Analysis of Mission Critical Voice over IP Networks

by

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Abstract

As government agencies and companies that deal with mission critical communications work toward modernizing their communications systems, the concept of using Internet Protocol (IP) for mission critical voice communication has and will continue to be a goal. IP has become so prevalent in our society that it is the only logical choice for end to end communications.

This work is focused on the feasibility of transporting mission critical voice over IP. In order to complete this study, an in-depth look at mission critical voice communications was completed to define mission critical voice communications. The U.S. Air Traffic Control system was used as an example of mission critical voice.

For Voice over IP to be deployed across the internet in an efficient manner, the issue of voice quality must be addressed. Models like the "E-Model" provide mechanisms to calculate voice quality based on network parameters. This research proposes an optimization algorithm that uses the "E-Model" to select network and voice parameters like coding scheme, packet loss limitations, and link utilization. This optimization can be used to determine the optimal configuration for a voice over IP network.

A discrete event simulation was used to created a performance model to determine the types of delay that could be expected by voice in a small remote network and across a high speed core network. In this model, many network parameters were studied. These parameters include router scheduling, maximum MTU size, link utilization, voice utilization, and data file size distribution. Many interesting observations were made from the results of this study. These are reported in this work.

Finally, a general method is proposed for analyzing mission critical voice over IP networks. As the field of mission critical voice over IP grows, tools and techniques developed in this paper and others will be valuable.

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Chapter 1

Introduction

Envision a situation where an airplane pilot is receiving a correction of a readback from an Air Traffic Controller. Only in this instance, the delay from the time the controller pressed his or her push to talk (PTT) key to begin speaking and the time that the voice arrived at the airplane, due to delays in the Internet Protocol (IP) based voice network, was such that another pilot decides that the airways are clear and begins transmitting. This distorts the controllers transmission (due to the fact that they are on the same air to ground VHF frequency). What occurs next is that the first pilot, assuming mistakenly that his readback was correct, proceeds on a course in opposition to what the controller thinks is going to happen. This could potentially lead the plane on a collision course.

What should occur is the controller (realizing that the correction was not readback) would attempt to contact plane #1 again to re-verify the transmission. But, this does not always occur. Studies have shown that simultaneous transmissions can cause problems like obliterated callsigns, induced misperceptions, blocked transmissions and acknowledgements and contributed to call sign similarities [NADLER]. Therefore, a potentially dangerous situation is possible. And in fact, the collision of two Boeing 747s in 1977 that killed 583 people was partially blamed on a “step-on” [NADLER]. A step-on is defined as blocking that occurred due to the fact that there was delay in the transmission of one party, another party mistakenly transmitted on the same frequency as the first party, effectively garbling the transmission of the first party [NADLER].

In the above example, a delay that may simply be annoying on a telephone conversation, can cause grave consequences in other situations. For the purpose of this paper, we will define certain types of communication as "mission critical". The obvious definition of mission critical voice is voice communications critical to the mission of an organization. This paper goes one step further and defines mission critical voice communications as voice communication critical to a mission that directly impacts the potential to cause injury or loss of life. This example points out the need to control delay and voice quality on mission critical voice services

Another example that illustrates a different point about mission critical voice communications is the possible situation as described by Beard and Frost in [BEARD]. In this situation, a natural disaster has occurred that has damaged a portion of a public switch telephone network (PSTN). In addition, the PSTN has been flooded with calls from friends and relatives looking for information about their loved ones. And the local media has also participated by calling the E-911 lines looking for information about the situation [BEARD].

The interesting aspect of this scenario is that the personnel that were paged to respond to the emergency cannot get through on the telephone to receive their assignment and therefore do not show up where they are needed. This could potentially slow the rescue operation endangering human lives. Even though it was pointed out in [BEARD] that this event was fictitious, it was also pointed out that these types of problems had occurred in actual disasters.

What the above example points out is the need for very high availability in mission critical voice communications. In addition, it also demonstrates that although these two examples are very different, they share certain characteristics. Specifically, the need for high voice quality, low delay, and high availability.

When we think of the problems associated with Voice over Internet Protocol (IP), Quality of Service (QoS) typically comes to mind. [BERNET] defines QoS as "the set of technologies that enables network administrators to manage the effects of congestion on application traffic by using network resources optimally, rather than by continually adding capacity".

This definition enables technologies like Differentiated Services (DiffServ), Integrated Services (RSVP), and Multiprotocol Label Switching (MPLS) to be used to help ensure that a service meets a set of parameters that are used to define QoS. Latency, bandwidth, packet loss, and de-sequencing are among the parameters used to describe if a service meets QoS parameters [HERSENT].

Taken individually, each of these technologies have shortfalls. RSVP was heralded at a way to provision a network based on the services requested. The main criticism was that it does not scale well and the "state" must be maintained on every router [BERNET]. DiffServ scales well but is thought to be not precise enough to be able to make strict QoS guarantees [BERNET]. MPLS addresses routing and has the ability to classify and aggregate flows, but still has management issues like which label distribution protocol will ultimately be used and how will label paths be engineered [ARMITAGE]. Several proposals [RFC2998][FAUCHEUR] are being studied to combine these technologies to derive the benefits of each to provide high QoS and incidentally, provide benefits with respect to survivability and availability.

In light of the array of options that are being proposed to address the problems related to mission critical voice over IP (QoS, availability/survivability, voice quality), tools are needed to quantify the various options. This research uses a collection of tools to analyze voice quality and delay characteristics in mission critical voice over IP networks.

1.1 Mission Critical Voice Communications over Internet Protocol

This thesis answers the following question: With respect to latency and voice quality, is it feasible to use Internet Protocol (IP) to transmit mission critical voice traffic? Two areas that are important to the transmission of mission critical voice over IP are latency and voice quality. These areas were chosen because, in a general sense, they represent some of the primary challenges (along with availability, survivability, and security) associated with the implementation of mission critical voice over IP. Although, availability, survivability, and security are not the subject of this research, this research does take note of the importance of availability, survivability, and security in mission critical networks. Several methods are used in this research to determine the feasibility of mission critical voice over IP:

- Review of existing research on Mission Critical Voice over IP. This includes research into delay bounds in queuing networks and models for voice quality.
- Performance models that use simulation to determine delay bounds in packet networks with voice as a primary component.

- Optimization work on the E-Model, a model that assesses voice quality given certain impairments.

