Quality of Service for Voice over Next Generation Networks

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Abstract

The Global communications transformation is currently in progress. Packet switched technology has moved from data – only applications into the heart of the network to take up the functions of traditional circuit – switched equipment. Voice over ATM(VoATM) and voice over IP(VoIP) are the two main alternatives for carrying voice packets over NGN's. ATM offers the advantage of its built in quality of service mechanisms. IP on the other hand could not provide QoS guarantees in its traditional form. IP QoS mechanisms evolved only in recent years. There are currently no QoS differences between Next Generation Networks based on VoATM or VoIP. However non QoS agreements are more in favour of VoIP instead of VoATM. This gives VoIP the leading edge bet the Voice over packet technologies.

In this thesis the E – Model was optimized and used to study the effects of delay, utilization and coder design on voice quality. The optimization was used to choose a coder and utilization levels given certain conditions. An optimization algorithm formed through the E – Model was used to assist with the selection of parameters important to VoIP networks. These parameters include the link utilization, voice coder and allowable packet loss. This research also shows us that different utilization, voice coder and packet loss levels are optimal in different situations. A remote and core VoIP Network simulation model was developed and used to study the complex queuing issues surrounding VoIP networks. The models look at some of the variables that need to be controlled in order to minimize delay.

Table of Contents

Chapter 1: NGN's and QoS

1.1. Features of Next Generation Networks (NGN's 1
1.2. Next Generation Networks – The Voice of the Future 1
1.3. Africa's Packet Telephony Network 2
1.4. NGN Migration - Key considerations in the evolution 3
1.4.1. Investment Protection 3
1.4.2. Operational and Capital Savings 4
1.4.3. Scalability 5
1.5. Migration architectures 5
1.5.1. Voice Trunking: Tandem Replacement 5
1.6. An Introduction to QoS 7
1.7. Voice Quality 8
1.7.1. End-to-end Delay 9
1.7.2. Jitter 11
1.7.3. Echo 11
1.7.4. Voice Compression 12
1.7.5. Packet Loss 13
1.8. QoS Management 13
1.8.1. Use of RTCP in Measuring QoS 14
1.8.2. Procedures for maintaining QoS 14
1.8.3. QoS Agreements 15
1.9. The Internet Integrated Services Architecture 16

Chapter 2: Voice transmission over ATM Networks

2.1. Introduction to VoATM 19
2.2. Voice Traffic in ATM Networks 20
2.2.1. CBR Voice Traffic 20
2.2.2. VBR Voice Traffic 21
2.3. Applications of Voice over ATM 21
2.4. ATM Adaptation Types for Voice over ATM 23
2.4.1. Service Classes for VoATM 24
2.4.2. ATM Adaptation Layers (AAL's) 25
2.4.2.1. ATM Adaptation Layer Type 0 (AAL0) 25
2.4.2.2. ATM Adaptation Layer Type 1 (AAL1) 26
2.4.2.3. ATM Adaptation Layer Type 2 (AAL2) 26
2.4.2.4. ATM Adaptation Layer Type 5 (AAL5) 30
2.5. Error Performance of AAL Protocols 31
2.6. Network Design Considerations 31
2.6.1. Technical Challenges 32
2.6.1.1. Echo 32
2.6.1.2. End-to-End Delay 32
2.6.1.3. Silence Suppression 34
2.6.2. Signalling 34
2.6.3. Synchronization 34

2.7. Voice over ATM to the Desktop.. 35
2.7.1. Reference Configuration..35

Chapter 3: Voice Transmission over IP Networks

3.1. Voice over IP Concepts..40
3.2. Why Voice over IP..40
3.3. Applications of Voice over IP..41
3.4. The IP Phone Terminal Equipment..41
3.5. IP Voice Gateways..42
3.6. IP Voice Related Standards and Protocols..43
3.6.1. The Internet Protocol - IPv4..43
3.6.2. The Next Generation of IP - IPv6..45
3.6.3. Transport Control Protocol..45
3.6.4. User Datagram Protocol..46
3.6.5. IETF Real-time Protocol..47
3.6.6. H.323..48
3.6.7. IP Switching..51
3.7. Speech coding..56
3.7.1. Speech and Audio Coders for VoIP and their Quality..57
3.8. Benefits of Voice over IP..59
3.9. Risk Factors..60

Chapter 4: QoS mechanisms for VoIP and VoATM

4.1. QoS supporting mechanisms..63
4.2. Integrated Services Architecture (IntServ)..66
4.2.1. Basic principles..67
4.2.2. RSVP..68
4.2.2.1. Introduction to RSVP..68
4.2.2.2. RSVP Protocol..68
4.2.2.3. Reservation Styles and OPWA..69
4.2.3. IntServ Router architecture..70
4.2.4. Evaluation of benefits and drawbacks..71
4.2.5. Technical effort and availability..74
4.3. Differentiated Services Architecture (Diffserv)..75
4.3.1. Basic principles..75
4.3.2. DiffServ Router architecture..76
4.3.3. Evaluation of benefits and drawbacks..77
4.3.4. Technical effort and availability..79
4.4. Multiprotocol Label Switching (MPLS)..80
4.4.1. Basic principles..80
4.4.2. Evaluation of benefits and drawbacks..82
4.4.3. Technical effort and availability..84
4.5. ATM QoS Mechanisms..85

Chapter 5: Estimation of Voice Quality

5.1. Impairment Calculations of the E Model 88

Chapter 6: The E – Model Optimization

6.1. Description of Optimization Problem 93
6.2. Results 98
6.2.1. Case 1 - Optimizing for Coder Selection 98
6.2.2. Case 2 - Optimizing for Coder and Packet Loss 100
6.2.3. Case 3 - Optimizing for Coder and Link Utilization 104
6.3. Discussion of E-Model Optimization Results 106

Chapter 7: Delay in VoIP networks

7.1. Objectives 109
7.2. Simulation Models 110
7.2.1. Case 1 - Effect of link rate in VoIP Networks 112
7.2.2. Case 2 - Effect of Voice/Data ratio in VoIP Networks 113
7.2.3. Case 3 - DRR queuing in VoIP Networks 115
7.2.4. Case 4 - DRR queuing with Pareto Distribution in VoIP Networks 117
7.3. Discussion of the Delay in VoIP networks 120

Chapter 8: Conclusions

8.1. Conclusions and recommendations 121

References 123

Appendix 1 - OPNET Model Description and Validation

A.1. Voice Model 127
A.2. Best Effort Model 128
A.3. Pareto Generator 129
A.4. Priority Router Model 131
A.5. DRR Router Models 133
A.6. Other Models 134

List of Figures

Figure 1.1 – Africa's Packet Telephony Network..2
Figure 1.2 – Traditional TDM Architecture..5
Figure 1.3 – Voice Trunking with Packet Transport..6
Figure 1.4 – E – model, Impact for Delay..10
Figure 1.5 – QoS Agreements..15
Figure 1.6 – IETF traffic service class hierarchy..17
Figure 2.1 – Procedure of New AAL2 Protocol..27
Figure 2.2 – Structure of new AAL2 Protocol..28
Figure 2.3 – AAL Type 2 connection..28
Figure 2.4 – Relation Between AAL – SAP and ATM – SAP..29
Figure 2.5 – Network configuration for Telephony between Workstations/ Set Tops..36
Figure 2.6 – Inter working with N – ISDN..36
Figure 2.7 – Inter working with Voice over Frame Relay..36
Figure 2.8 – Inter working with Voice over Wireless Phone..37
Figure 2.9 – Inter working with Voice over IP..37
Figure 2.10 – Inter working with PBX..38
Figure 2.11 – Inter working with Voice over Legacy Lans..38
Figure 3.1 – The Framework for IP voice..41
Figure 3.2 – Typical IP voice terminal with video conferencing capabilities..42
Figure 3.3 – IP voice to PSTN gateway..42
Figure 3.4 – The Voice over IP protocol stack..43
Figure 3.5 – IPv4 header..44
Figure 3.6 – IPv6 header..45
Figure 3.7 – TCP header..46
Figure 3.8 – UDP header..46
Figure 3.9 – H.323 terminal and H.323 scope..48
Figure 3.10 – H.323 entities..49
Figure 3.11 – Simple call setup signaling..50
Figure 3.12 – The Scope of H.225.0..51
Figure 3.13 – The structure of IP- switches and the protocols..53
Figure 3.14 – The Ipsilon protocol hierarchy..53
Figure 3.15 – The IP switch as a packet forwarding router..54
Figure 3.16 – Soft – state routing and flow control..54
Figure 3.17 – A switched flow in IP switching..55
Figure 3.18 – IP switching network..56
Figure 4.1 – Router architecture in the IntServ Model..71
Figure 4.2 – Router architecture in the Differentiated Service Model..77
Figure 5.1 – VoIP Voice Quality Levels vs. Delay..91
Figure 6.1 – Case 1, Link Rate = 256 kbps..99
Figure 6.2 – Case 1, Link Rate = 1.5 Mbps..100
Figure 6.3 – Case 2, Link Rate = 256 kbps..102
Figure 6.4 – Case 2, Link Rate = 1.5 Mbps..103
Figure 6.5 – Case 3, Link Rate = 256 kbps..105
Figure 6.6 – Case 3, Link Rate = 1.5 Mbps..105
Figure 7.1 – Delay Analysis of VoIP Network..109
Figure 7.2 – Remote Site (Edge) VoIP Model..110
Figure 7.3 – Backbone Network Model..111
Figure 7.4 – Case 1 – Remote VoIP Network - results..112
Figure 7.5 – Case 1 – Backbone VoIP Network - results..113
Figure 7.6 – Case 2 – Remote VoIP Network - results..114
Figure 7.7 – Case 2 – Backbone VoIP Network - results..115
Figure 7.8 – Case 3 – Remote VoIP Network- results..116

Figure 7.9 – Case 3 - Backbone VoIP Network- results 117
Figure 7.10 – Case 4 - Remote VoIP Network – Voice Delay - results 118
Figure 7.11 – Case 4 - Remote VoIP Network – Data Delay - results 118
Figure 7.12 – Case 4 - Backbone VoIP Network – Voice Delay - results 119
Figure 7.13 – Case 4 - Backbone VoIP Network – Data Delay - results 120
Figure A.1 – Node Model for Voice Packet Generator (VOIP_Gen4).... 127
Figure A.2 – ON vs. OFF Times for Voice Source....127
Figure A.3 – Node Model for BE Traffic Generator Model (BE_Frag_Gen)....128
Figure A.4 – Pareto File Size Generator 129
Figure A.5 – Pareto File Size Generator 130
Figure A.6 – Pareto File Size Generator 130
Figure A.7 – Edge Priority Router (DS_Edge_Router)....131
Figure A.8 – Core Priority Queue (core_priority_router)....132
Figure A.9 – Core DRR Router Model (core_drr_router)....133

List of Tables

Table 1.1 – Absolute Category Rating (ACR) System....8
Table 1.2 – E-model vs. MOS values....9
Table 1.3 – Voice Compression Parameters....12
Table 1.4 – E-Model Impairment (Ie) for Packet Loss vs. Coding Scheme....13
Table 2.1 – Benefits of utilizing different methods for transporting VoATM....19
Table 2.2 – Service Classes for Voice....24
Table 2.3 – AAL5 Payload Format for Voice Service....30
Table 2.4 – Delay Limits....33
Table 2.5 – Silence Suppression....34
Table 3.1 – Type of Service in IP....44
Table 3.2 – Flow Types....55
Table 3.3 – Speech and Audio Coders Quality....58
Table 5.1 – Default values for E-Model....88
Table 6.1 – Coding Parameters....95
Table 6.2 – Case 1, Link Rate = 256 kbps....99
Table 6.3 – Case 1, Link Rate = 1.5 Mbps....100
Table 6.4 – Case 2, Link Rate = 256 kbps....102
Table 6.5 – Case 2, Link Rate = 1.5 Mbps....103
Table 6.6 – Case 3, Link Rate = 256 kbps....105
Table 6.7 – Case 3, Link Rate = 1.5 Mbps....106
Table 6.8 – Results of E-Model Optimization....107
Table 7.1 – Remote Case 1 Utilization Levels....112
Table 7.2 – Remote Case 2 Utilization Levels....114
Table 7.3 – Backbone Case 2 Utilization Levels....115
Table A.1 – DRR Queue Performance....134

LIST OF ABBREVIATIONS

A

AIN	Advanced Intelligent Networks
ACR	Absolute Category Rating
ADPCM	Adaptive Differential Pulse Code Modulation
AAL	ATM Adaptation Layer
ABR	Available Bit Rate
AAL – SAP	AAL Service Access Point
AAL – SDU	AAL Service Data Unit
ANP	Advanced Networking Protocol
AMPL	A Mathematical Programming Language
ATM ARP	ATM Address Resolution Protocol
ATM LE	ATM LAN Emulation
ATM VCC	ATM Virtual Circuit Connection

B

B – TE	Broadband Terminal Equipment
BS	Base Station
B – ISDN	Broadband Integrated Service Digital Network
BGP	Border Gateway Protocol
BO	Bandwidth over provisioning

C

CES	Circuit Emulation Services
CLR	Cell Loss Ratio
CTD	Cell Transfer Delay
CDV	Cell Delay Variation
CAM	Cell Arrival Monitoring
CCS	Common Channel Signaling
CAS	Channel Associated Signaling
CIF	Cells in Frames
CELP	Code Exited Linear Prediction
CBQ	Class Based Queuing
CS – ACELP	Conjugate Structure Code Excited Linear prediction
CBR	Constant Bit Rate
CELP	Code Excited Linear Prediction
CBR	Constant Bit Rate
CDMA	Code Division Multiple Access
CPS – SDU's	Common part Sub layer – Service Data Unit
CSCW	Computer Supported Collaborative Work

D

DSI	Digital Speech Interpolator
DSCP	Differentiated Service Code Point Marking

E/ F

ETSI	European Telecommunications Standards Institute
FEC	Forward Error Correction
FRAD	Frame Relay Access Device

G/ I

GEO	Geosynchronous Earth Orbit
ISO 9000	Quality Management & Quality Assurance Standards – Guidelines for Selection
ITU	International Telecommunications Union
IIS	Internet Integrated Services
IFMP	Ipsilon Flow Management Protocol
IMT	Integrated Management Tool
ISP	Internet Service Provider

L

LSF	Low sampling frequency
LEO	Low Earth Orbiting
LANE	LAN Emulation
LLC/SNAP	Logical Layer Control / Standard Network Access Protocol
LPC	Linear Predictive Coding

M

MGCP	Media Gateway Control Protocol
MOS	Mean Opinion Score
MSC	Mobile Switching Centre
MGC	Media Gateway Controller
MBR	Memory Buffer Register
MPOA	Multi – Protocol over ATM
MPLPC	Multi Pulse Linear Predictive Coder
MP- MLQ	Multi Pulse Maximum Likelihood Quantization

N

NGN's	Next Generation Networks
NBR	Nominal Bit Rate
N – ISDN	Narrowband Integrated Services Digital Network

O

OPNET	Optimized Network Engineering Tool
OPWA	Open Path without advertising
OAM	Operations and Maintenance
OSS	Operation Support system
OSPF	Open Shortest Path First

P

PDU's	Protocol Data Unit

PLC	Packet Loss Concealment
PHB	Per Hop Behaviour

R

RTCP	Real Time transport Control Protocol
RTP	Real Time Protocol
RLR	Receiving Loudness Rating

S

SCTP	Simple Control Transport Protocol
SRTS	Synchronous Residual time stamp
SAR	Segmentation and Reassembly
SLA	Service Level Agreement

T

TDM	Time Division Multiplexing
TL 9000	Telecommunications Sector – specific ISO 9000 standard
TALI	Transport Adapter Layer Interface
TIA	Telecommunications Industry Association
TELR	Talker Echo Loss ratio
TDMA	Time Division Multiple Access

V/ W

VTOA	Voice Telephony over ATM
VAD	Voice Activity Detection
VPN	Virtual Private Network
WRED	Weighted Random early Detection
WDM	Wavelength Division Multiplexing
WAN	Wide Area Network
WEPL	Weighted Echo Path Loss

Chapter 1: NGN's and QoS

1.1 Features of Next Generation Networks (NGN's)

The demand on the telecommunication networks of today and in the future are reflected clearly in the societal changes around the world. Customers of all services have historically wanted more for less, more flexibility, and sometimes just more. This can certainly be seen in the rapid growth of mobile network usage. The mentality of anytime, anywhere is becoming prevalent around the world – the primary differences between regions being largely a matter of degree. Also, customers are seeing innovation across all areas of consumer goods and services with "just in time" services tailored to their specific needs and timeframes. These too will be key demands placed upon telecommunication networks as they evolve.

Key characteristics of NGN's are:

- Geographic transparency: boundaries are disappearing and economic benefits independent of service "density" must be realized.
- Transport efficiencies: transport costs (price/bit) are continuously declining; NGN's must share these efficiencies – for both bearer and signalling traffic.
- Internet technology economics: leverage services and service delivery through the Internet, as well as the "silicon economics" of Internet hardware (servers, etc.) as memory and processor price/performance improve.
- "Old World" to "New World" interoperability: existing PSTN infrastructure, and its associated investment must be fully utilized.

1.2 Next Generation Networks – The Voice of the Future

The term "Next Generation Networks" is very broad. Almost any development in networking could potentially be placed under this banner. The definition used by ETSI's NGN Starter Group places the emphasis on providing operators with a step-by-step manner to create, deploy and manage innovative services.

"NGN is a concept for defining and deploying networks, which, due to their formal separation into different layers and planes and use of open interfaces, offers service providers and operators a platform which can evolve in a step-by step manner to create, deploy and manage innovative services"

However, NGN is more commonly associated with voice – a vision for the future of packet-based voice networks as part of the evolution from today's TDM circuit switched voice. This makes sound commercial and technical sense. For the majority of operators offering a mix of voice and data services, voice is still contributing over 80% of revenues. Besides, data networks have embraced the requirements of packet-based networks from the outset.

1.3 Africa's Packet Telephony Network

The network of tomorrow will be the channel to resources around the world. Regardless of location, regardless of technology, the uniform availability of communications services will be the fundamental differentiator between today's network and networks of the future. This reality will be based upon standards based transport, signalling, services, and many more aspects. Figure 1-1 below illustrates this from a high level.

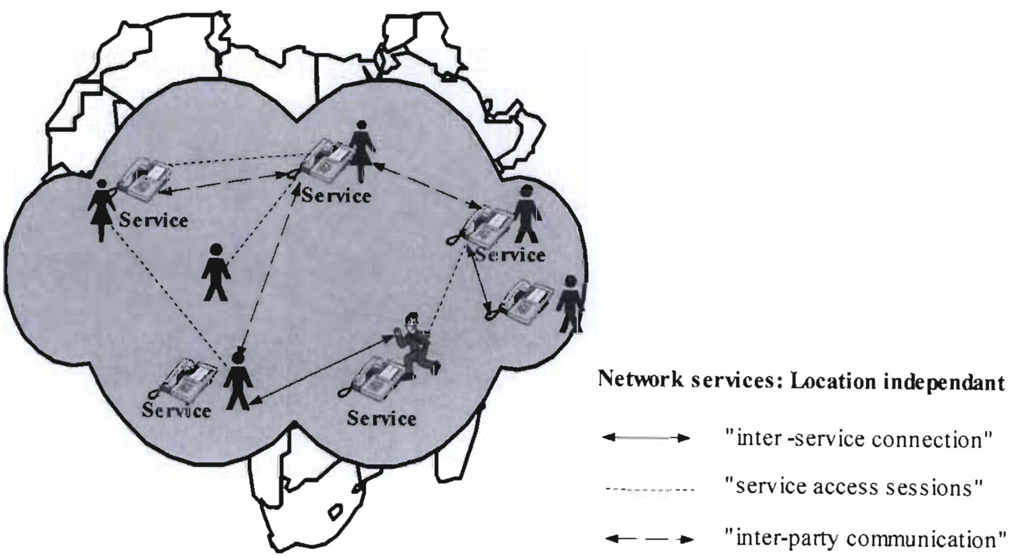

Figure 1-1: Africa' Packet Telephony Network

1.4 NGN Migration - Key considerations in the evolution

As carriers move towards solutions using new technologies and architectures, the success or failure of these solutions is dictated by many factors. In the case of the NGN's and "deconstructed" switch / packet network replacing the circuit switch / TDM network, it is important that these key benefits be delivered:

- Investment protection
- Operational and capital costs savings
- Carrier grade reliability
- Improved service creation capabilities
- Scalability

1.4.1 Investment Protection

The carriers of today have a lot of money invested in their existing networks. It would be fiscally irresponsible for their management to even consider installing a completely new network, and discarding their network. It is critical that any new network technology, whether it is a new network element (NE) or an entire "sub-network", interoperate and leverage existing capabilities. With this in mind, incorporating NGN components based upon standard, open protocols is the first step to protecting a carrier's investment. Compatibility with SS7 and intermachine trunk (IMT) requirements are fundamental tenants to supporting a smooth migration to a NGN. This compatibility ensures basic call setup and teardown, as well as access to existing Advanced Intelligent Network (AIN) services such as local number portability (LNP), free call, etc. Additionally, from an operational perspective, interoperability with existing Operational Support Systems (OSS) is required before the NGN can actually be placed into service.

1.4.2 Operational and Capital Savings

The primary driver behind NGN's is economics. Whether the horizon is short term or long term the fundamental aim is lower costs, higher revenues, or both. In as much as most carriers currently own and maintain both voice and data networks, it is reasonable to project possible savings on the order of 50% when the two networks are combined. From an acquisition cost perspective, the NGN equivalent of a circuit switch – soft switch/MGC and MGs together can be less than one third the cost. Furthermore, given the distributed nature of NGN's, and the incremental growth characteristic, capital budget management and growth planning are both simpler. Rather than large purchases (i.e. major switch upgrades), incremental upgrades to media gateways or additional media gateways should occur. Since NGN solutions are based upon open standards and are closely linked to Internet technologies, significant cost savings will occur over the life of the network. Open standards create choice and encourage competition - a strong determinate of pricing trends. With the leverage of Internet technologies, whether it be software ("web" technology, etc.) or hardware (server and mass storage technologies, etc.) there will be dramatic cost and innovation benefits realized – similar to those found in the data networking market with routers, switches, and PCs.

Carrier Grade Reliability - much of the success of today's telecommunications carriers revolves around the fact that in most industrialized countries the telephones always work. Carrier grade standards for availability are typified by "five nines" or 99.999% uptime. To achieve this high level of reliability, equipment manufacturers and their carrier customers have developed products, architectures, and processes whose mission is focused on maximizing network uptime. From a product perspective, reliability is typically increased by redundancy – redundant processors, links interfaces, hard disks, power supplies, etc. These components are thoroughly tested by manufacturers who have implemented the most stringent quality standards, such as ISO 9000 and TL 9000. Architecturally speaking, redundancy is often the approach of choice. System reliability is addressed by implementing "mated pairs", i.e. redundant systems often operating in synchronization but geographically separated with redundant, diversely routed links providing the interconnection. This continues to be the standard industry practice and is part of network proposals.

1.4.3 Scalability

Historically, telecommunications networks have scaled rather poorly and often at significant cost to the carrier. Switches were either "over provisioned" to support growth or they were upgraded to include additional line and trunk cards and additional call capacity. If capacity was required remotely from the serving switch, "remotes" or digital loop carriers were implemented. These solutions were usually expensive from both a capital and operating perspective. In contrast, NGN architectures support incremental growth in ports (lines or trunks), in call capacity, and in extension to new remote locations. With its distributed nature soft switch or media gateway controller, media gateways and signalling gateways all interconnected via a packet transport – an NGN solution offers both incremental growth and the ability to offer advances in technology without the "wholesale" changes that are typical of traditional circuit switches.

1.5 Migration architectures

1.5.1 Voice Trunking: Tandem Replacement

The first step supported by both the technology available and industry consensus is the migration of the voice trunking network from TDM transport to packet transport (ATM or IP). This is demonstrated in the following two illustrations, Figure 1-2 and Figure 1-3:

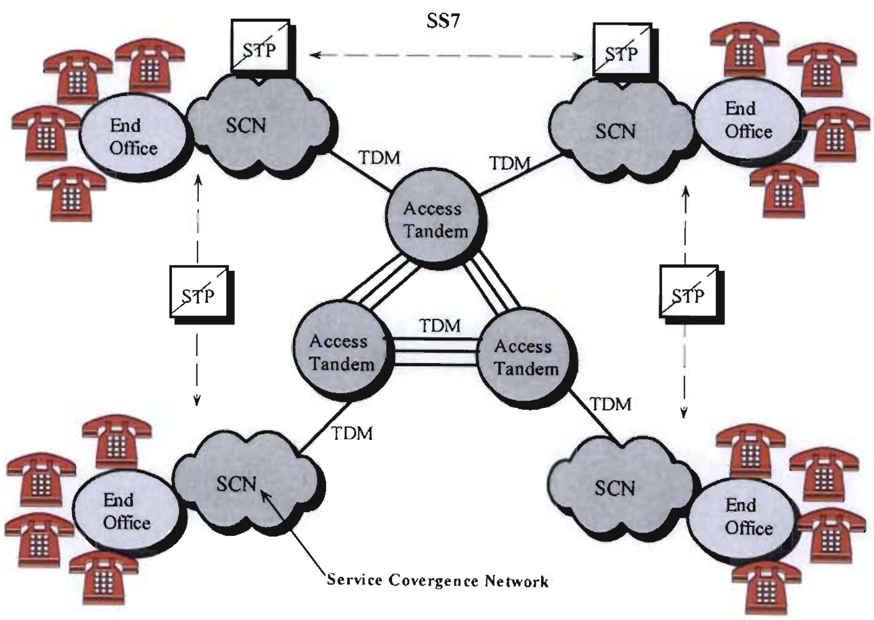

Figure 1-2: Traditional TDM Architecture

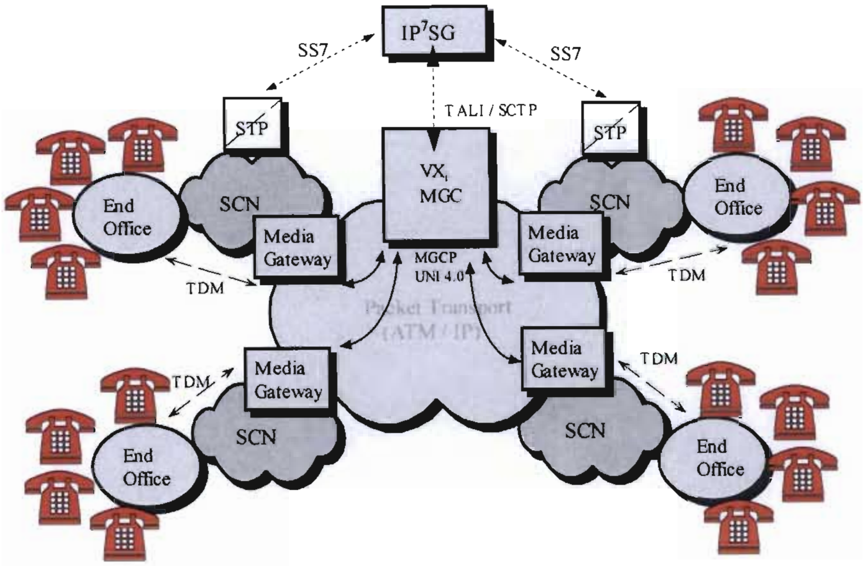

Figure 1-3: Voice Trunking with Packet Transport

Effectively, TDM trunks from End Offices, which typically connect directly to a Tandem or Transit switch network, now use a distributed transport and switching network, comprised of a packet transport and a soft switch / media gateway controller for call control. Examining the architecture, the three key components and their roles are:

- Media Gateways: TDM to Packet translation for the bearer channel, and ingress/egress to the packet network. These MGs are of the trunking gateway variety, meaning they terminate TDM trunks.
- Signaling Gateway: Conversion of SS7 signaling from TDM to packet, and management of SS7/ISUP country variants. Packetized SS7 information is forwarded to the MGC via standard interfaces. Today, this includes TALI, and in the future SCTP.
- Media Gateway Controller: Call control, trunk management, screening, number translations and correct trunk routing. The MGC controls the MGs through standards such as Q.2931 for ATM MGs and MGCP for IP MGs.

1.6 An Introduction to QoS

ITU gives us a generic definition of QoS: "The collective effect of service performance which determine the degree of satisfaction of a user of the service[1]" ETSI recommendation ETR300 003 further sliced and refined the ITU definition into sub definitions, that correspond to the requirements and viewpoints of the different parties taking part in the communication:

- QoS requirements of the user/customer
- QoS offered by the service provider
- QoS achieved by the service provider
- QoS perceived by the user/customer
- QoS requirements of the Internet Service Provider

If we move deeper into QoS at a more technical level, the quality issues of Voice over packets (VoP) fall under three concepts:

1. The characteristics of the Quality of Voice over packets. The characteristics of QoS are quantities that express the impairments caused by the terminals and by the network. Examples of these quantities are processing delay, transfer delay, delay variance and packet loss.
2. QoS management defines the control mechanisms in a packet network and the mechanisms the applications use to adapt to the changing conditions of the network.
3. QoS agreements are a step away from the best-effort service. Current efforts in R&D try to extend present protocols and switching disciplines to support desired requirements on the quality of service.

1.7 Voice Quality

Voice quality in the context of a VoIP scenario is difficult to quantify because the definitions of acceptable voice quality have changed in the last several years. This is complicated by the method in which voice quality is measured. Voice quality is most accurately measured by subjective opinion. The traditional measurement for voice quality measurement in telecommunications is the Mean Opinion Score (MOS). The MOS test is also called the Absolute Category Rating (ACR) test. The ACR is described in detail in ITU Recommendation P.800. Using the MOS method, listeners are asked to rate speech and classify it into categories. These categories are shown in Table 1-1 [2]:

Quality	Score
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

Table1-1: Absolute Category Rating (ACR) System

The MOS level 4.0 which is considered to be "good" quality of speech has traditionally been considered "Toll Quality". This is the quality that could be expected from Telkom for a connection in the South African public switched telephone network (PSTN). Many other tests are also used to measure quality for voice connections. These include the Percent Good or Better (%GOB), Percent Poor or Worse (%POW), Degradation Category Rating (DCR), and the E-model [2] [3]. The E-model tries to address in a qualitative manner several of the quality issues that will affect voice over packet systems. One of the driving forces behind the E-model is that the actual quality of the speech is not always as crucial as the perceived quality. Note that cellular phone users will tolerate degraded quality of voice that would not be tolerated for traditional wire line telephony [4]. The European Telecommunications Standards Institute (ETSI) developed the E model to address the needs of network planners [4]. The E-model is based on the premise that "Psychological factors on the psychological scale are additive" [5]. The E-Model defines the "R" value as the measure of voice quality. The International Telecommunication Union (ITU) and the Telecommunications Industries Association (TIA) has approved the E-Model for use [3] [6].

User Satisfaction	E-model-R	MOS
Very Satisfied	90	4
Satisfied	80	4
Some Users Dissatisfied	70	3
Many Users Dissatisfied	60	3
Nearly All Users Dissatisfied	50	2
Not Recommended	0	1

Table 1-2: E-Model vs. MOS values [3]

Comparing the MOS scale and the E-model we have a reference as to what is considered acceptable. Table 1.2 shows a comparison of the two scales [3]. More information about the E-Model is included in Chapter 5. One of the key components of the E-Model (and voice quality) is the end-to end delay.

1.7.1 End-to-end Delay

Delay is one of the most crucial limitations that affect voice over packet systems. This is because when people have conversations with each other they interact with each other. This interactivity dictates the delay by determining the amount of time that a sender will wait before assuming that his/her last response was not heard and repeat the response. If the delay is too high, the receiver will have just received the first response and will be responding at the same time. This creates confusion and unclear conversation and thus poor interactivity [4]. Interestingly enough, what is considered unacceptable varies significantly. For instance a one way delay up to 250 ms is acceptable, but 400 ms is the limit after which the conversation can be considered to be half duplex. Between 150 and 200 ms, conversation begins to be affected according to the Telecommunication Industry Association (TIA) [3]. This is due to the fact that when a person finishes speaking, there is approximately a 200 ms break before the other person starts speaking. This is called turn taking [3]. When an extra 150 ms is added, the turn talking rules begin to fail and the conversation rhythm must change. This can cause one person to talk before the other person is finished or both will try to speak simultaneously. The delay associated with analogue systems was typically minimal because the primary contributor was propagation time. The propagation time is approximately 25 ms. For most analogue systems, this amounts to a total delay of less than 50 ms. For digital systems however, there are several sources of delay. These are listed below:

- Encoding/Packetization Delay
- Switching and Queuing Delay
- Serialization Delay
- Propagation Delay
- Decoding Delay and De - jitter Buffer

Figure 1-4 (which is an implementation of the E-Model) shows the impact that delay has on the voice quality according to the E-model. The algorithms to complete these calculations are in ITU Recommendation G.108 [8].

