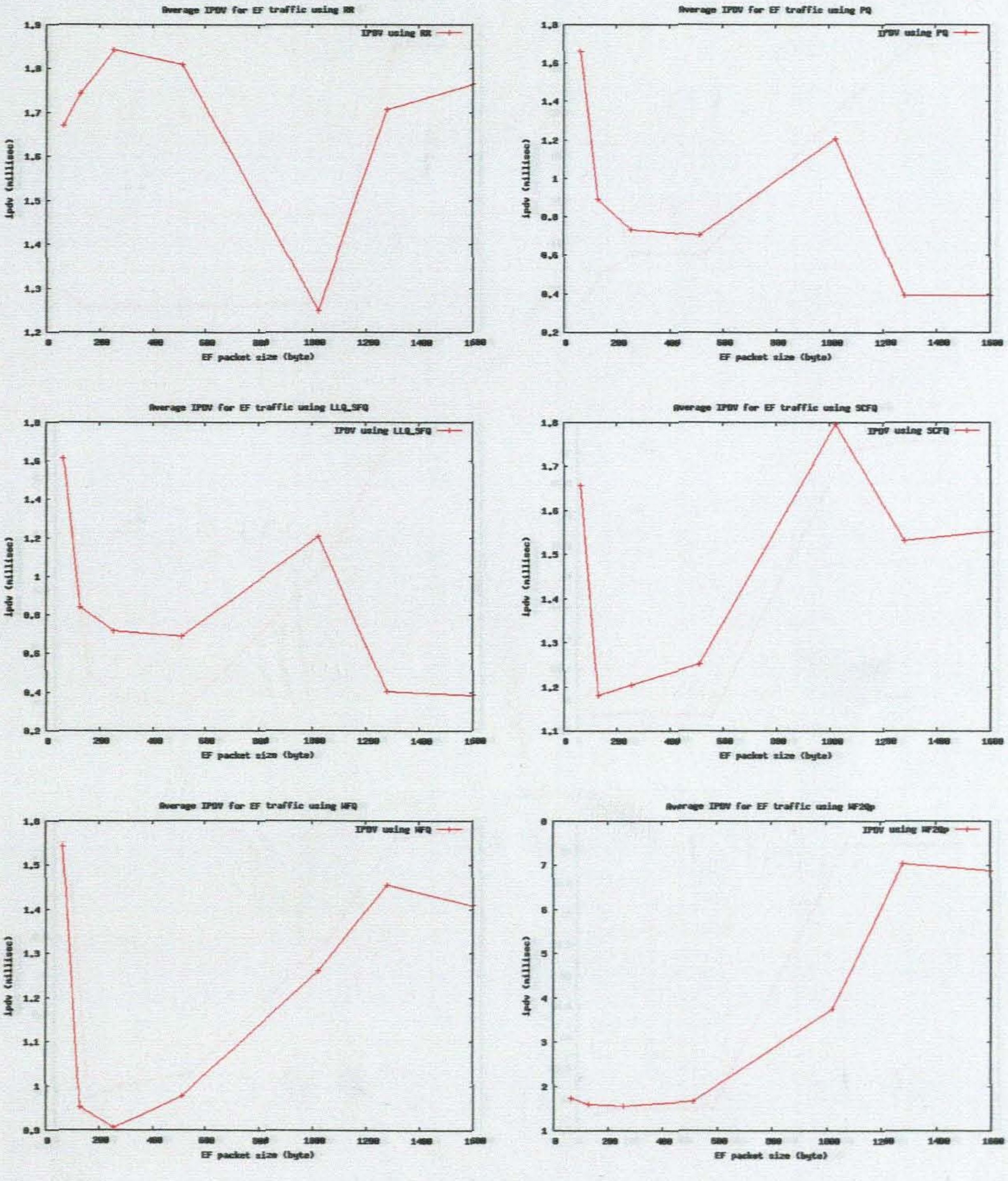
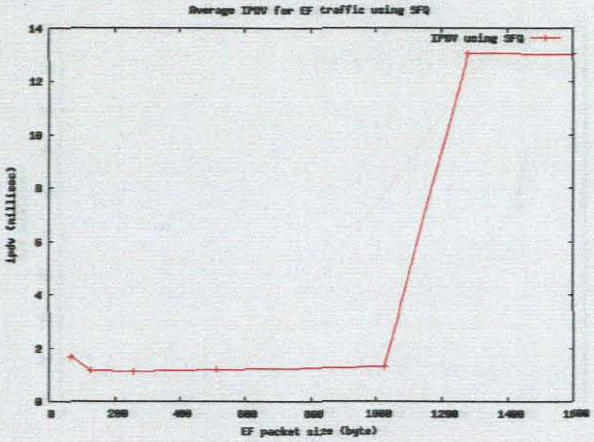
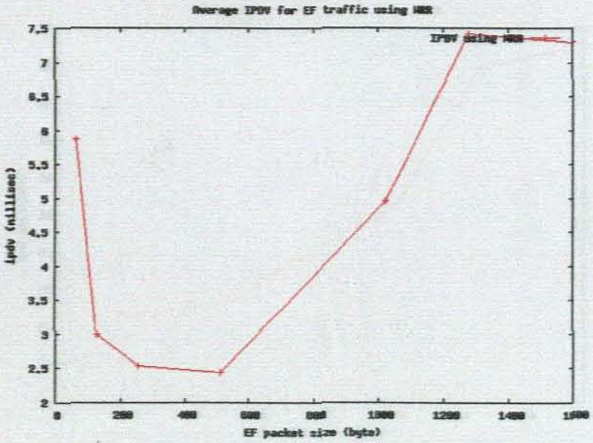
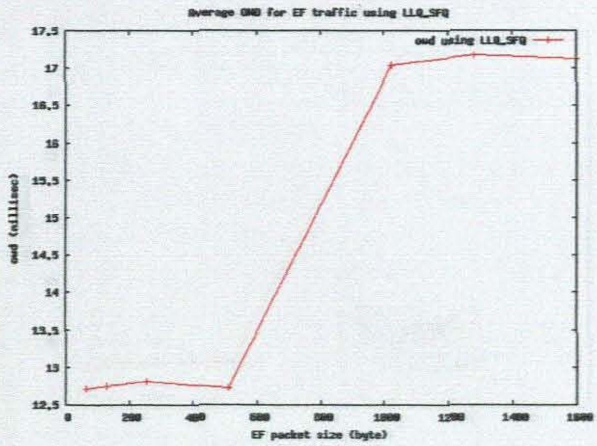
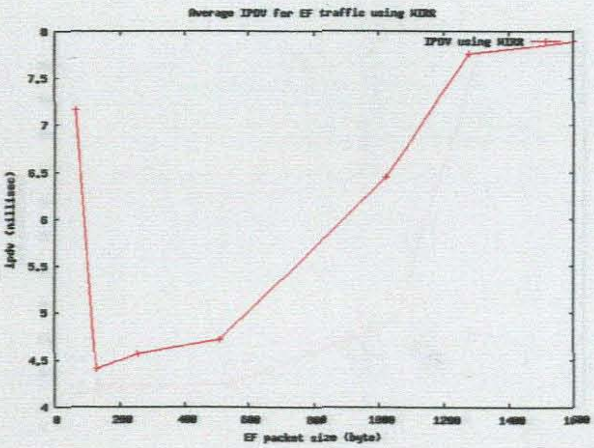