1.2 Why study Mission Critical Voice over IP?

Many private industries and government agencies are currently struggling with separate voice and data networks. Many mission critical voice networks are still using TDM style telecommunications for voice and separate IP networks for data. The lack of integration of mission critical voice and data leads to poor utilization of telecommunication facilities and frequently higher costs.

As new mission critical telecommunications systems are being implemented, more consideration is being given to integrated networks. This is being done for a couple of reasons. First, there are possibly costs savings involved with reducing two separate networks (one for voice and one for data) to a single integrated network [THOM]. Second, the management of a single provider network is significantly less expensive than maintaining multiple networks [THOM]. Several examples of mission critical networks that may benefit from this type of research are:

1. Air Traffic Control Communications. The U. S. Air Traffic Control system currently utilizes a combination of systems that deliver the data and voice necessary to operate the National Airspace System (NAS). The mission critical voice is delivered one of several ways:
    - Leased wireline connections that rely on vendor owned multiplexing.
    - Wireless analog and digital connections that utilize FAA owned terrestrial microwave systems.
    - Wireless leased connections that utilize a satellite link.
    - Wireline connections that utilize FAA owned copper or fiber optic networks (mostly on airports).

    Although, the combination of services is highly available (+.99999 for the most critical services) [FAA1], management of a disjointed network of services and systems is expensive and difficult to maintain.
2. Military Communications. In a situation where military troops are deployed in an area not previously occupied by these troops, the need to communicate (both with

voice and data) quickly and efficiently with the troops in the field is necessary. Integrated networks that can provide secure and reliable data and voice are desirable. These systems must operate in environments that are potentially hostile and not secure. Research is being done today into methods of integrating military communications using ATM and IP over wireless links [RDRN].

3. Emergency Communications. In a situation where a natural or man made disaster has occurred, typically normal communications has been disrupted [BEARD]. Both voice and data communications are required to control the situation by law enforcement personnel. A very reliable and capable network is required to handle the communications while normal infrastructure is being restored. This network has to be immune from interruptions caused by less important communication that is occurring as a result of the disaster [BEARD].

The three examples listed above illustrate why the study of mission critical voice over IP is important. The next section summarizes the findings of this research.

1.3 Research Contributions

Since IP is becoming the prevalent protocol for transport of data, there is interest in integrating voice and data over IP. This also applies to mission critical voice services, like air traffic control and military communications. Two of the primary challenges associated with voice communications over IP are latency and voice quality. This thesis attempts to answer the question (as stated above): With respect to latency and voice quality, is it feasible to use Internet Protocol (IP) to transmit mission critical voice traffic?

The primary contributions of this research are:

- Determined the feasibility of a mission critical voice over IP network that looked at delay and voice quality.
- Developed an optimization of the E-Model for parameters like coder type, packet loss level, and link utilization level. The E-Model is an ITU and TIA accepted model that, based on a collection of input parameters, calculates a value for voice quality.

- Determined and average and peak delays from a simulation model of a remote communications site. This analyzed the problems associated small networks connected a low bitrate links.
- Determined peak delays from a simulation model of a core network. This analyzed the problems associated networks containing multiple hops, low speed to high speed links, and high speed to low speed links.

The lessons learned from this research are numerous, indicating the complexity of transmitting mission critical voice over IP. The primary conclusion from the research is that a mission critical voice over IP network is feasible given that certain stringent requirements are met. First, an E-Model optimization was completed to determine delay, coder type, and other crucial parameters. Second, on the edge of the network (where link bitrates are small), packet size and the number of voice sources must be controlled. Third, the core of the network must be tightly controlled with respect to voice load and scheduling mechanisms.

This research showed that the E-model can be optimized for many parameters, enabling network designers to use analytical methods to select coding schemes, QoS parameters, and link load levels. An important finding of this portion of the research is that an optimization of the E-Model is possible and useful. The results illustrate that the optimization solved the problem. This is important because this model could be extended to more complex problems that cannot easily be analyzed by hand.

This research also used a discrete event simulation model to analyze delays in a sample network. During the portion of the simulation that dealt with low bit rate links (Remote Network), a requirement of 12 ms maximum network delay was established as a requirement. This was met in certain scenarios. Three assumptions proved to be very important. First, maximum data packet size (MTU) is highly correlated with worst case delay for voice. Second, since the number of voice sources is limited, the likelihood of all sources being "on" is not remote. Third, voice utilization is very important and becomes even more important when a Pareto distribution is used for data file sizes.

During the core network simulation, it was observed that it was possible to limit voice delay through the core of the network to 40 ms which was the requirement set up in Chapter 6. But, this was only observed with strict priority queuing. When DRR queues were used, maximum delays climbed to at least 60 ms. This presents problems with scheduling in the core of the network. Section 1.4 describes the organization of the thesis.

1.4 Organization

This paper is organized as follows. Chapter 2 defines and looks at the characteristics of mission critical voice. This includes latency, packet loss, and overall voice quality as well as other characteristics like availability and survivability. Chapter 3 reviews the problems that exist for mission critical voice over IP and what types of solutions are currently proposed to address these problems. Bounds on latency are explored as well as impacts on delay from link failure. Chapter 4 describes the E-Model and how it is used. An optimization of the E-model is explored in Chapter 5. In Chapter 6, a simulation study is presented that looks at latency in a voice over IP network. Both edge network latency and core network latency are explored. In Chapter 7, architectural issues for mission critical voice over IP are investigated. Finally, Chapter 8 draws some conclusions and proposes future research. Validation of the simulation model is included in the Appendix.

Chapter 2

Description of Mission Critical Voice

Having established the motivation for research into transporting mission critical voice over IP, what exactly is "mission critical voice"? Applications that are associated with fault tolerant computing are typically related to human safety, environmental cleanliness, or equipment protection [PRADHAN]. For the purposes of this research, mission critical voice refers to voice transmission could potentially cause injury or loss of life if not received correctly or in a timely fashion. This may include, but is not limited to 911 emergency service, military communications, and air traffic control. Looking at these examples, it is not difficult to see the need for high quality communications that is reliable and predictable. The qualities that are relevant to mission critical voice applications include availability, reliability, survivability, voice quality, latency, and security.

This research does not try to classify all types of mission critical voice communications. Instead, it attempts to look at the qualities that one would need to classify a mission critical application. For example, the FAA availability requirements for critical communications is .99999 [FAA1]. While, maybe on a military field of battle, that level of availability may not even be possible. For this reason, this research attempts to identify and discuss the qualities that a mission critical voice service may define as requirements.