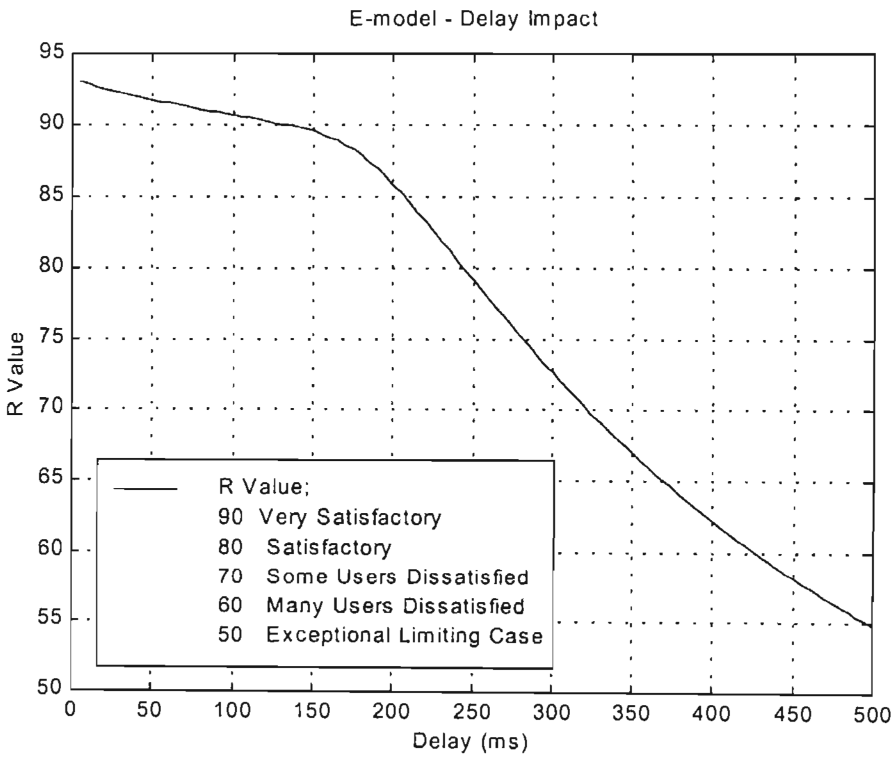

Figure 1-4: E-model, Impact for Delay

As Figure 1-4 shows, with defaults being used in the E-model, the "knee" in the curve is between 150 ms and 200 ms. This seems reasonable given the limits mentioned in the above discussion about turn taking [3].

1.7.2 Jitter

Jitter is a distortion that occurs when data is transmitted over a network, and packets do not arrive at its destination in consecutive order or on a timely basis, i.e. variations in delay. In packet switched networks, jitter is an aberration of the inter-packet arrival times compared to the inter-packet times of the original transmission. This distortion is particularly damaging to multimedia traffic. For example, the playback of audio or video data may have a jittery or shaky quality. [9]

1.7.3 Echo

On systems that have 2 to 4 wire converters (Hybrids), echo can become an issue. The echo occurs when a signal that goes through a hybrid gets partially reflected toward the sender due to imperfect impedance matching at the hybrid. The delay in the signal being reflected back toward the sender (if it is more than approximately 10 ms) can distort the message being received by the receiver. Another type of echo is acoustic echo. This occurs on "hands free" phones or "speaker" phones, where the received signal has the opportunity to be fed back into the microphone. Methods of dealing with echo include echo cancelling devices and echo suppressors. TIA specifies that for PSTN and Toll quality speech, echo cancelling devices should be installed where the delay exceeds 10 ms. The devices should be able to cancel at least 55 dB of the echo. The improvement that the echo cancellers have on the quality is also shown by the E-model [3].

1.7.4 Voice Compression

Voice compression techniques affect the quality of the speech. Most of these schemes affect several speech quality parameters. This is due to their specific encoding mechanisms. Many of the schemes use a type of linear prediction. This enables them to send only the necessary bits to reconstruct the original signal. Because of high correlation between adjacent packets, only the difference needs to be sent. This would end up propagating errors. Adaptive Quantizes use fixed steps and knowledge of the statistics of the signal to adapt the quantizer to the statistics of the signal. ADPCM is probably the best known use of this technology. Linear predictions schemes use either the previous sample or they "look ahead" and use the knowledge of those samples to encode a set of samples that can be decoded using a similar set of rules. To prevent error propagation, correlation coefficients are used that are based on the variance or root mean square of the sliding energy of the input signal. Long term and short term predictors are used. Code excited linear predictive (CELP) coders operate differently. They use the excitation in the samples to match a code out of a set of C codes of length L in a codebook. The code is used that offers the least error between the signal and the decoded signal. The index typically requires .2-2 bits/sample [10]. Some of the popular speeches coding schemes are noted in Table 1-3 with important parameters [4] [8].

Coding Standard	Bit-Rate (kbps)	Frame length (ms)	Codec delay(ms)*	Complexity(MIPS)	MOS	E-Model
G.711 PCM	64	.125	.125	.1	4.2	0
G.726 ADPCM	16/24/32/40	.125	.125	12	4.0	50/25/7/2
G.729 CS-ACELP	8	10	15	20	4.0	10
G.729 A CS-ACELP	8	10	15	10.5	~4.0	11
G.723.1 MP-MLQACELP	6.3/5.3	30	37.5	16	3.9/3.7	15/19

*This assumes a single frame with look ahead.

Table 1-3 - Voice Compression Parameters

1.7.5 Packet Loss

Packet loss is an unavoidable fact of the internet. Reliable protocols like TCP cannot be used due to the fact that retransmission can take multiple round trip times and voice transmission cannot withstand the delay. Packet loss may or may not be related to delay. If packet loss is caused by bit errors, these losses would be unrelated to delay. However, if a maximum delay for a voice stream is exceeded, typically those packets are dropped. In this case, there exists a tradeoff between delay and packet loss. If the maximum delay allowed by a voice stream is increased, packet loss will decrease. To correctly interpret this relationship, a good understanding of the delay distribution over the end-to-end link is required. To combat the detrimental effects of lost packets, several methods have been

1.8.1 Use of RTCP in Measuring QoS

RTCP sender reports are used for three main purposes: to allow synchronization of multiple RTP streams, to allow the receiver to know the expected data and packet rates and to measure the distance in time to the sender. The most important issue is the synchronization of multiple sources. The receiver reports are used to measure the QoS of the connection: fraction lost, cumulative packets lost, the extended highest sequence number received and the inter-arrival delay variance. The cumulative number of packets lost and the sequence number are used to compute the packets lost since the last receiver report. This can be used for determining the long term congestion in a LAN. If the state of congestion is higher than the value set by the terminal manufacturer (programmer) then the terminal should reduce the media rate. High interarrival jitter (the interarrival jitter field) and long intervals in sender reports can also be used as indicators of high state of congestion.

1.8.2 Procedures for maintaining QoS

The methods that can be used by voice terminals and gateways to respond to congestion can be grouped in two: those that respond to short term problems and those that respond to longer term problems. The methods do not seek to maintain QoS, but instead to provide an orderly degradation of service. Short term responses are responses to problems like lost or delayed packets. A typical long-term response would be that to the growing congestion on the LAN. The media degradation order is: video, data, audio, control. There are three typical short-term responses: reducing the frame rate for a short period of time, reducing packet rate by mixing audio and video in same packet, and packet rate reduction by video fragmentation at the H.261 macro block level. More sophisticated responses would be increasing the amount of redundancy information in packets or increasing the amount of information used in Forward Error Correction [12, 13, 14]. Long-term responses are: reduction of media bit rate, turning of media of least importance and returning a busy signal to the receiver as indication of LAN congestion. The busy signal sending can be combined with turning of media. In a multi-router configuration, reacting to delay variance can be difficult. It may be impossible to distinguish the source of delay variance when there is a lot of router incurred reordering and varying of the packet delay.

1.8.3 QoS Agreements

QoS agreements are a step away from the current best-effort model of the Internet. When QoS agreements are made, typically the application requests some QoS from the network and the network gives guarantees on QoS. In this case we call this the provision of QoS. The ATM Forum traffic classes are an example of this. The IETF has defined a similar contract-based model, the Internet Integrated Services Architecture for the Internet. In the IETF model the guarantees are reserved using RSVP. Another way of giving applications QoS is the policy-based approach. The Simplified ATM management proposed in [15] and [16], lies somewhere between the ATM Forum approach and the Internet approach. In this model each user is allocated a share of the link, a nominal bit rate (NBR). The NBR is a service provider-customer contract of a guaranteed bandwidth. The actual bit rate of connections is measured based on the exponential moving average. The measured bit-rate (MBR) is compared to the NBR, and the cells transmitted are given some priority from 1 to 7 based on the ratio of MBR/NBR in the access switch .

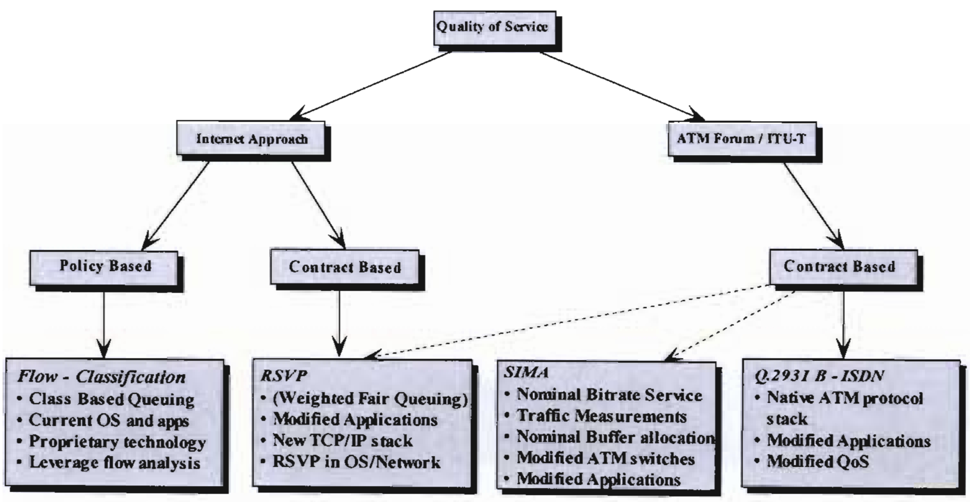

Figure 1-5: QoS Agreements

1.9 The Internet Integrated Services Architecture

The Internet Integrated Services Architecture is the IETF's approach for providing Internet Quality of Service. The goal was that real-time services could co-exist with the traditional non-real-time best-effort service in IP networks. The IIS is often mistaken to be the same as RSVP. The big picture includes many other elements, and is more a broad reference model, than just the RSVP. The core service model of IIS centres around the question of time-of-delivery of packets. The QoS commitments made by the network are related to per-packet delay, bounds on minimum and maximum delays being the sufficient parameters. Applications are grouped in two: real-time applications and elastic applications. Elastic applications always wait for data to arrive, where as real-time applications are time sensitive to data delivery. Real-time applications can be further grouped in two:

1) Applications that need perfectly reliable upper bounds for delay.
2) Applications that can tolerate and adapt to variations in delay and do not need perfectly reliable upper bounds for delay.

The service model for the intolerant applications is called *guaranteed service*. The service model for the tolerant applications is called predictive service [17, 18]. The predictive service gives the applications a fairly, but not perfectly reliable bound for delay, which can be calculated with properly conservative predictions of the behaviour of other flows. The service also tries to minimize the post maximum delay. It does not try to minimize the delay of every packet, but rather it tries to pull in the tail of the delay distribution. Applications can also adapt to changes in the state of congestion of the network by changing their bit rate and thus their traffic characterization. For example video conferencing application can easily change the coding scheme and reduce the frame rate. The service model for elastic applications is called best-effort service. Also the terms ASAP (as soon as possible) and datagram service are used. Elastic applications are sensitive to delay - excess delay often shows in poor application performance. However the performance of the applications is more dependent on the average delay than on the delay distribution. The same scheme that is used for predictive service can provide controlled link sharing. The objective is not to bound delay, but to limit overload shares of the link - thus giving the name for the new service *fair* share or controlled load. The technology behind this service and the other new services is WFQ - Weighted Fair Queuing. WFQ-scheduling in the way it is used in controlled load service, is available in commercial routers today, and is used to segregate traffic into classes based on things like protocol type or application.

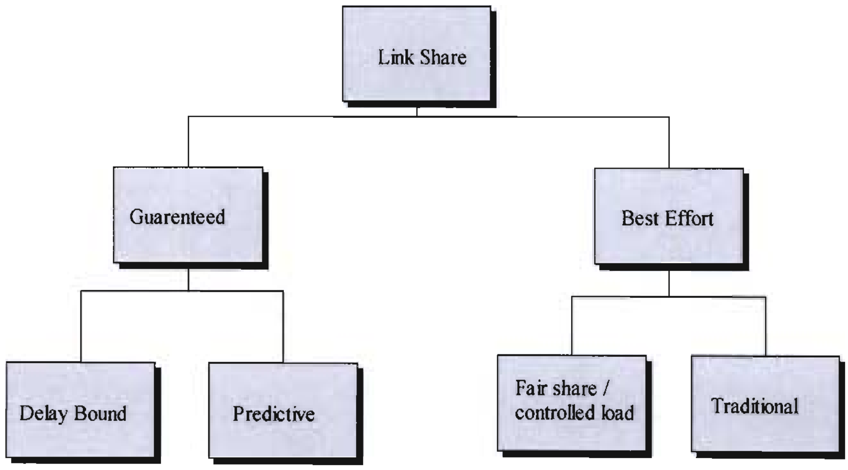

Figure 1-6: IETF traffic service class hierarchy.

Chapter 2: Voice transmission over ATM Networks

2.1 Introduction to VoATM

ATM from the start was designed to be a multimedia, multiservice technology. Although ATM has been accepted by service providers for its ability to deliver high speed data services, until recently its potential for deploying voice services was overlooked. With the competitiveness of today's market, network operators and service providers have been continuously striving to reduce operating costs and lift network efficiency and have turned to the ATM network to achieve these goals. With hundreds of millions of ATM equipment infrastructure in developed countries, service providers have recognized that significant economies of scale can be achieved if the data traffic and voice traffic are integrated onto a single network. In order to achieve this, service providers have started to use the circuit emulation services (CES) of ATM switches to carry full or fractional E1/T1 circuits between end points. These CES mechanisms treat voice as a constant stream of traffic encoded as a constant bit rate (CBR) stream. In actuality, voice is a combination of bursts of speech and silence and this increases the complexity of VoP. The ATM forum and the International Telecommunications Union came up with several advanced mechanisms to improve the efficiency of transporting voice traffic, including:

- AAL trunking using AAL-1 for narrowband services.
- IP over ATM (AAL-5).
- Loop emulation services using AAL-2.

Standards	Voice Compression	Silence Removal	Channel Suppression	Switched Concentration
CES	No	No	No	No
BD-CES	No	No	Yes	No
ATM trunking using AAL-1	No	No	Yes	Yes
VoIP over ATM	Yes	Yes	No	No
AAL-2	Yes	Yes	Yes	Yes

Table 2-1: Benefits of utilizing different methods for transporting VoATM.

2.2 Voice Traffic in ATM Networks

Voice over ATM refers to the transport of voice and voice-band data over ATM. In this context, voice refers to human speech, fax, and modem data. The ATM Forum started to work on voice transport over ATM in April 1995. The Voice and Telephony over ATM (VTOA) working group was formed to provide voice and telephony services over ATM. Voice services are the traditional simple two-way conversations and the telephony services include the increasingly popular additional uses for voice networks such as messaging, conferencing, advanced call routing etc. Voice can be transported over the ATM network as constant bit rate (CBR) or variable bit rate (VBR) traffic. In this section, we briefly describe the basic characteristics of CBR and VBR voice traffic.

2.2.1 CBR Voice Traffic

There is a user demand for carrying certain types of CBR or circuit traffic over ATM networks. For the CBR traffic, the required bandwidth is reserved/guaranteed on a per-connection basis within the network and the end terminals are synchronized. CBR services provide good voice quality and standards-based solution. However, with CBR services, there is no dynamic interleaving of voice and data, resulting in low/bandwidth efficiency. CBR voice traffic can be transported using the following:

- Pulse code modulation (PCM) at 64 kb/s (G.711)
- Adaptive differential PCM (ADPCM) at 64/40/32/24/16 kb/s (G.722, G.725, and G.726)
- Embedded ADPCM (EADPCM) at 40/32/24/16 kb/s
- Low-delay code excited linear prediction (LD-CELP) at 16/12.8/9.6 kb/s (G.728)
- 8.0 kb/s Conjugate-Structure Algebraic CELP (CS-ACELP) (G.729)
- 1-8 kb/s voice for IS-54 and IS-95 wireless voice communications
- Fax modulation
- Modem modulation
- Transport of voice at rates higher than 64 kb/s (in the future)

2.2.2 VBR Voice Traffic

When the network performs voice compression along with silence suppression scheme such as digital speech interpolation (DSI), the traffic becomes statistical and therefore, CBR service is not appropriate. Furthermore, with CBR, a lot of bandwidth is wasted since the entire channels should be provided regardless of whether or not all of the voice slots are occupied. On the other hand, VBR services can support bursty compressed voice efficiently using a proper ATM adaptation layer (AAL) and high bandwidth efficiency can be obtained by using a dynamic bandwidth allocation scheme. The bandwidth unused by voice can be used for the transport of data using available bit rate (ABR) or unspecified bit rate (UBR) services. VBR voice traffic can be created using popular ADPCM or other compression techniques (described above) with silence removal. When voice activity drops, voice samples are not generated or are generated with a small number of bits per sample. It is also referred to as gap mode CBR and voice activity detection (VAD).

2.3 Applications of Voice over ATM

This section briefly describes the key application areas which require VoATM.

a) ATM to the Desktop: Voice is required in multimedia applications where a single desktop workstation is used as the interface for applications involving multi-media including voice. Voice is also required in regular telephony applications where a single desktop workstation is used as the interface for both voice only connections and data-only applications.

b) Distributed Private Branch Exchange (PBX): High-bandwidth interconnection among distributed PBX modules is required for both voice and signalling traffic. ATM is an adequate choice for PBX vendors since it provides high-bandwidth connectivity, multimedia capability, and a standard interface to the public network.

c) Broadband Computer Telephony Integration: Broadband computer-telephony integration is associated with the trends toward ATM to the desktop and the distributed

PBX. Instead of a central computer and PBX, the computing functionality is distributed to the Broadband Terminal Equipment (B-TE), and the PBX functionality is also distributed to interconnected PBX modules. Broadband computer-telephony integration enables B-TE to control the switching of calls so as to flexibly manage multimedia, multiconnection calls.

d) Telephone Company Access Network: The current access architecture from a user to a telephone company involves multiple access lines for voice, video, leased line, and packet-switched data services. ATM is a technology that enables all these traffic types to be transmitted over a single access line, resulting in big cost saving.

e) Cable Company Telephony Service: The cable company provision of interactive services such as video on demand (VoD), home banking, home shopping, and telephony requires the replacement of the current unidirectional distribution network with a bi-directional one. This can be implemented by the installation of optical fibre. ATM is well suited to this application since it is standardized on fibre and can provide the interactivity and multimedia capabilities required by service mix.

***f) Cellular Company Access Network*:** The current cellular company access network takes calls from a mobile unit over a wireless interface to a base station (BS) and from there over a wired interface, typically T1/E1, to the mobile switching centre (MSC). The MSC is linked to the public switched telephone network (PSTN) via another wired interface, typically T3/E3. The cellular company can therefore use ATM to transport voice between the BS and the MSC as part of its internal network. ATM can also be used from the MSC to the public telephone network, provided the telephone company has an ATM interface in the Central Office (CO). On digital wireless interfaces the voice traffic is already packetized, using, for instance, IS-54 (TDMA) and IS-95 (CDMA), making it particularly suited for the ATM transport on the wired portion of the cellular company network.

g) ATM via Satellite: Geosynchronous Earth Orbit (GEO) satellite has been used to interconnect Broadband ATM islands, linking both local and wide are networks. Protocol parameters are adjusted to take into account the increased delay and error rate resulting from the use of satellite channel. The advantages of this type of interconnection include cost-effective wide area networking and the extension of communication link to wide geographic areas. The major requirement is to transport voice traffic multiplexed with video and data. Low and Medium Earth Orbit (LEO and MEO) satellites may also use ATM for inter-satellite communications as well as on the up and down links to the ground. In this case, the end-to-end delay is actually reduced compared to terrestrial transmission because the speed of light is faster in a vacuum than in optical fibre. The main motivation of ATM in LEO/MEO applications is the provision of a high performance switched network with an emphasis on multimedia and less of a requirement for voice-only traffic.

h) Long Distance Terrestrial Transport: Proprietary cell-based technologies have been used for long haul transport of frame relay traffic ever since the initial introduction of frame relay. ATM provides the advantages of a standardized interface to perform the same function and the capability to transport voice as well as data. It is therefore an adequate choice for the public network backbone.

2.4. ATM Adaptation Types for Voice over ATM

2.4.1 Service Classes for VoATM

Table 2-2 summarizes ITU service classes and also indicates the correspondence between service class and type of traffic requirement for VoATM.

Service/Classification	A	B	C	Y
Bit Rate	CBR	VBR	VBR	ABR
Timing Requirement	Yes	Yes	No	No
Possible Voice Service	• Single voice channel • N*64 kb/s • DSn/En • Q.931 N-ISDN D-Channel Signalling	• Voice channels with silence detection/removal	• Q.2931 BISN signalling	• Q.2931 BISDN signalling

Table 2-2: Service Classes for Voice

The requirements of any service can be specified quantitably in terms of the QoS (Quality of Service) factors. The basic QoS parameters which correspond to a network performance objective for VoATM may include cell loss ratio (CLR), maximum and mean cell transfer delay (CTD), cell delay variation (CDV). Voice is extremely sensitive to delay. Therefore, CTD is one of the most important QoS parameters which determine the voice quality. The mean CTD gives us a measure of the end-to-end delay incurred in an ATM network. CTD is a function of the propagation delay, the queuing delay and the switching delay. Usually, medium delays would require echo cancellers, and long delays would be unsuitable for most purposes. The delay usually depends on the delay at the intermediate nodes in addition to the propagation delay of the network.

The peak-to-peak cell delay variation (CDV) is another important QoS parameter. The CDV is a function of the number of multiplexed connections, the type of the multiplexed connections through the switches, and the switching variability. CDV varies with the mixture of the traffic. However, CDV is not a major concern for voice. ATM networks typically have a low cell-delay variation, and this can be taken care of by a play out buffer at the receiver. It may not always be possible to delay cells to compensate for the maximum network delay, in which case it is preferable to drop cells delayed by more than an acceptable value rather than attempting to handle the CDV. Cell Loss Ratio (CLR) is defined as the number of cells lost divided by the total number of transmitted cells. To guarantee high quality voice, it is desirable to keep the cell loss ratio to a minimum. The overall voice quality is a function of the cell-loss and the transfer delays. Voice can tolerate a low cell-loss-ratio. At 64 kb/s, a cell loss amounts to a loss of about 6 ms of voice content. The cell loss varies with the amount of buffer available at the switches. However, adding more buffers and thereby reducing the cell loss ratio does not necessarily improve the overall voice quality. The end-to-end delay requirements and the end-to-end delay variation requirements also need to be satisfied. At the receiving end, not all cells that are received are useful. The useful cells are only the ones that arrive within a specified time. The cells arriving later will be discarded.

2.4.2 ATM Adaptation Layers (AALs)

The AAL provides for the mapping of higher layer protocol data units (PDUs) into the information field of cells and the reassembly of these PDUs. The AAL is divided into AAL type 0, 1, 2, 3/4, 5, which were originally designed to suit the requirements of the service classes described above. However, there is nothing rigid about the assignment of AAL types to service classes for voice services except AAL 3/4 which is used for non-real time data traffic. Any service class can be transported on any adaptation type as long as the performance is acceptable to the voice user.

2.4.2.1 ATM Adaptation Layer Type 0 (AAL0)

AAL0 is a proposed voice specific AAL, which uses the full 48 bytes of the cell payload, giving a 2% improvement over AAL1. Synchronous Residual Time Stamp (SRTS) which is used in AAL1 for source clock recovery is not supported, but adaptive clock recovery may be used for end-to-end synchronization. AAL0 does not support cell sequence number monitoring, but may detect cell losses by monitoring the arrival times between cells. AAL0 may utilize a CAM (Cell Arrival Monitoring) mechanism to detect lost cells. The receiver monitors the time between arriving cells. At 64kb/s, the inter-cell arrival time is expected to be 6 ms +/- CDV (Cell Delay Variation). If a cell has not arrived within this time window then it is assumed lost and it is discarded. There is no means to detect cell misinsertions. This mechanism relies on the CDV being bounded within half a cell arrival time; otherwise it is not possible to discriminate between a late arriving cell and an early arriving cell. The delay associated with this mechanism is the

CDV value. AAL0 can be used for a single voice channel with a QoS specifying low cell delay variation.

2.4.2.2 ATM Adaptation Layer Type 1 (AAL1)

AAL1 defined by ITU-T I.363 uses 1 byte of the ATM cell payload to carry a cell sequence number and SRTS timing information. This leaves 47 bytes for user information. Although AAL0 mechanism can provide the 2% gain in bandwidth efficiency, AAL1 can support CBR voice better than AAL0 since it provides more stringent cell loss detection capabilities. Furthermore, AAL1 offers the following advantages over AAL0:

- Reduced complexity at the PSTN/N-ISDN Inter-Working Units (IWU).
- I.580 specifies the use of AAL1 at a B-ISDN/N-ISDN Interworking Unit
- Where switching systems are not synchronized to an ATM network, the SRTS timing recovery mechanism of AAL1 is useful to ensure that jitter values meet G.823 requirements.
- Fax and modem terminals and echo cancellers require reliable cell loss detection and compensation. AAL1 can therefore be used at the ATM interface.
- Common AAL for voice and circuit emulation services. This reduces the cost of supporting multiple service types at the terminal.
- Greater tolerance to CDV introduced by the network. AAL1 sequence number monitoring mechanisms have greater tolerance than AAL0 to CDV introduced by the network. The sequence number enables the receiver to discriminate between a late arriving cell and an early arriving cell. In summary, a voice specific AAL1 with a fast sequence number interpretation mechanism can be useful to support of CBR voice and voice-band data transport over an ATM network.

2.4.2.3. ATM Adaptation Layer Type 2

A new ATM adaptation layer (AAL type 2) protocol for low bit rate voice and data applications was defined by ITU-T. The motivation of a new AAL2 was to support low bit rate and delay sensitive applications. The new recommendation was named as I.363.2.

It has also been accepted by ATM Forum. This recommendation may form the basis of ATM Forum activities related to VTOA trunking (mobile or land line).

The objective of new AAL is to accommodate low bit rate traffic (below 64 kb/s) such as compressed voice and delay-sensitive applications into ATM networks such as cellular system. Therefore, the basic requirements should include short cell-assemble time and high bandwidth efficiency. ITU-T SG13 and ATM Forum initiated new AAL standardization activities and there have been many proposals for new AAL [19]. The provisional name for new AAL was composite user AAL (AAL-CU), but it was renamed as AAL Type 2 (AAL2). The key features of AAL2 protocol include short and variable length payload and user packet multiplexing capability. Figure 2-1 and 2-2 illustrate the basic procedure and the structure of AAL2 protocol, respectively. The AAL type 2 is subdivided into the Common Part Sublayer (CPS) and the Service Specific Convergence Sublayer (SSCS) as shown in the Figure 2-2.

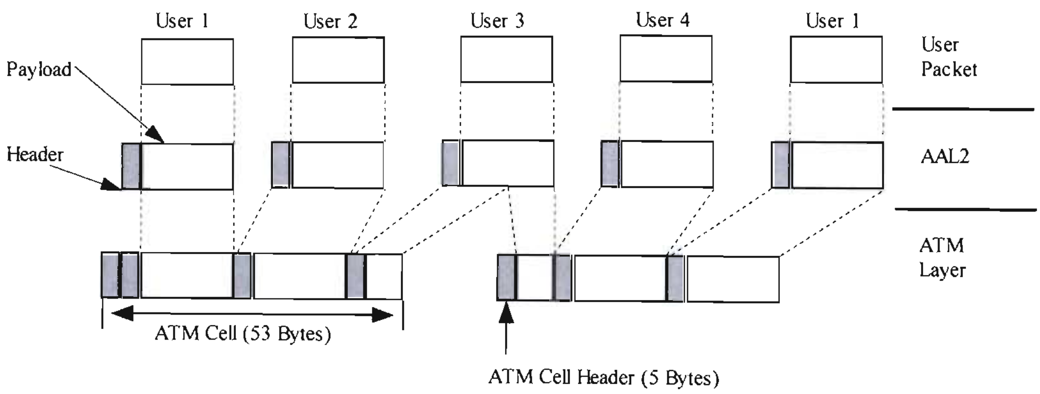

Figure2-1: Procedure of New AAL2 Protocol

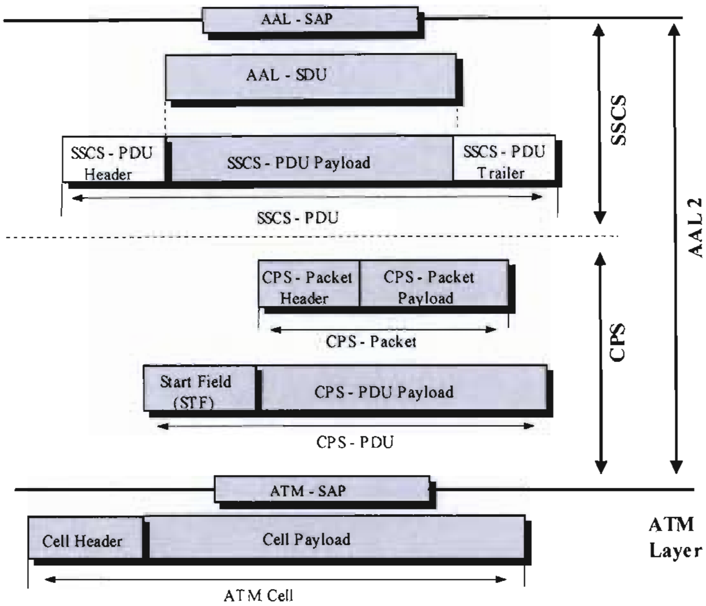

Figure2-2: Structure of new AAL2 Protocol

Different SSCS protocols may be defined to support specific AAL type 2 user services, or groups of services. The SSCS may also be empty, merely providing for the mapping of the equivalent AAL primitives to those of the CPS and vice-versa. The AAL type 2 provides the capabilities to transfer AAL-SDUs from one AAL-SAP to another AAL - SAP through the ATM network as shown in Figure 2-3.

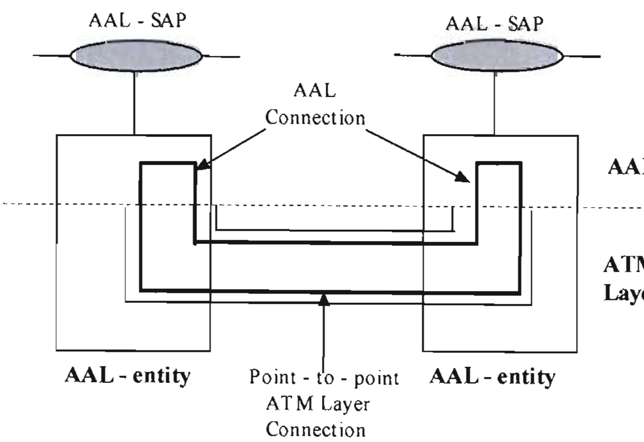

Figure2-3: AAL Type 2 Connection

The AAL type 2 users have the capability to select a given AAL-SAP associated with the QoS (for example, delay and loss sensitivity) required to transport that AAL-SDU as shown in Figure 2-4. The AAL type 2 makes use of the service provided by the underlying ATM layer. Multiple AAL connections may be associated with a single ATM layer connection, allowing multiplexing at the AAL; multiplexing in the AAL type 2 occurs in the CPS. The AAL user selects the QoS provided by the AAL through the choice of the AAL-SAP used for data transfer.