Figure A.1 Average IPDV for schedulers









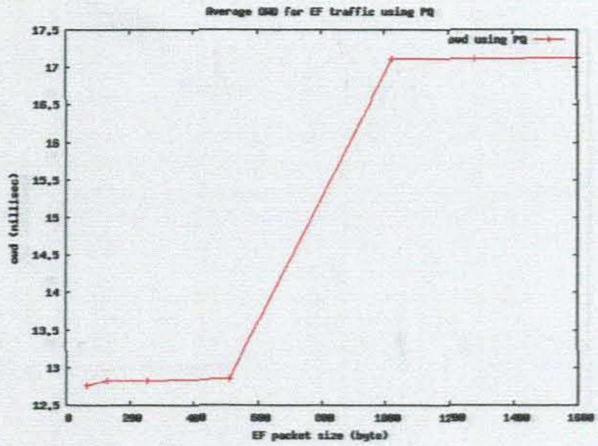


Figure A.1 Average IPDV for schedulers

Figure A.2 Average OWD for schedulers

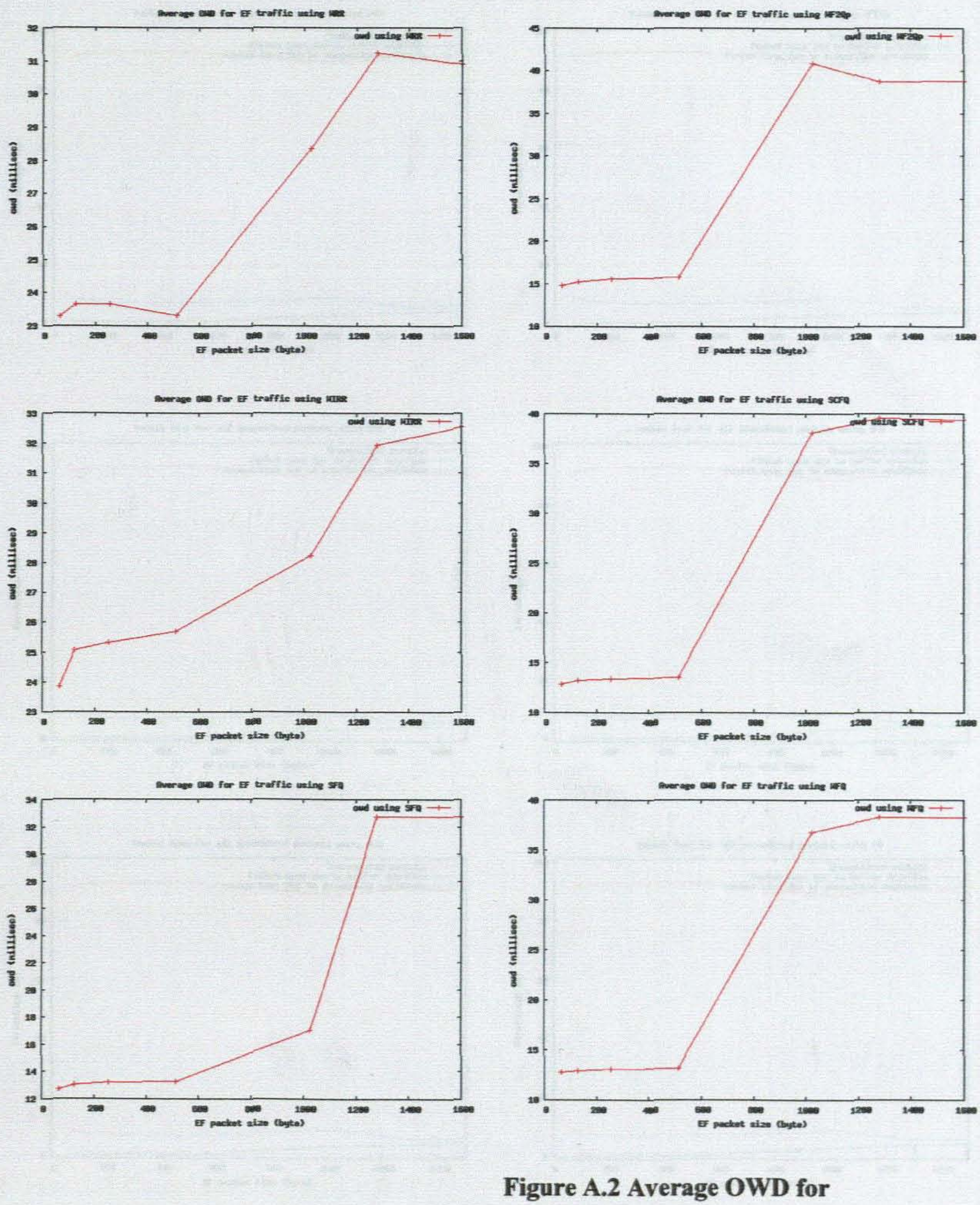
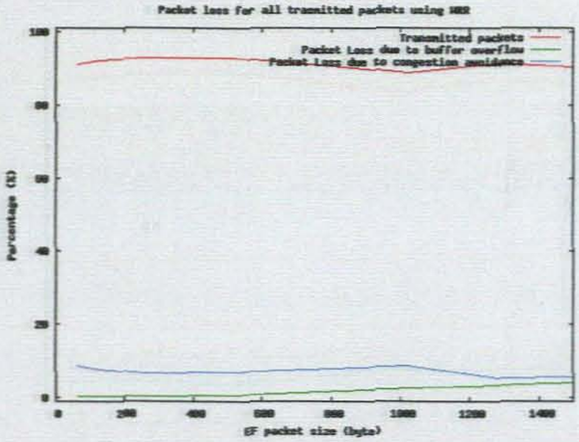
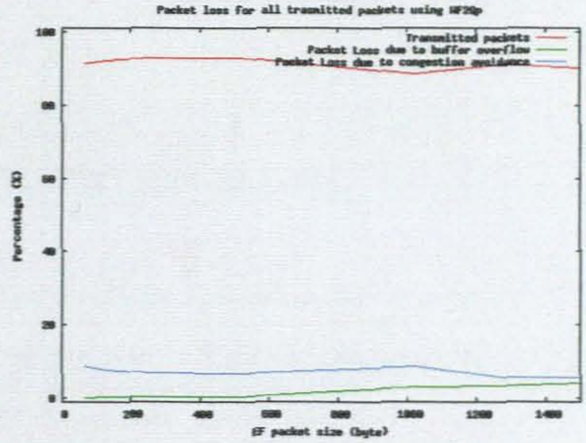
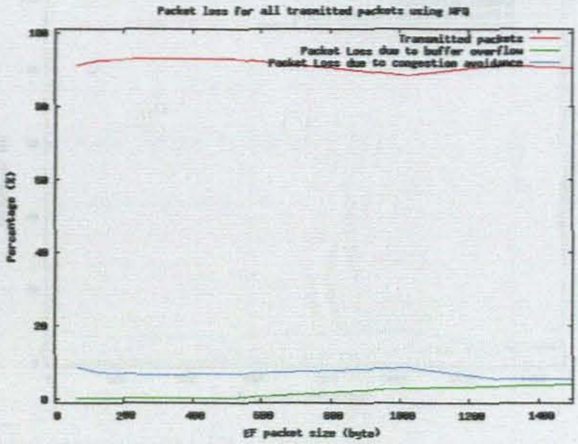
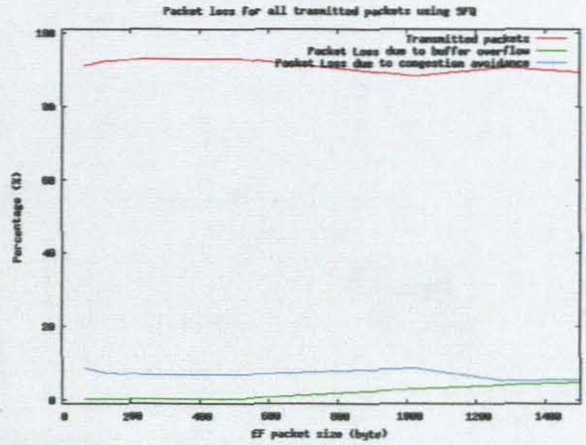


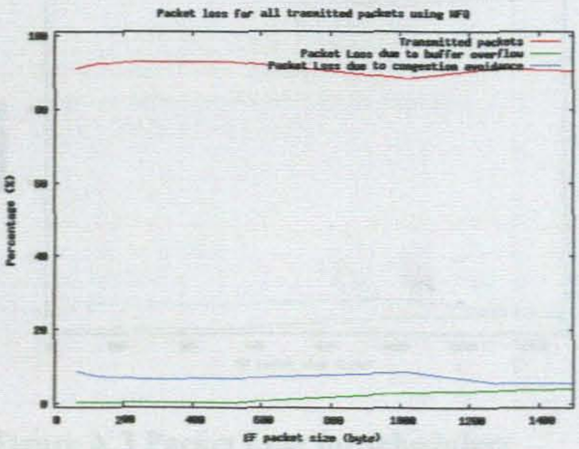
Figure A.2 Average OWD for schedulers

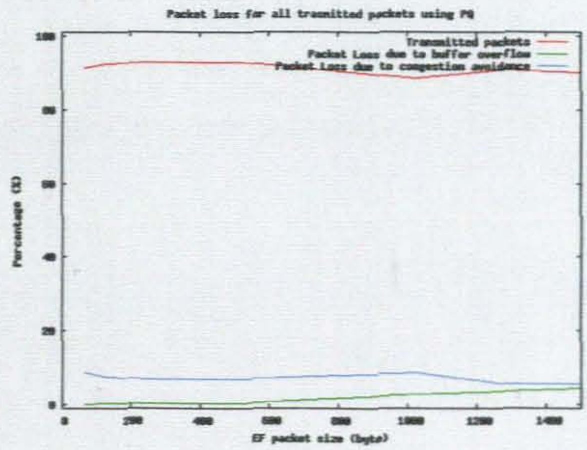

Packet loss for all trasmitted packets using WRR
Transmitted packets
Packet Loss due to buffer overflow
Packet Loss due to congestion avoidance
Percentage (%)
0
20
40
60
80
100
0
200
400
600
800
1000
1200
1400
EF packet size (byte)


Packet loss for all trasmitted packets using WF2Qp
Transmitted packets
Packet Loss due to buffer overflow
Packet Loss due to congestion avoidance
Percentage (%)
0
20
40
60
80
100
0
200
400
600
800
1000
1200
1400
EF packet size (byte)


Packet loss for all trasmitted packets using WFQ
Transmitted packets
Packet Loss due to buffer overflow
Packet Loss due to congestion avoidance
Percentage (%)
0
20
40
60
80
100
0
200
400
600
800
1000
1200
1400
EF packet size (byte)


Packet loss for all trasmitted packets using SFQ
Transmitted packets
Packet Loss due to buffer overflow
Packet Loss due to congestion avoidance
Percentage (%)
0
20
40
60
80
100
0
200
400
600
800
1000
1200
1400
EF packet size (byte)


Packet loss for all trasmitted packets using WFQ
Transmitted packets
Packet Loss due to buffer overflow
Packet Loss due to congestion avoidance
Percentage (%)
0
20
40
60
80
100
0
200
400
600
800
1000
1200
1400
EF packet size (byte)