The U.S. Air Traffic Control (ATC) system is used as a model for this research. There are a couple of reasons why the ATC system is a good choice to use as a model.

- The ATC system has a large number of mission critical voice services. Currently, there are over 2000 critical services (defined here as individual radio channels) from pilot to controller.
- The mission critical services, used in the ATC system, are used on a routine basis. This is important because it allows performance of the communications

to be recorded, analyzed, and impacts to the ATC system to be accurately measured. This is accomplished through the use of the ASRS. The Aviation Safety Reporting System (ASRS) is an independent third party that receives and analyzes reports as specified by the Aviation Safety Reporting Program. The ASRS is an FAA Program that "stipulates the free and unrestricted flow of information concerning deficiencies and discrepancies in the aviation system…a program intended to ensure the safest possible system identifying and correcting unsafe conditions before they lead to accidents." [PRINZO]. The ASRS is an example of a system that attempts to track problems in the transfer of information in the aviation system. In fact, during a one year period beginning in 1976, 70% of 28,000 reports submitted and controllers were related to information transfer and were "primarily related to voice communications." [PRINZO]

- Air Traffic Control communications have definite consequences that effect safety when the requirements for the service are not met. During a 29 month period, 417 hazardous incidents were noted due to misunderstandings, misinterpretations, mis-transmissions, or unheard numbers in ground-to-air communications [PRINZO]. This is significant, because it demonstrates the importance of high quality communications to the safety of the flying public. In 1983, Monan noted that problems with information transfer between controllers and pilots occurred with such frequency that it was considered a problem for air traffic safety [PRINZO].

It is important to realize that the data used in this thesis is useful when looking at all mission critical communications. That is because many of the requirements for the ATC system apply to other types of mission critical services. For example, when a 911 operator speaks, his or her voice must be clear and legible. So, the voice quality requirement may very well be the same as the similar requirement in the ATC system. But, since 911 calls travels over normal dial lines that do not carry the same security requirements as a voice grade ATC service, not all requirements that apply to ATC systems apply to all mission critical systems. Therefore, this is intended to be a guide for

ATC mission critical service, and a template for analyzing other forms of mission critical services.

2.1 Voice Quality

Voice quality in the context of a Voice over IP 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.80. Using the MOS method, listeners are asked to rate speech and classify it into categories. These categories are shown in Table 2.1 [ITU80]:

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

Table 2.1 – 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 was the quality that could be expected for a connection in the United States public switched telephone network (PSTN). Many other tests are also used to gauge 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 [ITU80][TIA116].

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 wireline telephony [HERSENT].

The European Telecommunications Standards Institute (ETSI) developed the E-model to address the needs of network planners [HERSENT]. The E-model is based on the premise that “Psychological factors on the psychological scale are additive” [ETSI205]. 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 [TIA116][ITU107].

User Satisfaction	E-model - R	MOS
Very Satisfied	90	4.3
Satisfied	80	4.0
Some Users Dissatisfied	70	3.6
Many Users Dissatisfied	60	3.1
Nearly All Users Dissatisfied	50	2.6
Not Recommended	0	1.0

Table 2.2 – E-model vs. MOS values [TIA116]

Comparing the MOS scale and E-model provides a reference as to what is considered acceptable. Table 2.2 shows a comparison of the two scales [TIA116]. More information about the E-Model is included in Chapter 4. One of the key components of the E-Model (and voice quality) is latency.

2.1.1 Latency

Latency 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 [HERSENT].

Both [TIA116] and [HERSENT] address the topic of voice delay. Interestingly enough, what is considered unacceptable varies significantly. For instance, [HERSENT] noted that an acceptable one way delay up to 250 ms is acceptable, but 400 ms is the limit after which the conversion can be considered to be half duplex. Between 150 and 200 ms, conversation begins to be affected according to the Telecommunication Industry Association (TIA) [TIA116]. 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 [TIA116]. When an extra 150 ms is added, the turn taking 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. An example that TIA gives is the very simple but important question, "Will you marry me?". If the answer is yes, but is delayed by a noticeable amount, the answer may be interpreted as a hesitation to reply rather than a positive answer.

The delay associated with analog systems was typically minimal because the primary contributor was propagation time. The propagation time across the U.S is approximately 25 ms [TIA116]. For most analog 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 [TIA810]:

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

Figure 2.1 (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 [ITU108].

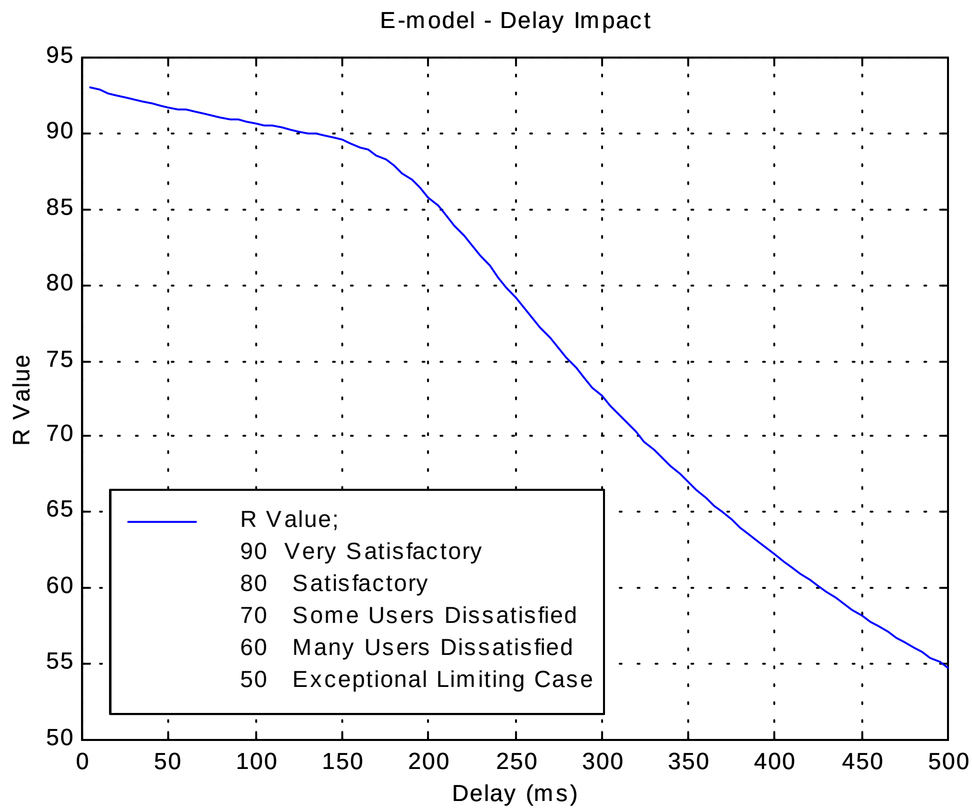

Figure 2.1 – E-model, Impact for Delay

As Figure 2.1 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 [TIA116].