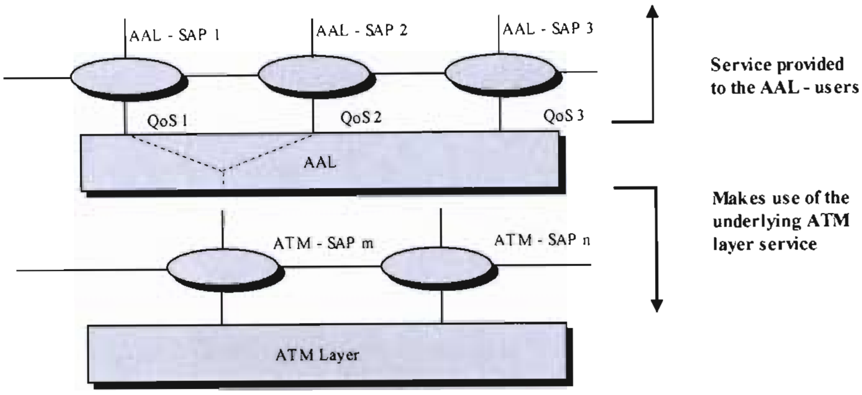

Figure2-4: Relation Between AAL – SAP and ATM- SAP

Note how QoS at the AAL-SAP is mapped into the ATM-SAP. The AAL type 2 CPS provides the capabilities to transfer CPS-SDUs from one CPS user to another CPS user through an ATM network. The AAL type 2 CPS possesses the following characteristics:

- The AAL type 2 connection is defined on an end-to-end basis as concatenated AAL type 2 links.
- The AAL type 2 channel is a bi-directional virtual channel. The same value of channel identification shall be used for both directions.
- The AAL type 2 channels are established over a Permanent Virtual Circuit (PVC) or Switched Virtual Circuit (SVC) of an ATM connection. Here, the AAL type 2 link is an ATM VCC with the ability to carry AAL type 2 channels. Most of the CPS functions have been specified by ITU-T SG13. However, SSCS part is yet to be fully defined. Other elements that need to be defined are as follows.
- AAL2 Negotiation Procedures (ANP)
- Additional details of SSCS
- Operation and Maintenance (OAM) functions.

2.4.2.4 ATM Adaptation Layer Type 5 (AAL5)

Cost is a very important criterion for the success of VoATM to the Desktop. A low cost option for voice services is to transport voice over the current AAL5 interface because many ATM workstations already have AAL5 adapters for data applications. AAL5 could be used efficiently if the QoS requirement for the peak-to-peak cell delay variation (CDV) is low. ATM Switches provided by IBM, Nortel and Stratacom use proprietary AALs which are similar to the AAL5 for dynamic data transfer with one exception: they furnish the timing synchronization that is required for transmitting very delay-sensitive voice calls. These switches can also convert traditional PBX signals into the Q.2931 signalling used on ATM networks and this helps prevent ATM voice circuits from being triggered when they are not needed. The payload format specified by ATM Forum is shown in Table 2-3. The voice data format is negotiated when the link is established but it is mandatory to support G.711, using either μ or A law coding, at 8kHz sampling. After further study, voice data formats other than G.711 may be supported by negotiation during the call setup. Active voice bytes per cell are controllable on a per VC basis via signalling or management to be in the range 1 to 40. The AAL-5 trailer length field will reflect the chosen value.

Fields	Number of Bytes	Description
Voice data	1 to 40 bytes	Voice samples
Local time stamp	4 bytes	32 bit unsigned starting sequence number
Wall clock time stamp	4 bytes	32 bit unsigned wall clock time stamp
AAL5 trailer	8 bytes	AAL5 trailer
Total	48 bytes	

Table 2-3: AAL5 Payload Format for Voice Service

AAL5 can be a possible means of transporting voice over an ATM network. However, it has a high overhead for short data units as shown in Table 2-3 and is therefore likely to be used to transport Q.2931 signalling, voice traffic that is already packetized, such as wireless data for IS-54 and IS-95, or new voice traffic in situations where the user has already invested in AAL5 interfaces for data communications.

2.5 Error Performance of AAL Protocols

Detection of lost cells is mandatory requirement for voice services. This is particularly necessary for modems and echo cancellers which require phase continuity. If bit count integrity is not maintained, modems will lose data or retrain, causing loss of data for 5-20 ms. Misinserted cells are sufficiently rare and can be ignored in ATM networks. It is possible to detect cell loss or misinsertion by the sequence number (SN) in AAL1 and AAL2, but not by AAL5. AAL5 provides bit error detection only and is therefore suited to the transport of signalling traffic which requires detection of bit errors. Cell loss may be detected in AAL0 by implementing CAM mechanism described above. A single cell loss affects 48 bytes of user data when AAL0 is used, 46 or 47 bytes when AAL1 is used, and approximately 46 bytes when AAL2 is used. Misinserted cells are simply discarded at the receiver. When a cell loss is detected, the receiver may insert a replacement cell. For example, the previous cell could be repeated or an interpolation made between the cells before and after the missing one. Because of the tight end-to-end delay requirements for voice, there is insufficient time to request a retransmission of missing cells. Forward Error Correction (FEC) is an option in AAL1 and is proposed as an option for AAL2 in the ATM Forum. In AAL1, it can be used to recover up to 4 lost cells in a block of 124 cells or to correct up to 2 error bytes out of 124. Assembling 124 cells and performing the error correction introduces a delay of approximately $47*8*124/64 = 728.5$ ms for a single 64 kb/s voice connection, which is unacceptable according to the delay requirements for voice calls specified in G.114, G.131, and G.126. Therefore, FEC is inappropriate for voice unless multiplexing has already been performed on the individual voice calls. For instance, a DS1 emulation incurs a delay of approximately $728.5/24 = 30$ ms delay as a result of performing FEC [20, 21]

2.6 Network Design Considerations

Though ATM is equipped with transferring voice over the network efficiently, ATM onto the desktop will not be popular enough. The reasons being, there are enough competing technologies, like 100-Mbps Ethernet/Gigabit Ethernet that provide similar services with minimal infrastructure upgrade to be done. But, when WAN is considered, ATM has its own niche over its competing technologies. While designing and engineering voice over an ATM WAN, there are set of design issues which needs to be addressed. Some of them are discussed below.

2.6.1 Technical Challenges:

A packetized approach to transmit voice faces a number of technical challenges, which spring from the real-time or interactive nature of the voice traffic. Some of the challenges that need to be addressed are:

2.6.1.1 Echo

It is a phenomenon, where the transmitted voice signal gets reflected back due to unavoidable impedance mismatch and four-wire/two-wire conversion between the telephone handset and the communication network. It can, depending on the severity, disrupt the normal flow of conversation. Its severity depends on the round-trip time delay. It is found if the round-trip time delay is more that 30 ms, the echo becomes significant, which makes the normal conversation difficult.

2.6.1.2 End-to-End Delay

Voice is most sensitive to delay and mildly sensitive to variations in delay (jitter). It is highly critical that the delay is kept at a bare minimum to hold an interactive communication end-to-end. It has been found that delay can have two effects on communication performance. Delay can interfere with the dynamics of voice communication, in the absence of noticeable echo, whereas in the presence of noticeable echo, increasing delay makes echo effects worse. When the delay reaches above 30 ms, echo canceller circuits are required to control the echo. Once the echo canceller circuits are in place, network delays can be allowed to reach up to 150 ms without further degrading the voice quality. According to the ITU-T Recommendation G.114 the following delay limits for one-way transmission time for connections with adequate echo control are considered allowable.

Delay	Acceptability
0 – 150 ms	Acceptable to most user applications
150 – 400 ms	Acceptable when the impact on quality is aware of
> 400 ms	Unacceptable

Table 2-4: Delay Limits

Delay occurs in ATM networks because of one or more of the following reasons.

a) Packetization Delay (or cell construction delay):

It is the time taken to fill in a complete packet/cell before it is transmitted. Normal PCM (Pulse Code Modulation) encoded voice samples arrive at the rate of 64Kbps, which means it takes around 6 ms to fill the entire 48 byte payload of the ATM cell. The problem can be addressed either with partially filled cells or by multiplexing several voice calls into a single ATM VCC (Virtual Circuit Channel).

b) Buffering Delay

Sometimes, due to the delay in transit some cells might arrive late. If this happens, SAR (Segmentation and Reassembling) function provided by the Adaptation layer might have to under-run with no voice data to process which would result as gaps in the conversation. To prevent this, the receiving SAR function would accumulate a buffer of information before starting the reconstruction. In order to ensure no under-runs occur the buffer size should be kept in such a way that it exceeds the maximum predicted delay. The size of the buffer translates into delay, as each cell must progress through the buffer on arrival at the emulated circuit's line rate. This implies that the Cell Delay Variation (CDV) has to be controlled within the ATM network.

c) Encoding Delay

This is the processing time taken by the compression algorithms to encode the analogue signal to digital form.

2.6.1.3 Silence Suppression

Voice in its inherent nature is variable. It is found that on an average, human voice has a speech activity factor of about 42%. There are pauses between sentences and words with no speech in either direction. Also voice communication is half-duplex. i.e., one person is silent while the other speaks. One can take advantage of these two characteristics to save bandwidth by halting the transmission of cells during these silent periods. This is known as silence suppression.

Algorithm	Bandwidth	MIPS (C5x DSP)	Total Codec Delay (msec)	Application
ADPCM (G.726)	64 kbit/s		10 – 20ms	Digital Networks
CS – ACELP (G.729)	16 kbit/s		10 – 40ms	Most telephony applications

Table 2-5: Silence Suppression

2.6.2 Signalling

This relates to the efficient utilization of resources and the transfer of control and signalling information. There are two parts in a voice call - the actual voice samples and the signalling information, like dialled number, the on-hook/off-hook status of the call, and other routing and control information. This signalling can be encoded and may be sent as Common Channel signalling (CCS), where signalling information from different channels is aggregated into a single signalling channel, and Channel Associated signalling (CAS), where signalling information is embedded within each discrete voice channel.

2.6.3 Synchronization

The transport of voice demands that the data be synchronized between the speaker and the listener. There are two standard mechanisms that are used to achieve synchronization between point- to-point applications. They are Adaptive Clocking and the Synchronous Residual Time Stamping (SRTS). These mechanisms work by adjusting the clock rate at one end of the circuit based on the clock rate of the other end. The above mentioned mechanisms work effectively only in the master-slave environment or point-to-point communication. When multipoint services are in operation, it is not possible for a slave to adjust its clock based on two or more different signals coming from different master sites. For multipoint service, it is easy to adopt an externally synchronized model where each node in the network is synchronized to some external clock source.

2.7 Voice over ATM to the Desktop

The goal of the VoATM to the desktop is to enable voice services to the desktop using ATM technology. To achieve this goal, the following three technical objectives must be met:

- Both data and voice traffic must have a single connection to the desktop (ATM cannot offer a two connection solution to the desktop, particularly when network consolidation and support for multimedia applications are a core ATM goal)
- Desktop voice must provide the capability to call any existing telephone from the ATM desktop (universal telephone connectivity is essential)
- Desktop voice should support voice features and services (e.g., call transfer).

2.7.1 Reference Configuration

Access and inter working requirements for desktop voice give rise to some interesting technical challenges. To address all the technical challenges, various inter working scenarios should be considered because the development of multimedia networks and equipment introduces voice and telephony in a large variety of configurations.

(a) Telephony between workstations or set tops.

This configuration permits to set a voice call between workstations or set tops.

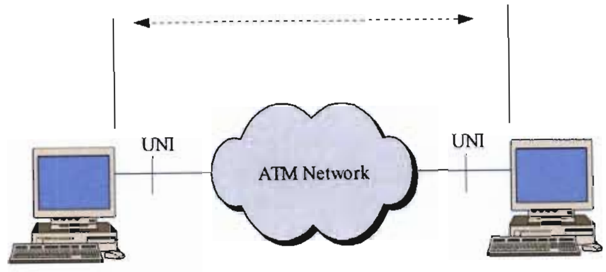

Figure 2-5: Network configuration for Telephony between Workstations or Set Tops

(b) Inter working with N-ISDN

For the support of ATM wide area network access, basic N-ISDN supplementary services should be supported along with SVC signalling as shown in Figure 2-6. This configuration permits to set a voice call between a TE and a legacy phone through the N-ISDN network.

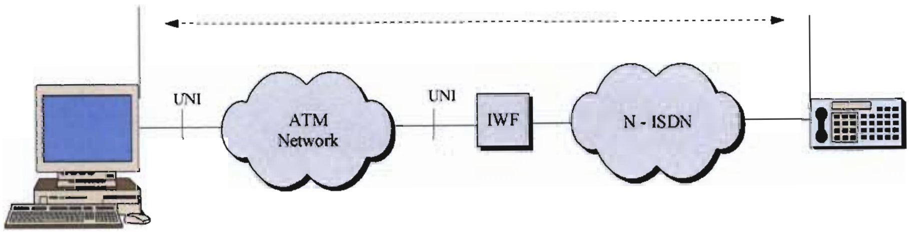

Figure 2-6: Inter working with N- ISDN

(c) Inter working with voice over Frame Relay

It is also needed to inter work with Frame Relay network to support Frame Relay users. This configuration permits to set a voice call between a TE and a phone attached to a Frame Relay Access Device (FRAD). Frame Relay forum is defining an IWF between a PBX and a Frame Relay Network to transport compressed voice and signalling through a Frame Relay network.

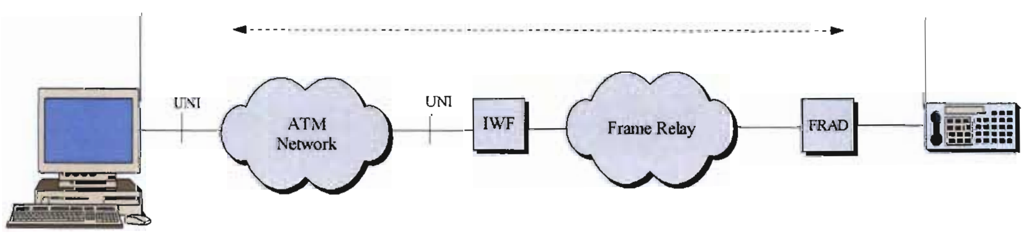

Figure 2-7: Inter working with Voice over Frame Relay

(d) Inter working with voice over wireless phone

This configuration is needed to permit to set a voice call between a TE and a wireless phone. A Telephone Mobile station sends and receives compressed voice to the Base Stations. The overall voice delay, quality and bit rate between an ATM telephony workstation and a wireless phone will be optimum if the ATM workstation also send and receives compressed voice with the same compression algorithm.

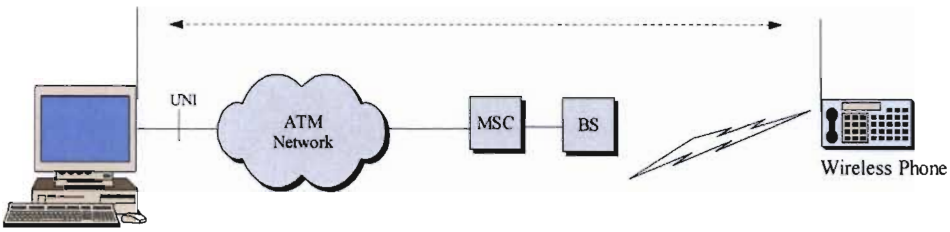

Figure 2-8: Inter working with Voice over Wireless Phone

(e) Inter working with voice over Internet

This configuration is needed to permit to set a voice call between a TE and a workstation attached to an IP network. IETF has defined RTP to transport real time traffic over IP. Telephony applications over Internet will emerge soon with highly compressed voice to reduce the bit rate, and the jitters.

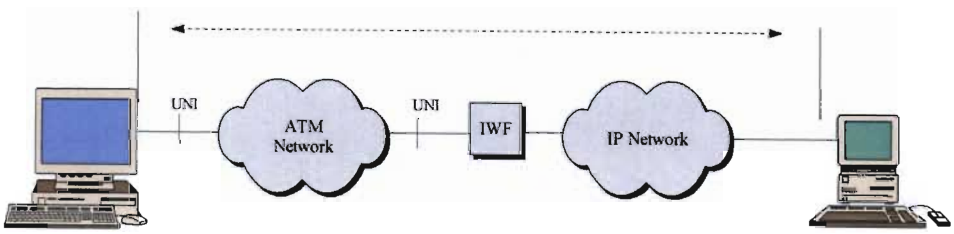

Figure 2-9: Inter working with Voice over IP

(f) Inter working with PBX

Enterprise, namely multimedia business applications, is considered essential to continued growth of ATM. Most large enterprises deploy private PBX voice networks; hence, the need to achieve the inter working with PBX voice networks. For inter working with PBXs, private network signalling such as QSIG and general PBX services need to be supported. TDM/ATM payload conversion is also needed as with public telephone network inter working.

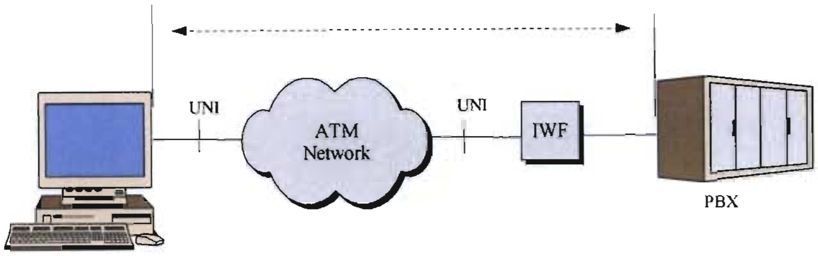

Figure 2-10: Inter working with PBX

(g) Inter working with voice over legacy LANs.

This configuration is needed to provide voice services to legacy LAN users. The real complexity of inter working with legacy LANs comes in three areas:

- Connecting the end-station application software to the ATM network to enable it to take advantage of QoS and to process voice with acceptably low software delays.
- Connecting voice calls between the ATM LAN and circuit-switched public networks and PBXs, preserving clocking relationships and furnishing signalling protocols.
- Implementing an open scheme for call control that matches or improves upon what PBXs deliver (directory services, hunt groups, etc).

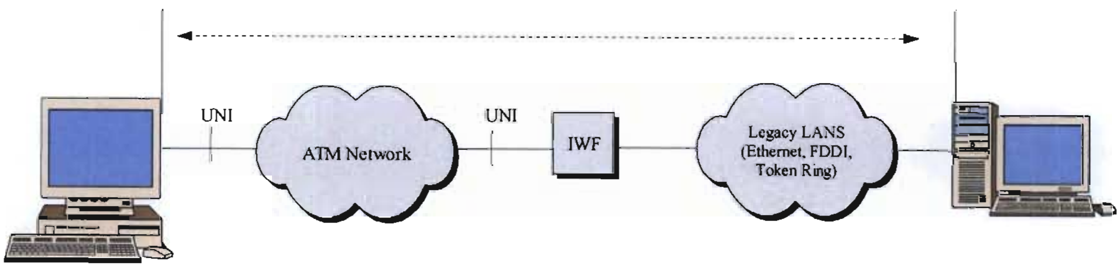

Figure 2-11: Inter working with Voice over Legacy Lans

Desktop ATM is likely to remain a minority taste for some time to come. When voice and data integration offers significant rewards, such as in some call centres, then voice will be a powerful driver of ATM to the desktop. Most likely there won't be much migration in general office environments until there are good solutions for extending quality of service to Ethernet and Token Ring desktops via an ATM backbone. Such solutions may be based on the recently adopted cells-in-frames (CIF) specification, which defines a method of encapsulating ATM cells in Ethernet or Token Ring frames. Another possibility is mapping the IETF's resource reservation protocol (RSVP) requests to ATM QoS, either as an extension to LAN emulation (LANE) or multi-protocol over ATM (MPOA).

Chapter 3: Voice Transmission over IP Networks

3.1 Voice over IP Concepts

Voice over IP (VoIP) is the transmission of voice over networks using the Internet Protocol. IP networks have become increasingly popular in the past few years, the exponential growth of the public Internet leading the way in to the IP-world. In this chapter the basic concepts relating to VoIP will be presented: what is VoIP, what it is used for and why, the applications, the main standards and protocols relating to VoIP: IP, UDP, TCP, RTP, H.323 and IP Switching. At the end of the chapter we will present the basics of packetized voice: coding and voice source modelling.

3.2 Why Voice over IP

Voice is real-time and requires real-time handling from the network. Most data networks currently do not provide real-time service. Still IP-voice has found a market, the main driver being surpassing the costly long-distance telephony. The popularity of virtually free long-distance calls have proven the fact that even poor quality is satisfactory if the price is right. In the future long-distance call charges will sink, not because of VoIP, but because of the increasing competition. The cost advantage of VoIP will diminish, but still market experts are predicting a bright future for it. Because of statistical multiplexing and advanced compression methods VoIP should in principal be more cost effective than circuit switched voice transmission. Video conferencing and CSCW (Computer supported collaborative work) applications are the new drivers of VoIP.

3.3 Applications of Voice over IP

There are three basic network scenarios for VoIP:

- **Computer to Computer**

  This is the basic scenario: both the A and B subscribers are using computers attached to an IP network as terminals.

- **Computer to Phone and Phone to Computer**

  In this scenario one of the subscribers is using a computer for IP-voice and the other uses a phone on a PSTN/ISDN/GSM/TDM network. A gateway on the edge of the IP network translates IP- voice to voice and takes care of the signalling between the two networks.

- **Phone to Phone**

  Both subscribers are using conventional phones in this option, and the IP network is used for the long distance connection. Gateways on both ends take care of translations between networks.

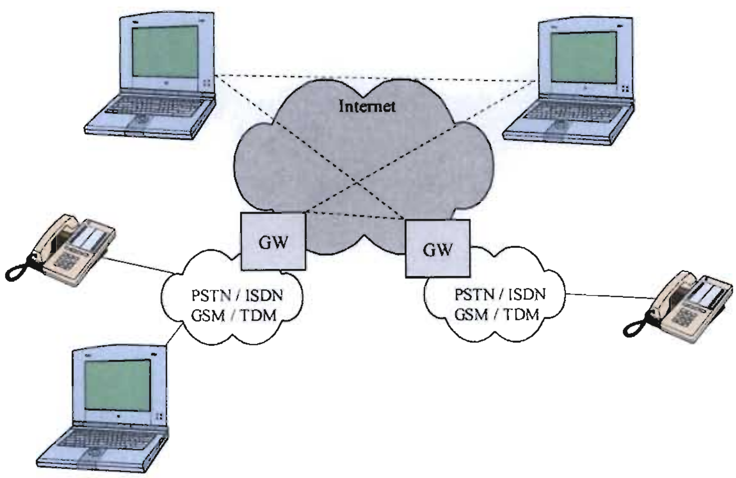

Figure 3-1: The framework for IP voice

3.4 The IP Phone Terminal Equipment

The IP phone terminal equipment can be workstations or personal computers equipped with IP phone software, sound card, speakers, a microphone and a network interface. The first experimental terminal software was designed for UNIX workstations with plenty of processing capability and memory, but the Pentium equipped PCs of today can do the job as well. An ordinary telephone handset gives more comfort and better audio quality than the speaker/microphone combination.

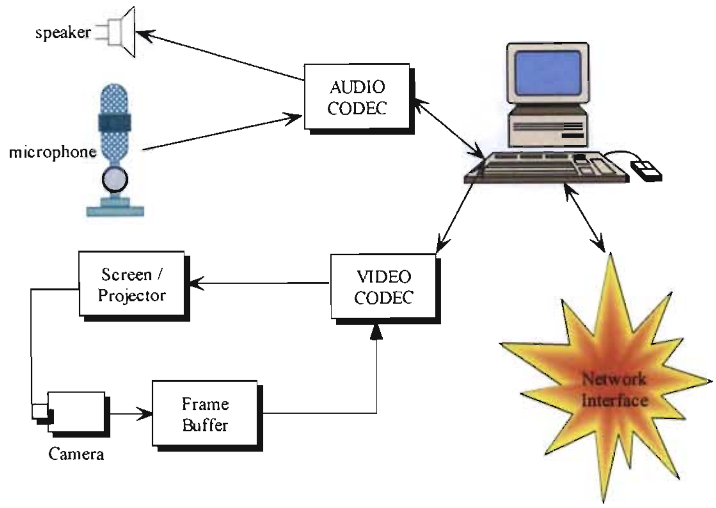

Figure 3-2: Typical IP voice terminal with video conferencing capabilities.

3.5 IP Voice Gateways

An IP voice gateway is an interworking unit capable of translating IP-phone signals to ordinary telephone signals, IP address to telephone number conversion and foremost can signal open a connection from the IP network to a terminal in the telephone network. So far the gateways have been built around pc hardware. The processing intensive coding and decoding of voice is done with special DSP-equipped telephony boards. The control functions are run in main memory using the CPU. There are scaling and reliability problems in architectures like these, but the cost is also relatively low compared to for example a PBX.

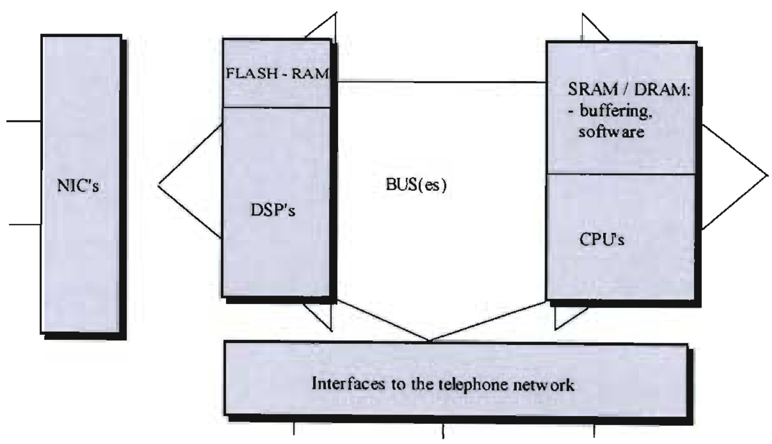

Figure 3-3: IP Voice to PSTN gateway

3.6 IP Voice Related Standards and Protocols

In this section the most important protocols and standards related to VoIP are presented. We will start with Internet Protocol: the handling of types of services in IP networks, fragmentation and reassembly and IP options. This will be followed by an introduction to UDP and TCP protocols, the two transport layer protocols of IP. UDP is mostly used for real-time transmission of voice and TCP in relation to VoIP for streaming control and applications where added buffering delay is acceptable, e.g. audio broadcast. To make use of TCP retransmissions, buffering is needed. Waiting for retransmissions is not appropriate for real-time communications, where interactivity needs to be maintained. RTP is the session layer protocol used for synchronization, multicast session participant information relaying and recipient network quality monitoring purposes.

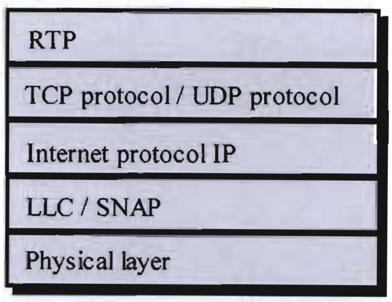

Figure 3-4: The Voice over IP protocol stack

3.6.1 The Internet Protocol - IPv4

[25] defines the latest revised version of Internet Protocol, better known as IPv4. The internet protocol is used for interconnected systems of packet switched computer communication networks. The services internet protocol provides for are: transmitting blocks of data called datagram's from source to destination and fragmentation and reassembly of long datagram's, if necessary. The IP header carries Internet source and destination addresses as well as a number of parameters needed for the routing of packets. Figure 3-5 shows the IPv4 header [25]. As can be seen the header comprises of fixed fields present in every IP packet, and several options. Each row in the header is 32 bits long and the bits are transmitted with most significant bit first. The first field in the header is the version field, used for distinguishing between other "IP compatible" protocols and previous versions of IP. The current IP version number is four. The time to live (TTL) field was ini-

tially an indication of the maximum lifetime of the packet in the network in seconds. IP multicast uses the TTL field to set the scope of the session. The multicast traffic is not sent outside of the site if the TTL is 15 or lower. With TTL of 63 the session will cover a region, and session with TTL of 127 or larger will cover the world.

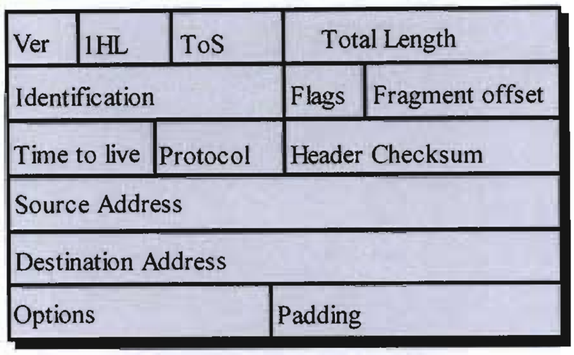

Figure 3-5: IPv4 header

The second octet of the packet header defines the "type of service", ToS. It is divided in two fields, the "precedence" and the "type of service". Precedence is an indication of the priority and the "type of service" is an indication of routing. The routing protocols are supposed to compute a default route when no ToS bit is set, a shortest route (D set), a largest throughput route (T set), a most reliable route (R set) and a cheapest route (C set). The bits should not be combined. In general the ToS is not used in the Internet because of fear of misuse, although the "type of service" routing is defined in some routing protocols (BGP and OSPF) and some routers are capable of priority queuing (e.g. Cisco routers). A network manager may choose to use ToS within a private network [26].

Parameter	Options
Precedence (3bit)	8 values
D, Delay (1bit)	Low (0) and high (1)
T, Throughput (1bit)	Low (0) and high (1)
R, Reliable route (1bit)	Low (0) and high (1)
C, Cheapest route (1bit)	Low (0) and high (1)

Table 3-1: Type of Service in IP

3.6.2 The Next Generation of IP - IPv6

IP next generation (IPng) [27], [28] is a newer version of the Internet Protocol designed as a successor to the IP version 4. The version number assigned for IPng is 6 and it is formally called IPv6. It was not designed to be a giant leap away from IPv4 - many of functions of IPv4 were kept. The primary motivation for the design of IPng was the exponential growth of the Internet leading to running out of IP address space. IPng offers: expanded routing and addressing capabilities, auto-configuration of addresses, improved scalability of multicast routing by adding a "scope" field to multicast addresses, a simplified header format (despite the increased address size, headers have only doubled), improved support for options, quality-of-service capabilities (flow labelling), authentication and privacy capabilities. The IPng protocol consists of two parts, the basic IPng header and IPng extension headers.

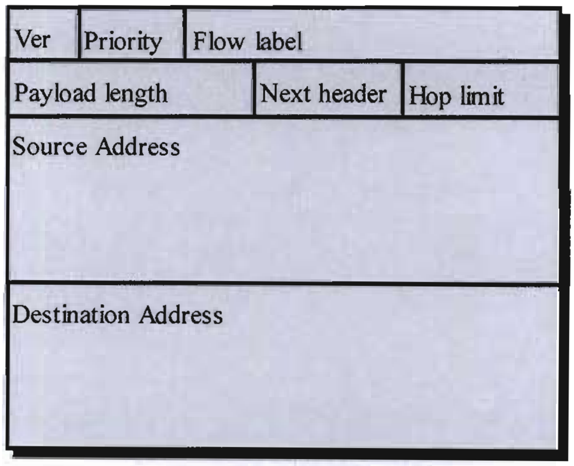

Figure 3-6: IPv6 header

3.6.3 Transport Control Protocol

IP provides best-effort service. There is no guarantee that all datagram's arrive at the destination. Local congestion may cause discarding of queued packets, and transmission errors may cause packet loss. Packets may also be spread across several parallel routes and arrive in disorder. The transport control protocol, TCP was designed to hide these errors. A TCP connection is set between two ports. A TCP port is identified by an IP address and a 16-bit port number, which identifies an application within the host. TCP is used as an alternative to user datagram protocol for VoIP. When a packet is lost (or delayed long enough) the sender re-transmits the packet. For a real-time application the re-transmitted packet is usually worthless, because the play out point of the packet (or samples in a packet) has already been missed. Therefore TCP is only appropriate in regard to VoIP for applications where timely delivery is not required, e.g. signalling of a IP telephony connection, Internet audio broadcasts and audio on demand. The TCP header is shown on fig 3.7

Source Port		Destination Port	
Sequence Number			
Acknowledgement Number			
Offset	Reserved	Control	Window
Checksum		Urgent Pointer	
Options		Padding	
Data			

Figure 3-7: TCP header

3.6.4 User Datagram Protocol

The user datagram protocol is an alternative for TCP. Through UDP the applications get direct access to the datagram service of IP. Applications post packets to UDP ports identified by an IP address and a 16-bit port number. The UDP header (Figure 3-8) is lightweight and therefore appropriate for the transmission of short voice packets. As mentioned before real-time voice mainly uses UDP.