Packet loss for all trasmitted packets using PQ
Transmitted packets
Packet Loss due to buffer overflow
Packet Loss due to congestion avoidance
Percentage (%)
0
20
40
60
80
100
0
200
400
600
800
1000
1200
1400
EF packet size (byte)

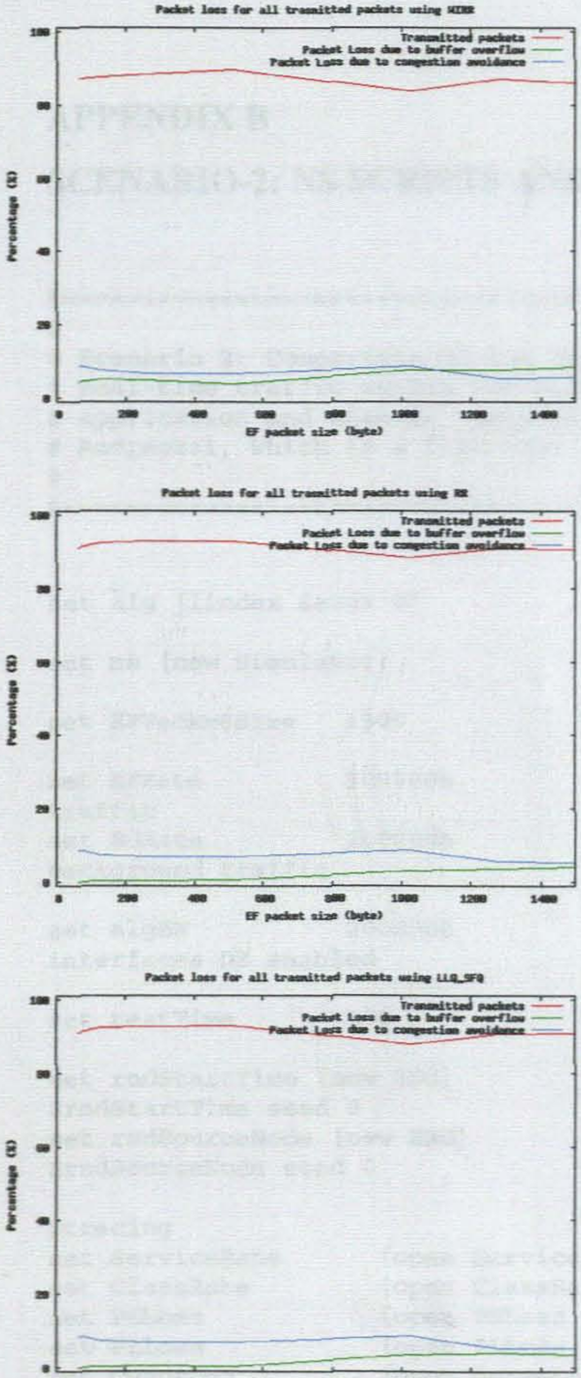


Figure A.3 Packet Loss for schedulers

APPENDIX B

SCENARIO-2: NS SCRIPTS AND SIMULATION PLOTS

```
#==========================================================================
#
# Scenario 2: Comparison of low latency and low jitter schedulers for
# Real-time traffic within the DiffServ environment with more
# application and classes. Adapted from example-2 by Sergio
# Andreozzi, which is a file that is part of DiffServ4NS
#
#==========================================================================


set alg [lindex $argv 0]

set ns [new Simulator]

set EFPacketSize   1300                      ; # EF packet size in bytes

set EFRate         300000b                   ; # 300kbit/s rate for EF
traffic
set BGRate         100000b                   ; # 1Mbit/s   rate for
background traffic

set algBW          2000000                   ; # Bandwith between
interfaces DS enabled

set testTime       100

set rndStartTime [new RNG]
$rndStartTime seed 0                    ; # seeds the RNG heuristically
set rndSourceNode [new RNG]
$rndSourceNode seed 0                   ; # seeds the RNG heuristically

#tracing
set ServiceRate        [open ServiceRate.tr       w]
set ClassRate          [open ClassRate.tr         w]
set PELoss             [open PELoss.tr            w]
set PLLoss             [open PLLoss.tr            w]
set QueueLen           [open QueueLen.tr          w]
set VirtualQueueLen    [open VirQueueLen.tr       w]
set Goodput            [open Goodput.tr           w]
set OWD                [open OWD.tr               w]
set IPDV               [open IPDV.tr              w]

# Set up the network topology shown at the top of this file:
set s(0)  [$ns node]
set s(1)  [$ns node]
set s(2)  [$ns node]
set s(3)  [$ns node]
set s(4)  [$ns node]

set e1  [$ns node]
```

```
set core [$ns node]
set e2 [$ns node]

set dest(0) [$ns node]
set dest(1) [$ns node]
set dest(2) [$ns node]
set dest(3) [$ns node]
set dest(4) [$ns node]

$ns duplex-link $s(0) $e1 100Mb 1ms DropTail
$ns duplex-link $s(1) $e1 100Mb 1ms DropTail
$ns duplex-link $s(2) $e1 100Mb 1ms DropTail
$ns duplex-link $s(3) $e1 100Mb 1ms DropTail
$ns duplex-link $s(4) $e1 100Mb 1ms DropTail

$ns simplex-link $e1 $core 2Mb 5ms dsRED/edge
$ns simplex-link $core $e1 2Mb 5ms dsRED/core

#set txTime         [expr $EFPacketSize*8.0/2000]; # transmission time
for a EF packet in millisec

$ns duplex-link $core $e2 5Mb 3ms DropTail

$ns duplex-link $e2 $dest(0) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(1) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(2) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(3) 100Mb 1ms DropTail
$ns duplex-link $e2 $dest(4) 100Mb 1ms DropTail

set qE1C [[$ns link $e1 $core] queue]
set qCE1 [[$ns link $core $e1] queue]

# Set DS RED parameters from Edge1 to Core:
$qE1C set numQueues_ 3

if {($alg=="PQ")} {
$qE1C setSchedularMode PRI
}

if {($alg=="RR")} {
$qE1C setSchedularMode RR
}

if {($alg=="SCFQ")} {
$qE1C setSchedularMode SCFQ
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1    10
$qE1C addQueueWeight 2     7
}

if {($alg=="WFQ")} {
$qE1C setSchedularMode WFQ
$qE1C addQueueWeight 0     3
$qE1C addQueueWeight 1    10
$qE1C addQueueWeight 2     7
}
```

```
if {($alg=="LLQ")} {
$qE1C setSchedularMode LLQ SFQ 1700000
$qE1C addQueueWeight 1    10
$qE1C addQueueWeight 2     7
}


# Premium Service

# queue and precedence levels settings
$qE1C setQSize    0  50
$qE1C setNumPrec 0    2           ;# Premium Service queue 0, two
levels of precedence

#classifying and marking
$qE1C addMarkRule 46 [$s(0) id] -1 any any ;# packets coming from s0
are marked for premium service
$qE1C addMarkRule 46 [$s(1) id] -1 any any ;# packets coming from s1
are marked for premium service

#metering
$qE1C addPolicyEntry   46 TokenBucket 500000 100000 ;# cir in bit/s,
cbs in bytes
$qE1C addPolicerEntry  TokenBucket  46 51           ;# packets out of
the bucket are marked with DSCP 51
$qE1C addPHBEntry 46 0 0
$qE1C addPHBEntry 51 0 1


#shaping/dropping
$qE1C setMREDMode DROP 0
$qE1C configQ 0 0  30    ;#max  in-profile packets in queue 0
$qE1C configQ 0 1  -1    ;#drop all out-of-profile packets


# Gold Service

# queue and precedence levels settings
$qE1C setQSize    1 150
$qE1C setNumPrec 1    3   ;# Gold Service  queue 1, two levels of
precedence, AF11 for telnet,  AF12,AF13 for ftp,

#classifying and marking
$qE1C addMarkRule 10 -1 -1 any telnet          ;# telnet packets are
marked with DSCP=10
$qE1C addMarkRule 12 -1 -1 any ftp             ;# FTP packets are
marked with DSCP=12

#metering
$qE1C addPolicyEntry  10 Dumb  ; #no policy for telnet
$qE1C addPolicerEntry    Dumb 10

$qE1C addPolicyEntry  12 TSW2CM [expr $algBW/2/2]  ; #when ftp exceeds
0.5Mbit/s, stronger drop
$qE1C addPolicerEntry    TSW2CM 12 14

$qE1C addPHBEntry 10 1 0
```