2.1.2 Noise and Level Impairments

Most traditional voice systems have a series of losses and gains throughout the network. In the telecommunications world, a reference audio power level is set in the network. This is referred to as the Zero Transmission Line Power (0TLP) point. Gains and losses are referred to with respect to the 0TLP point. Likewise, noise is measured in similar manner. dBrn, which is reference noise, is equal to the noise level (in dBm) + 90 [FAA3].

For example, the term –7 dBm0 means that the power at the 0 TLP point is –7 dBm. With –7 dBm0, a +5 TLP is equal to –2dBm. Similarly, a 30 dBrn noise reading would represent a noise power level of –60 dBm. Noise testing is typically measured using a weighting curve. The reason for this is that noise in the voice band will have a more significant effect on a circuit than noise outside of the voice band. The most widely used weighting curve is the C-message weighting. The reference level is dBrnC.

Improper noise or level adjustments on a circuit can have adverse effects on the speech quality. The optimum range for overall loudness rating (OLR) is 6 to 10 dB [ITU108]. The overall loudness rating is the combination of the Send Loudness Rating (SLR), Network Loss, and Receiving Loudness Rating (RLR). The SLR and RLR are overall gains/losses of the terminal devices. The Network Loss is the net loss (or gain) imposed by the network. A zero loss line imposes no net loss or gain on the circuit.

Noise can come from several sources. Room noise at either the sender or receiver end can significantly affect quality as well as poor signal to noise ratio and noise contrast levels that are too high [ITU108]. Most quiet offices have noise levels of between 30 and 50 dBrnC These values are affected by the environment and the type of headsets that are used. Other sources for noise include circuit noise and quantization noise.

2.1.3 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 garble 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 [HERSENT].

Methods of dealing with echo include echo canceling devices and echo suppressors. TIA specifies that for PSTN and Toll quality speech, echo canceling 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 [TIA116].

2.1.4 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. Typically, because of high correlation between adjacent packets, only the difference needs to be send. Of course, this would propagate errors

Adaptive Quantizers 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 [HERSENT]. 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 [HERSENT]. 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 [KROON]. Some of popular speech coding schemes are noted in Table 2.3 with important parameters [HERSENT][ITU108].

Coding Standard	Bit Rate (kbps)	Frame Length (ms)	Codec Delay (ms)*	Complexity (MIPS)	MOS	E-Model 'Le'
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-MLQ-ACELP	6.3/5.3	30	37.5	16	3.9/3.7	15/19

*This assumes a single frame with lookahead.

Table 2.3 - Voice Compression Parameters

2.1.5 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 typically cannot withstand the delay [KOSTAS]. Packet loss may or may not be related to latency. If packet loss is caused by bit errors, these losses would be unrelated to latency. However, if a maximum delay for a voice stream is exceeded, typically those packets are dropped. In this case, there exist 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 devised to minimize the effects. In [PETR], the quality of voice was explored when used with priority discarding, which allowed the least significant bits of voice frames to be discarded first (as opposed to randomly). The result yielded an increase in the level of packet loss that could be tolerated.

ITU G.711, G.729, and G.723.1 compression standards both have packet loss concealment (PLC) methods to deal with packet loss. For example, G.729 uses the Line Spectral Frequencies (LSF) of the previous frame with attenuated codebook gains from the previous frame. [ITU729].

Packet Loss %	G.711 without PLC	G.711 with PLC Random Packet Loss	G.711 with PLC Bursty Packet Loss	G.729A + VAD	G.723.1 + VAD (6.3 kbits/s)
0	0	0	0	11	15
0.5				11	15
1	25	5	5	15	19
1.5				17	22
2	35	7	7	19	24
3	45	10	10	23	27
4				26	32
5	55	15	30		
7		20	35		
8				36	41
10		25	40		
15		35	45		
16				49	55
20		45	50		

Table 2.4 - E-Model Impairment (Ie) for Packet Loss vs. Coding Scheme

Table 2.4 compares E-model impairment given a certain packet loss level and coding scheme [TIA116]. G.729 and G.723.1 is shown with Voice Activity Detection (VAD). Since the maximum in the E-model is R = 100 and any values below 70 are not generally considered acceptable, it is easy to see the impact of packet loss using this model.

2.2 Other Mission Critical Voice Qualities

This description of Mission Critical Voice focuses on a few specific qualities that are in the scope of this work. There are many other qualities that must be considered when designing a mission critical voice network. A few of these qualities are security, availability, and survivability. Section 2.2.1 briefly looks at availability and survivability.

2.2.1 Availability and Survivability

Availability and Survivability are both important to the transmission of mission critical voice. But, these qualities are different and impose different requirements on a mission critical network.

High availability is one of the most important aspects of mission critical communications. This can be demonstrated by example. If an emergency occurs and the 911 service is not available, human safety may be at risk due to a low service availability. Logic would dictate that high availability is important to 911 emergency service.

Availability is defined as the probability that a system is operating correctly and is available to perform it functions at the instant of time [PRADHAN]. This is important because a system can be inoperable frequently for very short periods of time and still have high availability.

Survivability is quite different from availability because it focuses more on a particular event and its potential effect on the network. Various groups have defined survivability in a multitude of different ways [ARPA][AYAN]. Advanced Research Projects Agency (ARPA) defined survivability as the ability to continue performing critical tasks at an adequate level even when there are successful attacks launched [ARPA]. This definition assumes that the “event” is an enemy attack. The important aspect of their definition is that the event was successful and the system continued to perform the most critical tasks. It may seem that their definition is directed at intentional attacks. However, the theory given in [ARPA] is equally applicable to natural events.

Ayanoglu and Gitlin describe survivability as the capability of the network to resist interruptions/disturbance of service due to physical or natural catastrophes rather than electromagnetic interference or crosstalk [AYAN]. Webster's dictionary defines survivability as the ability "to continue living or existing, as after an event, …". These definitions are interesting because they all point to preventing an event (natural or manmade) from causing a loss of service.