Source Port	Destination Port
Length	Checksum
Data octets	

Figure 3-8: UDP header

3.6.5 IETF Real-time Protocol

The IETF audio-video transport group started work on a real time transport protocol in 1993. The aim of the protocol was at providing services required by interactive multimedia conferences, such as play out synchronization, demultiplexing, media identification and active-party identification. However, not only multimedia conferencing applications can benefit from RTP, but also storage of continuous data, interactive media distribution, distributed simulation, active batch, and control and measurement applications could take advantage of the possibilities RTP brings. [29]

The design goals of RTP were [29]:

1. Content flexible - RTP should not be limited to only voice and video conference;
2. Extensible - RTP should be able to accommodate new services as operational experience accumulates;
3. Independent of lower layer protocols - RTP should work with UDP, TCP, ST-II and ATM;
4. Bridge/RTP gateway compatible - it should be possible to aggregate several media streams into a single stream and possibly retransmit it with different encoding;
5. Bandwidth efficient - header overhead in short voice packets can be as much as 100%. For example with a 65ms packetization interval using 4800 bit/s encoding produces 39 byte packets. IPv4 incurs 20 bytes of headers, UDP an additional 8 bytes and the data link layer at least an additional 8 bytes. With RTP headers around 4 to 8 bytes the total of headers is around 36 or 40 bytes per packet. This could stand in the way of running RTP over low-speed links;
6. International - a and μ law encoding as well as non US-ASCII character sets should be included;
7. Processing efficient - even the longest packetization intervals give packet arrival rates of 40 per second for a single voice channel. Per packet processing overhead may become a concern;
8. Implementable now - the protocol is more or less experimental and the lifetime of the protocol was not anticipated long, so it must be implementable with the current hardware and software.

3.6.6 H.323

ITU-T H.32x recommendations define visual telephone terminals and how they are run over various networks. H.320 applies to N-ISDN while H.321 applies to B-ISDN (ATM). H.322 and H.323 apply to LANs. The difference between the latter two is in that H.323

applies to LANs without QoS guarantees and H.322 to LANs with QoS guarantees. H.323 is applicable to any packet-switched network regardless of the underlying physical layer. The network is expected to provide both an unreliable and a reliable delivery mechanism. In IP networks the former is Transmission Control Protocol, TCP and the latter User Datagram Protocol, UDP. H.323 is independent of the network topology: H.323 terminals can communicate through hubs, routers, bridges, and dial-up connections. The scope of recommendation H.323 is in Figure 3-9.

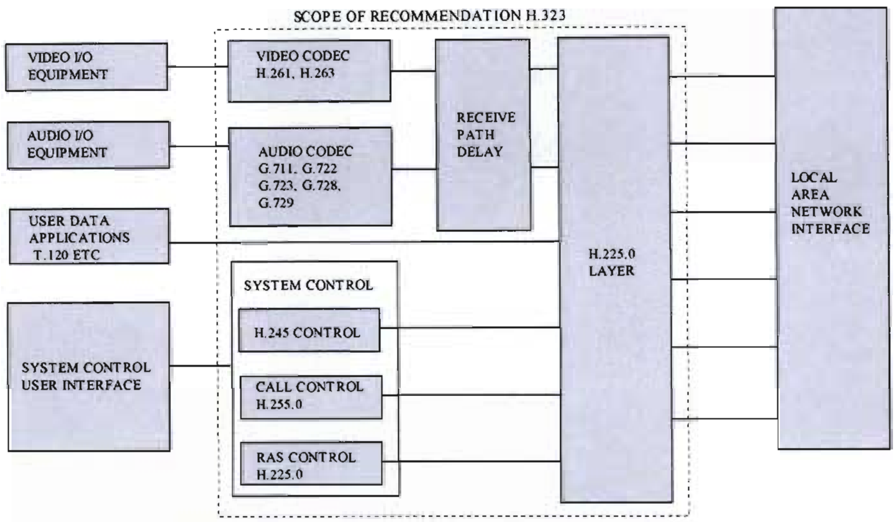

Figure 3-9: H.323 terminal and H.323 scope.

H.323 is an umbrella standard, in the sense that it is a collection of many ITU-T recommendations. H.323 provides system and control descriptions, as well as call model descriptions and call signalling procedures. H.225.0 describes media (audio and video) packetization, media stream synchronization, control message packetization, and control message formats. Recommendation H.245 describes the messages and procedures used for opening and closing logical channels for audio, video, data, camera mode requests, control, and indications. T.120 series of recommendations are used for data applications, such as shared whiteboard. For audio coding G.711 is mandatory, while G.722, G.728, G.723.1 and G.729 are optional. For video H.261 QCIF mode is mandatory, while H.261 CIF and H.263 modes are optional. H.323 entities are:

- H.323 terminal, that provides real-time bi-directional audio, video and data communications;

- gatekeeper that provides security functions and the means to control H.323 traffic on the net through admission control. It also provides address translation services, e.g. it converts telephone numbers to network addresses.
- Multipoint Control Unit (MCU), needed in centralized and hybrid multipoint conferences. MCU is used for distribution of media streams. All terminals send their media streams to the MCU, which then distributes selected or mixed streams back to the terminals. In a decentralized multipoint conference each terminal distributes its media streams to all other terminals in the conference.
- gateway, provides translation of call signalling, control messages, and multiplexing techniques between H.323 and other ITU-T terminal types.

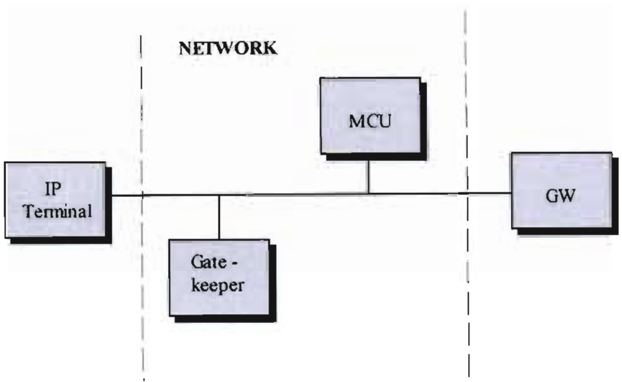

Figure 3-10: H.323 entities.

a) Call models and call setup in H.323

H.323 call signalling can be either directly between endpoints (terminals) or it can be gatekeeper routed, in which case the gatekeeper relays all signalling between endpoints. The routed signalling is needed for terminals that do not contain a multipoint controller (MC) that provides conference control (e.g. establishment of common communication mode and media channels). The call setup takes place in several steps (see Figure 3-11: Simple call setup signalling.). If the implementation of H.323 uses gatekeepers, the calling endpoint must first request permission to place the call from the gatekeeper using H.225.0 ARQ (1). If the call is allowed the gatekeeper responds with ACF (2). A rejected call response is ARJ (2). If no gatekeeper is used, or after permission the endpoint 1 sends a Setup (3) to endpoint 2. The called endpoint acknowledges the call setup with Call Proceeding (4). If the called endpoint is able to accept the incoming call and the implementation uses a gatekeeper, it must request permission to receive a call with the same ARQ (5) /ACF/ARJ (6) procedure as above. If the gatekeeper allows the call to be received, the called endpoint sends an Alerting (7) message to the calling endpoint indicating that the user is notified of the incoming call. If the user answers the call, the called endpoint sends a connect message to the calling endpoint[31].

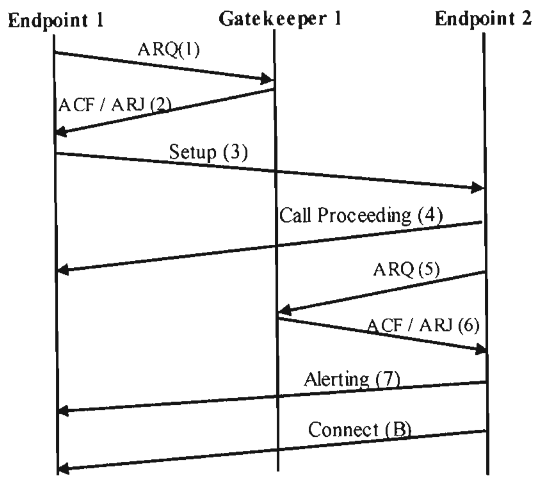

Figure 3-11: Simple call setup signalling.

The communication between H.323 terminals takes place over logical channels after call setup. The logical channels are opened using procedures defined in H.245. H.245 control always takes place on logical channel 0. Separate logical channels are opened for audio, video and data. Multiple channels can exist for each media type through the use of H.225.0. Data communications typically use T.120 series recommendations to define procedures, protocols and applications.

b) H.255.0

H.225.0 covers protocols and message formats. It is designed to operate over various LANs, such as IEEE 802.3 and IEEE.802.5. Acting as a convergence layer, above the transport layer, H.225.0 is protocol independent and can be used over LANs with QoS guarantees as well. The scope of H.225.0 is: *communication between H.323 terminals and gateways in the same LAN using the same transport protocol.* H.225.0 may be used over interconnected LANs or even over the Internet, but the performance is acceptable only when the network load is low. The H.323/H.225.0 protocol stack is depicted in Figure 3-12.

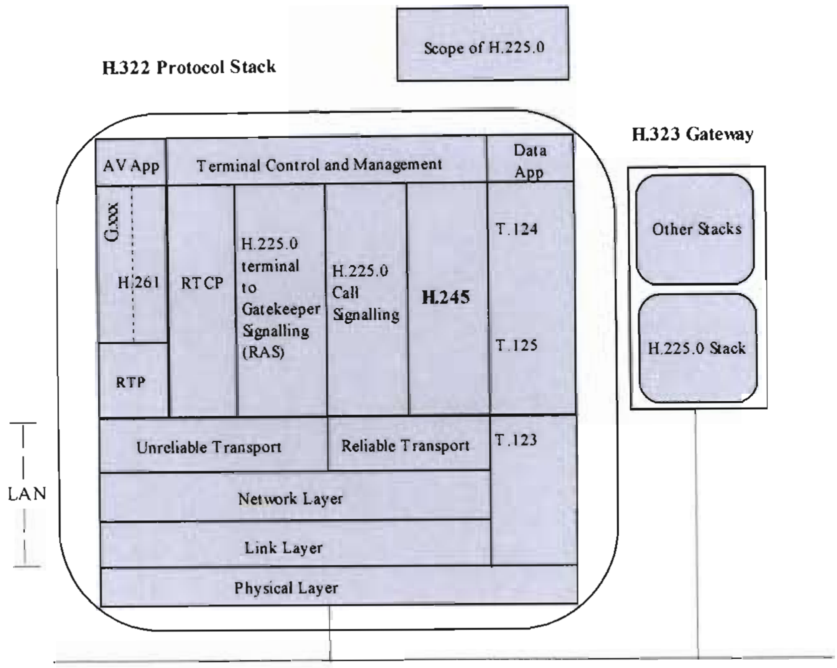

Figure 3-12: The scope of H.225.0.

In H.323 audio and video packets are formatted as defined in H.225.0 using IETF RTP. RTP payload format specifications are used for specifying the carriage of audio and video in RTP. A separate logical channel for RTCP QoS feedback to the source of media is also opened. The feedback information can be used by the media source to adapt encoding or buffering schemes.

3.6.7 IP Switching

The rapid growth of the Internet has resulted in congestion in many areas of the Internet. At the same time access speeds to Internet are increasing and thus new IP routing capacity is needed. On the corporate side, a typical user has a 10BaseT Ethernet LAN connection. LANs are increasingly divided into workgroup sized sub-nets forming a campus network, where routing is used to connect individual sub-nets and the sub-nets to the corporate intranet and to the Internet. This requires some heavy-duty routing at wire speeds. A number of incentives to deliver high-speed routing are underway: the Multigigabit routing, IP/ATM [32], the Cell Switch Router (CSR)[33], Tag Switching [34], Aggregate Route-based IP-switching [35], Multiprotocol Label Switching (MPLS) [36] and Multiprotocol over ATM (MPOA)[37]. Here I will present one of these, Ipsilon's IP switching[38], [39].

a) The Architecture of IP Switching

The components of a router are: line-cards to physically interface the backplane bus to the links, switch fabric to interconnect the various components of the router, forwarding engine and a network processor. The network processor runs the routing protocols and computes the routing tables, and it typically processes the packets needing special handling. The forwarding engine inspects the packet headers and decides which line-card the packet should be forwarded to. In high-speed routers the backplane is replaced by a switching fabric, which offers a much higher aggregate capacity. IP switching differs from the routing approach in that IP switching makes the routing decision per flow not packet per packet. Once a sequence of packets with common properties is identified, it is switched through the switch fabric, and all the subsequent packets of the flow traverse the switch fabric without having to go through the forwarding engine. The switch fabric used with IP switching is an, ATM switch.

The same approach of using ATM architecture to speed IP routing is used in [32], [33], [34], [35],[36] and [37]. One of the functions of flow classification is to select the flows to be switched through the ATM switch and those that should be forwarded through the forwarding engine. The flow classification decisions can be based on source and receiver IP addresses and TCP and UDP port numbers. Different IP flows may require different QoS. The QoS reservations would be signalled from the application to the IP switches using IETF's RSVP. In the ATM switch the classified flows that have been decided to be switched are assigned to a separate VC. ATM switching requires that all traffic must be labelled with a VCI indicating the VC it belongs to. The information of the association between a flow and VCI label is distributed upstream using Ipsilon Flow Management Protocol, IFMP. Another protocol, the General Switch Management Protocol is used for switch control between the ATM switch and the IP switch controller implemented as separate units. This is depicted in Figure 3-13.

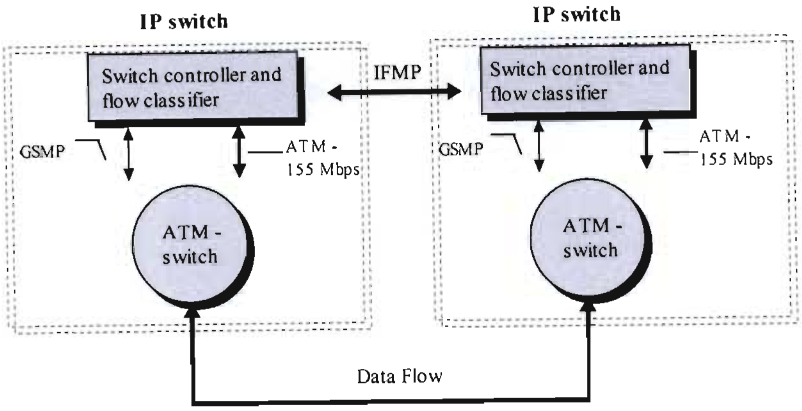

Figure 3-13: The structure of IP-switches and the protocols.

The protocol hierarchy of IP switching is in Figure 3.14. The Ipsilon solution uses AAL-5. LLC/SNAP encapsulation for IP packets is only used on the default channel. On the flows redirected to a specific VC the LLC/SNAP and IP header fields are removed. The ATM ARP and LES - protocols have been replaced with Ipsilon's own IFMP.

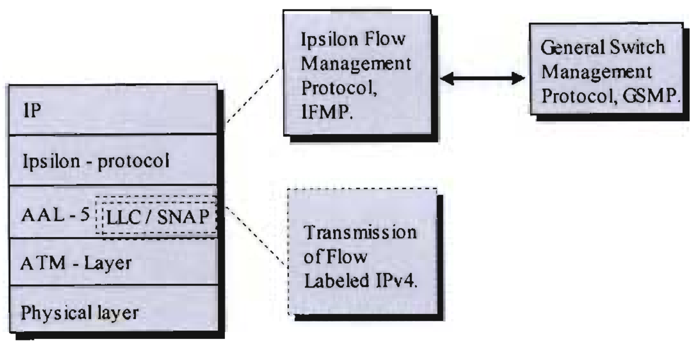

Figure 3-14: The Ipsilon protocol hierarchy.

b) IP Switch Operation

In this section we will briefly go through the operation of IP switching. A good presentation on the operation and performance issues of IP switching can be found in [18]. The senders and receivers considered here can be either other IP-switches or terminals. The sender is located upstream from the switch and the receiver downstream. When a new IP packet X enters the ATM switch fabric of the IP switch it uses the default connection (VP=0, VC=15). From the ATM switch the packet is directed to the connection X to the control processor. Using pre computed routes and routing protocols the packet is forwarded to the appropriate default connection W on the upward link, See Figure 3-15.

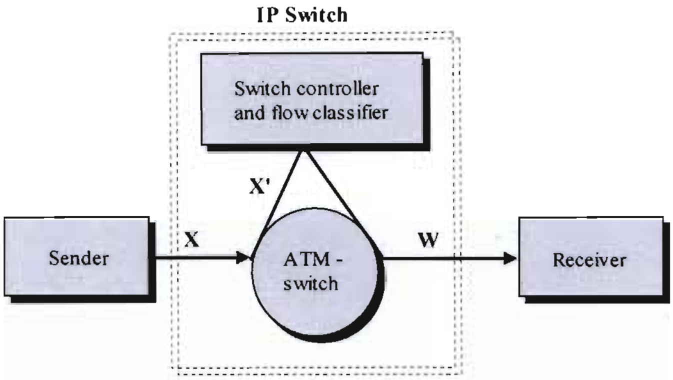

Figure 3-15: The IP switch as a packet forwarding router.

If in the process of packet forwarding the flow classifier identifies a flow (see Figure 3-16), the flow is switched to a separate virtual channel. First the switch controller using GSMP opens a new connection between Y and X, then using IFMP it signals upstream to the closest neighbouring element a request to direct the flow to the VC Y. The redirect request is not acknowledged. The first packet on the new connection is an indication of successful redirect. From this on all subsequent packets flow through Y to X, and the routing information is cached thus speeding the forwarding of packets.

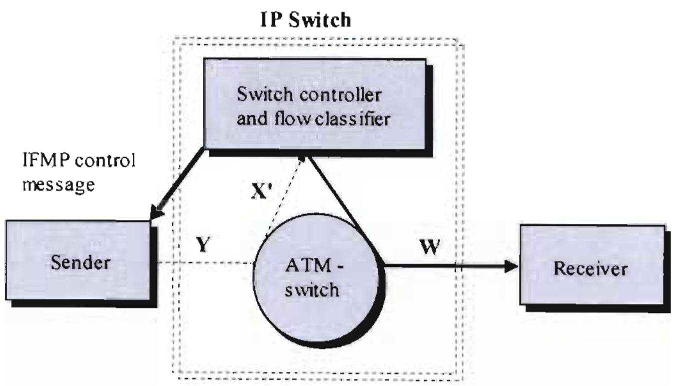

Figure 3-16: Soft-state routing and flow control.

We further consider the situation of Figure 3-16. Now, if the downstream closest neighbouring element (the receiver) sends an IFMP requests to direct the flow on connection W to a new connection (Figure 3-17), and the flow is redirected on that connection, we have a direct connection through the ATM switch.

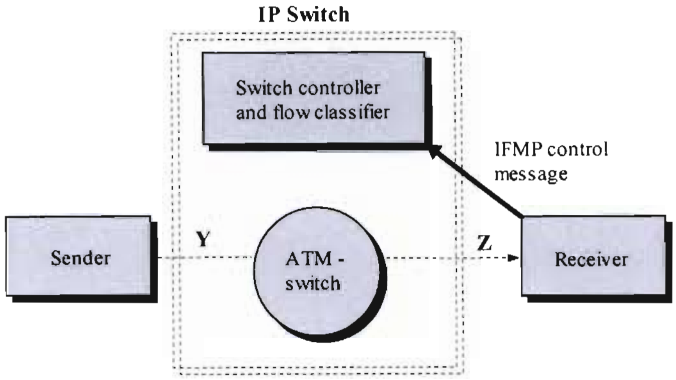

Figure 3-17: A switched flow in IP switching.

An IP switching network is in Figure 3-18. If we now consider a case where the IFMP connection requests have been issued first from the neighbouring element of switch1 to switch1, then from switch1 to switch2 and from switch2 to the edge device we have the situation of Figure 3-17 in both of the switches for some particular flow. The edge device will insert a corresponding VCI in all packets of the flow, and the headers of the packet are formatted according to some flow type. The flow types are in Table 3-2 below.

Flow type	Contents	Adaptation	MTU	Control
Default	TCP/IP	LLC/SNAP + AAL-5 CPCS-PDU	var	Default
Type 0	TCP/IP	AAL-5 CPCS-PDU	1500 bytes	IFMP Flow Type 0- redirect message
Type 1	modified TCP/IP	AAL-5 CPCS-PDU	1484 bytes	IFMP Flow Type 1- redirect message
Type 2	modified TCP/IP	AAL-5 CPCS-PDU	1492 bytes	IFMP Flow Type 2- redirect message

Table 3-2: Flow types.

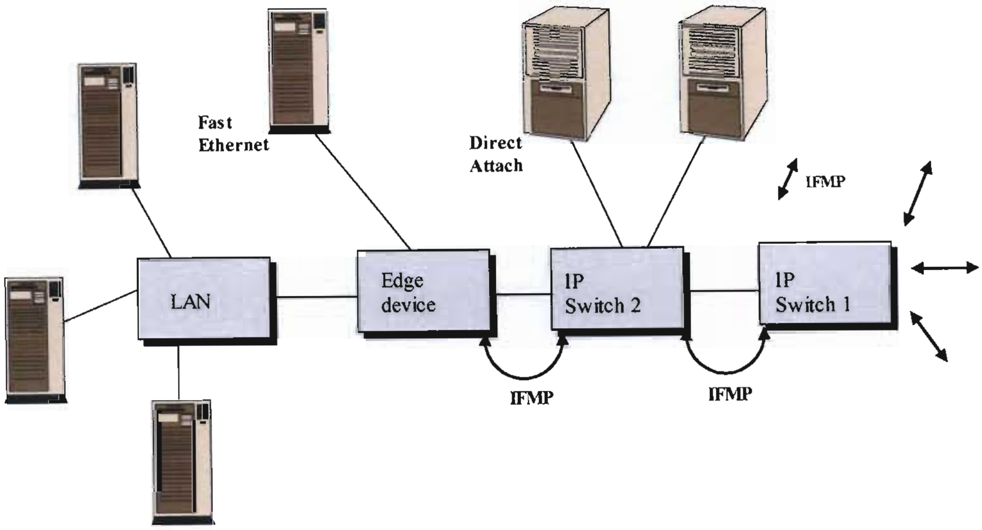

Figure 3-18: IP switching network.

3.7 Speech coding

Speech coding is conversion of a speech signal to a digital representation. The simplest way to do this is by applying sampling theorem directly. This means sampling the wave-form at a rate of twice the highest frequency present in the signal, and then digitizing the resulting samples to some desired degree of accuracy. The telephony nominal bandwidth is 4 kHz, so the speech signals need to be sampled at a rate of 8000 samples per second. The desired signal-to-noise ratio is dependent of the encoding precision, usually either 8 or 16 bits per sample. The total rate is thus 64000 or 128000 bits/s [40]. One of the goals of speech coding is to reduce the bit rate. To do this speech waveform specific properties have to be exploited. Adaptive quantizes vary their characteristics over time. This is to match the dynamic range variations of the speech signal. Time-varying filters exploit the short-term and long-term correlations of the signal. Coding methods can also take advan-tage of the property that in human hearing noise can be masked by the speech signal, if the spectral level of the noise is below the spectral level of the speech. A typical speech coder consists of two modules: an analysis module and a synthesis module. The Analysis module extracts from the speech waveform the time varying excitation waveform and the time varying filter parameters. The Synthesizer module recreates the perceptually best match to the original speech waveform. The most common speech synthesis models are the LPC *vocoder* model, the *multipulse* model and the *stochastic* model. The vocoder or voice coder is the most traditional of these models. Examples of the multipulse and stochastic coders are the MPLPC , multipulse linear predictive coder, and CELP, the code excited linear prediction[40].

3.7.1 Speech and Audio Coders for VoIP and their Quality

The design parameters of all speech coders are bit rate, quality, signal delay and complexity. The bit rate is a measure of the extent to which the speech model has been exploited in the coder. The lower the bit rate the greater the reliance on the speech production model. Quality is a measure of degradation of the speech signal and is measured in subjective tests of speech intelligibility and naturalness[40]. Signal delay is a measure of the duration of the speech signal used to estimate coder parameters in both encoding and decoding, plus the delay of the transmission channel. The longer the allowed delay in the coder, the better it can estimate the synthesis parameters. Complexity is a measure of the computation required to implement the coder algorithms in hardware or software. An ideal coder would have a low bit rate, high perceived quality, low signal delay and low complexity. Real coders make tradeoffs among the attributes.

The quality of speech coders is measured in subjective listening tests. The diagnostic Rhume Test (DRT) measures intelligibility of speech in terms of distinguishing minimally distinct pairs of rhyming words. Intelligibility for coder bit rates from 64 kbit/s to 4.8 kbit/s is close to that of natural speech. At lower bit rates a slight degradation is observed. The MOS test of speech quality uses a five point scale:

- 5, excellent quality, no noticeable impairments.
- 4, good quality, only very slight impairments.
- 3, fair quality noticeable but acceptable impairments.
- 2, poor quality, strong impairments.
- 1, bad quality, highly degraded speech.

Coder standard	Algorithm	Complexity %100MHZ Pentium / MIPS (DSP)	Frame size/ look ahead ms)	Codec delay using DSP (ms) *	Comp.	Rate (kb/s)	MOS (Clear channel, single encoding) **
PCM	PCM	0 MIPS		0	1	128	4.5
G.711	PCM μ/A law	0 MIPS	0.125 /0	0.225	2	64	4.10
G.726 (G.721) G.728	ADPCM LD-CELP LD-CELP	<1% / 1 MIPS 65% / 30 MIPS	0.125/0 0.625/0 0.625/0	0.225 3 3	8/5.3/4 /3.2 8	16/24/32/4 0 16	3.85 3.61 3.61
G.729 (IS-641, GSM EFR)	CS-ACELP	50% /20 MIPS	10 /5	25	16	8	3.92
G.729A	CS-ACELP	25-30% / 11 MIPS	10/5	25	16	8	3.7
G.723.1	ACELP&M Q-CLP	35-40% /16 MIPS	30/7.5	67.5	24.2/2 0.3	5.3 / 6.3	3.9
GSM 06.10	RPE-LTP	10 MIPS	20/0	40	9.7	13	3.50
IS-54 (TIA)	VSELP	24 MIPS	20/5	45	16	8	3.54
IS-96 (TIA)	QCELP		20/5	45	15/32/ 64/160	8.5/4/2/0.8	
	LC-CELP	10 MIPS			8	16	
	CELP+	30 MIPS			18.8	6.8	
	RCELP	16 MIPS			26.7	4.8	
FS-1016	CELP	30 MIPS			26.7	4.8	3
	TFI	150 MIPS			32/53. 3	4 /2.4	
FS-1015	LPC10E	15 MIPS			53.3	2.4	2.4

Table 3-3: Speech and Audio Coders Quality

*) Note that this only includes the delay of the algorithm, i.e. look ahead and the frame delay in both ends. The total system delay would also include processing delay for computation, multiplexing delay, buffering delay and transmission delay. For example a GSM cellular system using GSM 06.10 coder has a system delay of 76.7 ms.

**) The mean opinion scores for coders under background noise and over degraded channels vary. Particularly the low bitrate coders (<16 kb/s) have been in the past sensitive to background noise. The newer cellular coders, such as G.729 (IS-641) give better scores than the old (e.g. GSM06.10) in these conditions.

3.8 Benefits of Voice over IP:

a) Lower Recurring Transmission Charges: By directing voice calls over the corporate data network, rather than through a carrier, companies can significantly reduce their monthly phone bills. These savings are obviously dependent on several factors, including the volume of intracompany calls and the distances between company offices. Companies with overseas offices, obviously, can experience the greatest savings, since they can eliminate a great deal of international long-distance charges. These charges are often particularly high when the call originates in a foreign country that still has a highly monopolistic telecom market. In some configurations, these savings can be extended to calls outside of the company as well using PSTN gateways.

b) Economic Factors: The economic appeal of transmitting voice calls over the data network arises from two technical factors. First, data networks almost always have spare capacity. Network managers typically over-provision IP networks to allow room for growth and to avoid congestion during periods of peak utilization. At the same time, voice calls consume relatively little bandwidth. The characteristics of human speech, especially the comparatively large amount of silence that takes place during conversations, allows for a great deal of compression in the digitized transport of the call. This makes it possible for voice to be carried on existing data network connections without requiring investments in adding to the capacity of those connections. Even when such additions have to be made to the existing network because of call volumes, those costs are insignificant compared to the recurring costs charged by carriers to carry that same calling volume.

c) Reduced Long-term Network Ownership Costs: In addition to reducing a company's monthly phone bills, converged network architecture also reduces the ongoing costs of owning two separate networks - one for voice and one for data. These costs include the need to buy two separate sets of equipment, the staff time dedicated to the operation and maintenance of that equipment, the licensing of any software relating to the management of that equipment, and the monitoring of traffic on the two networks. With the Internet revolution in full swing, the demand for skilled, experienced technicians far outstrips supply. This has driven salaries for voice and data network staff through the roof, and has also made it difficult to recruit and retain such engineering talent. Companies that are able to reduce their need for technical staff by streamlining their network operations can therefore eliminate many of the human resource management headaches that plague their competitors.

d) Advanced Applications: The most compelling aspect of converged voice/data networking may well be the new generation of applications it enables. These applications include Web enabled call centres, unified messaging and real-time collaboration. Other examples include real-time multimedia video/audio conferencing, distance learning, and the embedding of voice links into electronic documents. Three or four years ago, the Internet was not ready for prime time as a medium for commerce. But now it is. VoIP will be a valuable enhancement.

3.9 Risk Factors:

a) Loss of Voice Quality: Technologists understand that data networks are very different from voice networks. On the data network, especially on those Ethernet transports that dominate corporate computing environments, packets bounce around somewhat indeterminately. They can collide and get distorted or even lost. Error correction mechanisms in Ethernet hardware and the IP protocol itself can readily compensate for these phenomena on the data side. But such problems can adversely affect voice calls, which require a good quality, real-time flow of packets from one end of the network to the other. And, while the human brain can comprehend human speech even when there is a lot of distortion, users have become accustomed to a certain level of call quality.

b) Loss of Reliability: Data networks are not yet as reliable as voice networks. We all know what it is like to have our computer freeze or to be told that the network is "down". But this rarely happens with our phones or our telephone carriers. Immediate and uninterrupted access to others over the phone is such an essential aspect of conducting business that few executives want to put voice communications at risk, regardless of how attractive the potential savings may be.

c) Vendor Architecture Dependence: There are really two aspects that trouble most decision makers. One is the possibility that another, superior solution for VoIP will come along shortly after a commitment has been made to a particular vendor's product. If the investment in that product is substantial, it's usually impractical to scrap it and switch to the better approach. Of even greater concern for technology managers, however, is the fact that selection of one vendor's approach to voice/data convergence may cause a lock-in that extends far beyond the VoIP solution itself, forcing a long-term commitment to that vendor's overall networking architecture. This concern is exacerbated by the lack of clear standards in the VoIP market. In the absence of such standards, technology managers have legitimate concerns about committing their companies to any proprietary architecture.

d) Lack of Expertise and Experience: VoIP technology is new. Every new tool must be tested and mastered. This takes time. Without the proper expertise and careful planning the technology can work against the customer.

Chapter 4: QoS mechanisms for VoIP and VoATM

4.1 QoS supporting mechanisms

For some time, manufacturers of network hardware, primarily routers and switches, have offered a range of basic features to address the need to make the network operate more efficiently. Generically, these features can be categorized as elementary QoS. Because network efficiency diminishes when network devices receive more packets than they can handle some of these methods can also be called congestion-avoidance techniques. But QoS is now the predominant term. These features can be grouped into categories, based on the nature of the way they accomplish the task of improving network performance:

- **Rate Limiting:** Given that some applications tend to use more bandwidth than they are are allowed to; the rate-limiting component of Policer (formerly called Committed Access Rate, or CAR) can be an important bandwidth management tool for setting up and enforcing policies about traffic flows. With rate limiting, network operators dictate a maximum amount of bandwidth that a particular resource can consume. Then, network operators can configure Policer with several levels of policy that dictate what action should be taken on traffic that exceeds that limit. For example, the first 100 kbps of traffic generated by a particular resource could be rated "first-class" traffic in an enterprise's policy management scheme and receive a top priority marking. Traffic above the first 100 kbps generated by that same resource could drop to a lower priority class or could be discarded altogether.

- **Queuing:** improves network performance by improving the ability of a router or switch to handle the packet load it receives. To accomplish this, routers and switches maintain queues in which packets can wait for their turn to be forwarded.