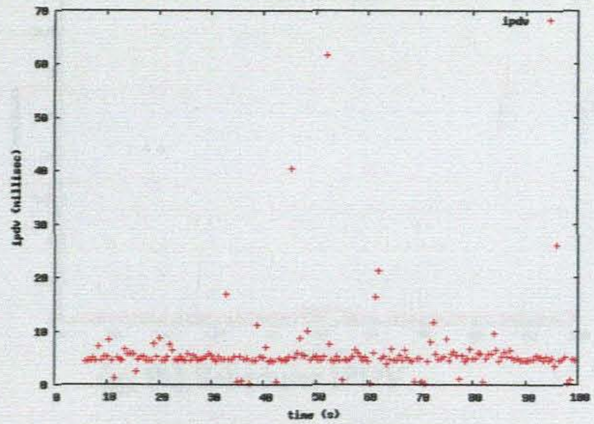
```
$qE1C addPHBEntry 12 1 1
$qE1C addPHBEntry 14 1 2

#shaping/dropping
$qE1C setMREDMode RIO-C 1
$qE1C meanPktSize 1300              ;# needed by the setQueueBW
$qE1C setQueueBW  1 1000000         ;# the alg is related to the bw
assigned to the service
$qE1C configQ      1 0 60 110 0.02
$qE1C configQ      1 1 30  60  0.6
$qE1C configQ      1 2  5  10  0.8


# Best Effort Service

# queue and precedence levels settings
$qE1C setQSize   2 100
$qE1C setNumPrec 2   2   ;# Best Effort Service     queue 2, two levels
of precedence

#classifying and marking
#no rules applied, all packets that do not match any rules are marked
with default codepoint DSCP=0

#metering
$qE1C addPolicyEntry   0 TokenBucket 700000 100000 ;# cir in bit/s, cbs
in bytes
$qE1C addPolicerEntry TokenBucket  0 50
$qE1C addPHBEntry  0 2 0
$qE1C addPHBEntry 50 2 1

#shaping/dropping
$qE1C setMREDMode DROP 2
$qE1C configQ 2 0  100   ;#max  in-profile packets in queue 0
$qE1C configQ 2 1  -1    ;#drop all out-of-profile packets


# Set DS RED parameters from Core to Edge1:
$qCE1 setMREDMode DROP
$qCE1 set numQueues_     1
$qCE1 setQSize           0 60
$qCE1 setNumPrec  1
$qCE1 configQ            0  0 50
$qCE1 addPHBEntry       10  0  0
$qCE1 addPHBEntry        0  0  0

# TRAFFIC ACTIVATION
source utils2.tcl

# EF - VoIP traffic
set startTime [$rndStartTime uniform 1 3]
VoIP_connection 0 0 $s(1) $dest(1) 1 $startTime $testTime
puts "EF : s(1)->d(1) - Traffic: VoIP - Start $startTime"

set startTime [$rndStartTime uniform 1 3]
VoIP_connection 0 0 $s(0) $dest(0) 1 $startTime $testTime
puts "EF : s(0)->d(0) - Traffic: VoIP - Start $startTime"
```

```
set startTime [$rndStartTime uniform 1 3]
VoIP_connection 0 0 $s(2) $dest(2) 1 $startTime $testTime
puts "EF : s(2)->d(2) - Traffic: VoIP - Start $startTime"

# EF - Video traffic
set startTime [$rndStartTime uniform 0 3]
video_connection 0 0 $s(0) $dest(0) 0 $startTime $testTime
puts "EF : s(0)->d(0) - Traffic: VIDEO FROM TRACEFILE - Start
$startTime"

set startTime [$rndStartTime uniform 0 5]
video_connection 0 0 $s(1) $dest(1) 0 $startTime $testTime
puts "EF : s(1)->d(1) - Traffic: VIDEO FROM TRACEFILE - Start
$startTime"

set startTime [$rndStartTime uniform 0 7]
video_connection 0 0 $s(4) $dest(4) 0 $startTime $testTime
puts "EF : s(4)->d(4) - Traffic: VIDEO FROM TRACEFILE - Start
$startTime"


# TELNET TRAFFIC
for {set i 0} {$i < 12} {incr i} {
    telnet_connection [expr $i+1000] $s([expr $i/5+1]) $dest([expr
$i/5+1]) 0 $i
}

# FTP TRAFFIC
for {set i 0} {$i < 12} {incr i} {
    ftp_connection [expr $i+2000] $s([expr $i/5+1]) $dest([expr
$i/5+1]) $i
}

# creating background traffic with different flows
set BGPacketSize 64
set i 0

while {$i<23} {
      set startTime    [$rndStartTime uniform 0 2]
        set sourceNode   [expr [$rndSourceNode integer 4]+1]
        set destNode     [expr [$rndSourceNode integer 4]+1]
      set flowBW    100000
      cbr_connection [expr $i+10] 1 $s($sourceNode) $dest($destNode) 0
$BGPacketSize $flowBW $startTime $testTime
        puts "flow $i: cbr connection ($sourceNode -> $destNode)
starting at time $startTime, rate $flowBW Kbps, packet size
$BGPacketSize"
      set i [expr $i+1]
      set BGPacketSize [expr $BGPacketSize+64]
}


#proc record_goodput {} {
 # global qE1C Goodput

  #Get an instance of the simulator
```

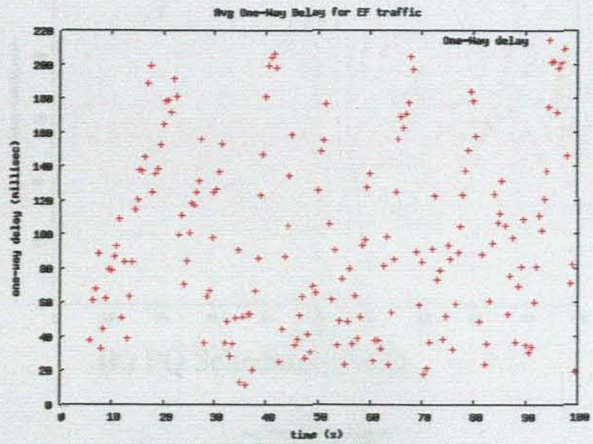
```
  #set ns [Simulator instance]

  #Get the current time
  #set now [$ns now]

  #Set the time after which the procedure should be called again
  #set time 1

  #set   r10       [$qE1C getStat TCPbReTX 10]
  #set   r12       [$qE1C getStat TCPbReTX 12]
  #set   r14       [$qE1C getStat TCPbReTX 14]
  #set   g10       [$qE1C getStat TCPbGoTX 10]
  #set   g12       [$qE1C getStat TCPbGoTX 12]
  #set   g14       [$qE1C getStat TCPbGoTX 14]

  #puts  $Goodput  "$now [expr $g10/($g10+$r10)] [expr
$g12/($g12+$r12)]" ;# [expr $g14/($g14+$r14)]"

  #Re-schedule the procedure
  #$ns at [expr $now+$time] "record_goodput"
#}

proc record_departure_rate {} {
          global qE1C ServiceRate ClassRate
          #Get an instance of the simulator
          set ns [Simulator instance]

          #Get the current time
          set now [$ns now]

          #Set the time after which the procedure should be called
again
          set time 1

          set EFRate        [expr [$qE1C getDepartureRate  0 0]/1000 ]
          set TelnetRate    [expr [$qE1C getDepartureRate  1 0]/1000 ]
          set FtpRate       [expr ([$qE1C getDepartureRate 1 1]+[$qE1C
getDepartureRate 1 2])/1000 ]
          set BERate        [expr [$qE1C getDepartureRate  2  ]/1000 ]
          puts $ClassRate   "$now $EFRate $TelnetRate $FtpRate $BERate"

          set PremiumRate   [expr [$qE1C getDepartureRate 0  ]/1000 ]
          set GoldRate      [expr [$qE1C getDepartureRate 1  ]/1000 ]
          set BERate        [expr [$qE1C getDepartureRate 2  ]/1000 ]
          puts $ServiceRate "$now $PremiumRate $GoldRate $BERate"

              #Re-schedule the procedure
          $ns at [expr $now+$time] "record_departure_rate"
}

proc record_delay {} {
          global Sink_ OWD IPDV

          #Get an instance of the simulator
          set ns [Simulator instance]
          #Set the time after which the procedure should be called
again
```

```
          set time 0.5

          #Get the current time
          set now [$ns now]

          set EF_OWD      [expr 1000*[$Sink_(0) set owd_]]
          puts $OWD       "$now $EF_OWD"

          set EF_IPDV      [expr 1000*[$Sink_(0) set ipdv_]]
          puts $IPDV       "$now $EF_IPDV"

              #Re-schedule the procedure
          $ns at [expr $now+$time] "record_delay"
  }


proc record_queue_len {} {
          global QueueLen VirtualQueueLen qE1C

          #Get an instance of the simulator
          set ns [Simulator instance]
          #Set the time after which the procedure should be called
again
          set time 0.5

          set TelnetQueue     [$qE1C getVirtQueueLen 1 0]
          set FTPinQueue      [$qE1C getVirtQueueLen 1 1]
          set FTPoutQueue     [$qE1C getVirtQueueLen 1 2]

        set PremiumQueue    [$qE1C getQueueLen     0]
        set GoldQueue       [$qE1C getQueueLen     1]
          set BEQueue       [$qE1C getQueueLen     2]

          #Get the current time
          set now [$ns now]

          puts $QueueLen "$now $PremiumQueue $GoldQueue $BEQueue"
          puts $VirtualQueueLen  "$now $TelnetQueue $FTPinQueue
$FTPoutQueue"

              #Re-schedule the procedure
          $ns at [expr $now+$time] "record_queue_len"
}

proc record_packet_loss {} {
          global PELoss PLLoss qE1C

          #Get an instance of the simulator
          set ns [Simulator instance]
          #Set the time after which the procedure should be called
again
          set time 2

          #Get the current time
          set now [$ns now]
  set edropsTelnet [expr [$qE1C getStat edrops 10]*100.0/[$qE1C getStat
pkts 10]]
```