If a voice service is designed so that all the parts of the system providing the service are redundant, a failure of an individual component will not disable the system and cause an outage of the service. Survivability is improved by eliminating a single event as a cause of a service outage. Availability is improved by reducing the probability that a combination of events will cause a service outage. Section 2.3 looks at the U.S. Air Traffic Control systems as an example of a mission critical service.

2.3 Mission Critical Voice in Air Traffic Control

The FAA defines critical communications as air traffic control (ATC) discussions between a pilot and controller or between two controllers. According to [FAA1], mission critical voice is defined as communications that carries real-time traffic that is used to control aircraft [FAA1]. The FAA considers this critical service which must be available 99.999% of the time. In addition, the minimum latency considered acceptable is 50 ms and the minimum quality of voice is considered to be Toll Quality (or MOS = 4.0).

Several tests have been done to show that latency in air traffic communications up to 150 ms is tolerable without "step on" incidences occurring [NADLER]. A "step on" is an instance where an Air Traffic controller believes that the pilot is not speaking (due to the delay) and speaks. The controller in this instance, steps on the pilot. The result is that the controller does not hear the pilot transmission and possibly the pilot does not hear the controller. This can result in operational errors. Therefore the “step on” has been used as the primary method to determine if delay is a problem in an ATC network.

The important distinction between normal speech and mission critical voice is that with mission critical voice, delays that amount to more than a certain threshold are regarded as a loss of service [FAA2]. This is because the potential errors as a result of the delay are typically more dangerous than a complete loss of service. This demonstrates the relationship of availability to latency.

Although, most of this discussion is based on voice communications, the other types of traffic that the network is carrying is also important. The QoS requirements of the remaining traffic will dictate how it is treated in the network. This can also the affect high priority voice traffic. For example, many queuing schemes like weighted round robin and deficit round robin do not give total priority to voice traffic. In the U.S. ATC system, the types of traffic that may share the network could vary widely. Radar signals may have latency requirements of up to a second, while other types of data may have very short latency requirements.

Availability has traditionally been achieved a number of ways. At remote sites, Availability is achieved by providing completely separate and redundant analog paths. For example, at a remote site that delivers radar data to a control center, only one communications path may be available from the Local Exchange Carrier (LEC) at a practical cost. A microwave or satellite link may provide the redundant route and thus the high availability. Packet switched communications provide additional challenges. One of these is to prove availability with an IP type network that reroutes packets based on a number of variables (least costs, least hops, or other means). Chapter 3 explores the concept of Quality of Service in mission critical networks, including a discussion of current research in this field.

Chapter 3

Quality of Service in Mission Critical Voice over IP Networks

Even though the concepts of voice over IP and mission critical IP networks are not new, there are still significant issues surrounding both of these topics. This chapter explores the difficulties associated with the implementation of voice over IP networks and mission critical networks. It also presents current research that deal with solving these issues.

Most of the research presented looks at Voice over IP QoS issues. Also presented here is some research that analyzes QoS in networks under failure. Although this research does not address survivability, it is important to mission critical networks. Extending the concepts presented in this research to networks under failure is a topic for future research.

3.1 Issues Surrounding VOIP and Mission Critical Network Implementation

This section looks at how the current internet performs with respect to QoS and other network related impairments. The performance of the current internet is important because it gives an indication of what can be expected from the current internet.

In [PAXSON], a large survey of internet connections was studied (20,000 TCP transfers between 35 sites) with the intent of analyzing packet dynamics. In this study, several network pathologies occurred with some frequency. Out-of-order delivery occurred with widely varying percentages and was considered by the author to be "fairly prevalent". During one instant, 15% of the packets to a certain router were delivered out of order in one direction, while only 1.5% were delivered out of order in the opposite direction. Approximately 1 in 5000 packets were found to have checksum errors (not an insignificant amount). Packet loss rates varied from 3.6% to 17% (although the 17% case was unusual). Geography played an important role in the loss rate (the 17% figure was "into Europe"). Loss durations were also studied varying from the bottom 10% having a loss duration of 33ms or less and the top 10% having loss durations of 3.2 secs or more.

Probably the most significant statistic is delay. Delay primarily occurred on time scales of .1 sec to 1 sec. It was noted that it frequently extended to much larger times including values up to and exceeding 65 sec [PAXSON].

It is clear from this study that serious problems exist in the current internet with regards to QoS. Most of the values mentioned above would not meet even the basic QoS requirements for VOIP (like delay). In [LABOVITZ], a similar study was completed that was looking for internet stability and backbone failures. The study was on a medium size network of 33 backbone routers and several hundred customer routers.

The general conclusion from the study was that the PSTN was "significantly" more reliable and had higher availability than the internet. In addition, internet failures were not as hard to detect as the source of the failure partly caused by non-standard reporting of faults. In general, all backbone routers had availabilities over 99%. Problems with availability seemed to be caused by "persistent circuit or hardware faults" [LABOVITZ]. However when studying individual routes, it was shown that availability was an order of magnitude less than the PSTN, which has an average availability of 99.999%. The most startling statistic is that 10% of the routes that had less than 95% availability. Only 25% to 35% of the routes from each ISP had greater than 99.99% availability [LABOVITZ]. One final point, it was noted that failure rates had increased slightly since a similar study was conducted in 1994, which questions the argument that as the providers upgrade their infrastructure, availability will increase.

It is not hard to see from these studies that the current internet is not suitable for mission critical voice. The rest of this chapter looks at different ways that researchers are trying to address the problems described above.

3.2 Delay Bounds in Queuing Networks

Due to the complex nature of calculating delays in IP networks, many models have been used to provide estimates of worst case delay. In [TIA116], the recommendation is to use an approximation to estimate the worst case queuing time.

Average Queuing Time $t_{Q-av} = \frac{t_{dls}\rho}{2(1-\rho)}$ 3.1

Max Queuing Time (approx 95% of distribution) $t_{A\text{-}wo} = 2(t_{Q\text{-}av})$ 3.2

Where ρ is representative of link utilization and t_{dls} is the voice packet serialization delay. [TIA116] assumed that an M/D/1 queue would be a good representation of a priority queue that serves fixed length voice packets as the first priority. The model also adds a delay element for a maximum size data packet (MTU) that has already started transmission on the link.