- **Traffic Shaping:** A traffic shaper delays some or all outgoing IP packets in order to stream into compliance the traffic profile associated with the output link it will go out on. Typically, traffic shaping happens at the egress of a network domain. A shaper uses a buffer to store the incoming packets in excess of allowed bursts and delays their transmission to the output interface to make the traffic conforming to the contracted policies.

- **Bandwidth Reservation:** improves network performance by allowing devices to reserve bandwidth. To accomplish this, source and destination devices negotiate with routers and switches to establish a reservation for transmission capacity.

- **First-in-first-out (FIFO) queuing:** in which packets are buffered in a queue and handled in the order received. Buffering allows packets to be stored until they can be handled, providing a limited form of congestion management. But this method is crude and does not distinguish between different types of traffic.

- **Weighted Fair Queuing (WFQ):** provides a more sophisticated solution. In addition to the basic congestion management provided by queuing, WFQ requires devices to monitor traffic levels—a capability that is not present in all devices—and to ensure that no single application monopolizes network capacity. However, because WFQ does not distinguish between traffic types according to their importance, it effectively gives priority to low-bandwidth (and not necessarily mission-critical) traffic.

- **Discarding:** is another QoS technique, typically implemented in addition to queuing, in which devices discard packets before becoming congested. Discarded packets are not received, and therefore not acknowledged, by the destination device. This lack of acknowledgement causes the originating device to retransmit the packets and usually to slow the path of transmission.

- **Random Early Detection (RED):** Allow devices to discard packets when queue thresholds are reached. When a source device fails to receive acknowledgement of packets it has sent, it assumes the presence of congestion and slows the pace of its transmissions.

This process is designed to recover from congestion and minimize the chance of its recurrence. By discarding queued packets before the device becomes congested, RED simulates congestion on the router, and triggers natural congestion remedies. RED does not require flow-state capability in network devices, nor does it differentiate between types of traffic.

The following implementations typically use the precedence bits in IP headers, which are currently used within the IETF DiffServ working group and will be introduced in more detail in the following sections. Application traffic is classified by source and/or destination port, and user traffic is classified by address. Each traffic class is assigned a priority; each priority level is assigned a queue, with associated handling instructions. There is more work involved in reading header information and classifying packets than there is in reading the precedence bits or DSCP. So typically specialized devices, often at a WAN link or other "edge" of the network, classify and mark the DSCP in packets before forwarding them to the next hop. All other devices then read the markings and handle the packets according to the instructions associated with queue definitions:

- **Class-based Queuing (CBQ):** provides QoS through the use of multiple, prioritized FIFO queues. Administrators configure the devices to assign packets to queues according to markings in the IP header. A high priority queue is serviced first, and a low priority queue is serviced last. So CBQ can expedite important traffic at the expense of less important traffic. This is the most basic form of QoS that can distinguish between different types of traffic. This technique does not require devices to incur the overhead of monitoring the amount of traffic associated with various applications, so it does not prevent an application from monopolizing network capacity. But it does differentiate between traffic types, and therefore offers a more sophisticated level of QoS.

- **Class-based Fair Queuing (CFQ):** enhances CBQ to prevent an application from monopolizing network capacity. As with WFQ, CFQ requires network devices to monitor traffic levels in each administrator-defined queue and guarantees that an ill-behaving, high priority application will not monopolize bandwidth.

- **Weighted Random Early Detection (WRED):** combines CBQ and RED to randomly drop packets according to traffic type (as configured by an administrator). Using this method, administrators can set up priority queues for each traffic type and configure thresholds for discards. As with RED, WRED triggers the congestion control mechanisms in the network protocols.

Each of these techniques can improve network performance and deliver an enhanced quality of service; however, capabilities vary by vendor and model, and typically network administrators must use vendor-specific tools to configure and monitor their network devices. So these methods by themselves constitute a basic QoS solution that is labour-intensive, and therefore not particularly powerful.

- **Policies:** policy-based management leverages the power of directory services, providing solutions that deliver significant ease of use and allow administrators to standardize and automate configuration management. Both the Distributed Management Task Force (DMTF) and the Internet Engineering Task Force (IETF) are defining a group of compatible standards in this area. In this paradigm, a policy is a virtual object in the directory. A policy object contains a group of configuration parameters. The policy object is then associated with a group of devices or services, which will receive those configuration parameters when the policy is enforced. Because of this one-to-many relationship, policies provide a level of abstraction that magnifies the power of directory-enabled management solutions. Administrators can manage a single set of configuration parameters—QoS configuration, for example—for a group of devices or services.

4.2 Integrated Services Architecture (IntServ)

4.2.1 Basic principles

To give better treatment to applications with more demanding requirements appropriate mechanisms have to be provided. Applications need to be able to state their requirements, either ahead of time in some sort of service request function, or on the fly, by means of fields in the IP packet header. In the case of IntServ [41] the former approach is chosen.

It provides more flexibility in stating requirements, and it enables the network to anticipate demands and deny new requests if the required resources are unavailable. This approach implies the use of some sort of resource reservation protocol. Another requirement is that elastic traffic, like the traffic generated by Web Access (HTTP), remote logon (TELNET) and file transfer (FTP) for example, must still be supported. Since it is desirable to use the Internet as a common infrastructure to support both non-real time and real-time traffic, a reservation protocol, like RSVP, can help control this situation by denying service requests that would leave too few resources available to handle current elastic traffic. The IntServ model proposes two service classes in addition to best-effort service. They are:

- **Guaranteed Service:** This service provides an assured capacity level, or data rate. There are specified upper bounds on the queuing delay through the network and there are no queuing losses.

- **Controlled Load:** This service tightly approximates the behaviour visible to applications receiving best-effort service under low traffic conditions. The service ensures that a very high percentage of the packets do not experience delays, but there is no specified upper bound on the queuing delay or loss through the network. The IntServ model is characterized by resource reservation. Before data are transmitted, an application requiring QoS must first set up paths and reserve resources. The Resource ReSerVation Protocol (RSVP) is a signalling protocol for setting up paths and reserving resources and has been chosen as signalling protocol for IntServ [42]. Each IP packet is associated with a flow, a distinguishable stream of related IP packets that result from a single user activity and require the same QoS. Typically, an IP packet is identified as a member of a flow on the basis of source and destination IP addresses, port numbers, and protocol type. To allow different sessions with different QoS requirements between source and destination, a given interarrival time between any two packets must not be exceeded. Otherwise, the next packet belongs to a different flow. RSVP offers applications the means to request the level of QoS on a per-flow basis. The request dictates the level of resources (e.g., bandwidth, buffer space) that must be reserved along with the scheduling behaviour that must be installed in the routers to provide the desired end-to-end QoS commitment for the data flow. Furthermore it defines the maximum delay and delay variation that must not be exceeded along the chosen path.

4.2.2 RSVP

4.2.2.1 Introduction to RSVP

RSVP [43] was designed to enable the senders, receivers and routers of communication sessions (either multicast or unicast) to communicate with each other in order to set up the necessary router state to support QoS. RSVP is unidirectional and receiver-oriented. The receiver of the data flow initiates the resource reservation. This design decision enables RSVP to accommodate heterogeneous receivers in a multicast group. RSVP also provides several reservation styles that allow applications to specify how reservations for the same multicast group should be aggregated at the intermediate switches. By using "soft state" in the switches, RSVP supports dynamic membership changes, automatically adapts to routing changes and leaves the responsibility for maintaining states to senders and receivers. This approach follows the IP philosophy, where all functionality of an end-to-end connection has to be at the sender and the Receiver.

4.2.2.2 RSVP Protocol

RSVP identifies a communication session by the combination of destination address, transport layer protocol type and destination port number. It is important to note that each RSVP operation only applies to packets of a particular session and as such every RSVP message must include details of the session to which it applies. All RSVP messages include the session identification. At each intermediate switch, RSVP distinguishes between two kinds of state information: path state and reservation state. Each data source periodically sends a Path message that establishes or updates the path state, and each receiver periodically sends a Reservation message that establishes or updates the reservation state (which is attached to the path state). Path messages are forwarded from the sender to the receiver using the network's routing protocol. Each RSVP sender host transmits Path messages downstream along the unicast or multicast paths provided by the routing protocol. These messages enable intermediate nodes to maintain path state information about the previous hop. Assuming the Path message is valid, the router creates or updates the path entry and (re)sets cleanup timers which trigger the deletion of the entry if it is not refreshed in time. Each RSVP receiver host transmits Resv messages as reservation requests along the reverse path. All the intermediate nodes that process these requests create and update reservation state information accordingly. An RSVP reservation request contains a so-called flow descriptor composed of a flowspec and a filterspec. While the flowspec specifies the desired QoS, the filterspec defines the flow of the particular session, which the QoS is specified for. With the receiver making the reservation, there is greater flexibility in handling multicast flows. This receiver-based model allows for a distributed solution enabling heterogeneous receivers to make reservations tailored to their needs. If the sender were making the reservations, they would have to know the addresses and characteristics of all the possible Receivers. With the approach in this example, each receiver needs to understand only its own capabilities.

4.2.2.3 Reservation Styles and OPWA.

RSVP allows reservations for a single sender (Fixed-Filter style, FF) or groups of senders (Shared-Explicit style and WildCard-Filter style, SE and WF respectively). With these styles resource reservation is separated from whom the resources are reserved for. FF style is a distinct style of reservation, where the sources are explicitly specified. SE allows selecting particular senders and WF creates a single reservation shared by all flows from all upstream senders. Path messages may include an ADSPEC field that gathers the information along the path to predict the potential end-to-end QoS and arrive as advertisements at the receivers. With this information the receivers can decide to change the reservation (if resources are available) to get the desired QoS. That scheme is called One Path With Advertising (OPWA).

4.2.3 IntServ Router architecture.

To obtain the main characteristics of the IntServ model, great changes are proposed in the router architecture of the existing best effort service Internet model. One important assumption made in the development of the IntServ model is that the network resources can be explicitly controlled. This means that admission control and reservation are basic blocks of the IntServ model. This also means that the intermediate routers need to maintain flow specific information. As mentioned before, to manage congestion and to provide QoS transport, IntServ use the following components: packet scheduler, microflow classifier, admission control and a reservation setup protocol. For the purpose of traffic control, each incoming packet must be mapped into some class; all packets in the same class get the same treatment from the packet scheduler. The microflow classifier based on IP packet header fields does this mapping. Every packet is identified as belonging to a certain session or flow. Once the classification of the packets is done they are sorted according to their class in the packet scheduler.

The packet scheduler manages the forwarding of different packet streams using a set of queues and other mechanisms like timers. It should be implemented at the point where packets are queued. The classifier and the packet scheduler develop functions that accomplish forwarding. Because the forwarding decision of the routers is made for every packet, it must be highly optimized. The control plane is mainly built from the following two blocks: admission control and the reservation setup protocol. They come into action during the connection setup. If the admission control accepts a reservation request (made by the resource reservation protocol, e.g., RSVP), then appropriate changes are made to the classifier and the packet scheduler. Admission control implements the decision algorithm that a router uses to determine whether a new flow can be granted the requested QoS without impacting earlier guarantees. This is invoked at each node along the path to make a local accept/reject decision, at the time when a host requests a real-time service along some path through the Internet. The reservation protocol is responsible for maintaining flow-specific state information at the end systems and at the routers along the path of the flow and also responsible of configuring classifiers, policers and schedulers. Figure 4-1 shows the described router architecture for the IntServ Model:

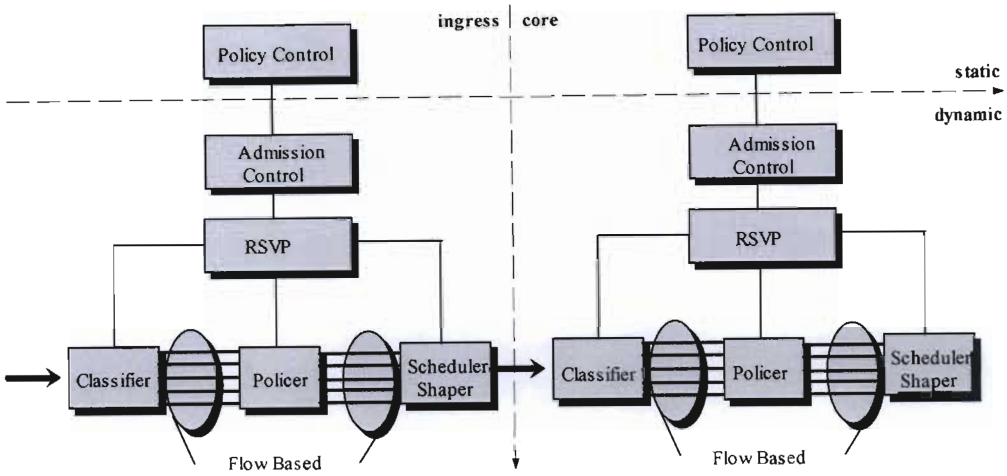

Figure 4-1: Router architecture in the IntServ Model

4.2.4 Evaluation of benefits and drawbacks

IntServ with its advanced signalling protocol has considerable benefits:

- With Guaranteed Services and Controlled Load, there are two classes that offer either delay and bandwidth guarantee or bandwidth guarantee only.

- Both classes offer a fine-grained approach that provides QoS to individual applications or flows. Not only can senders exactly specify their requirements, but the receiver can also announce its capability, thereby possibly reducing the necessary reservation. For point-to-multipoint connections, different receivers can announce different reservations and these reservations are aggregated in a way that minimizes network resource utilization.

- If routers implement Weighted Fair Queuing and if the nature of a traffic source can be characterized there will be an absolute upper bound on the network delay for the traffic in question [44].

- IntServ guarantees that reservations are not held longer than necessary. Either a reservation is maintained (and thus used) by periodic sending of refresh messages, either it is torn down explicitly, or it times out and is deleted. Thereby no unused (or unpaid) bandwidth will be reserved

- RSVP allows renegotiation. Applications that do not need their initial bandwidth anymore can return part of their reservation. This can lead to a much better utilization of the network in terms of real traffic (which is different from the utilization in terms of reserved - and thus paid - bandwidth). Further, this leads to a higher satisfaction among the users. However, returning bandwidth also contains a risk. It is not guaranteed that an application can increase its reservation at a later point if the need occurs.

- RSVP signalling messages are forwarded from the source to the destination and back. Requested QoS parameters are absolute and cannot be interpreted by nodes of different domains in a different way. When tunnelling another non- IntServ domain:

  – the content of the messages remains and will be reused at the end of the tunnel and
  – the messages can be mapped into an appropriate QoS class.

Therefore the user might notice little or no degradation. On the other side there are also some partially severe drawbacks, some of them result from IntServ being a very "heavy-weight" architecture.

- Scalability in large networks is not very good. As mentioned above each router at the border and in the core — maintains flow specific information. In core networks where routers easily see thousands of flows simultaneously (measurements suggest approx. 250,000 source-destination pairs per minute on an OC-3 link [45] due to the short lifetime of a flow) this leads to long processing times because of the table lookup for classification and routing. There is as well a high memory demand to keep information about all the flows. Particularly with respect to IPv6 and its long address and header structure this might pose a problem. However, in smaller campus and access networks, this restriction is not a real issue. With much lower numbers of different flows IntServ can perform very well.

- All communication is connection-based, i.e., even for short transmissions a connection has to be set up. For a sender that sends short bursts with large intervals between bursts, this means that each time packets are ready for transmission, the time-consuming process of making a reservation has to be repeated and it is not guaranteed that the connection will be accepted. The sender can set up the connection once and keep it as long as necessary, but utilization of this reservation will be low and thus uneconomical.

- Router performance is not only needed for connection setup but also for all other RSVP processing. Connections are torn down if they are not regularly refreshed, so senders have to send PATH messages in certain intervals.

- Applications generally do not know how much bandwidth they need. Exceptions are voice applications, where depending on the decoder the upper limit for bandwidth needs are well known. Video applications can know their requirements if they know in advance what they will send (this can be true, e.g., for video-on-demand where the content of the video stream is already known, but becomes difficult for live transmissions). This means, that either applications have to be modified or that users have to define the requirements by themselves. Modifying applications in a way that they can detect their needs is difficult as can be seen from the example. However applications can offer different coarse-grained QoS levels and the user can decide which one he wants. Then, of course, IntServ's advantage of fine granularity is not used optimally. Another approach is the use of an agent or middleware that measures the traffic and makes the reservations for the application based on the measurement. There is no need to modify existing applications, but finding a good algorithm for measurement might be difficult. Using a bad algorithm leads to inefficient reservations. At present, there exist hardly any applications that allow users to select their quality level. The same is true for applications that can determine traffic parameters by themselves. This means the modification or reprogramming of many of today's existing programs — a very futile undertaking.

- IntServ works best if all nodes along the path support RSVP. Nevertheless, nodes not supporting RSVP let pass RSVP messages. On the other hand, this means that from the sender and receiver point of view the nodes guarantee something, they cannot comply with. Therefore, a connection setup might succeed, but service parameters cannot be guaranteed end-to-end.

4.2.5 Technical effort and availability

Gradual upgrading within a domain does not improve much because end-to-end QoS is only as good as the worst section on a selected path. Thus upgrading needs to cover at least an administrative area or sub domain. Still, users attached to this domain cannot expect end-to-end QoS if the intermediate networks and the network of the communication partner do not support IntServ. Therefore, for every user to be able to use IntServ, all routers and end-systems (at least those that allow the use of time-critical applications) need to support RSVP. While this is very costly in monetary terms, fortunately most routers can be upgraded in a simple way via software extensions or updates. Changes are required in the router's forwarding path and the background functions (routing, CAC, reservation setup) [46].

The router must already support scheduling mechanisms like Weighted Fair Queuing. IntServ support and RSVP are available today on all manufacturers' equipment. Administrative costs are relatively small, because all reservations are user-initiated and need no active management by the network provider. Weighted Fair Queuing, as mentioned in the benefits of IntServ, is available in commercial routers today and is used to segregate traffic into classes based on criteria like protocol type or application. RSVP and therefore IntServ are available on most of today's end-systems. Operating systems like Windows [47] and Linux [48, 49] offer an application programming interface (API), at least in newer versions. Nevertheless there are currently only few applications that support this QoS approach. It is uncertain, as to when this software will ever be available on the commercial market and if so, there might be additional costs for the user to buy it or to upgrade existing versions. Another problem that reduces availability is compatibility.

Trials in other test beds e.g., [50] have shown, that there are sometimes problems when using different RSVP implementations, e.g., when connecting Windows and Linux systems.

4.3 Differentiated Services Architecture (Diffserv)

4.3.1 Basic principles

Although IntServ has made significant strides toward providing QoS, it is widely considered too heavyweight to scale across a large internetwork like the Internet. DiffServ addresses the issue of scalability by defining QoS mechanisms that operate on aggregates of flows with similar QoS requirements. Differentiated Services (DiffServ) is an IETF specified QoS architecture [51] that was proposed to provide service differentiation by creating service classes using either the ToS field of the IPv4 or the priority bits of the IPv6 headers [52]. The DiffServ architecture achieves its scaling properties by defining a small number of simple differentiated packet forwarding treatments, known as per-hop-behaviours (PHBs). Individual network elements implement these PHBs through a variety of mechanisms and queuing disciplines. The essence of DiffServ is to combine these differentiated PHBs with carefully configured traffic policing mechanism at the edge of the network to provide a variety of services. By concentrating policy enforcement activities at the edge and providing simple aggregate data handling in the core, a network operator can ensure that new IP services do not require excessive state information or expensive forwarding decisions in core network routers. Each data packet that enters a DiffServ ingress router is marked with a DiffServ codepoint (DSCP) in a newly defined IP header field (the DS field) to indicate which PHB should be applied to the packet. Packets marked with the same code point are considered a behaviour aggregate and they all receive the same PHB treatment, regardless of the micro-flow to which they belong.

4.3.2 DiffServ Router architecture

In addition to forwarding engines capable of implementing the emerging PHB standards, the DiffServ architecture requires edge devices that include traffic conditioning components that are able to classify, mark, shape, and drop packets as they enter and leave the DiffServ domain. Each DiffServ enabled edge router implements a traffic conditioning function which performs metering, shaping, policing and marking of packets to ensure that the traffic entering a DiffServ network conforms to the traffic conditioning rules specified within an SLA. At the edge of the network or administrative boundary, the classifier sets the values of the DS field for each incoming flow. The classifier selects packets based on the combination of one or more predefined set of header fields. The mapping of network traffic to the specific behaviours that result in different classes of service is indicated by the DS field. Each DS field uniquely identifies the per-hop-behaviour or the treatment given to the traffic at each hop along the network path. Each router sorts the packets into queues based on the DS field. The queues might get different treatment based on their priority, share of bandwidth, discard policies, etc. As marked packets flow downstream, they are merged with equally marked DiffServ packets into a behaviour aggregate and all subsequent traffic conditioning is performed on aggregate traffic.

As conforming traffic traverses a DiffServ domain, it may acquire unacceptable burst characteristics as a side effect of various queuing delays or increased aggregation. In order to ensure that the burst characteristics of aggregate data conform to the traffic specification for the provisioned service, each DiffServ domain must have the ability to perform traffic shaping on each aggregate as it exits the domain, and should shape each aggregate traffic flow as it exits the domain. The router implementations in the intermediate nodes use the 6-bit PHB field to index into a table for selecting a particular packet-handling mechanism. This forwarding policy determines how routers will handle the packets in terms of providing a class of service by combining traffic management functions such as packet queuing, scheduling, and buffer reservations at each node. DiffServ expects advance provisioning and reservations made in each of the intermediate nodes along the network path. If a network path crosses multiple DS domains or multiple ISPs, the ISPs must support the same PHBs to provide a consistent end-to-end service.

The Differentiated Service Router architecture is depicted in Figure 4-2:

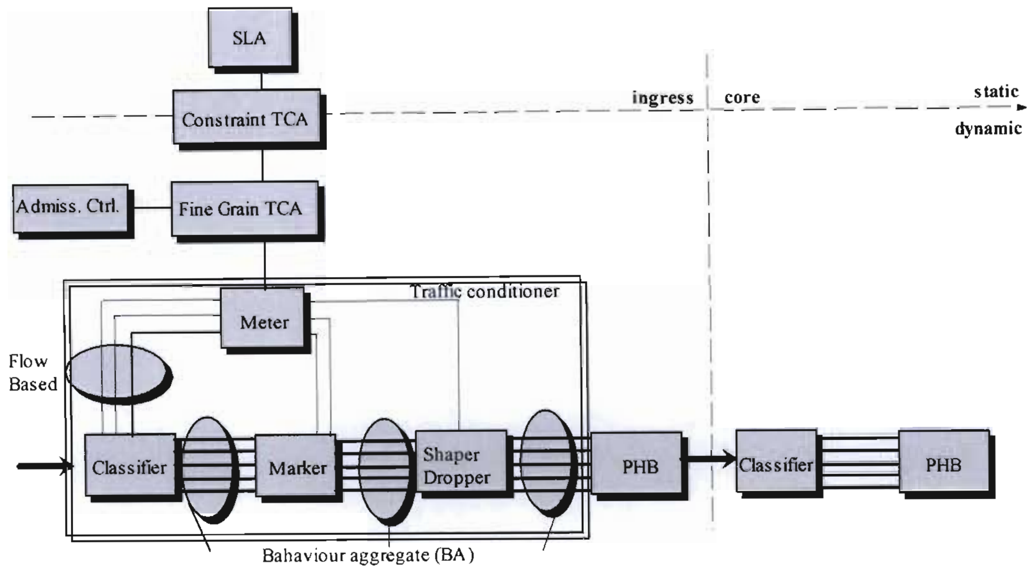

Figure 4-2: Router architecture in the Differentiated Service Model

4.3.3 Evaluation of benefits and drawbacks

The DiffServ architecture has been developed in response to some of the drawbacks of the IntServ architecture. Benefits of the DiffServ approach are:

- The core nodes are much simpler. The classifier keeps no state information of the flows so the scaling problem of IntServ simply does not exist here. Classifying is done accordingly to the content of the DS field.

- There is no initial setup delay due to reservation signalling.

- Currently only a small fraction of all applications need strong QoS guarantees. Therefore, giving relative precedence for certain flows over others might be enough, provided there is a good traffic classification and use of high priority classes is reserved for traffic that needs reservation [53].

- DiffServ does not require the applications to be able to identify their needs. Classifying of unclassified packets is done at the ingress node to the Differentiated Service domain. Depending on protocol information in the packet header (e.g., TCP/UDP or even higher protocol information like FTP/RTP), the appropriate codepoint can be assigned.

- If the classification is not done at the ingress router but by the user (i.e., the user can choose his favoured QoS level), he will not be bothered with different parameters (bandwidth, delay, etc.). Instead, he just chooses from a very limited number of quality levels, like e.g., the Olympic Services (gold, silver, and bronze). So users do not need to have insight into the technical aspects of the QoS architecture but have a rather intuitive interface. Incentives to use lower level services can be given by the pricing system.

However, there are also some drawbacks:

- DiffServ does not guarantee absolute bandwidth or delay but only a relative level of QoS.
- Since DiffServ works without reservation, local congestion in the core cannot be prevented.
- To prevent violations of the service parameters (core nodes cannot detect violations due to the lack of per-flow information) border nodes need meters to retrieve information about the flows. The shaper and dropper delays or discards packets that do not confirm to the service parameters. However, this is done on a service class level only. Single flows still can use more than their fair share of the resources. To prevent this, further Usage Parameter Control (UPC) has to be implemented, at least at the ingress nodes.

- As with all QoS providing architectures, DiffServ works best, if all nodes along the path support the appropriate mechanisms. However, as DiffServ does not provide strict but only relative guarantees, the user might not experience any visible degradation, even when mapping of classes between different administrative domains will cause a loss in QoS. A problem that occurs, if there are nodes in the path that do not support DiffServ, is that flows will not know whether nodes do classify packets and have per-hop-behaviours implemented. Instead flows with higher precedence and best effort traffic will be treated equally in the absence of such mechanisms. Therefore no end-to-end guarantees on loss, delay or delay variation can be given.

- Classifying flows can be very complicated. The ingress router might have to analyze protocol information of higher layer protocols to determine which DS code point to assign. Looking for higher protocol information not only contradicts the layered model, but means at the same time a higher effort to decode the (variable-sized) packet header.

- If the classification is not done at the ingress router but at the end system (i.e., by the application)the same problems as with IntServ occur. There are currently no applications that can perform that task.

- Tunneling of DiffServ classes can be difficult, because a service class is an aggregate of many flows and has large needs for resources. SLAs need to be defined at the domain border to assure that the classes receive the correct QoS in the other domain. This is valid for all domain borders, even if both sides use DiffServ, because it cannot be assumed that the same classes are available in each domain. Furthermore, even if the class model is the same, parameterization of the different classes can be different.

4.3.4 Technical effort and availability

Gradual upgrading might not improve much because end-to-end QoS is only as good as the worst section along a given path. Thus upgrading needs to cover at least an administrative area or subdomain. Still, users attached to this domain cannot expect end-to-end QoS if the intermediate networks and the network of the communication partner do not support any kind of QoS architecture. While this can be very costly in monetary terms, fortunately most routers can be upgraded in a simple way via software extensions or updates [46]. Changes are required in the router's forwarding path and the traffic control (routing, classifying, metering, and marking). DiffServ is available on most of today's end-systems. Operating systems like Windows2000 [54] and Linux [55] offer an application programming interface (API). Nevertheless there are currently only few applications that support this QoS approach. It is uncertain, as to when this software will ever be available on the commercial market and if so, there might be additional costs for the user to buy it or to upgrade existing versions.

4.4 Multiprotocol Label Switching (MPLS)

MPLS [56] differs in some ways from the QoS architectures mentioned above:

- MPLS is not a QoS architecture on its own. First of all, it provides means for traffic engineering. Secondly it provides functionality to support other architectures like Diff-Serv.

- MPLS works on layer 2 and can be used on different layer 1 protocols.

- MPLS can not only be used for IP but as the name implies also for other protocols like IPX, AppleTalk and others. However, [56] and most drafts focus on the use of MPLS together with IP exclusively.

4.4.1 Basic principles

In connectionless layer 3 networks, each packet is routed independently. Therefore, each router along the packet's path has to analyze the address in the packet's header and decide upon which output port to take. The fact that multiple packets belong to the same flow and are forwarded to the same port can not be exploited. Furthermore, other information from the header, like IP precedence or Virtual Private Network (VPN) membership can often not be considered due to performance reasons. The basic idea of MPLS is, to replace layer 3 routing by layer 2 switching. Instead of address-based packet routing, the output port is determined by a short label that all packets carry and that can be looked up in a table. The label is not an encoding of the destination address, but summarizes different information about the packet:

- Destination
- Precedence

- VPN membership
- QoS information (from RSVP)
- The route for the packet, as chosen by traffic engineering.

For this purpose special Label Switching Routers (LSR) are needed that provide the MPLS functionality. Even though, the initial determination of an appropriate output port is still done by routing, there is still the need for classical routing protocols like OSPF or IS-IS. With the use of MPLS there is — like in ATM networks — a mesh of so-called Label Switched Paths (LSP) that connect the LSRs with each other. The logical view of these LSPs can be quite different from the physical view of the network. The LSPs are set up via some kind of signalling and a label is assigned for each hop on the path. A label is valid only on this single hop and LSPs are unidirectional. Therefore bi-directional connections need two LSPs. LSPs are set up at the beginning of a connection and persist until its end. In general, the mechanism of packet forwarding consists of two steps:

1. All packets are assigned to one of several Forwarding Equivalent Classes (FEC). A Forwarding equivalent Class F is represented in the packet header by a label L for the next hop.
2. One port is assigned to each FEC (i.e, several FECs can use one port, but all packets of one FEC always use the same port).

With respect to the next hop, packets within a FEC can not be distinguished anymore. They all follow the same path throughout the whole MPLS domain. Such a bundle of packets is called a stream. The label L does not necessarily represent FEC F on all hops in the network. The binding of L to F is local. However, a router has to ensure that its binding of L to F is unique and may not be confused with another label for a different FEC from another router. Even though step 1 is performed in each node within the MPLS domain, its realization differs in border and in core nodes. At the ingress router packets do not have an MPLS label, so the border nodes assign the label of the packet according to its destination (and maybe other criteria). Therefore, that stream is chosen whose destination address prefix matches best with the destination address of the packet. Within the domain, core nodes only consider the label and do a lookup in their switching tables to determine the label for the next hop (label swapping). Here it can happen, that packets from different incoming FECs are aggregated in a new outgoing FEC with the same new label (label merging). The swapping of labels can be compared to the VP/VC swapping mechanism of ATM. However different FECs can also be transported over multiple hops by use of an MPLS-tunnel. The headers of all packets are extended by the same label (building a label stack) that is removed at the end of the tunnel. Then, even though the packet is still in the MPLS domain, the different streams can be easily distinguished by their original label.

4.4.2 Evaluation of benefits and drawbacks

Even though MPLS is intended to work on different layer 1 technologies, it works especially well on ATM networks. MPLS can take advantage of the optimized hardware of ATM switches that allows fast forwarding of packets. MPLS' benefits are:

- Traffic engineering is well supported by MPLS. With the use of LSPs traffic can be directed across the network in a more efficient way, i.e. the usage of single links is maximized and delays can be minimized.

- MPLS' CoS feature supports DiffServ very well by providing differential treatment for higher layer classes. By appropriately mapping DiffServ classes at the ingress node of the MPLS domain, higher prioritized layer 3 packets can experience the treatment of high prioritized layer 2 packets.

- MPLS can also be used to provide different service levels for different layer 4 protocols i.e. mail and web traffic can be routed differently. This might support e.g., different charging, because more expensive traffic (like time-critical traffic) is routed over reserved and thus more expensive lines. MPLS can also be used to provide different service levels to different users. Large corporate customers can be separated from private users. Here, we also have the possibility to charge different prices for different routes, i.e., service levels.

- Some advantages arise from the concept to assign a label once at the ingress node and not to analyze it anymore in subsequent routers:

- Forwarding can be done by switches that cannot do routing or are too slow to analyze the header at adequate speed.

- By assigning the label at the ingress node, information about its origins can be forwarded to nodes in the core that would otherwise not know about it. The packet's entry point to a network is known to all nodes in the domain.

- Assignment to a FEC at the ingress node can be changed or refined (i.e., made more complex) without impact on the core nodes.

- Packets are forced on a certain route rather than letting each node decide about the next hop. This is desirable if different routes have different prices or precedence. However, there are also severe drawbacks:

- MPLS is not a QoS architecture but a mechanism for traffic engineering. Whereas traffic engineering maximizes the usage of the network, it also causes the average length of paths to be slightly longer. Furthermore, by increasing the mean rate on a link and by increasing the number of interfaces to pass (longer path), keeping strict bounds with respect to delay and delay variation becomes more difficult. Only by the use of an overlaying QoS architecture (like DiffServ) it is possible to accomplish QoS goals.