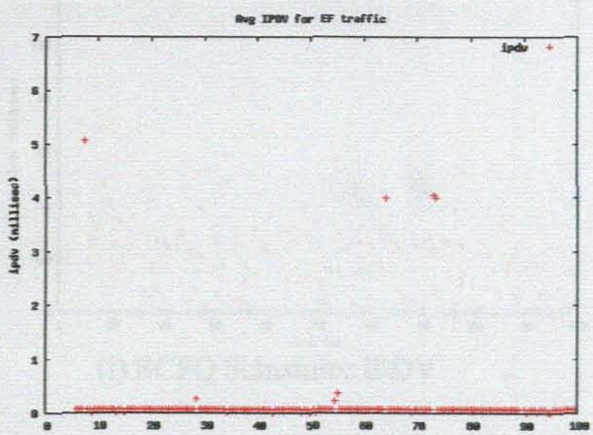
```
  set edropsFTP    [expr ([$qE1C getStat edrops 12]+[$qE1C getStat
edrops 14])*100.0/([$qE1C getStat pkts 12]+[$qE1C getStat pkts 14])]
          puts $PELoss "$now  $edropsTelnet $edropsFTP"

  set dropsTelnet [expr [$qE1C getStat drops 10]*100.0/[$qE1C getStat
pkts 10]]
  set dropsFTP    [expr ([$qE1C getStat drops 12]+[$qE1C getStat drops
14])*100.0/([$qE1C getStat pkts 12]+[$qE1C getStat pkts 14])]
          puts $PLLoss "$now  $dropsTelnet $dropsFTP"

          #Re-schedule the procedure
          $ns at [expr $now+$time] "record_packet_loss"
}

proc finish {} {
global ns Sink_ OWD IPDV EFPacketSize  alg QueueLen VirtualQueueLen
qE1C PELoss PLLoss ServiceRate ClassRate

    $ns flush-trace

    close $OWD
    close $IPDV
    close $PELoss
    close $PLLoss
    close $ServiceRate
    close $ClassRate
    close $QueueLen
    close $VirtualQueueLen

    exec   gnuplot ServiceRate.p
    exec   gnuplot ClassRate.p
    exec   gnuplot owd.p
    exec   gnuplot ipdv.p
    exec   gnuplot queue.p
    exec   gnuplot virqueue.p
    exec   gnuplot pktLoss.p

    exit 0
}

puts "EF packet size: $EFPacketSize"
$qE1C printPolicyTable
$qE1C printPolicerTable
$ns at 0.0 "record_departure_rate"
$ns at 5 "record_packet_loss"
$ns at 6   "record_delay"
$ns at 6 "record_queue_len"
$ns at [expr $testTime/2]     "$qE1C printStats"
$ns at [expr $testTime - 0.1] "$qE1C printStats"
$ns at $testTime "finish"

$ns run
```