Other texts [MCDYSAN] recommend using maximum delay calculations for a Weighted Fair Queue (WFQ), which stem from the General Processing Sharing (GPS) model. The delay bound for an end-to-end WFQ network is [MCDYSAN]:

$$Max[t] \le \frac{b + 2(N-1)L}{r} + \sum_{i=1}^{N} \frac{L_{Max}}{R_i} \qquad 3.3$$

Where r and b are the token bucket rate and burst size for the flow, L is the packet MTU size, L_{Max} is the maximum MTU size, R_i is the bit rate of the ith link, and N is the number of nodes traversed.

Recently there has been renewed interest in delay bounds in IP networks. [BOUDEC] and [CHARNY] have derived delay bounds for networks with aggregate scheduling. For the case of general topology, equation 3.4 is the delay bound found.

$$D = \frac{h}{1-(h-1)u\alpha}(\Delta + u\tau) \qquad \alpha < \min\left(\frac{\Gamma(l)}{(\Gamma(l)-C(l))(h-1)+C(l)}\right) \qquad 3.4$$

Where $\Gamma(l)$ is the total peak incoming bit rate on that queue, $C(l)$ is the link capacity on link l, h is the max hop count approaching the link under question, and $u(l) = (\Gamma(l) - C(l))/(\Gamma(l) - \alpha C(l))$. Δ is defined as *max(Kj/Sj)* for a queue, where *Kj* and Sj are the service deficit bound and guaranteed service rate of the queue at link *j*. τ is a parameter that relates the burst parameter (B) of the each flow with the rate on the link *C(l)* as defined by the following relationship: $\sum_{F \in S(l)} B_F \le \tau C(l)$.

The derivation for this bound is interesting because it uses worst case arrival and departure curves to arrive at the delay bound [BOUDEC]. The reader is referred to [BOUDEC] for a complete derivation. A couple of points about this delay bound [BOUDEC]:

- The condition that the utilization (α) must meet in equation 3.4 depends heavily on the total peak input and output rates of the queue. In fact, on a 6 hop network with a total peak input rate of 1.536 Mbps, and output link speed of 128 kpbs (not unusual at an edge router), the priority utilization required to have this delay bound exist is limited to 21.4%.
- Since *u(l)* grows with increases in *Γ(l)*, it is easy to see that the delay bound explodes as h grows, implying that burst sizes can increase with additional hops. In [CHARNY], it is pointed out that the potential token bucket burst size does increase by *Dr*, where *D* is the delay across a node and *r* is the rate of the flow. So this claim is justified.
- If the input characteristics are not known (or are very large), Γ*(l)* is infinity. This causes the limit on α (and still have a delay bound) to become α<h/(h-1) and the delay bound to change. The new delay bound is shown in equation 3.5. It is easy to see the effect that the hop count has on the delay bound.

$$D = \frac{h}{1-(h-1)\alpha}(\Delta + \tau) \qquad 3.5$$

Since equation 3.5 reflects the delay bound for general topology with any input traffic characteristics (other than token bucket rate and burst), it is conceivable that controlling the topology and input traffic would produce tighter traffic bounds. In [CHARNY], the delay bound for monotonic networks was derived in equation 3.6. For the monotonic requirement to be met, the following must occur. If *j* represents the maximum number of hops of any flow entering a node and *h* represents the maximum hop count of any flow in the network, then $j \le h$.

$$D = \frac{(1+\alpha)^h - 1}{\alpha}\left(\frac{\alpha b_{\max}}{r_{\min}} + \Delta\right) \qquad 3.6$$

The parameters r_{max} and b_{max} are the maximum token bucket rate and burst size of any flow on the link. α is the link utilization. It is also easy to see the effect of the hop count and link utilization in this delay bound. Figure 3.1 shows an example of the various delay bounds compared to an increasing hop count.

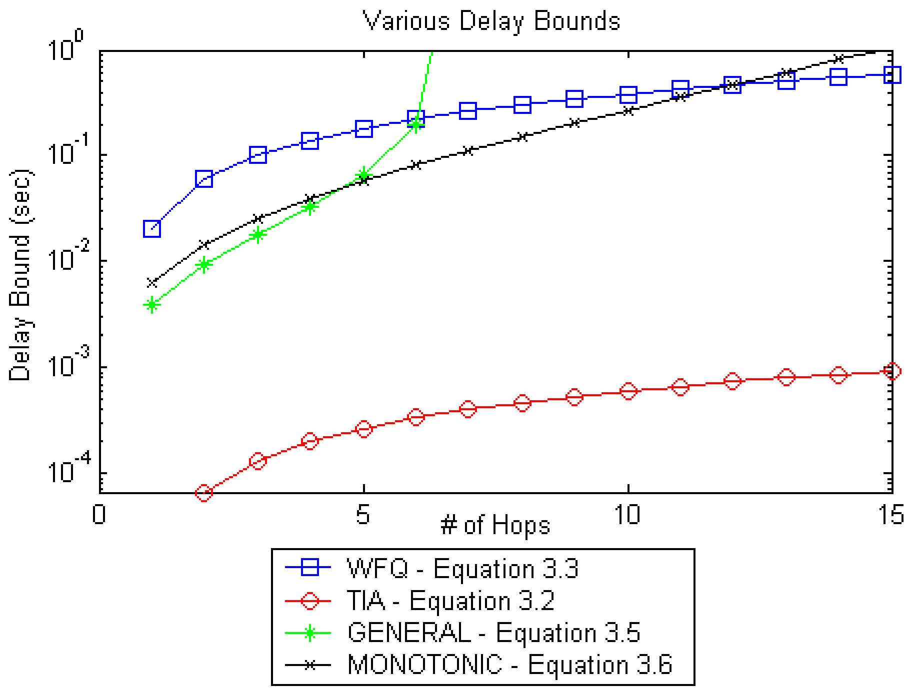

Figure 3.1 - Various Delay Bounds vs. Hop count

Figure 3.1 illustrates the difference between the various methods of estimating delay bounds in IP networks. For this example, the link speed is 1.536 Mbps and the flow rate (bit rate required for each flow) is 24 kbps. The MTU, Max MTU, and token bucket burst is 480 bits. The link utilization for the top priority traffic is .3. The queue is assumed to be a class based priority queue.

The bound for general topology explodes at hop #6. This is where the utilization violates the requirement for Equation 3.4. The general topology delay bound becomes unbounded at that point. The delay bound used in [TIA116] is insignificant compared to the other bounds. This is due to the fact that the estimate used for maximum delay is not the maximum possible delay, but rather 95% of the delay distribution. In addition to that fact, it is a tight bound assuming an M/D/1 queue. Figure 3.1 illustrates that estimating delay bounds is not simple nor well understood. Another area that is not well understood is the assessment impact to delay of link failure. The next section reviews a couple of studies that address that topic.