- Route selection in MPLS is based on two options:
- hop by hop routing which brings the drawbacks that current routing in IP networks has:
    - care must be taken to prevent loops,
    - it is not guaranteed that the best path is found.
- explicit routing, which allows QoS routing by selecting the best path at the ingress

node.

However, current IP routing protocols do not provide appropriate information to find a path that satisfies the traffic's requirements. Therefore, MPLS cannot set up its LSPs in a way that reflects the QoS needs of certain streams.

- MPLS needs its LSP database. In large networks and especially with a high user and service differentiation level (i.e., many different classes), these databases can become very large. Besides the necessary routing tables in the ingress nodes, this leads to another entity that consumes memory and processing power.

- Protocol or user separation as mentioned above increases complexity. Protocols do not pose much of a problem, because their number is rather limited. User separation poses more of a problem. Providers might be tempted to offer only a small number of different user profiles, but these might not fit every customer. Especially for corporate customers, individual profiles might be necessary.

- MPLS is very strict in terms of homogeneity. Only if all nodes within a domain support MPLS the system works, i.e., an MPLS domain cannot contain any non-MPLS nodes. Whereas IntServ or DiffServ nodes do at least forward packets (even though only as a best effort packet), non-MPLS capable routers cannot forward MPLS packets, because they do not recognize the MPLS header. A solution might be to gradually extend the MPLS domain within a network.

4.4.3 Technical effort and availability

MPLS only works if all nodes within a domain support MPLS; otherwise no LSPs can be set up. Therefore a network has to be upgraded in one step. MPLS switches and switch routers are available by all manufacturers. On Cisco products. MPLS is delivered mainly in conjunction with ATM switches. However POS cards are available too that have MPLS switching capability and different queuing/scheduling systems for differentiated CoS for high bandwidth (up to OC48). Provided, that an already existing switch has support for the necessary queuing and scheduling mechanisms, it is possible to upgrade hardware like IP routers and ATM switches to support MPLS.

4.5. ATM QoS mechanisms

ATM QoS is defined at the time of VC establishment by means of signalling messages exchanged during this process. Once the VC has been successfully established, its QoS characteristics are guaranteed. The guarantees are provided using the following mechanisms.

- Connection Admission Control (CAC) to decide whether or not the QoS requested by a new connection can be supported by the network. The connection is admitted if there are sufficient resources to guarantee the level of service. Otherwise, it is rejected.

- Usage Parameter Control (UPC) to ensure that the traffic injected into the network by a particular connection does not exceed the limits specified in the traffic contract negotiated during connection setup.

- Queuing and scheduling mechanisms at each node to ensure that each connection gets its share of bandwidth.

The CAC function obtains the following information from the traffic contract –

(i) The values of the traffic attributes specified in the traffic descriptor – i.e. PCR, SCR, MBS, and MCR etc.
(ii) The requested QoS class and values of the requested QoS attributes in either direction. The QoS attributes include maximum CTD, maximum CDV, and CLR etc.
(iii) The requested conformance definition for traffic of that connection. This includes PCR and SCR values used to determine whether or not a cell is conforming.

Using this information, the CAC mechanism determines whether to accept the connection. CAC also allocates resources for the connection at each node along the route. The UPC function is responsible for monitoring a connection to determine whether the traffic injected into the network is in compliance with the traffic contract. Appropriate actions are taken based on whether the traffic is compliant or not. UPC must ensure that non-compliant traffic of a connection does not affect the QoS provided to other connections that comply with the traffic contract. UPC also checks if the VPI and VCI values of the incoming traffic are valid. Bandwidth allocation is accomplished by setting up queues within an ATM node and servicing these queues in accordance with a scheduling algorithm. Assigning priorities to the queues can provide different levels of service. A scheduling algorithm that allocates bandwidth to the queues based on priority is employed. The ATM specifications do not define the scheduling algorithm or how these queues are set up.

Chapter 5: Estimation of Voice Quality

In Chapter 1, the E-Model was introduced. It is a model that uses a number of parameters to assess voice quality. R is the overall parameter that defines voice quality. Table 1.2 (in Chapter 1) shows a comparison of the R value and the MOS levels. R is represented in [3] [5] [8] as:

$$R = Ro - Is - Id - Ie + A \qquad 5.1$$

In this Equation 5.1, Ro represents the basic signal-to-noise ratio. This is based on send and receive loudness rating as well as circuit and room noise. Is represents impairments that are associated with voice signals, like incorrect loudness levels, quantization noise, and incorrect sidetone levels. Id represents impairments associated with delay. This includes end-to-end delay effects, and increased echo impairment due to delay. Ie represents equipment related impairments that are associated with specific equipment. Examples are coding schemes and packet loss levels. Finally, A represents an advantage factor that is assigned based on "advantage of access". These include (but are not limited to) mobile or cellular telephony, and telephony in hard to reach regions that may be reached only with satellite hops.

Table 5.1 is the default values and the permitted ranges for the parameters used as inputs to the E-Model [6]. Since the parameters of primary concern of this research involve delay, voice compression, and packet loss, this discussion will be centred around equations involving those parameters. For more information about the details of each parameter the reader is referred to [6] and [8]. These references include a full implementation (including source code) of the E-model.

Parameter	Abbr.	Unit	Default Value	Permitted Range
Sending Loudness Rating	SLR	dB	+8	0...+18
Receiving Loudness Rating	RLR	dB	+2	-5...+14
Sidetone Masking Rating	STMR	dB	15	10...20
Listener Sidetone Rating	LSTR	dB	18	13...23
D-Value of Telephone, Send Side	Ds	-	3	-3...+3
D-Value of Telephone, Receive Side	Dr	-	3	-3...+3
Talker Echo Path Loss	TELR	dB	65	5...65
Weighted Echo Path Loss	WEPL	dB	110	5...110
Mean one-way Delay of Echo Path	T	ms	0	0...500
Round Trip Delay in a 4-wire Loop	Tr	ms	0	0...1000
Absolute Delay in echo-free Connection	Ta	ms	0	0...500
Number of Quantization Distortion Units	qdu	-	1	1...14
Equipment Impairment Factor	Ie	-	0	0...40
Circuit Noise referred to 0 dBr-point	Nc	dBm0p	-70	-80...-40
Noise Floor at the Receive Side	Nfor	dBmp	-64	cannot be modified
Room Noise at the Send Side	Ps	dB(A)	35	35...85
Room Noise at the Receiver Side	Pr	dB(A)	35	35...85
Advantage Factor	A	-	0	0...20

Table 5.1 – Default values for E-Model

5.1 Impairment Calculations of the E - Model

As mentioned earlier, the E-Model relies on a system of impairments. The Simultaneous Impairment (*Is*) equations affected are [6]:

$$Is = Iolr + Ist + Iq \qquad 5.2$$

Iolr and *Iq* are impairments that are related to low values of overall gain and impairments due to quantization (separate from impairments related to coding) respectively. *Ist* is the impairment due to non-optimum sidetone.

$$Ist = 10\left[1+\left(\frac{STMRo-12}{5}\right)^{6}\right]^{1/6} - 46\left[1+\left(\frac{STMRo}{23}\right)^{10}\right]^{1/10} + 36 \qquad 5.3$$

$$STMRo = -10\ln\left[10^{\frac{STMR}{10}} + e^{\frac{T}{4}} 10^{\frac{TELR}{10}} \right] \qquad 5.4$$

STMR is the Sidetone Masking Rating, *TELR* is the Talker Echo Path Loss, and *T* is the mean one way delay of the echo path. As one can easily see, as *T* increases, the second term of Equation 5.4 actually becomes minimized. This is due to fact that lack of sidetone is the most noticeable when there is no delay in the echo path. Therefore, this term goes to a default value as *T* increases. The impairment due to delay is the Delay Impairment factor (*Id*). *Id* is defined as [6]:

$$Id = Idte + Idle + Idd \qquad 5.5$$

Idte is impairment due to talker echo.

$$Idte = \left[\frac{Roe - \mathrm{Re}}{2} + \sqrt{\frac{(Roe - \mathrm{Re})^2}{4} + 100} - 1 \right]\left(1 - e^{-T}\right)$$

$$Roe = -1.5(No - RLR) \qquad 5.6$$

$$\mathrm{Re} = 80 + 2.5(TERV - 14)$$

$$TERV = TELR - 40\ln\left(\frac{1 + \frac{T}{10}}{1 + \frac{T}{150}} \right) + 6e^{-0.3T^{2}}$$

Where *RLR* is the Receiving Loudness Rating. *Idle* represents Listener Echo.

$$Idle = \frac{Ro - Rle}{2} + \sqrt{\frac{(Ro - Rle)^2}{4} + 169} \qquad 5.7$$

$$Rle = 10.5(WEPL + 7)(Tr + 1)^{-0.25}$$

Where *WEPL* is the Weighted Echo Path Loss and *Tr* round trip delay in a four wire loop. The E-Model has tried to capture the impairment due to absolute delay with the following calculations [6]. Figure 1.4 in Chapter 1 illustrates the impairment that absolute delay imparts to voice quality. Note: If one way delay (*Ta*) < 100 ms, *Idd* is assumed to be 0.

$$Idd = 25\left\{ \left(1 + X^{6}\right)^{1/6} - 3\left(1 + \left[\frac{X}{3} \right]^{6} \right)^{1/6} + 2 \right\} \qquad 5.8$$

$$X = \frac{\ln\left(\frac{Ta}{100}\right)}{\ln(2)}$$

Where *Ta* is the delay in an echo free environment. *Ie* is the impairment due to equipment using codecs and voice encoding schemes. These values do not have mathematical relationships, but rather are dependent on mean opinion score test results that are converted to impairment values. [57] contains the latest recommended values for *Ie*. Table 1.3 and 1.4 in Chapter 1, summarize selected coders, packet loss percentages and their corresponding *Ie* values. Finally, the Advantage (*A*) factor is used to reward technologies that are so novel that users will tolerate more degradation than normal. An example may be a wireless connection in an area that previously was unable to have such access. TIA used the E-Model to arrive at a set of recommendations for Voice over IP use. This set of recommendations is published as [3]. First, a set of classifications for voice over IP were defined. Low, Medium, and High Voice Quality were defined at minimum *R* values of 50, 70, and 80. Figure 5.1 shows the levels defined compared to the graph of delay versus *R* value (similar to Figure 1.4).

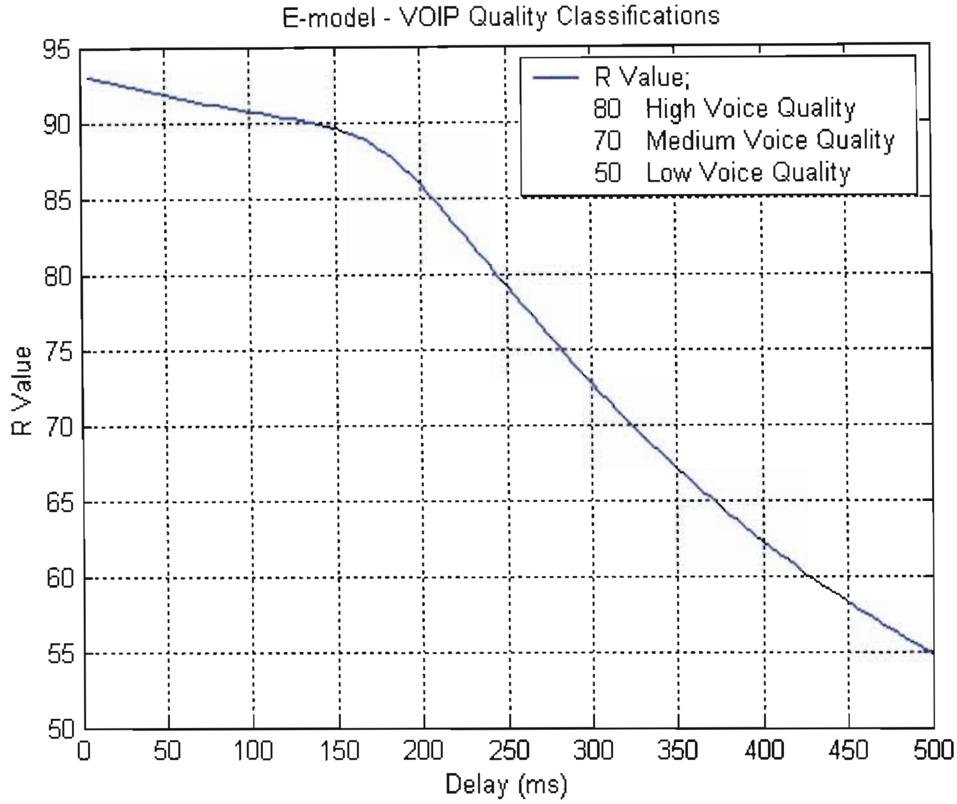

Figure 5.1 - VoIP Voice Quality Levels vs. Delay

The following are a summary of the recommendations [3]:

- Delay Rule 1: "Use G.711 end-to-end because it has the lowest *Ie* value and therefore it allows more delay for a given voice quality level."
- Delay Rule 2: "Minimize the speech frame size and the number of speech frames per packet."
- Delay Rule 3: "Actively minimize jitter buffer delay."
- Delay Rule 4: "Actively minimize one-way delay."
- Delay Rule 5: "Believe in the E-Model, which permits longer delays for low *Ie* value codec's, like G.711, for a given R-value."
- Delay Rule 6: "Use priority scheduling for voice-class traffic, as well as RTP header compression and data packet fragmentation on slow-links to minimize the contribution of this variable delay source."
- Delay Rule 7: "Avoid using slow serial links."
- Speech Compression Rule 1: "Use G.711 unless the link speed demands

compression."

- Speech Compression Rule 2: "Speech compression codecs for wireless networks and packet networks must be rationalized to minimize transcoding issues."
- Packet Loss Rule 1: "Keep packet loss well below 1%."
- Packet Loss Rule 2: "Use G.711 with PLC."
- Packet Loss Rule 3: "If other vocoders and codecs are used, then use vocoders and codecs that have a built-in or an add-on PLC."
- Packet Loss Rule 4: "New PLCs should be optimized for less than 1% of packet loss."
- Transcoding Rule 1: "For interoperability, IP gateways must support wireless codecs or IP must implement unified Transcoder Free Operation with wireless codecs."
- Transcoding Rule 2: "Maximum number of transcodes =1."
- Transcoding Rule 3: "DCME equipment must be eliminated."

The recommendations strongly recommend using the E-model and using G.711 whenever the bandwidth exists without regard to economics or efficiency. It seems more reasonable to use the E-Model to maximize the number of flows and still maintain a certain level of voice quality. This is explored further as an optimization problem in Chapter 6.

Chapter 6: The E – Model Optimization

The E-Model, as discussed in the previous Chapter is a model that allows users to relate network impairments to voice quality. This model allows impairments to be introduced and voice quality to be assessed. The E – Model optimization was previously carried out in [59]. It was based on an Air Traffic communications environment. The Model has since been revised by the ITU. The research carried out here is based on the current version of the E – Model ITU recommendations G.107 & G.108 .The current version provides an enhanced modelling of the voice quality. Three cases are considered to demonstrate the effectiveness of the E-Model optimization. The objective function for all cases is to maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (R). The cases considered are:

1. Optimization case 1: Find voice coder given link utilization, packet loss % and link bandwidth.
2. Optimization case 2: Find voice coder and packet loss % given link utilization and link bandwidth.
3. Optimization case 3: Find voice coder and link utilization level given packet loss % and link bandwidth.

Optimization and E – Model software which provides an E – Model extension to Voice quality measurement was used to solve the problem [58], [60], [61].

6.1 Description of the Optimization Problem

The E-Model has several qualities that interact with each other. For example, subjective tests have shown that there is a trade off between bandwidth and quality when looking at compression methods [3]. There is also a relationship between delay and quality [3]. Refer to Chapter 5 for a detailed description of the variables and constants used in this model. The E-Model is extremely complex with 18 inputs that feed interrelated components. These components feed each other and recombine to form an output (*R*). An example of this complexity is that the delay parameter is used to generate the simultaneous impairment component as well as delay impairment component. Several assumptions are made with this optimization. First, we can assume 1 quantization unit (*qdu*) for this model. Since the Talker Echo Path Loss (*TELR)* is the default 65 dB and the Weighted Echo Path Loss (*WEPL)* is the default 110 dB, the impairments due to echo (*Idte* and *Idle)* are not significant (typically < 3). The impairment due to delay (*Idd)* is the primary component that affects the model as implemented for this project. It is the impairment due to absolute delay (even with perfect echo cancellation). In our model, we assume that the absolute delay in echo free conditions (*Ta)* is always 100 ms or more. Also, the Advantage (*A)* is assumed to be equal to zero. One portion of the Simultaneous Impairment factor (*Is*) is influenced by round trip delay. Since we assume that high quality echo suppression devices are used, this impairment is very minimal for all cases. However, the full set of equations for the *Is* parameter are implemented in this model. Due to the complexity of the E-Model, the approach used here is to try to identify which E-Model parameters are fixed and which parameters are not. In the context of this research the only parameters of the E-Model that are not fixed are:

- *T, Ta*, and *Tr* – Delay variables
- *Ie* – Equipment Impairment

Where *T* is the mean one way delay of the echo path, *Ta* is the absolute delay in echo free conditions, and *Tr* is the round trip delay in a 4-wire loop [6]. In addition, parameters that affect delay and *Ie* are introduced.

- *PL* - Packet Loss %
- ρ - Link Utilization
- Coder Type

Next, the relationship between these parameters are identified. Since, we are making the assumption that the echo cancellers are on the end and are very good, we can say that $T = Ta = (1/2)\ Tr$. The relationship between *Ie* and *T* is not as straightforward. Since *Ie* is directly related to a particular coding scheme, information can be derived from that coding scheme that can relate T to Ie. For example, each coding scheme has a coding delay, bandwidth, and MTU size. Table 6.1 shows the parameters used for this research for selected coding schemes.

Parameter	Voice Packet Length (ms)	Frame Length (ms)	Rate (b/s)	Ie Value no packet loss)	MTU Size With Overhead (bits)	Code Delay (ms)
G.711	.125	20	85,600	0	1712	20
G.726-32	.125	20	53,600	7	1072	20
G.726-16	.125	20	37,600	50	752	20
G.729A	10	20	29,600	11	592	35
G.723.1	30	30	20,800	15	624	67.5

Table 6.1 – Coding Parameters

A couple of assumptions were made to arrive at these parameters. First, G.711 and G.726 are assumed to have a 20ms packet length [3]. Second, G.729A is assumed to have two 10 ms frames to form a 20 ms packet. And G.723.1 is assumed to have a single 30 ms frame to form a 30 ms packet. The Code Delay was taken from [3], which is shown in Equation 6.1, which only applies to high speed links (link speed > coder rate) [3].

$$\textit{Total Delay} = (N+1) * \textit{frame length} + \textit{code look-ahead} \qquad 6.1$$

N is the number of frames per packet. The look-ahead delay is a code specific amount of time that the code must look forward prior to coding the current samples. In addition, increasing utilization and decreasing packet loss (by increasing buffer size) increases delay. To address the delay caused by utilization and packet loss, the M/M/1 queuing function was used to establish variable delay on a link as a function of utilization and packet loss. This allowed increased traffic at increased delay costs. Packet loss is assumed to be the result of delays associated with the tail of a distribution and therefore should be a part of the delay equation. The following set of equations is used to calculate delay. In Equation 6.2, $S(X)$ is the distribution of total delay of the M/M/1 queue [59], μ is the service rate, and ρ is the link utilization. Equation 6.3 is derived from Equation 6.2. *Td* represents delay for a given packet loss, utilization, and service rate. PL% is representative of packet loss percentage.

$$S(X) = 1 - e^{-\mu(1-\rho)T_d} \qquad 6.2$$

$$Td = \frac{\ln(PL\%)}{-\mu(1-\rho)} \qquad 6.3$$

$$\text{T} = \text{Hop Count} * \text{Td} + \text{Code Delay} + \text{Propogation Delay} + \text{Misl.Delay} \qquad 6.4$$

The total one way delay is represented as *T* in Equation 6.4. All cases that were run assumed a 5 hop system. We assumed that the propagation delay is 25 ms and miscellaneous delays are 6 ms. Miscellaneous delays include switching delays and delays caused by echo cancellers. The code delay is dependant on the coding algorithm and is shown in Table 6.1. The approach used to set up this optimization is to define a "set" that includes the items that are varied. For example, in Case 1, the set includes the coders. For Case 2, the set is the combination of coders and packet loss percentages. For Case 3, the set include the combination of coders and possible utilization levels. This allows the algorithm to search the "universe" of possibilities and define its working set based on the constraints. The optimization problem is set up as follows.

- The *Set* of system configurations is defined.
- The parameters are calculated. This includes all E-Model parameters with fixed inputs and variable inputs based on the combinations in the *Set*.
- The objective is to maximize the number of calls on a link.
- The first constraint is that the minimum *R* value (voice quality) is 70.
- The second constraint is that the sum of the variable *Portion* is equal to 1.0.

This yields the following optimization algorithm (shown for Case 1) [59]:

Set: CODE;
Parameters:
T {CODE}, E-Model Parameters, Ie {CODE}, MTU {CODE}, Fixed Data Delay,
Calculation of E-Model Parameters,
Variables:

Portion {CODE}
Code_Feas {CODE} binary
Objective:
Maximize Calls: sum {i in CODE}
*(Code_Feas[i]*portion[i]*LinkBW*util/Rate[i])*
Subject to: #1
*Minimum R {i in CODE}: Ro – (Id[i] + Is[i] + Ie[i])*Code_Feas[i] + A*
>= 70;
Subject to: #2
Total Code: sum {i in CODE} portion [i] = 1;

Since the number of calls will be maximized with one of the *Set* combinations, this problem can be considered an "assignment" type optimization. The variable *Portion* is used to assign the calls to a particular combination in the *Set*. Strict assignment would require *Portion* to be a binary integer (1 or 0). To avoid this non-linearity, the integer requirement was relaxed and the program was allowed to make fractional assignments. The *Code_Feas* variable is a binary variable that penalizes coders that do not meet the constraints and limits the working set to $R > 70$. If the algorithm is attempting to make the minimum R constraint work and is having a problem, the only variable that it has at its disposal is the variable *Code_Feas*. The algorithm simply switches *Code_Feas* for that coder from a 1 to a 0 which eliminates the impairment portion of the equation and satisfies the constraint. But, by setting *Code_Feas* to zero for that coder, it eliminates it from participating in the objective. It removes that coder from its working set. During the first attempt, the optimization would not attempt to set the *Code_Feas* to "1" on all variables. Being non-linear, the algorithm found one coder that met the constraints and did not look for others that could produce a better objective function. This problem was solved by setting all *Code_Feas* variables to "1" during program initialization. For the algorithm to meet the $R > 70$ constraint, it must look at all *Code_Feas* variables and reverse them if necessary. If we wrote out the equations for Case 1 (with the link bitrate equal to1.5 Mbps), calling *Code_Feas 'c'* and *Portion 'p'* we have (note: the values for *Is* and *Id, Ro* came from [58]):

Set Code (c1, c2, c3, c4, c5)
s.t. 3.64c1 + 23.21c2 + 53.27c3 + 10.39c4 + 14.31c5 <= 24.75 #1
s.t. p1 + p2 + p3 + p4 + p5 = 1.0 #2
s.t. p >= 0; 1 >= c >= 0 (and it is binary)
Maximize 1.53(c1)(p1) + 2.5(c2)(p2) + 3.64(c3)(p3) + 4.7(c4)(p4) + 6.7(c5)(p5);

The code implemented for the optimization of the E – Model was modified from [60]. Details of the optimization algorithm can be found in [60]. This optimization algoritm was used in order to simplify calculations. The following section shows the results of the optimization tests.

6.2 Results

The results are divided into three general cases. For all cases, the objective function is to maximize the number of calls that can be carried on a link while maintaining a minimum voice quality level ($R > 70$).

- Case 1 - Optimizing for Coder Selection
- Case 2 - Optimizing for Coder and Packet Loss Level Selection
- Case 3 - Optimizing for Coder and Link Utilization Level Selection

6.2.1 Case 1 - Optimizing for Coder Selection

Case 1 was run for two different link speeds: 256 kbps and 1.5 Mbps. When the link speed was 256 kbps, the objective returned was 4.4 calls. G.729A was the coder chosen by the optimization. We can see from Figure 6.1 that G.729A was the only coder that was feasible. The data for Figure 6.1 was generated by the optimization program.

R Value, Calls vs. Coder - Link BW = 256 kbps

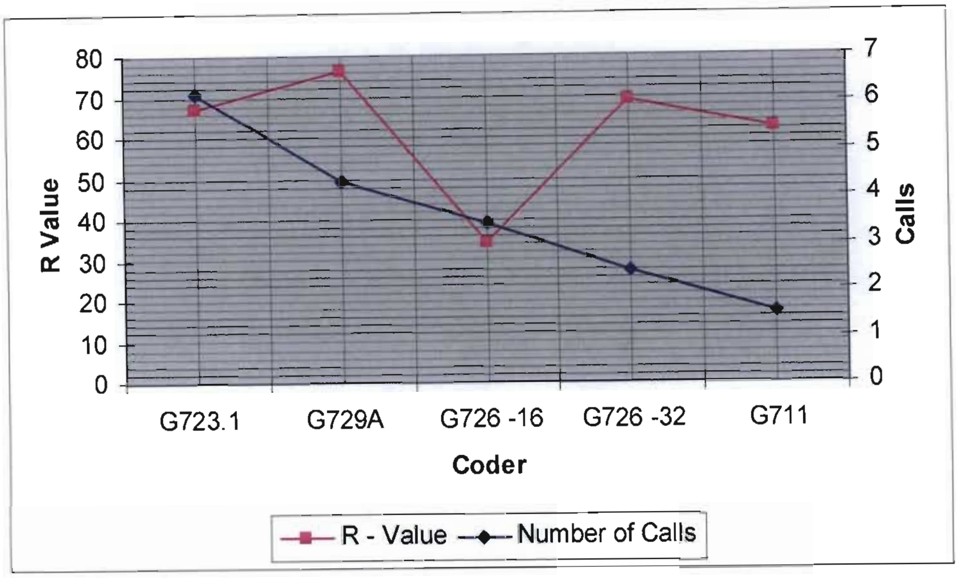

Figure 6.1 – Case 1, Link Rate = 256 kbps

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	67	6.2	237
G729A	76	4.3	199
G726-16	34	3.4	217
G726-32	69	2.4	283
G711	62	1.5	415

Link Bit rate	Link Utilization	Packet Loss Level
256000	0.5	0.005

Table 6.2 - Case 1, Link Rate = 256 kbps

For Case 1 with a link speed of 1.5 Mbps, G.723.1 was selected as the optimum coder by the optimization routine. The objective function found 38 calls. Figure 6.2 contains data collected from the optimization program. It clearly shows that the optimization selected the correct coder. The results of this case are not surprising, as G.723.1 is more efficient but with a lower quality of voice. The significant decline in the *R* value of G.726-16 as compared to the other coding schemes demonstrates the weakness of

ADPCM compared to other low bit rate coding schemes like MP-MLQ (used by G.723.1 - 6.3 kbps) and CS-ACELP (used by G.729).

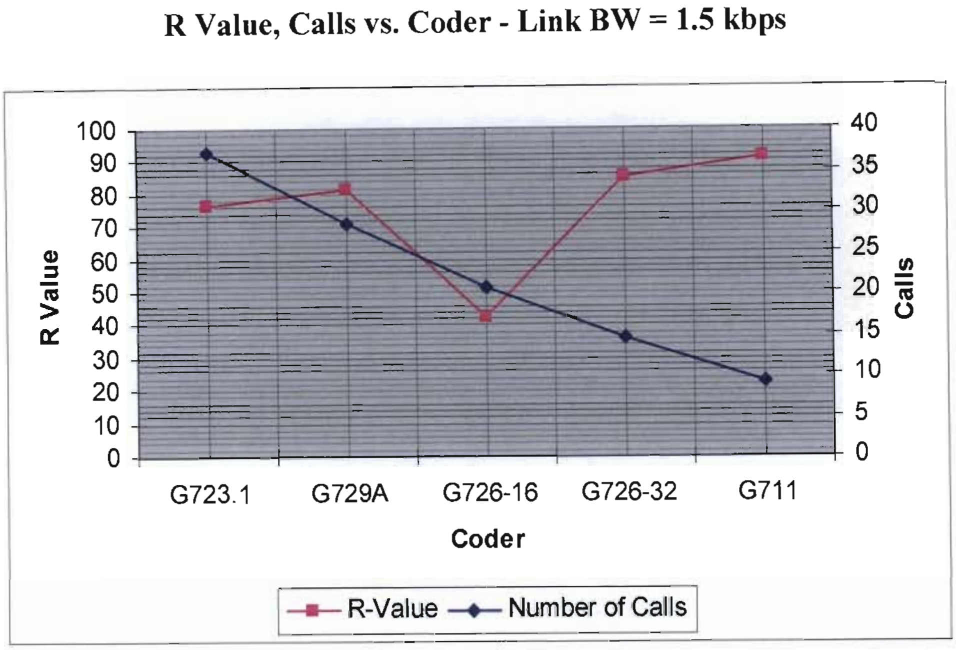

Figure 6.2 – Case 1, Link Rate = 1.5 Mbps

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	76	37.1	122
G729A	81	28.4	88
G726-16	42	20.5	78
G726-32	85	14.4	89
G711	91	9	111

Link Bit rate	Link Utilization	Packet Loss Level
1500000	0.5	0.005

Table 6.3 - Case 1, Link Rate = 1.5 Mbps

An interesting observation is that the results generated for both the *R* value and numbers of calls are similar except for the scale. Even though the results are similar, the G.723.1 coder came into a feasible position on the 1.5 Mbps test.

6.2.2 Case 2 – Optimizing for Coder and Packet Loss

Case 2 required additional analysis because not all of the packet loss percentages have been tested and recorded (see table 1.4). A polynomial fit was completed for each of the three coders missing *Ie* values. The effective equipment impairment factor was then calculated using the following formula [57].

$$Ie_{eff} = Ie + (95 - Ie) * \frac{P_{pl}}{P_{pl} + B_{pl}} \qquad 6.5$$

where I_e = *Equipment impairment factor from* [57]

P_{pl} = *Packet loss percentage*

B_{pl} = *Packet loss Robustness factor*

Specific impairment factor values for codec operation under packet loss have formerly been treated using tabulated, packet loss dependant *Ie* – values. Now the packet loss Robustness factor B_{pl} is defined as codec specific value. The packet – loss dependant Effective Equipment factor Ie_{eff} is derived using codec specific value for the Equipment Impairment factor at zero packet loss *Ie* and the Packet loss Robustness factor B_{pl}, both listed in ITU recommendation G.113 for several codec's. With the Packet loss probability P_{pl}, Ie_{eff} is calculated using formula 6.5 [6].

Case 2 was then run for two different link bandwidths. First it was run with a link bandwidth of 256 kbps. It was then run with a link bandwidth of 1.5 Mbps. For the 256 kbps link speed, the objective returned was 4.3. G.729A with packet loss of 2% was the combination chosen. There were only two coder/packet loss % combinations that were feasible.