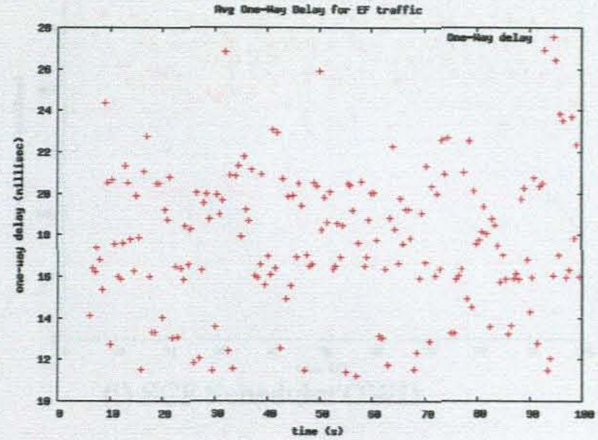
(a) RR Scheduler IPDV



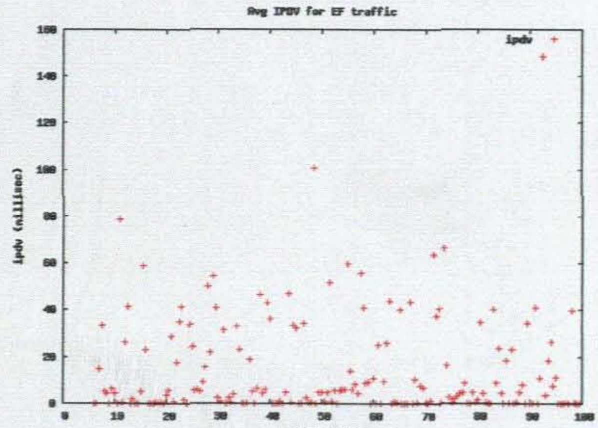
(b) RR Scheduler OWD



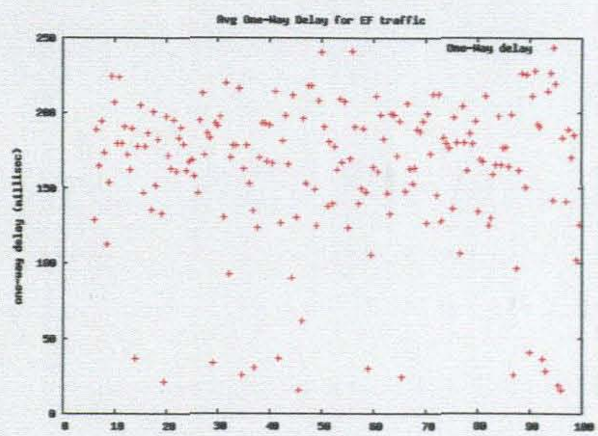
(c) LLQ Scheduler IPDV



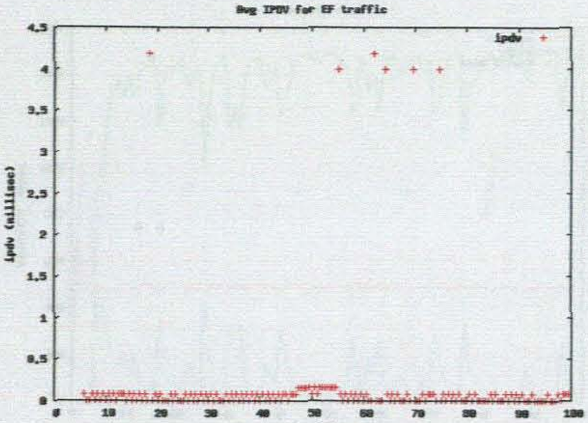
(d) LLQ Scheduler OWD



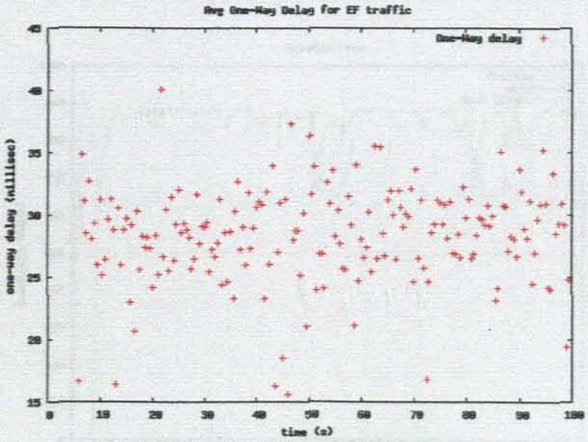
(e) WFQ Scheduler IPDV



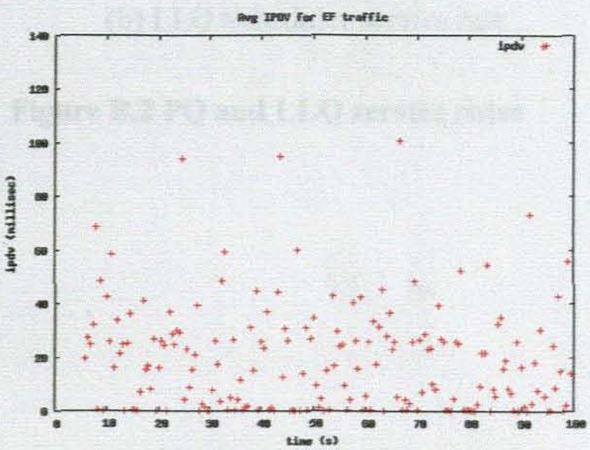
(f) WFQ Scheduler OWD



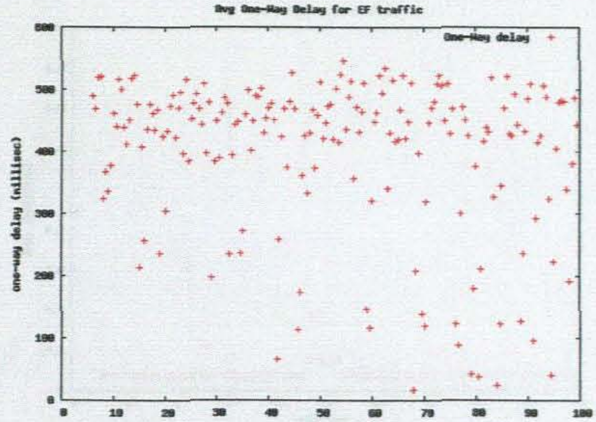
(g) PQ Scheduler IPDV



(h) PQ Scheduler OWD

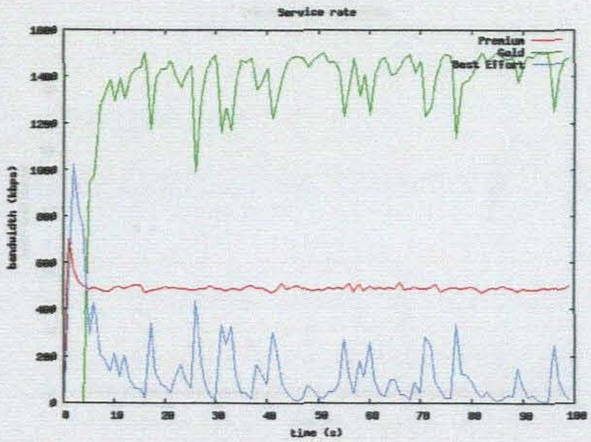


(i) SCFQ Scheduler IPDV

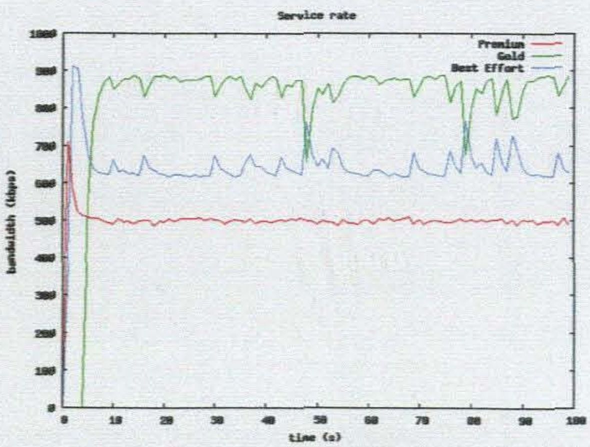


(j) SCF Scheduler OWD

Figure B.1 Scenario-2 IPDV and OWD

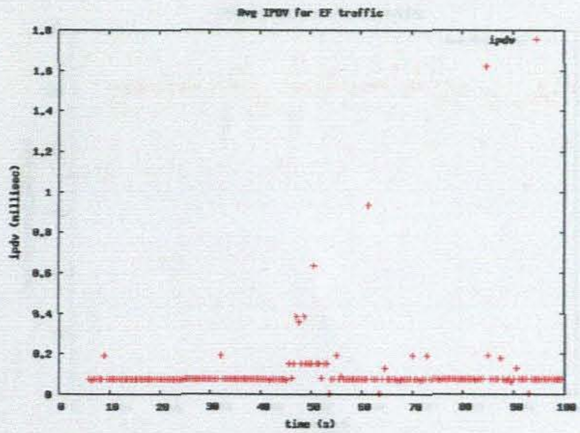


(a) PQ Scheduler service rate

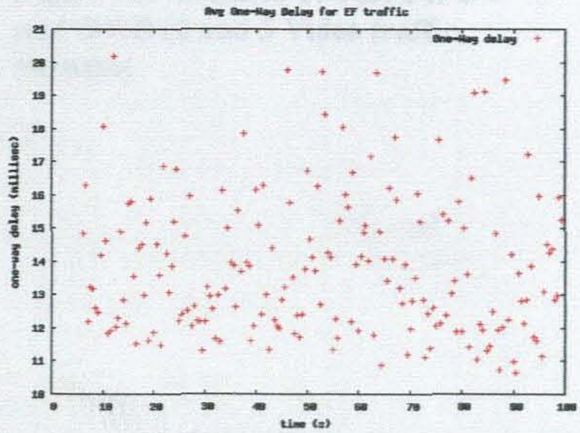


(b) LLQ Scheduler service rate

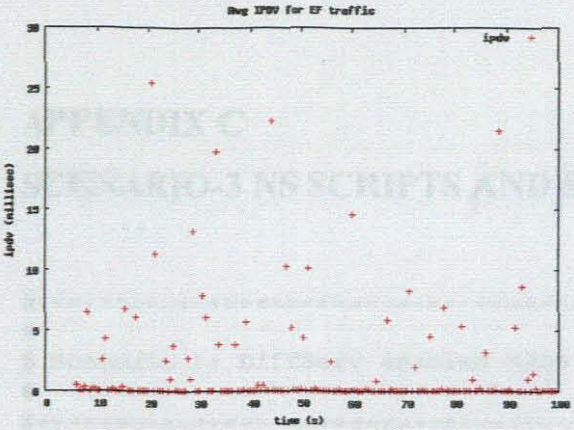
Figure B.2 PQ and LLQ service rates



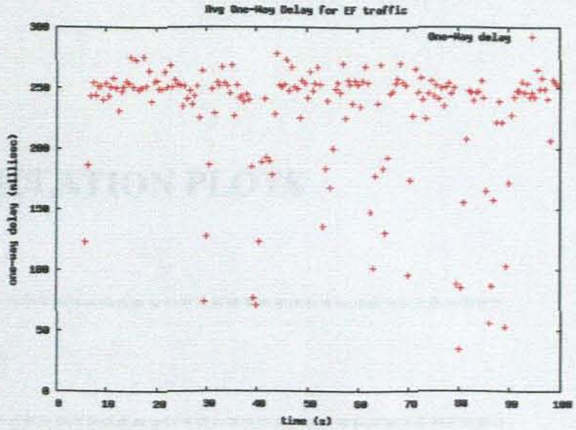
(a) IPDV for 2 Video Traces



(b) OWD for 2 Video Traces



(c) IPDV for 3 Video Traces



(d) OWD for 3 Video Traces

Figure B.3 Scenario-2: Test 2 IPDV and OWD (2 and 3 Video traffic streams)