3.3 The Impacts of Link Failure on Delay

There are several performance evaluation studies and models that have been completed that look at delay and congestion when links in the network have failed. This report looks at three papers.

The first paper [TIPPER] uses a network simulation tool to implement a network running Resource Reservation Protocol (RSVP). RSVP is extended here to include the ability to set up explicit paths. The simulation involves various links failing and network performance being assessed. Two approaches are used. First, a working path and protection path are set up and the working path is failed. Next, a dynamic search is used to restore the working path when it is failed. The first approach led to slightly faster restoration times. But, it also led to poor utilization (resulting in call blocking) and severe "transient" congestion following restoration [TIPPER]. The recommendation was to use the first approach for only a small amount of the traffic that must have that level of protection.

The second paper [ROGERS] looked at the performance of a Differentiated Services (Diffserv) network under failure. A simulator called the MoMaRS (Missouri - Maryland Routing Simulator) was used to perform this research. The result of this research was that the priority-based routing algorithm that was used did provide the required protection for the high priority traffic, but if excess network resources did not exist, the lower priority may be "starved".

The final paper [THIR] used the concept of Multiprotocol Label Switching (MPLS) Book Ahead Guaranteed (BAG) Services to investigate restoration of three types of survivable traffic with the goal of optimizing the network based on different objectives. The traffic types were fully survivable (guaranteed full bandwidth under normal and failure conditions), fractionally survivable (guaranteed full bandwidth under normal conditions and a reduced level of service in the event of a major link failure), and zero survivable (guaranteed full bandwidth only under normal conditions). Four possible objectives were investigated. These are listed here:

- "maximize residual capacity in the network for best effort traffic."
- "minimize the total demand routing cost for BAG services"
- "minimize the penalty if a demand cannot be met..."
- "maximization of revenue"

The research showed that the objectives can be combined into a single objective to accomplish the goal of representing all traffic classes (even best effort) and solved. The problem was solved by relaxing the integrality requirement of the variables turning the problem into a linear programming problem.

The research in this section is included because of the impact of failure to mission critical networks. This research does not further investigate the topic of delay under failure, but rather leaves it for future research. The next section looks at the E-Model, a model used to assess voice quality when several types of impairments are present.

Chapter 4

The E-model: A Model for Integrating Voice Quality

In Chapter 2, 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 2.2 (in Chapter 2) shows a comparison of the R value and the MOS levels. *R* is represented in [TIA116][ETSI205][ITU108] as:

$$R = Ro - Is - Id - Ie + A \qquad 4.1$$

In this Equation 4.1, *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 4.1 is the default values and the permitted ranges for the parameters used as inputs to the E-Model [ITU107]. Since the parameters of primary concern of this research involve delay, voice compression, and packet loss, this discussion will be primarily centered around equations involving those parameters. For more information about the details of each parameter the reader is referred to [ITU107] and [ITU108]. 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 4.1 – Default values for E-Model

4.1 E-Model Impairment Calculations

As mentioned earlier, the E-Model relies on a system of impairments. The Simultaneous Impairment (*Is*) equations affected are [ITU107]:

$$Is = Iolr + Ist + Iq \qquad 4.2$$

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.

$$Ist = 10\left[1+\left(\frac{STMRo-12}{5}\right)^{6}\right]^{1/6} - 46\left[1+\left(\frac{STMRo}{23}\right)^{10}\right]^{1/10} + 36 \qquad 4.3$$

$$STMRo = -10\ln\left[10^{-\frac{STMR}{10}} + e^{-\frac{T}{4}}10^{-\frac{TELR}{10}}\right] \qquad 4.4$$

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 4.4 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 [ITU107]:

$$Id = Idte + Idle + Idd \qquad 4.5$$

Idte is impairment due to talker echo..

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

$$Roe = -1.5(No - RLR)$$

$$\mathrm{Re} = 80 + 2.5(TERV - 14) \qquad 4.6$$

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

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

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

$$Rle = 10.5(WEPL + 7)(Tr + 1)^{-0.25} \qquad 4.7$$

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 [ITU107]. Figure 2.3 in Chapter 2 illustrates the impairment that absolute delay imparts to voice quality. Note: If one way delay (*Ta*) < 100 ms, *Idd* is assumed to be 0.

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

$$X = \frac{\ln\left(\frac{Ta}{100}\right)}{\ln(2)} \qquad 4.8$$

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. [ITU113A] contains the latest recommended values for *Ie*. Table 2.3 and 2.4 in Chapter 2, 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 [TIA116]. 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 4.1 shows the levels defined compared to the graph of delay versus *R* value (similar to Figure 2.3).