R Value vs. Packet Loss - Link BW = 256 kbps

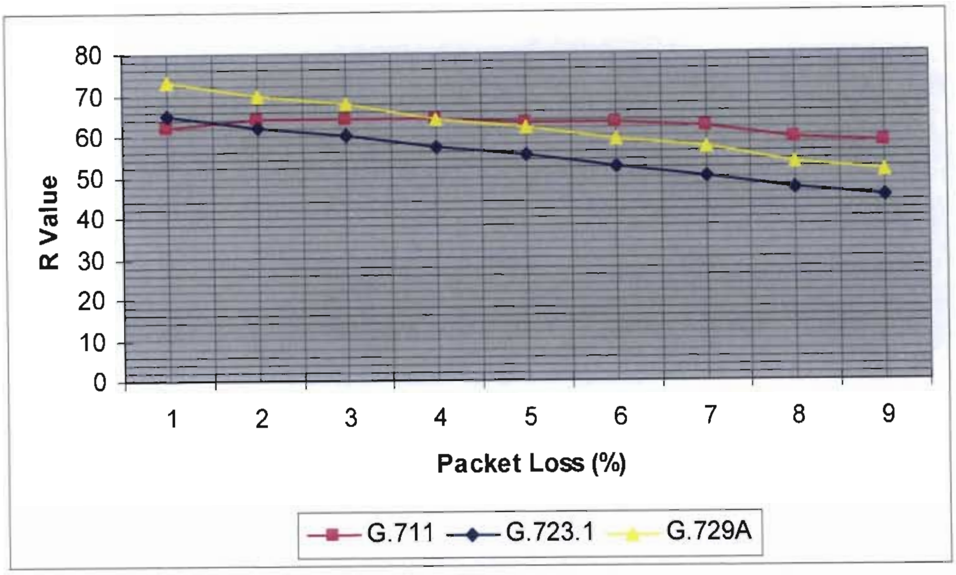

Figure 6.3 – Case 2, Link Rate = 256 kbps

Packet Loss %	R - Value		
	G.711	G.723.1	G.729A
1	62	65	73
2	64	62	70
3	64	60	68
4	64	57	64
5	63	55	62
6	63	52	59
7	62	50	57
8	59	47	53
9	58	45	51

Number of Calls	Link Bit rate	Link Utilization Level
1.5 (G.711)	250000	0.5
6.15 (G.723)		
4.3 (G.729)		

Table 6.4 - Case 2, Link Rate = 256 kbps

Next, the test was run with a link speed of 1.5 Mbps. The objective returned was 37.1. G.723.1 with packet loss of 1% was the combination chosen. Looking at Figure 6.7, we can see that G.711 and G.729A were feasible at least a portion of the packet loss spectrum.

R Value vs Packet Loss - Link BW = 1.5 Mbps

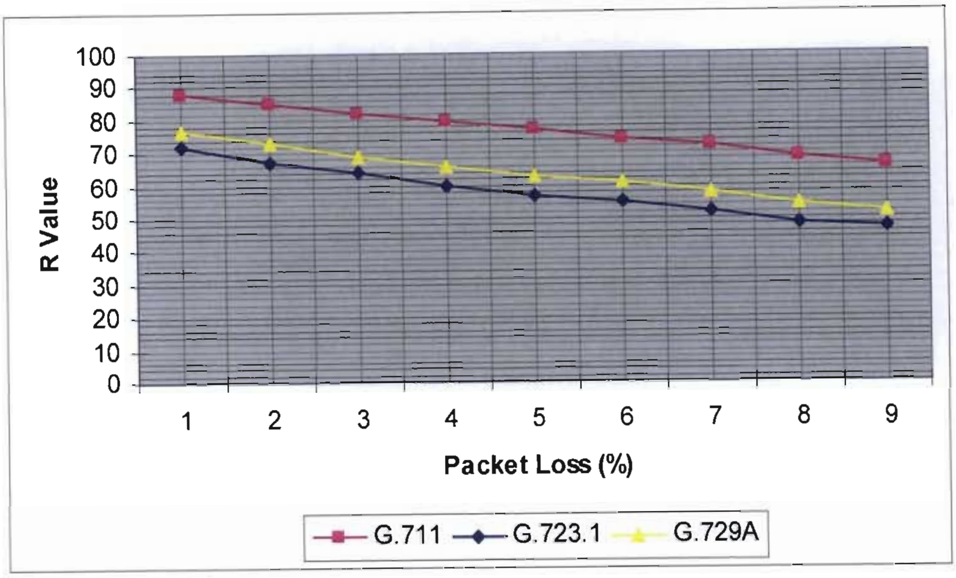

Figure 6.4 – Case 2, Link Rate = 1.5 Mbps

Packet Loss %	R - Value		
	G.711	G.723.1	G.729A
1	88	72	77
2	85	67	73
3	82	64	69
4	79	60	66
5	77	57	63
6	74	55	61
7	72	52	58
8	68	48	54
9	66	47	52

Number of Calls	Link Bit rate	Link Utilization Level
9.0 (G.711)	1500000	0
37.1 (G.723)		
26.1 (G.729A)		

Table 6.5 - Case 2, Link Rate = 1.5 Mbps

The algorithm for Case 2 is interesting because it has a modification that allows it to favour higher packet losses (given the choice). The modification is accomplished by adding a small benefit to the objective function (*+portion [i, j]*Packet Loss[j]*). This benefit consists of the packet loss level multiplied by the portion assigned to that coder. For example, if a coder is feasible for 1% to 4% packet loss, the objective for the 4% packet loss will be slightly higher than the 1% objective. This was done to bias the optimal solution toward a minimum requirement for a link (rather than the most stringent requirements). This modification was demonstrated when the G.729 coder with 2% packet loss was chosen over the G.729 coder with 1% packet loss (Case 2 with the link speed equal to 256 kbps). After reviewing the data from this test, it is apparent that the degradation from packet loss to the audio (via the *Ie* factor) typically outweighs any gains that may be derived from reductions in delay (at least using M/M/1 assumptions). This is due to the relatively steep *Ie* curves.

6.2.3 Case 3 – Optimizing for Coder and Link Utilization

Case 3 was run for two cases. First it was run with a link bandwidth of 256 kbps. The objective returned was 5.6. G.729A running with 60% link utilization was the combination chosen. Looking at Figure 6.5, we can see that the G.729A coder was in a feasible range till link utilization reached approximately 62%.

R, Calls vs. Link Utilization (Voice) - Link BW = 256 kbps

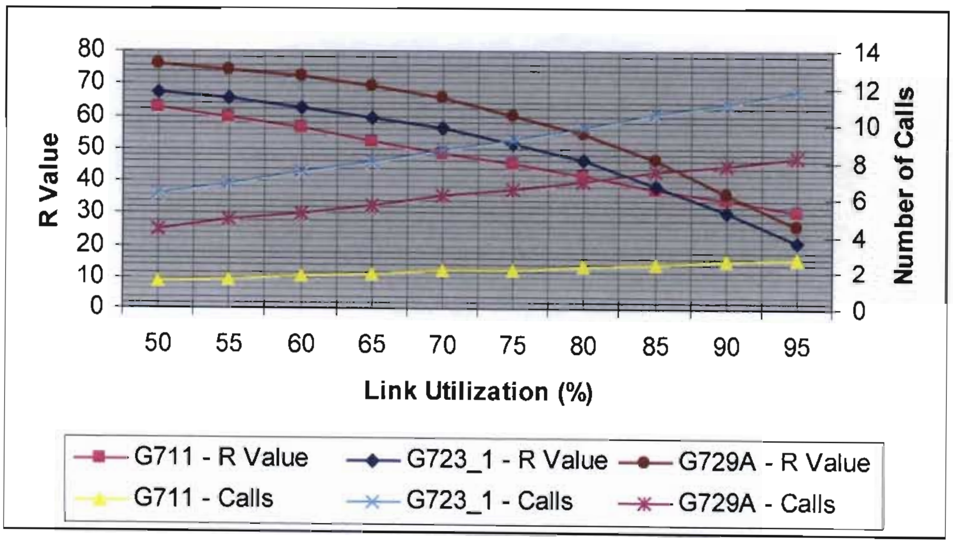

Figure 6.5 – Case 3, Link Rate = 256 kbps

Link Utilization %	G711 - R Value	G723_1 - R Value	G729A - R Value	G711 - Calls	G723_1 - Calls	G729A - Calls
50	62	67	76	1.5	6.2	4.3
55	59	65	74	1.6	6.8	4.8
60	56	62	72	1.8	7.4	5.2
65	52	59	69	1.9	8.0	5.6
70	48	56	65	2.1	8.6	6.1
75	45	51	60	2.2	9.2	6.5
80	41	46	54	2.4	9.8	6.9
85	37	38	46	2.5	10.5	7.4
90	34	30	36	2.7	11.1	7.8
95	30	21	26	2.8	11.7	8.2

Link Bit rate	Packet Loss Level
256000	0.005

Table 6.6 - Case 3, Link Rate = 256 kbps

R Value, Number of Calls vs. Link Utilization - Link BW = 1.5 Mbps

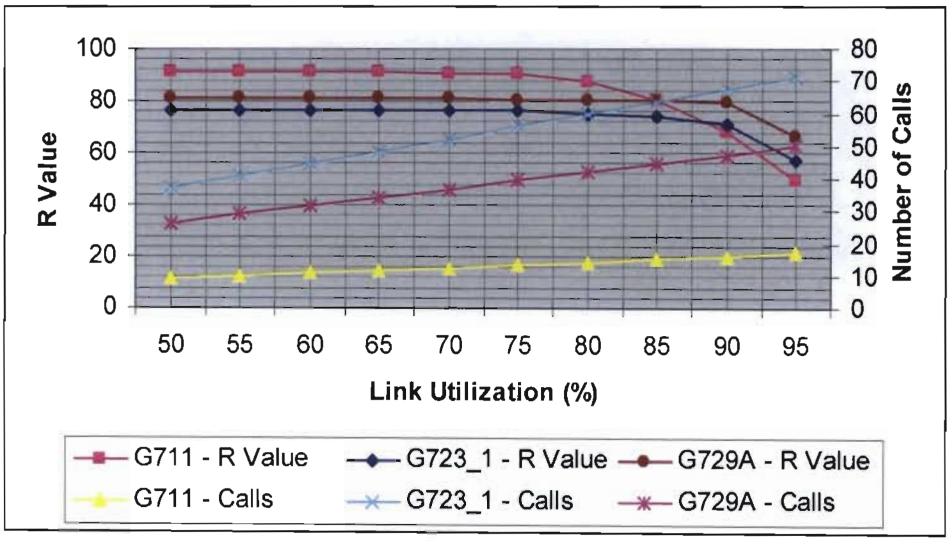

Figure 6.6 – Case 3, Link Rate = 1.5 Mbps

Link Utilization %	G711 - R Value	G723_1 - R Value	G729A - R Value	G711 - Calls	G723_1 - Calls	G729A - Calls
50	91	76	81	9.0	37.1	26.1
55	91	76	81	9.9	40.8	28.7
60	91	76	81	10.8	44.5	31.3
65	91	76	81	11.7	48.3	33.9
70	90	76	81	12.6	52.0	36.5
75	90	76	80	13.5	55.7	39.1
80	87	75	80	14.4	59.4	41.7
85	80	74	80	15.3	63.1	44.3
90	68	71	79	16.2	66.8	46.9
95	49	57	66	17.1	70.5	49.6

Link Bit rate	Packet Loss Level
1500000	0.005

Table 6.7 - Case 3, Link Rate = 1.5 Mbps

For Case 3 with a link bit rate of 1.5 Mbps, the objective returned was 66.8. G.723.1 running with 90% link utilization was the combination chosen. Looking at Figure 6.9, we can see that all three coders were in a feasible range till link utilization reached approximately 85%. An important observation can be made here. G.723.1 utilizes bandwidth better than the other coders here and although it has high overhead and coder delay, with enough bandwidth, it is still the best choice for this application. But also note that very little change in the downward direction of G.723.1 would have brought it completely out of feasibility.

6.3 Discussion of the Optimization Results

This chapter utilized the E-Model to assist with the selection of the QoS parameters important to VoIP networks. These include the voice coder, allowable packet loss, and allowable delay. It was based on the concept that maximization of the link usage with respect to the number of calls is important to the user. An important finding of this research is that an optimization of the E-Model is possible and useful. The figures and tables in this chapter illustrate that the optimization chose the correct combination every time. Table 6.8 reviews some of the results of the optimization problem.

Case Number	Variables	Link Bit rate (b/s)	Optimum Solution
1	Coder	256000	G.729A
1	Coder	1500000	G.723.1
2	Coder, Packet Loss %	256000	G.729A with 2% PL
2	Coder, Packet Loss %	1500000	G.723.1 with 1% PL
3	Coder, Link Utilization	256000	G.729A with 60% Load
3	Coder, Link Utilization	1500000	G.723.1 with 90% Load

Table 6.8 - Results of E-Model Optimization

All three cases found that G.729A and G.723.1 both are optimal depending on the circumstances. G.729A was a better coder with lower bandwidth links. This is because the quality of the voice using G.729A is relatively good and the delay imposed by the coder is not as severe as G.723.1. However, as the bandwidth increases, G.723.1 looks more favourable due to the fact that G.723.1 uses less bandwidth per audio stream. Given the objective of maximizing calls subject to a minimum voice quality level, G.723.1 becomes the more optimal coder at higher bandwidth levels. Case 2 found that G.729.A with 2% loss was optimal when the link bit rate was 256 kbps. When the link bit rate was 1.5 Mbps, G.723.1 with 1% packet loss was optimal. The relevant finding from Case 2 was that packet loss typically hurt the voice quality more than the benefit derived from smaller packet delay. Case 3 with a link bit rate of 256 kbps found that G.729A was the better coder and that the maximum possible voice load is approximately 60%.

For the higher link rate of 1.5 Mbps, the G.723.1 coder was selected. This is significant because the load chosen was 90%. G.729A at 95% load did not deliver as many calls. In all tests, a delay was assumed. Since the optimization always drove the *R* Value down to 70, the delay that was calculated using the addition of coding delays and network delays is the delay bound allowable. This model was tested with default values producing an *R* value of 93.2 (the correct answer). This helped to validate the equations used in this model. The ability to analyze various coders, delay and packets loss is vital to assessing voice quality in networks. The E-Model provides a tool that is useful for this purpose. Chapter 7 looks at delay issues in VoIP networks.

Chapter 7: Delay in VoIP networks

7.1 Objective

The objective of the following simulation study is to look at the end-to-end delay experienced by a voice over IP network. In particular we will investigate and compare the delays experienced across both a Remote Network and a Backbone Network. The goal for this simulation study is to determine if it is possible to meet the criteria for delay in a typical voice Network shown.

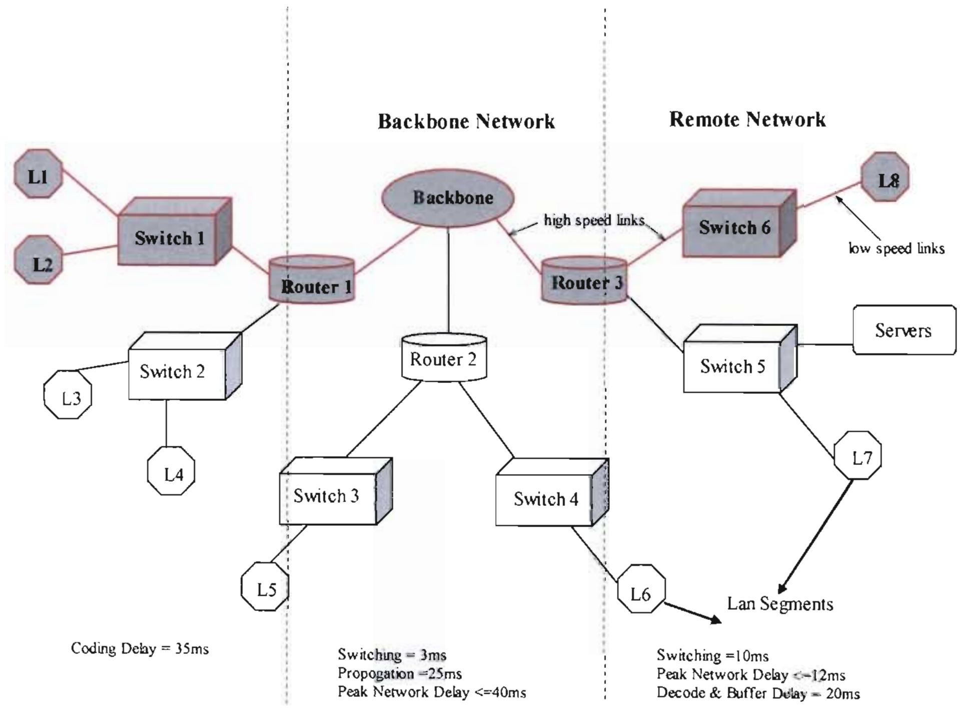

Figure 7.1 - Delay Analysis of a Voice Network

The data used for typical delay times was derived from recommendations in [3] for a network using G.729 coding. This simulation was designed with a 145ms peak delay as the goal for a single hop edge network and a three hop backbone network. The software that was used to implement the simulation models was OPNET IT Guru™ and OPNET Modeller™ by Opnet Technologies, Inc [65].

7.2 Simulation Models

The simulation model for the Remote Network is shown in Figure 7.2. The Model simulates an edge VoIP router that has 6 voice inputs and 2 data inputs. This is a possible network design for a network that has remote facilities. The Remote model focuses on delays associated with the low bit rate links leaving the edge router. Differentiated service is used as the mechanism to implement priority and deficit round robin queuing. All packets are marked with a Diffserv Code Point (DSCP) that the routers can use to classify the packet. For voice, the packets are marked as EF (expedited forwarded) packets. For data, the packets are marked as best effort (BE). The intent of the Remote Network simulation is to analyze delays associated with slow links that connect the remote network with the backbone network.

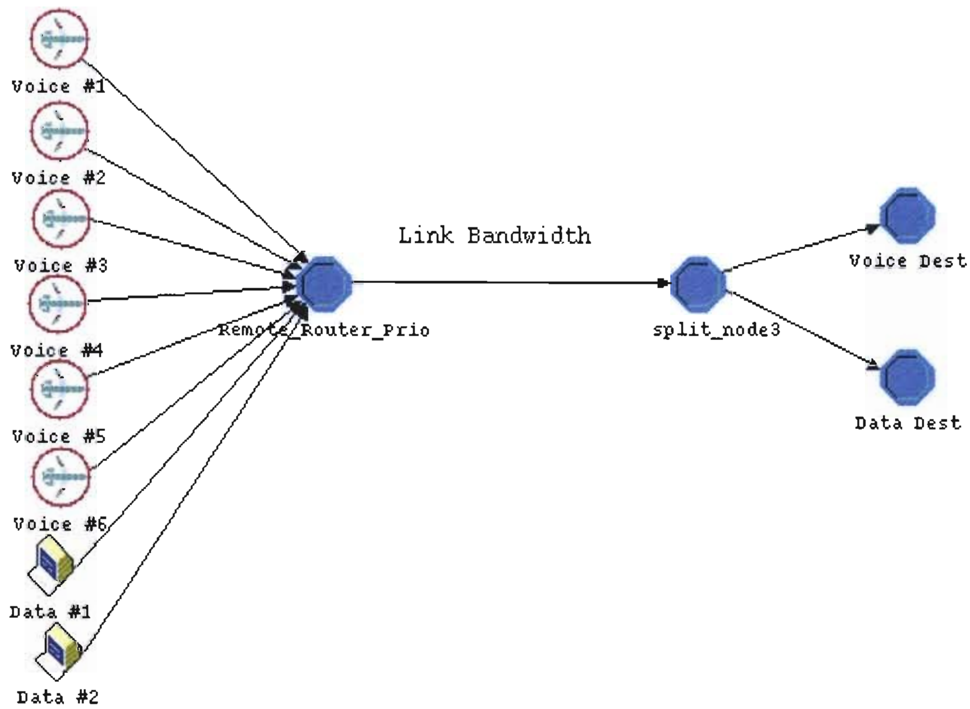

Figure 7.2 - Remote (Edge) VoIP Model (Priority Queue)

The goal of the Backbone Network Simulation is to analyze delays associated with the core of the network and determine if it is possible to meet the 40ms delay criteria. This model includes many voice and data sources and high speed links that end on low speed links. Figure 7.3 shows the Network diagram used to simulate a backbone VoIP network. To simulate the Backbone network, routers were modelled that can pass traffic out of a second output port based on the destination address of the IP packet. This network had a total of 1280 voice sources and 32 best effort data sources. The voice and data sources are the same as in the Remote Network model (fig 7.2). The average inter arrival times and file sizes are exponentially distributed unless otherwise specified. The DSCP is used by the routers to determine the scheduling. Referring to figure 7.3, we can see that each ingress router has 40 voice sources and one data source that feeds the router. At each core router, there are a total of 8 x 40 = 320 voice sources feeding each core router. Half of the traffic is passed out of the network at that core router. An average of 50% of the data packets leaves the network at each core router. This leaves 640 voice sources and half of the data traffic entering the egress router. One voice source in each group of 40 is sent to the main destination i.e. 32 voice sources. The remaining 608 voice sources are passed out of the network at the egress router.

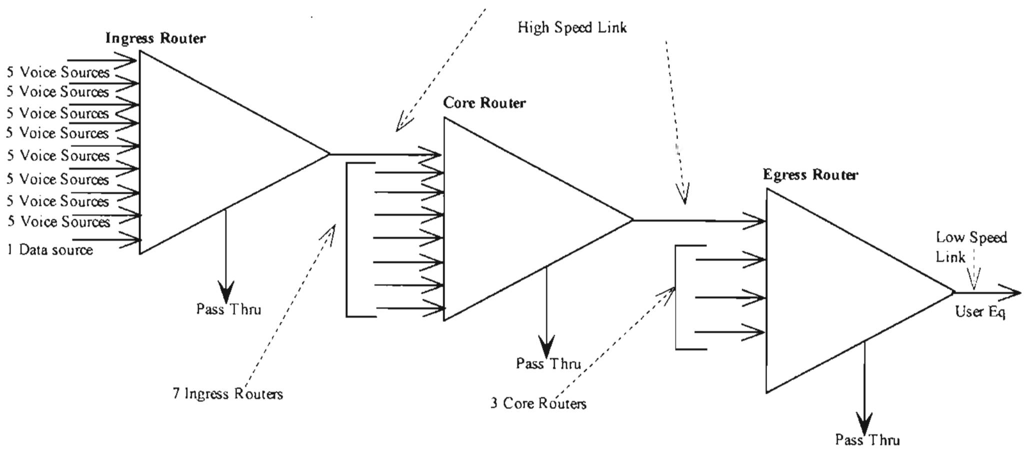

Figure 7.3 - Backbone Network Model (Variable Link Speeds and Data Sources)

Appendix 1 contains more information about the components of the Models in addition to validation data. The next section contains the results of the simulations. The overall load, load of voice versus data, queuing mechanisms and data source distribution was varied to provide the different scenarios.

7.2.1 Case 1 – Effect of Link Rate in VoIP Networks

This case varies the rate of the link from 235,8kbps to 314,45kbps for the Remote Network. As the link rate increased the total load on the link decreased from 89% to 67%. This also had the effect of varying the load of voice traffic and data traffic. Table 7.1 shows the voice and data load with varying link rates. Figure 7.4 shows the results for the Remote network. The interesting observation from this case is that the voice delay was relatively constant across the varying load levels, even when there was congestion on the link. There was a very slight increase in the average voice delay when the load was 0.89. Congestion occurred when the overall load was nearing .89, as can be seen by the rapid increase in data delays.

Link Rate (bps)	Total Load	Voice Load	Data Load
235,840	0.89	0.24	0.64
269,531	0.78	0.21	0.56
314,543	0.67	0.18	0.48

Table 7.1 - Remote case 1 Utilization Levels

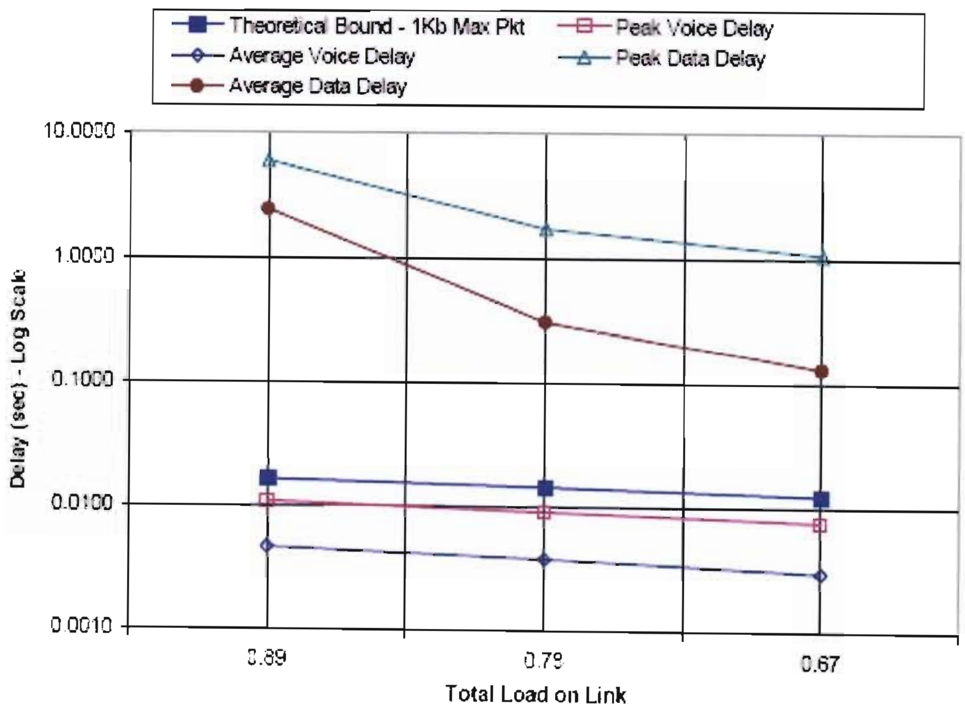

Figure 7.4: Case 1 – Remote VoIP Network

For the Backbone Network the voice load on all links was at 0.5 while the data loads were changed from .16 to.47. The overall load on the links varies from approximately 0.66 to 0.97. Figure 7.5 shows the delay observed in this case.

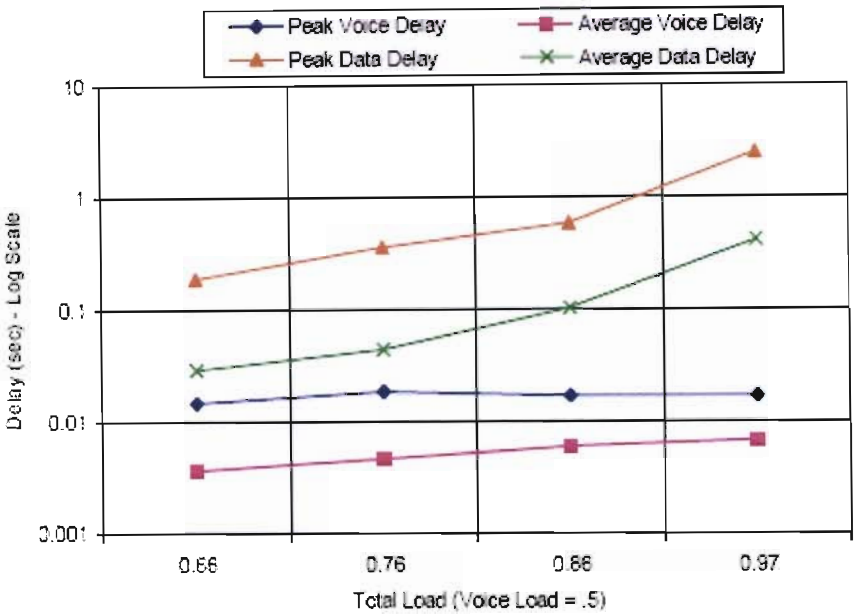

Figure 7.5: Case 1- Backbone VoIP Network

As can be seen from figure 7.5 the data delay climbs with increased data (and overall load). One interesting point is that the peak voice delay remains fairly constant at approximately the 0.76 load and maintains that level. The average voice delay does continue to climb. This can be logically explained. If there are more data packets, it is more probable that a voice packet or group of voice packets will have to wait on a data packet. Therefore the average voice delay should climb while the peak voice delay should not.

7.2.2 Case 2 – Effect of Voice/Data ratio in VoIP Networks

This case looks at the delay caused by different proportions of data loads to voice load. The total load was 0.84 for the Remote case. Table 7.2 shows the data, voice and total loads.

Link Rate (bps)	Data Interarrival Time	Total Load	Voice Load	Data Load
144,000	0.3203	0.80	0.40	0.40
192,000	0.1836	0.83	0.30	0.53
288,000	0.1016	0.83	0.20	0.63
576,000	0.043	0.85	0.10	0.75

Table 7.2 – Remote case 2 Utilization levels

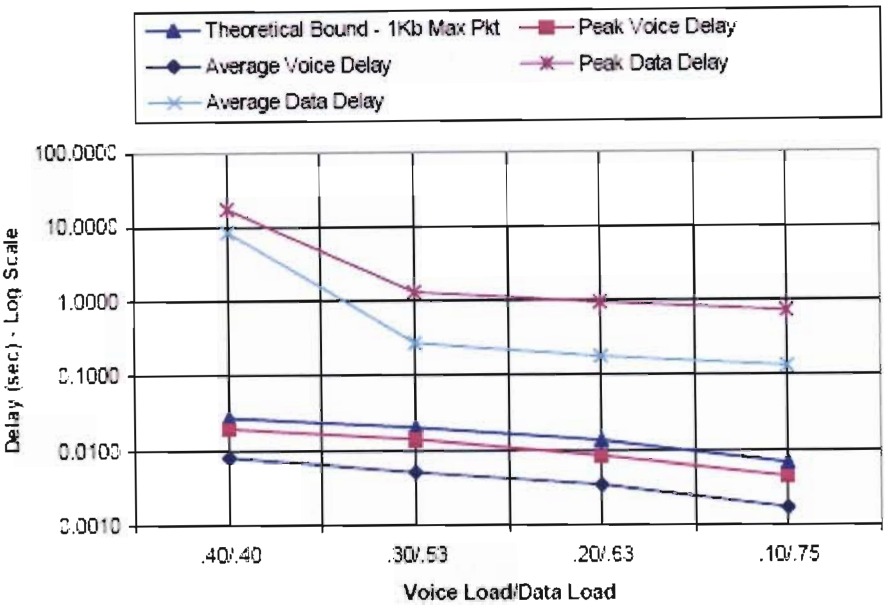

Figure 7.6: Case2 – Remote VoIP Network

Figure 7.6 shows the results of the Remote case. When the voice utilization climbed above 30% the data delay suffered by a factor of 10. The reason for this can be explained by the fact that for the .40/.40 case, if all voices are "on", the voice sources would saturate the link (6 x 2400 = 144kbps). A 20% voice load is chosen to be the standard percentage of the load for voice in order to remain under the 12ms delay for the Remote VoIP Network.

For the Backbone Network a constant total load of (.85 to .88) was maintained and the portion of load that is voice to the portion of load that is data was varied. Table 7.3 shows the load configuration. Figure 7.7 shows the results for the Backbone Network.

Voice Load	Data Load	Total Load
.10	.78	.88
.30	.57	.87
.50	.36	.86
.70	.16	.86

Table 7.3: Backbone Case 2 Utilization levels

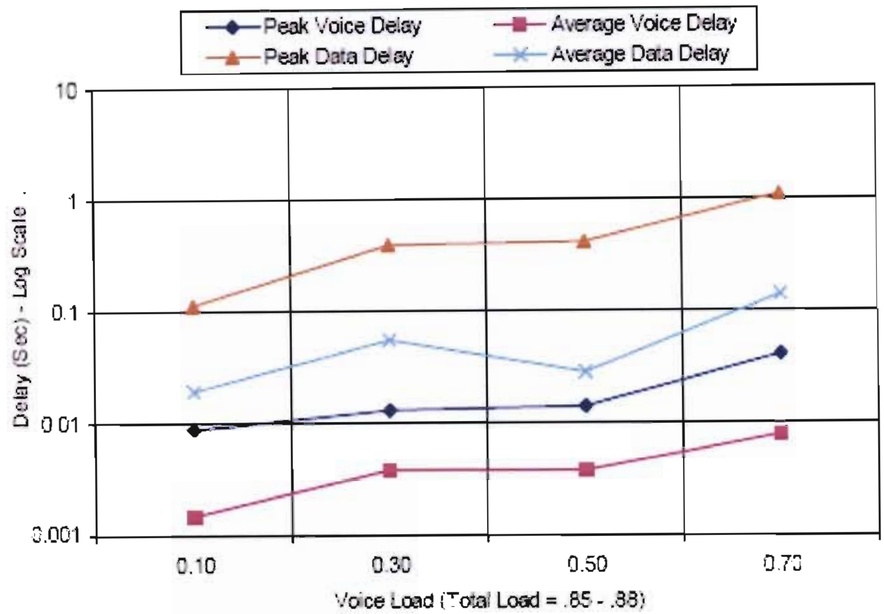

Figure 7.7: Case 2 – Backbone VoIP Network

The data packet delays tend to track with voice delays. This makes sense because the voice packets have priority , so if they suffer from queuing delays, then the data packets will be impacted from the same delays. The delays for voice and data levelled of or decreased when the voice load was 50% and data load 36%. A possible reason is that since the data load is decreasing and voice load increasing, this point has low enough voice load that the voice traffic is not starving the data and therefore we see a decrease in data delay.