APPENDIX C

SCENARIO-3 NS SCRIPTS AND SIMULATION PLOTS

```
#=====================================================================
#
# Scenario 3: DiffServ enabled MPLS
#
#=====================================================================


set alg [lindex $argv 0]

set ns [new Simulator]

set EFPacketSize   1300                    ; # EF packet size in bytes

set EFRate         300000b                 ; # 300kbit/s rate for EF
traffic
set BGRate         100000b                 ; # 1Mbit/s   rate for
background traffic

set algBW          2000000                 ; # Bandwith between
interfaces DS enabled

set testTime       60

#Define different colors for data flows within NAM
$ns color 1 gold
$ns color 2 green
$ns color 3 Red
$ns color 4 blue
$ns color 5 yellow
$ns color 6 brown

set rndStartTime [new RNG]
$rndStartTime seed 0                ; # seeds the RNG heuristically
set rndSourceNode [new RNG]
$rndSourceNode seed 0               ; # seeds the RNG heuristically

set nf [open out.nam w]
$ns namtrace-all $nf

#tracing
set ServiceRate       [open ServiceRate.tr       w]
set ClassRate         [open ClassRate.tr         w]
set PELoss            [open PELoss.tr            w]
set PLLoss            [open PLLoss.tr            w]
set QueueLen          [open QueueLen.tr          w]
set VirtualQueueLen   [open VirQueueLen.tr       w]
set Goodput           [open Goodput.tr           w]
set OWD               [open OWD.tr               w]
set IPDV              [open IPDV.tr              w]
```

```
#=============================================
# Set dynamic distance-vector routing protocol
#=============================================
$ns rtproto DV

#=========================================================
# Define trigger strategy, Label Distribution Control Mode
# and Label Allocation and Distribution Scheme
#=========================================================
Classifier/Addr/MPLS set control_driven_ 1
#Classifier/Addr/MPLS enable-on-demand
#Classifier/Addr/MPLS enable-ordered-control

#
# Turn on all traces to stdout
#
Agent/LDP set trace_ldp_ 1
Classifier/Addr/MPLS set trace_mpls_ 1
#
# use 'List' scheduling of events
#
$ns use-scheduler List
#=============================================================
# define nodes and MPLS LSRs (in case of a LSR, the [$ns node]
# command has to be preceded by node-config -MPLS ON
# and succeeded by node-config -MPLS OFF
#=============================================================


# Set up the network topology shown at the top of this file:
set s(0) [$ns node]
set s(1) [$ns node]
set s(2) [$ns node]
set s(3) [$ns node]
set s(4) [$ns node]
set e5 [$ns node]

$ns node-config -MPLS ON
set LSR6 [$ns node]
$LSR6 label "ingress_router"

set LSR7 [$ns node]
set LSR8 [$ns node]
$LSR7 shape "box"
$LSR8 shape "box"

set LSR9 [$ns node]
$LSR9 label "egress_router"
$ns node-config -MPLS OFF

set dest(0) [$ns node]
set dest(1) [$ns node]
set dest(2) [$ns node]
set dest(3) [$ns node]
set dest(4) [$ns node]
```

```
$ns duplex-link $s(0) $e5 100Mb 1ms DropTail
$ns duplex-link $s(1) $e5 100Mb 1ms DropTail
$ns duplex-link $s(2) $e5 100Mb 1ms DropTail
$ns duplex-link $s(3) $e5 100Mb 1ms DropTail
$ns duplex-link $s(4) $e5 100Mb 1ms DropTail


$ns simplex-link $e5 $LSR6 5Mb 5ms dsRED/edge
$ns simplex-link $LSR6 $e5 5Mb 5ms dsRED/core
$ns duplex-link $LSR6 $LSR7 5Mb 5ms DropTail
$ns duplex-link $LSR6 $LSR8 5Mb 5ms DropTail


#set txTime          [expr $EFPacketSize*8.0/2000]; # transmission time
for a EF packet in millisec

$ns duplex-link $LSR7 $LSR9 5Mb 3ms DropTail
$ns duplex-link $LSR8 $LSR9 5Mb 3ms DropTail

$ns duplex-link $LSR9 $dest(0) 100Mb 1ms DropTail
$ns duplex-link $LSR9 $dest(1) 100Mb 1ms DropTail
$ns duplex-link $LSR9 $dest(2) 100Mb 1ms DropTail
$ns duplex-link $LSR9 $dest(3) 100Mb 1ms DropTail
$ns duplex-link $LSR9 $dest(4) 100Mb 1ms DropTail

#==================================================================
# The default value of a link cost (1) can be adjusted
# Notice that the procedure sets the cost along one direction only!
#==================================================================
#$ns cost $LSR3 $LSR4 3
#$ns cost $LSR4 $LSR3 3

#========================================================
# Install/configure LDP agents on all MPLS nodes,
# and set path restoration function that reroutes traffic
# around a link failure in a LSP to an alternative LSP.
# There are 2 options as follows:
# "new": create new alternative path if one doesn't exist
# "drop": do not create any new alternative path
#
# Adjust loop length to address all LSRs (MPLS nodes).
#========================================================
for {set i 6} {$i < 10} {incr i} {
set a LSR$i
for {set j [expr $i+1]} {$j < 10} {incr j} {
set b LSR$j
eval $ns LDP-peer $$a $$b
}
set m [eval $$a get-module "MPLS"]
$m enable-reroute "new"
}


#============================
# Set ldp-message color in NAM
#============================
$ns ldp-request-color blue
```

```
$ns ldp-mapping-color red
$ns ldp-withdraw-color magenta
$ns ldp-release-color orange
$ns ldp-notification-color yellow
#


set qE1C1 [[$ns link $e5 $LSR6] queue]
set qC1E1 [[$ns link $LSR6 $e5] queue]

# Set DS RED parameters from Edge1 to Core:
$qE1C1 set numQueues_ 3

if {($alg=="PQ")} {
$qE1C1 setSchedularMode PRI
}

if {($alg=="SCFQ")} {
$qE1C1 setSchedularMode SCFQ
$qE1C1 addQueueWeight 0     3
$qE1C1 addQueueWeight 1    10
$qE1C1 addQueueWeight 2     7
}

if {($alg=="LLQ")} {
$qE1C1 setSchedularMode LLQ SFQ 1700000
$qE1C1 addQueueWeight 1    10
$qE1C1 addQueueWeight 2     7
}


# Premium Service

# queue and precedence levels settings
$qE1C1 setQSize   0  50
$qE1C1 setNumPrec 0   2            ;# Premium Service            queue
0, two levels of precedence

#classifying and marking
$qE1C1 addMarkRule 46 [$s(0) id] -1 any any ;# packets coming from s0
are marked for premium service
#$qE1C1 addMarkRule 46 -1 -1 any trace
#$qE1C1 addMarkRule 46 -1 -1 any exponential

#metering
$qE1C1 addPolicyEntry   46 TokenBucket 500000 100000 ;# cir in bit/s,
cbs in bytes
$qE1C1 addPolicerEntry  TokenBucket  46 51           ;# packets out of
the bucket are marked with DSCP 51
$qE1C1 addPHBEntry 46 0 0
$qE1C1 addPHBEntry 51 0 1


#shaping/dropping
$qE1C1 setMREDMode DROP 0
$qE1C1 configQ 0 0  30   ;#max  in-profile packets in queue 0
$qE1C1 configQ 0 1  -1   ;#drop all out-of-profile packets
```

```
# Gold Service

# queue and precedence levels settings
$qE1C1 setQSize   1 150
$qE1C1 setNumPrec 1   3   ;# Gold Service  queue 1, two levels of
precedence, AF11 for telnet,  AF12,AF13 for ftp,

#classifying and marking
$qE1C1 addMarkRule 10 -1 -1 any telnet          ;# telnet packets are
marked with DSCP=10
$qE1C1 addMarkRule 12 -1 -1 any ftp             ;# FTP packets are
marked with DSCP=12

#metering
$qE1C1 addPolicyEntry  10 Dumb  ; #no policy for telnet
$qE1C1 addPolicerEntry    Dumb 10

$qE1C1 addPolicyEntry  12 TSW2CM [expr $algBW/2/2]  ; #when ftp exceeds
0.5Mbit/s, stronger drop
$qE1C1 addPolicerEntry    TSW2CM 12 14

$qE1C1 addPHBEntry 10 1 0
$qE1C1 addPHBEntry 12 1 1
$qE1C1 addPHBEntry 14 1 2

#shaping/dropping
$qE1C1 setMREDMode RIO-C 1
$qE1C1 meanPktSize 1300               ;# needed by the setQueueBW
$qE1C1 setQueueBW  1 1000000          ;# the alg is related to the bw
assigned to the service
$qE1C1 configQ     1 0 60 110 0.02
$qE1C1 configQ     1 1 30  60  0.6
$qE1C1 configQ     1 2  5  10  0.8


# Best Effort Service

# queue and precedence levels settings
$qE1C1 setQSize   2 100
$qE1C1 setNumPrec 2   2   ;# Best Effort Service    queue 2, two
levels of precedence

#classifying and marking
#no rules applied, all packets that do not match any rules are marked
with default codepoint DSCP=0

#metering
$qE1C1 addPolicyEntry   0 TokenBucket 700000 100000 ;# cir in bit/s,
cbs in bytes
$qE1C1 addPolicerEntry TokenBucket  0 50
$qE1C1 addPHBEntry  0 2 0
$qE1C1 addPHBEntry 50 2 1

#shaping/dropping
$qE1C1 setMREDMode DROP 2
```

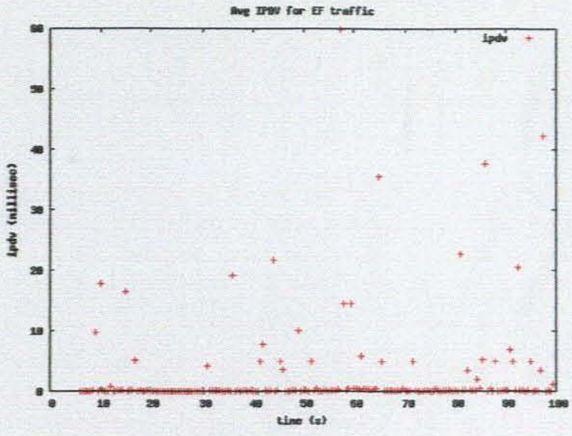
```
$qE1C1 configQ 2 0  100   ;#max  in-profile packets in queue 0
$qE1C1 configQ 2 1  -1    ;#drop all out-of-profile packets


# Set DS RED parameters from Core to Edge1:
$qC1E1 setMREDMode DROP
$qC1E1 set numQueues_    1
$qC1E1 setQSize           0 60
$qC1E1 setNumPrec        1
$qC1E1 configQ            0   0 50
$qC1E1 addPHBEntry       10   0   0
$qC1E1 addPHBEntry        0   0   0

# TRAFFIC ACTIVATION
source utils2.tcl

# # EF - VoIP traffic
set startTime [$rndStartTime uniform 1 3]
VoIP_connection 0 0 $s(1) $dest(1) 1 $startTime $testTime
puts "EF : s(1)->d(1) - Traffic: VoIP - Start $startTime"

set startTime [$rndStartTime uniform 1 5]
VoIP_connection 0 0 $s(0) $dest(0) 1 $startTime $testTime
puts "EF : s(0)->d(0) - Traffic: VoIP - Start $startTime"

set startTime [$rndStartTime uniform 0 9]
VoIP_connection 0 0 $s(2) $dest(2) 0 $startTime $testTime
puts "EF : s(2)->d(2) - Traffic: VoIP - Start $startTime"

# EF - Video traffic
set startTime [$rndStartTime uniform 0 3]
video_connection 0 0 $s(0) $dest(0) 0 $startTime $testTime
puts "EF : s(0)->d(0) - Traffic: VIDEO FROM TRACEFILE - Start
$startTime"

set startTime [$rndStartTime uniform 0 5]
video_connection 0 0 $s(1) $dest(1) 0 $startTime $testTime
puts "EF : s(1)->d(1) - Traffic: VIDEO FROM TRACEFILE - Start
$startTime"

set startTime [$rndStartTime uniform 0 8]
video_connection 0 0 $s(4) $dest(4) 0 $startTime $testTime
puts "EF : s(4)->d(4) - Traffic: VIDEO FROM TRACEFILE - Start
$startTime"


# TELNET TRAFFIC
for {set i 0} {$i < 5} {incr i} {
    telnet_connection [expr $i+1000] $s([expr $i+0]) $dest([expr $i+0])
0 $i
}

# FTP TRAFFIC
for {set i 0} {$i < 5} {incr i} {
    ftp_connection [expr $i+2000] $s([expr $i+0]) $dest([expr $i+0]) $i
}
```

```
# creating background traffic with different flows
set BGPacketSize 64
set i 0

while {$i<23} {
      set startTime   [$rndStartTime uniform 0 2]
        set sourceNode  [expr [$rndSourceNode integer 4]+1]
        set destNode    [expr [$rndSourceNode integer 4]+1]
      set flowBW    100000
      cbr_connection [expr $i+10] 1 $s($sourceNode) $dest($destNode) 0
$BGPacketSize $flowBW $startTime $testTime
        puts "flow $i: cbr connection ($sourceNode -> $destNode)
starting at time $startTime, rate $flowBW Kbps, packet size
$BGPacketSize"
      set i [expr $i+1]
      set BGPacketSize [expr $BGPacketSize+64]
}

$ns at 0.10 "[$LSR9 get-module MPLS] ldp-trigger-by-withdraw 7 -1"
$ns at 0.30 "[$LSR6 get-module MPLS] make-explicit-route 9 6_7_9 1000 -
1"
$ns at 0.30 "[$LSR6 get-module MPLS] make-explicit-route 9 6_8_9 1005 -
1"

# comment the following line when simulating plain IP routing
# only uncomment the following line if simulating ER-LSP
#$ns at 0.50 "[$LSR2 get-module MPLS] flow-erlsp-install 7 -1 1000"
# comment the following two lines when simulating plain IP routing
# only uncomment the following lines if simulating PBER-LSP
$ns at 0.50 "[$LSR6 get-module MPLS] flow-erlsp-install 7 -1 1000"
$ns at 0.50 "[$LSR6 get-module MPLS] flow-erlsp-install 7 -1 1001"



proc record_departure_rate {} {
          global qE1C1 ServiceRate ClassRate
          #Get an instance of the simulator
          set ns [Simulator instance]

          #Get the current time
          set now [$ns now]

          #Set the time after which the procedure should be called
again
          set time 1

          set EFRate          [expr [$qE1C1 getDepartureRate  0 0]/1000 ]
          set TelnetRate      [expr [$qE1C1 getDepartureRate  1 0]/1000 ]
          set FtpRate         [expr ([$qE1C1 getDepartureRate 1
1]+[$qE1C1 getDepartureRate 1 2])/1000 ]
          set BERate          [expr [$qE1C1 getDepartureRate  2  ]/1000 ]
          puts $ClassRate     "$now $EFRate $TelnetRate $FtpRate $BERate"

          set PremiumRate     [expr [$qE1C1 getDepartureRate 0  ]/1000 ]
          set GoldRate        [expr [$qE1C1 getDepartureRate 1  ]/1000 ]
          set BERate          [expr [$qE1C1 getDepartureRate 2  ]/1000 ]
          puts $ServiceRate "$now $PremiumRate $GoldRate $BERate"
```