7.2.3 Case 3 – Deficit Round Robin queuing in VoIP networks

This case looks at the delay in VoIP Networks when a DRR queue is used instead of a priority queue. Queue weights were assigned to voice and to data. In DRR queues, the quantum indicates the number of bits each class is given each round. If a class does not have enough traffic to use its quantum or weight, that quantum is given to other classes. Appendix 1 contains a description of the DRR queue as implemented in the Remote case. For example 95/5 indicates that the voice weight is 95 bits and data weight is 5 bits. The following weight combinations were tested: 95/5, 80/20, 65/35, 50/50, 35/65, 20/80, 5/95. The link speed was 288kbps for the remote case and the voice load was 0.2 while the data load was 0.65. Figure 7.8 shows the results in the remote case.

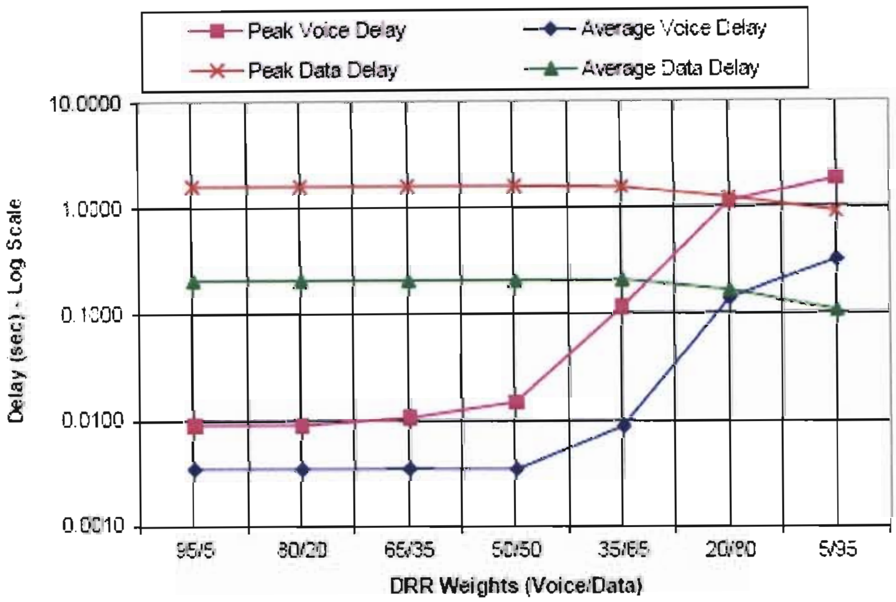

Figure 7.8: Case 3 – Remote VoIP Network

As can be seen the voice delay begins to climb very rapidly at the 50/50 weight setting. This can be explained by the fact that with a 288kbps link speed, 50% of the link is 144kbps, which is the bandwidth needed if all 6 voice sources are "on". This implies that during this time, the data traffic was utilizing all of its bandwidth. This is reasonable since the average data load is 0.65 of 288kbps or 188kbps. The voice sources can only use 144kbps (all sources on), so any bandwidth the voice traffic has above 50/50 will be given back to the data traffic. Therefore the data delay is relatively flat.

For the Backbone Network a constant voice and data load was queued with a DRR queue. The DRR weights for voice/data were 95/5, 80/20, 65/35, 50/50 and 35/65. The voice load was 0.5 while the data load was 0.36. Figure 7.9 shows the results. As can be seen the voice delays increased sharply between DRR weights of 65 and 50. Since voice traffic is 50% of the traffic on the link, if the bandwidth available to voice is reduced to near 50%, delays should rise sharply. The flatness of the delays below the 65/35 DRR weight load is related to the fact that there is only a limited amount of voice traffic possible. Any bandwidth above the level that corresponds to the weight given to the voice is given back to the data traffic which gives both the voice and data the same amount of bandwidth below 65/35 weighting level.

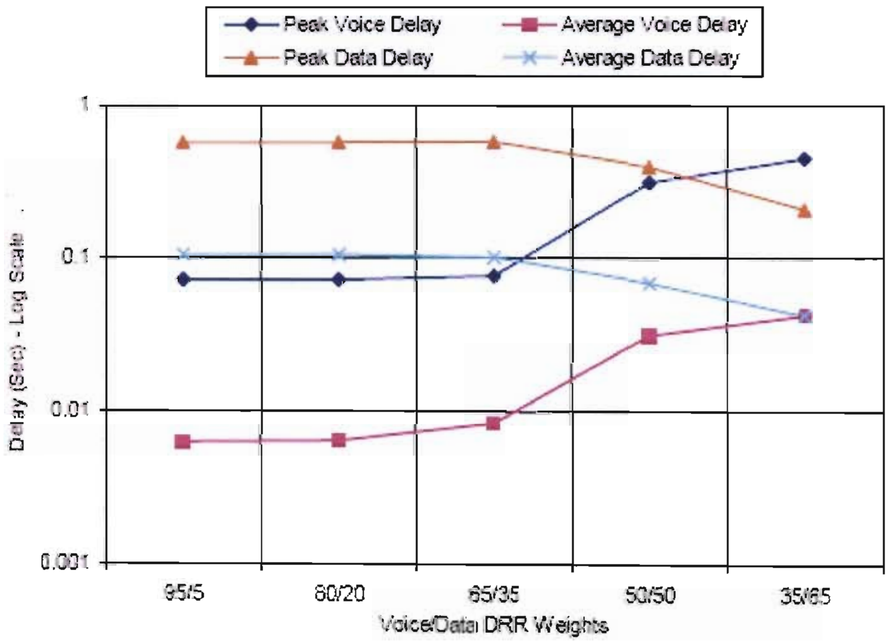

Figure 7.9: Case 3 – Backbone VoIP Network

7.2.4 Case 4 – DRR queuing with Pareto distribution in VoIP

This case is a repeat of case 3 using a Pareto distribution for data file size instead of the exponential distribution. Figure 7.10 and Figure 7.11 compares the data collected for the exponential(m/m/1) case compared to the Pareto case for the Remote Network

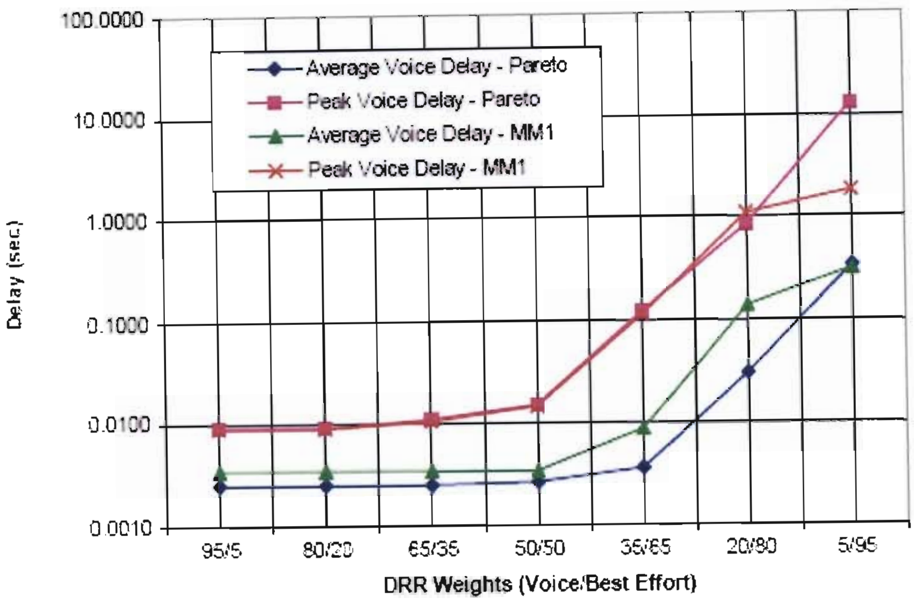

Figure 7.10: Case 4 – Remote VoIP Network – voice delay

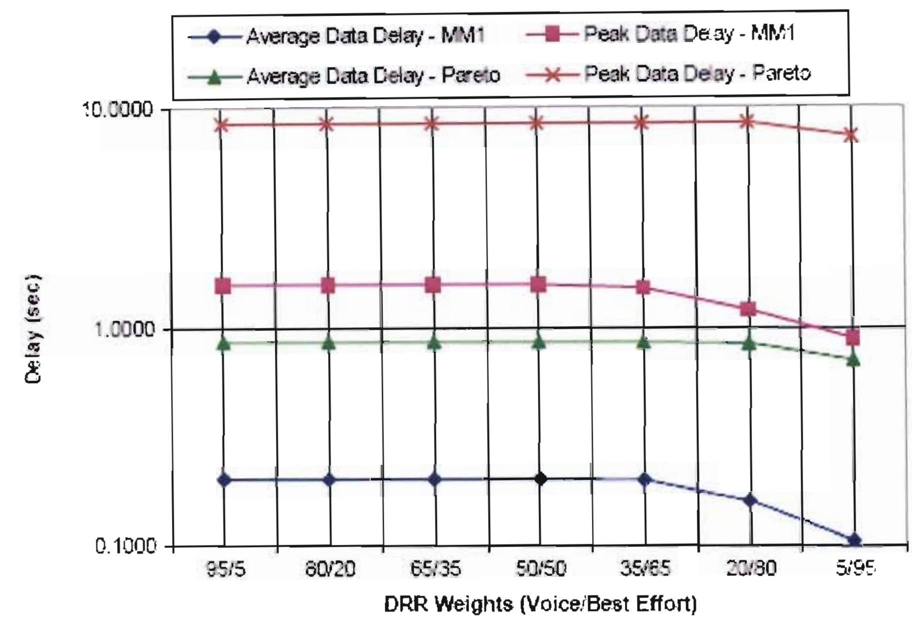

Figure 7.11: Case 4 – Remote VoIP Network – data delay

The following observations were made:

- The peak voice delay did not change significantly between the Pareto case and the exponential case except for the 5/95 case. When the voice traffic has the least amount of extra bandwidth (5/95) is when the ability of the pareto distribution to make a difference occurs.

- The average delay for voice was less with the Pareto case than the exponential case.
- The peak data delay and the average data delay were almost a magnitude greater with the pareto case as compared to the exponential case.

The results for the Backbone VoIP scenario is shown in figure 7.12 and figure 7.13. The following observations were made:

- The general shape of the remote scenario graphs shown above and the following graphs are similar. The rapid rise in voice delays are due to the fact that voice traffic encounters congestion.
- The voice delays, when the Pareto data traffic generator is used rises at a rapid rate when the voice weight is being reduced as opposed to the exponential (m/m/1) traffic generator which levels of slightly.

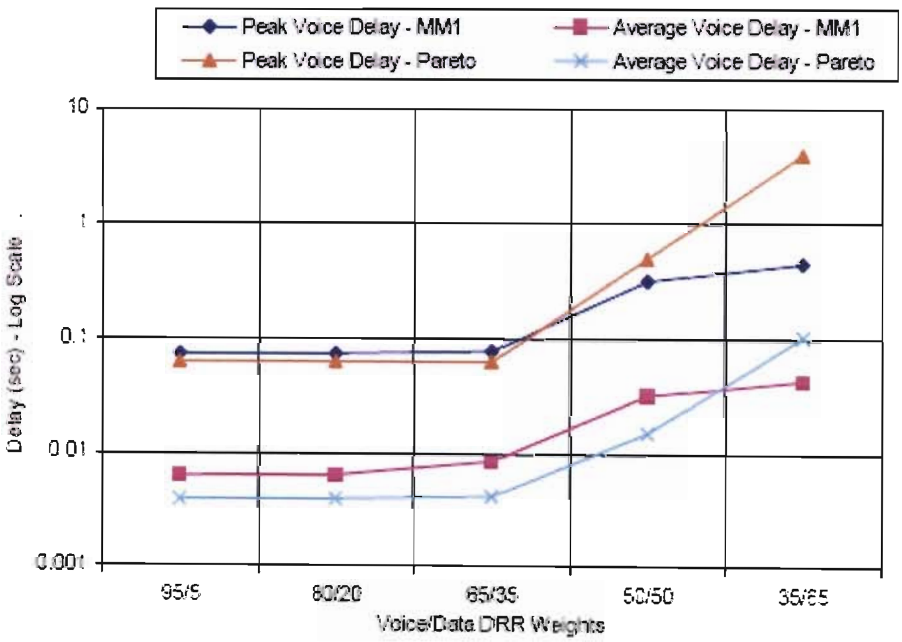

Figure 7.12: Case 4 – Backbone VoIP Network – voice delay

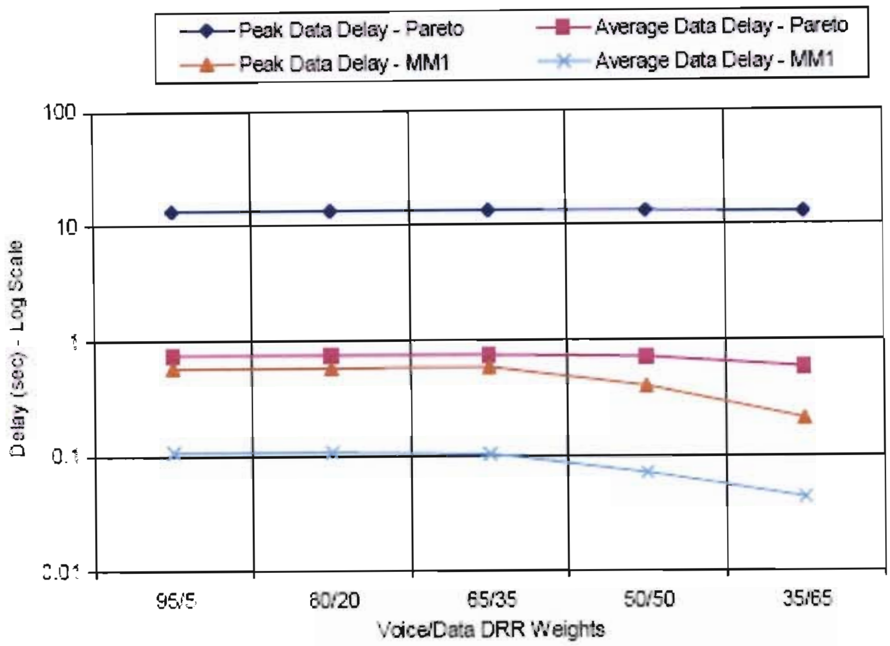

Figure 7.13: Case 4 – Backbone VoIP Network – data delay

7.3 Discussion of Delay in VoIP Networks

These simulations have led to a number of important observations and conclusions.

- The inter arrival time between voice packets (20ms) is important in the determination of load and the calculation of delay. In both the remote and backbone network we saw that if enough voice sources are transmitting, the peak voice delays can rise sharply.
- DRR routers have a place in VoIP networks because they can protect data as shown in Figure 7.8 and figure 7.9. Most of the quantum should be given to the voice source with an adequate amount given to the data so that it does nor starve and suffer higher delays.
- Figure 7.1 allowed 12 ms maximum delay in the Remote Network. In figure 7.10 we can see that with 65/35 DRR (voice/data) weighting or higher, delays did not exceed 12ms, but in figure 7.6 if voice utilization rises above 0.2 voice delay can climb beyond 12ms.
- Overall, it appears that it is possible to use IP to transport voice. It has to be reliant on strict control of the network and voice coding parameters(inter arrival time, overall utilization and voice utilization).

Chapter 8: Conclusions and Recommendations

8.1 Conclusions from the Analysis

This research investigated several aspects of QoS for voice over next generation networks. In chapter, one many key characteristics of NGN's were reviewed and the various voice quality parameters were discussed. In chapter two, voice over ATM was discussed. We found out that there is nothing strict about the assignment of AAL types to service classes for voice. Any service type can be transported on any adaptation type as long as the performance is acceptable to the voice user. However the proper AAL (i.e. AAL2) should be selected according to user requirements such as QoS and error performance. The selection of AAL may have impact on the quality of voice service over an ATM network.

Chapter 3 discussed the issues around the transmission of voice in IP networks. The various IP related standards and protocols were discussed including both the benefits and risk factors of VoIP. Over the years various QoS mechanisms have been added, which further enhance the quality of VoIP networks. At the moment there is no significant difference for NGN's based on VoIP or VoATM. Although ATM has inherent QoS capabilities, non QoS arguments are more in favour of IP than ATM. Firstly VoIP offers the best platform for an all IP service integration. Secondly IP is the most economical way to provide a backbone network for voice, multimedia and data services because the majority of traffic will anyhow be IP.

Chapter 4 looks at the various methods and means of enhancing QoS for voice over packet technologies. We looked at both Integrated Service and Differentiated Service architectures and evaluated both their benefits and drawbacks. We also looked at quality of service supporting mechanisms like Multiprotocol Label Switching. Efforts to enhance the QoS capabilities of VoIP will continue in the future in order to make it possible to transport real time voice across large networks like the internet.

An optimization of the E – Model was used in order to study the effects of end-to-end delay, utilization and coder design on voice quality. This optimization was used in order to choose a coder and utilization levels given certain conditions. This has significant application with VoIP network design. The E – Model assists with the selection of parameters important to VoIP networks. An optimization can correctly select combinations of these parameters and allow network resources to be used in an efficient manner. Simulations were also carried out in order to study the delay in VoIP networks. The delay requirements for both the remote and backbone network were also met. Different link utilization, voice utilization and queuing methods were studied in order to look at the feasibility of transporting VoIP.

There are many areas of research in voice communications that are not covered in this research. More accurate methods to model large VoIP networks are needed. Ways to extend the E – Model to include more variables and better estimates of network delay will help this model be more accurate in the future.

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Appendix 1

OPNET Model Description

A.1 Voice Model

The voice traffic source is an EF voice source that is modelled using an On-Off source with G.729 coding (20 bytes) plus 40 bytes for the IP/UDP/RTP header. These packets are delivered every 20 ms. The EF traffic is marked with a DiffServ Code Point (DSCP) of 101110 (46) which is recommended by the IETF [66] for EF traffic. The node model is shown in Figure A.1.

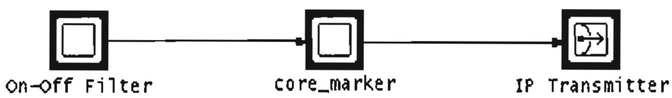

Figure A.1 - Node Model for Voice Packet Generator (VoIP_Gen4)

Using a mean "ON" time of .4 seconds and mean "OFF" time of .6 seconds, the average bit rate is 9600 bps. When the source is "ON", the bit rate is 24 kbps. The "ON" times and "OFF" times are exponentially distributed. This model was tested as a standalone model and generated packets when "ON". When "OFF", no packets are generated. Figure A.2 shows a graph of "ON" time versus "OFF" time. Value 1.0 indicated "ON" time and value 0.0 indicates "OFF" time.

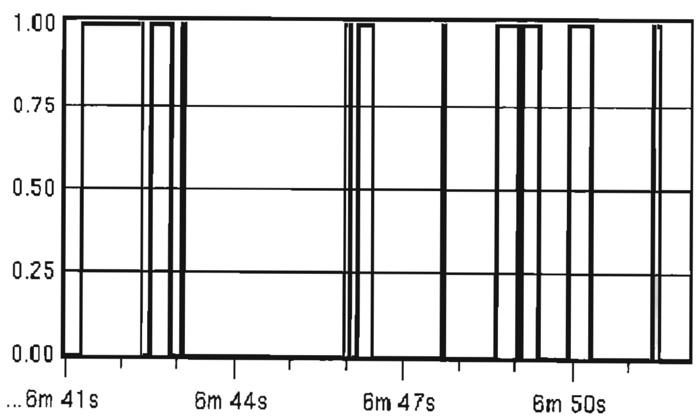

Figure A.2 - ON vs. OFF Times for Voice Source

A.2 Best Effort Model

The Best Effort (BE) traffic source is a data source which generates IP packets. The MTU Generator creates files using exponentially distributed interarrival times and file sizes that are either exponentially distributed or use a Pareto distribution. The IP_fragment module splits the file up into MTUs (the size is specified by the user) and attaches a 20 byte IP header to the packet. The marker can mark in the IP address fields, TOS field, and other fields as required by the model. The TOS field for BE traffic is marked with a DSCP of 000000 (0).

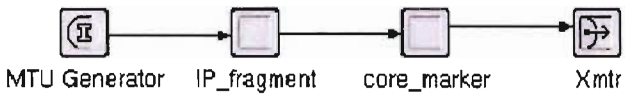

Figure A.3 - Node Model for BE Traffic Generator (BE_Frag_Gen)

A.3 Pareto Generator

The Pareto Generator that is included in version OPNET 7.0, is a stand alone generator that was used to create the data that is shown in Figures A.4 - A.6. This is the file sizes generated with a minimum file size of 1000 bits and a shape parameter of 1.10. Using these parameters, the mean is 11,000 bits. The probability mass function is [62][67]:

$$p(x) = \alpha k^{\alpha} x^{-\alpha-1} \qquad \alpha, k > 0, x \geq k \qquad A.1$$
$$E(x) = k\alpha/(1-\alpha) \qquad A.2$$

Where α is the shape and k is the smallest possible value. Two features of the Pareto distribution should be noted. A shape factor less than 2 will result in infinite variance and a shape factor less than 1 will result in an infinite mean [67]. The research in [67] looked at what factors of k and ρ could be expected in internet traffic. This research for World Wide Web files showed that when file sizes are greater than 1000 bytes, an α of 1.06 is possible. Using this research as a guide, this model used a ρ of 1.06 and set the smallest packet size to 1000 bits for these examples and 453 bits for the model. 453 bits was used for these reasons. Figures A.4, A.5, and A.6 show the file sizes generated from a Pareto file size generator using different random number generator seeds, a shape parameter of 1.06, a k parameter of 1000, and a constant inter arrival time of 1 second.

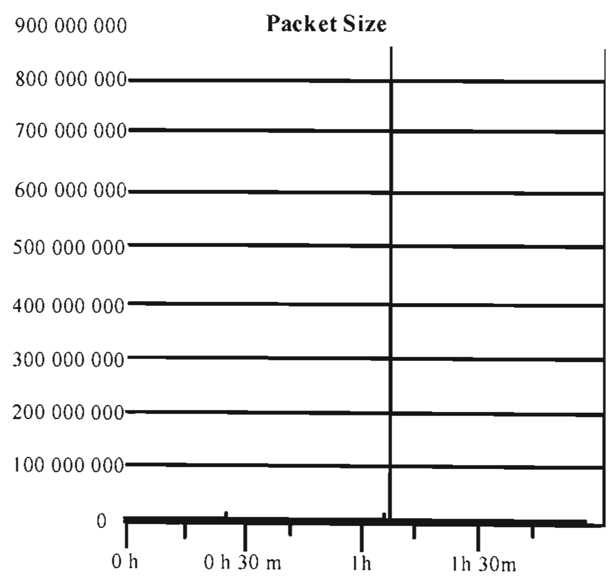

Figure A.4 - Pareto File Size Generator

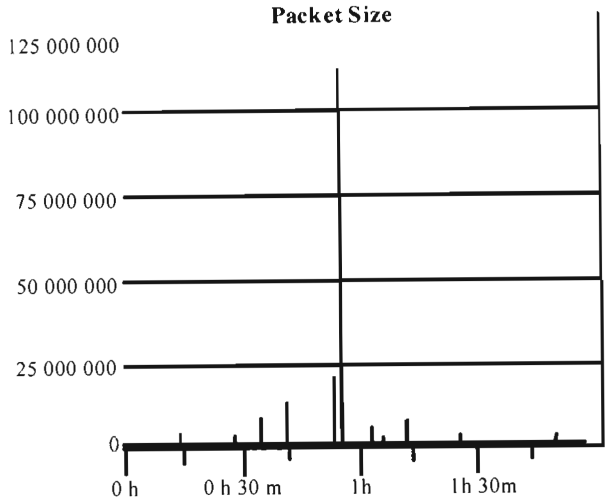

Figure A.5 - Pareto File Size Generator

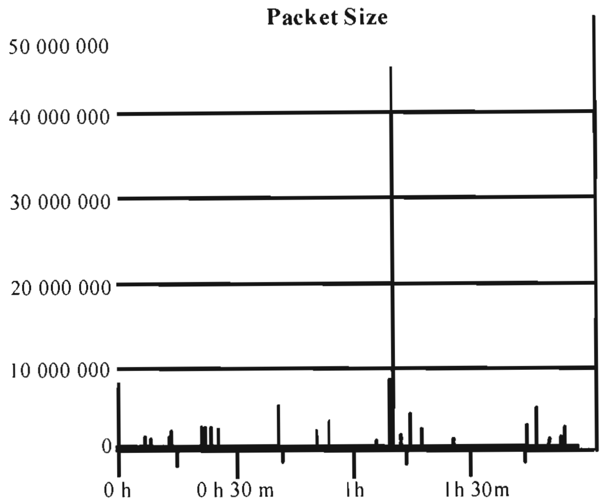

Figure A.6 - Pareto File Size Generator

Figures A.4 - A.6 shows that it is not likely (1 in 5400) that a very large file can be expected in 5400 attempts and that this file will contribute heavily to the average of the utilization of the link.

A.4 Priority Router Model

The priority router model used in the Remote Network Model simulation is the DS_Edge_Router. In the Core Network Model, it was also used for the Ingress Router. It includes token bucket shapers on the inputs, a module that forwards based on the output address (this feature is not exercised in the Remote Network Model as all packets are sent to the link).

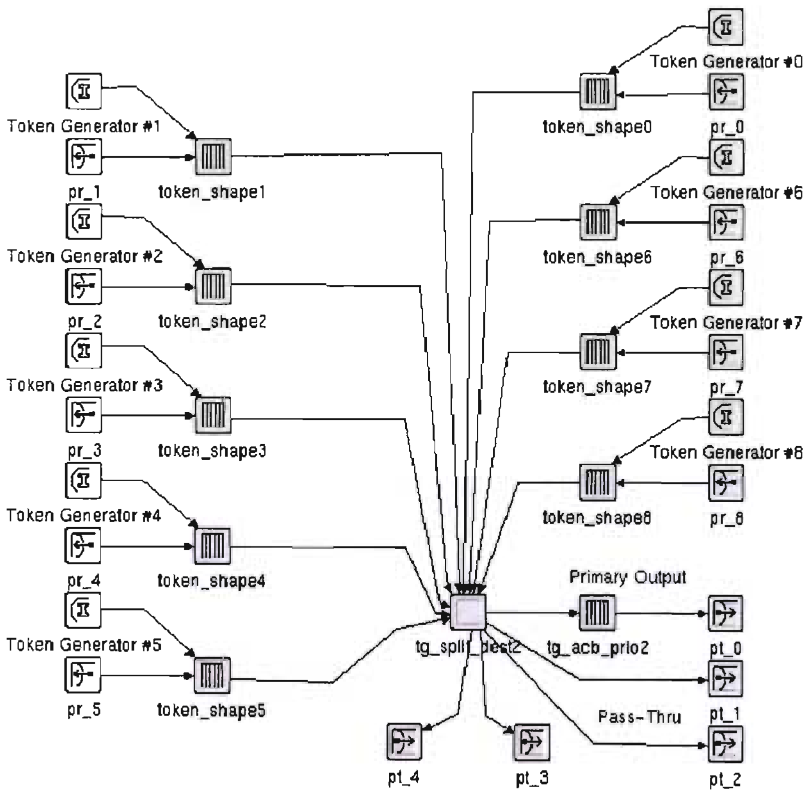

Figure A.7 - Edge Priority Router (DS_Edge_Router)

The node diagram for core_priority_router is shown in Figure A.8. As a packet arrives, it is classified based on its DSCP. The priority for that packet is set based on the DSCP. DSCP mapping to priority and service rate can be set by the user either at the node level or the network level.

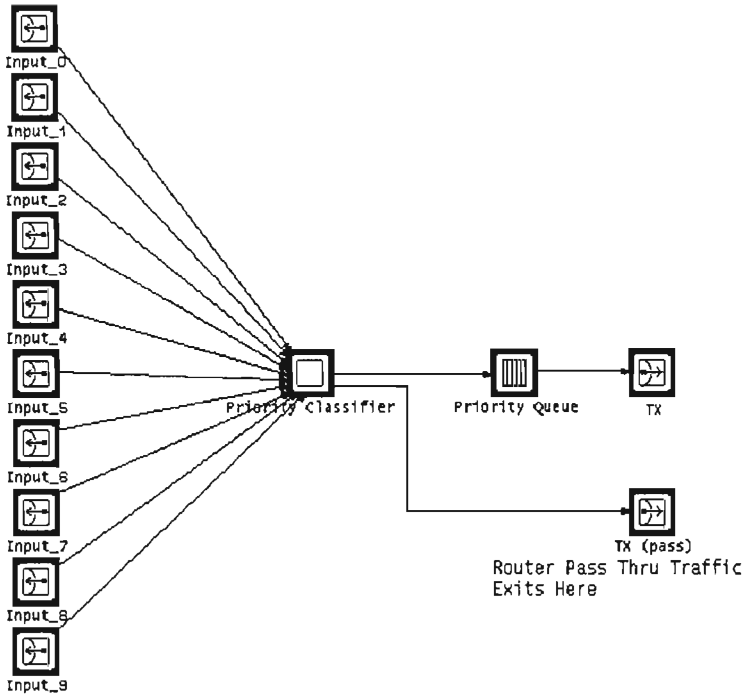

Figure A.8 - Core Priority Queue (core_priority_router)

A.5 Deficit Round Robin (DRR) Router Models

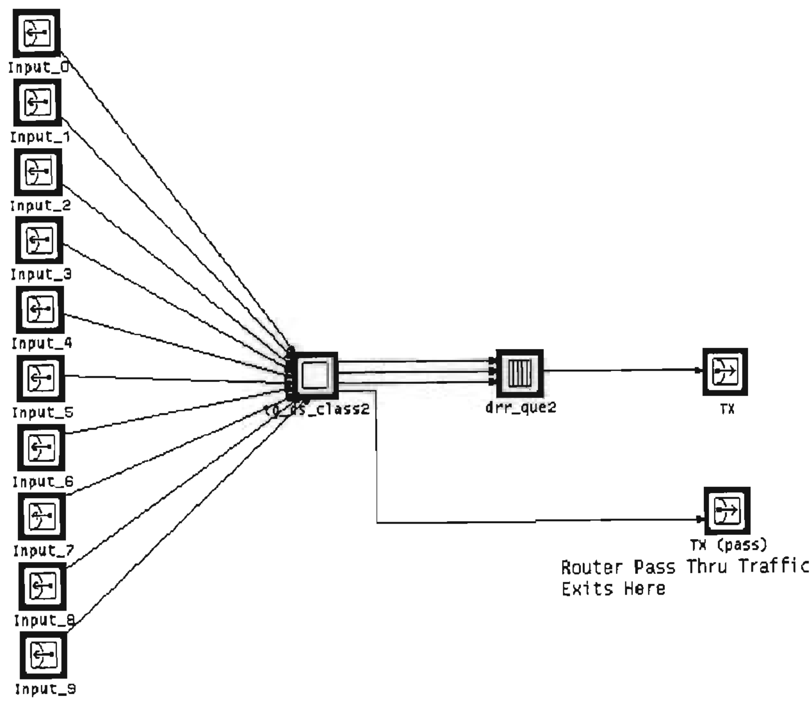

Figure A.9 - Core DRR Router Model (core_drr_router)

The DRR scheduler uses the Deficit Round Robin algorithm as described in [69]. Briefly, the algorithm works as follows [69]. When a packet arrives on a particular input (this implementation), it is queued to a separate queue set up specifically for that input (i). If queue (i) is not already on the Active_List, it is added. Each queue contains a deficit variable (DCi) that is zero if the queue is empty. To schedule packets, the dequeuing module looks at the DCi variable for each queue in a round robin fashion based on the Active_List. If (DCi > = Packet Size) for the packet at the top of the ith queue, then that packet is sent and the DCi variable is decremented by the amount of the packet size. As the dequeuing module passes by each queue (i) it adds a quantity Qi to each DCi. The portion of the link that a particular queue receives is equal to Qi/Q (in the long run) where the Quantum (Q) is the total bits that can be given out per round. If a queue delivers all the packets that are in that queue, the queue is taken off the active list. If a queue cannot deliver a packet because its DCi is not large enough, it is skipped till next round.

Queue number	Quantum (Q)	Min Service Rate	Average Delay	Peak Delay
1	1024	111111	0.0146	0.1077
2	1024	111111	0.0302	0.1780
3	512	55556	0.0395	0.2407
4	512	55556	0.0486	0.4993
5	512	55556	0.0461	0.3992
6	512	55556	55556	0.4027
7	256	27778	17.2423	33.1007
8	256	27778	16.7888	30.6073

Table A.1 - DRR Queue Performance

A.6 Other Models

There are many other models, nodes, and processes that were used in the creation of this model. These were primarily administrative, performing functions like calculating delay, printing results, and marking packets.