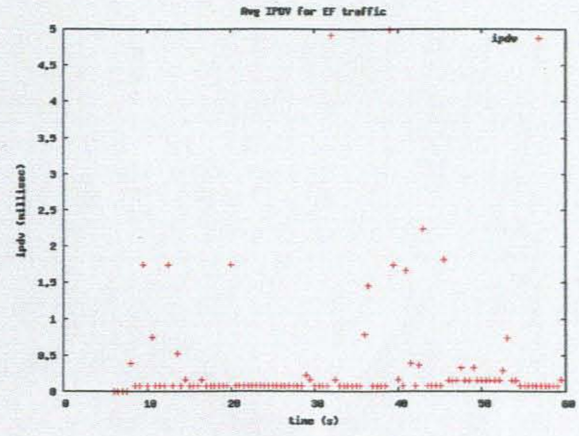
```
		#Re-schedule the procedure
	$ns at [expr $now+$time] "record_departure_rate"
}

proc record_delay {} {
	global Sink_ OWD IPDV

	#Get an instance of the simulator
	set ns [Simulator instance]
	#Set the time after which the procedure should be called
again
	set time 0.5

	#Get the current time
	set now [$ns now]

	set EF_OWD	[expr 1000*[$Sink_(0) set owd_]]
	puts $OWD	"$now $EF_OWD"

	set EF_IPDV	[expr 1000*[$Sink_(0) set ipdv_]]
	puts $IPDV	"$now $EF_IPDV"

		#Re-schedule the procedure
	$ns at [expr $now+$time] "record_delay"
  }

proc record_queue_len {} {
	global QueueLen VirtualQueueLen qE1C1

	#Get an instance of the simulator
	set ns [Simulator instance]
	#Set the time after which the procedure should be called
again
	set time 0.5

	set TelnetQueue		[$qE1C1 getVirtQueueLen 1 0]
	set FTPinQueue		[$qE1C1 getVirtQueueLen 1 1]
	set FTPoutQueue		[$qE1C1 getVirtQueueLen 1 2]

      set PremiumQueue	[$qE1C1 getQueueLen	0]
      set GoldQueue		[$qE1C1 getQueueLen	1]
	set BEQueue		[$qE1C1 getQueueLen	2]

	#Get the current time
	set now [$ns now]

	puts $QueueLen "$now $PremiumQueue $GoldQueue $BEQueue"
	puts $VirtualQueueLen  "$now $TelnetQueue $FTPinQueue
$FTPoutQueue"

		#Re-schedule the procedure
	$ns at [expr $now+$time] "record_queue_len"
}
```

```
proc record_packet_loss {} {
          global PELoss PLLoss qE1C1

          #Get an instance of the simulator
          set ns [Simulator instance]

      #Set the time after which the procedure should be called again
          set time 2

          #Get the current time
 set now [$ns now]set edropsTelnet [expr [$qE1C1 getStat edrops
10]*100.0/[$qE1C1 getStat pkts 10]]
set edropsFTP    [expr ([$qE1C1 getStat edrops 12]+[$qE1C1 getStat
edrops 14])*100.0/([$qE1C1 getStat pkts 12]+[$qE1C1 getStat pkts 14])]
          puts $PELoss "$now  $edropsTelnet $edropsFTP"set dropsTelnet
[expr [$qE1C1 getStat drops 10]*100.0/[$qE1C1 getStat pkts 10]]
set dropsFTP    [expr ([$qE1C1 getStat drops 12]+[$qE1C1 getStat drops
14])*100.0/([$qE1C1 getStat pkts 12]+[$qE1C1 getStat pkts 14])]
          puts $PLLoss "$now  $dropsTelnet $dropsFTP"

          #Re-schedule the procedure
          $ns at [expr $now+$time] "record_packet_loss"
}

proc finish {} {
global ns Sink_ OWD IPDV EFPacketSize  alg QueueLen VirtualQueueLen
qE1C1 PELoss PLLoss ServiceRate ClassRate

    $ns flush-trace

    close $OWD
    close $IPDV
    close $PELoss
    close $PLLoss
    close $ServiceRate
    close $ClassRate
    close $QueueLen
    close $VirtualQueueLen

    exec   gnuplot ServiceRate.p
    exec   gnuplot ClassRate.p
    exec   gnuplot owd.p
    exec   gnuplot ipdv.p
    exec   gnuplot queue.p
    exec   gnuplot virqueue.p
    #exec   gnuplot pktLoss.p
    exec nam out.nam &

    exit 0
}


puts "EF packet size: $EFPacketSize"
$qE1C1 printPolicyTable
$qE1C1 printPolicerTable
$ns at 0.0 "record_departure_rate"
#$ns at 5 "record_packet_loss"
```

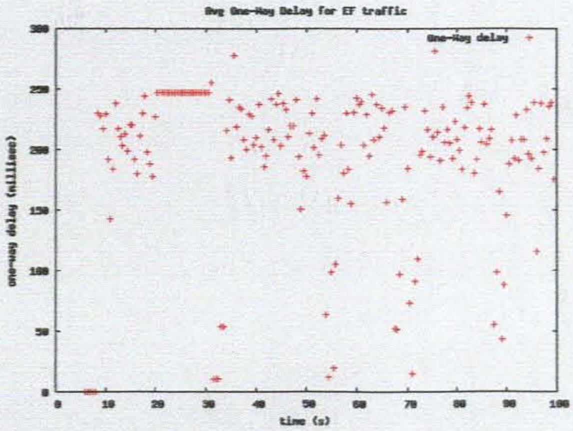
```
$ns at 6   "record_delay"
$ns at 6 "record_queue_len"
$ns at [expr $testTime/2]      "$qE1C1 printStats"
$ns at [expr $testTime - 0.1] "$qE1C1 printStats"
#
# Define the first link failure
#
$ns rtmodel-at 1.5 down $LSR6 $LSR7
$ns rtmodel-at 5.5 down $LSR6 $LSR8

#
# Define when the link have to be restored
#
$ns rtmodel-at 4.5 up   $LSR6 $LSR7
$ns rtmodel-at 6.5 up   $LSR6 $LSR8

$ns at $testTime "finish"

$ns run
```



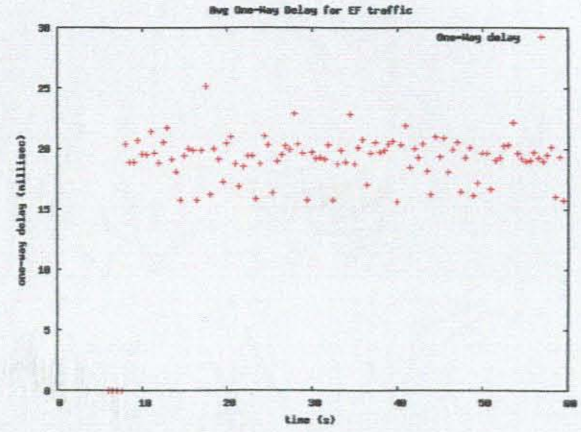
(a) IPDV for DiffServ only



(a) IPDV for DiffServ enabled MPLS



(b) OWD for DiffServ only



(b) OWD for DiffServ enabled MPLS

Figure C.1 IPDV and OWD for DiffServ only

Figure C.2 IPDV and OWD for DiffServ enabled MPLS