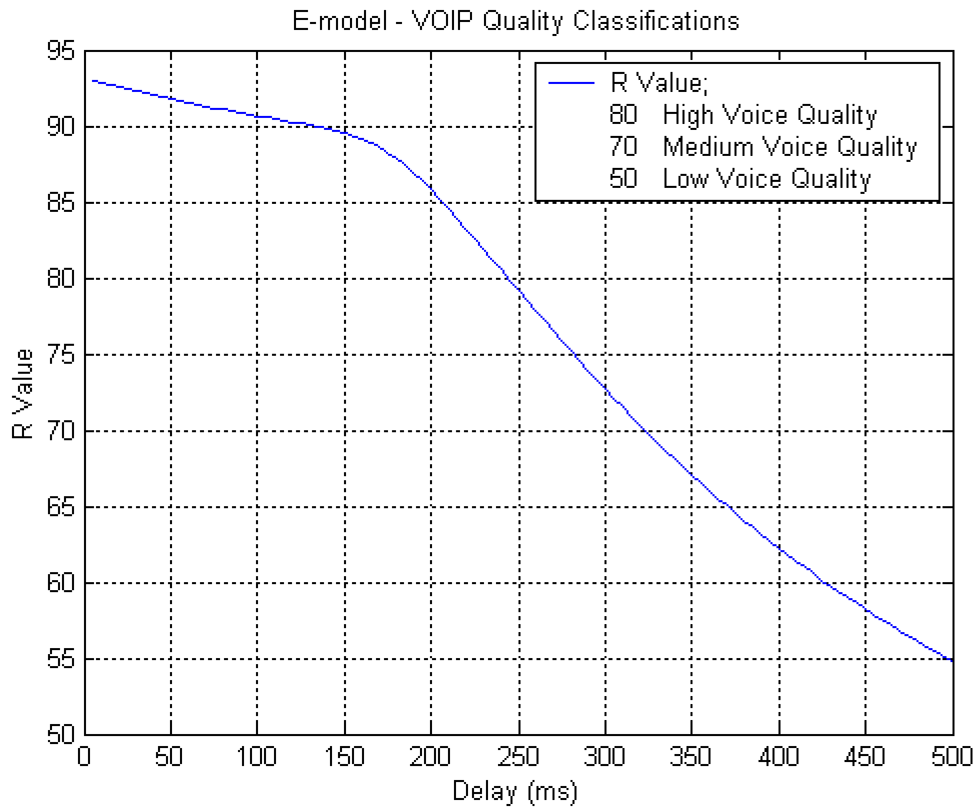

Figure 4.1 - VOIP Voice Quality Levels vs. Delay

The following are a summary of the recommendations [TIA116]:

- 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 codecs, 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."
- Transcoding Rule #2: "Maximum number of transcodes = 1."
- Transcoding Rule #3: "DCME equipment must be eliminated."

It interesting that the recommendations strongly recommend using the E-model and using G.711 whenever the bandwidth exist 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 5.

Chapter 5

E-Model Optimization

The E-Model, as discussed in Chapter 4 is a model that allows users to relate network impairments to voice quality. Mission Critical Voice requires certain levels of voice quality as stated in Chapter 2. This model allows impairments to be introduced and voice quality to be assessed. Three cases are considered to demonstrate the effectiveness of optimizing the E-Model. 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: Find voice coder given link bandwidth, packet loss level, and link utilization level.
2. Optimization: Find voice coder and packet loss level given link bandwidth and link utilization.
3. Optimization: Find voice coder and link utilization level given link bandwidth and packet loss level

AMPL [KERN2] is the optimization software that is used to solve the problem.

5.1 Description of E-Model Optimization Problem

The E-Model has several qualities that interact with each other. For example, subjective tests have shown that there is a tradeoff between bandwidth and quality when looking at compression methods [TIA116]. There is also a relationship between delay and quality [TIA116]. Refer to Chapter 4 for a detailed description of the variables and constants used in this model.

While the problem seems like it should lend itself to optimization, there are several challenges. First, 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 [ITU107]. In addition, parameters that affect delay and *Ie* are introduced.

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

Next, the relationship between these parameters is 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 5.1 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 5.1 – 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 [TIA116]. 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 [TIA116], which is shown in Equation 5.1, which only applies to high speed link speeds (link speed > coder rate) [TIA116].

$$\textit{Total Delay} = (N+1)*\textit{frame length} + \textit{code look-ahead} \qquad 5.1$$

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 5.2, *S(X)* is the distribution of total delay of the M/M/1 queue [KLEIN], μ is the service rate, and ρ is the link utilization. Equation 5.3 is derived from Equation 5.2. *Td* represents delay for a given packet loss, utilization, and service rate. PL% is representative of packet loss percentage.

$$S(X) = 1 - e^{-\mu(1-\rho)Td} \tag{5.2}$$

$$Td = \frac{\ln(PL\%)}{-\mu(1-\rho)} \tag{5.3}$$

$$T = Hop\ Count * Td + Code\ Delay + Propagation\ Delay + Misl.\ Delay \tag{5.4}$$

The total one way delay is represented as *T* in Equation 5.4. 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 5.1.

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):

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, we relaxed the integer requirement and allowed the program to make fractional assignments. Despite allowing fractional values, the assignment theorem [LUEN] ensures that the solution produced will always exhibit an assignment of 1 or 0 for every *Portion* variable.

The *Code_Feas* variable is a binary variable that penalizes coders that do not meet the constraints and limits the working set to $R \geq 70$. 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. In essence, it removes that coder from its working set. This is an implementation of a penalty function as described in [KERN2]. One pitfall of using the hard (binary) penalty function is that it is nonlinear. 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 $R \geq 70$ constraint, it MUST look at all *Code_Feas* variables and reverse them if necessary.

Further looking into the complexity of non-linear programming, suppose we wrote out the equations for Case 1 (with the link bitrate equal to1544 Mbps), calling *Code_Feas* *‘c’* and *Portion ‘p’* we have (note: the values for *Is* and *Id, Ro* came from AMPL):

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 logical direction from this point would be to assume that the constraint $R \geq 70$ is not active and choose a starting point (like **p =** [1 0 0 0 0} and **c =** [1 1 1 1 1]). Next a system of active constraints would be put into matrix form (forming the ***A*** matrix). A direction matrix **d** would be formed from **A** and **f** (the objective) using the following relationship (gradient projection method) [KERN]:

$$P = I - A^T(AA^T)^{-1}A \qquad 5.4$$

$$d = -PDf(x)^T \qquad 5.5$$

Unfortunately, since *c* is binary, it will always be active non-linear and will never be linear (not even in a small subspace). MINOS, the non-linear solver in AMPL [KERN2], has an interesting way of dealing with this. The solver looks for a couple of clues. First, is the objective non-linear? Second, are the constraints non-linear? Since c is binary, both are non-linear. Knowing this, the solver takes the following approach. To account for the non-linear objective function, the solver uses a reduced gradient approach. The solver then defines a system of basic variables and superbasic variables. It uses a quasi-Newton algorithm to find the search direction and step length [KERN2].

To deal with non-linear constraints, the solver approximates the non-linear constraint with a linear constraint and adds a penalty and Lagrangian term to the objective to compensate for the approximation. The resulting sub-problem is then solved using a reduced gradient approach. The following section shows the results of the optimization tests.

5.2 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 ($R \geq 70$).

- 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

This algorithm was not set up to determine the number of calls as an integer. This was done for a technical reason. The MINOS solver that AMPL uses to solve non-linear problems will not solve non-smooth (integer) problems reliably. Therefore, if two combinations produce the same integer number of calls, the highest actual number will be considered the best selection.

5.2.1 Case 1 - Optimizing for Coder Selection

Case 1 was run for two different link speeds: 256 kbps and 1.544 Mbps. When the link speed was 256 kbps, the objective returned was 4.3 calls. G.729A was the coder chosen by the optimization. We can see from Figure 5.1 that G.729A was the only coder that was feasible. The data for Figure 5.1 was generated by the optimization program.

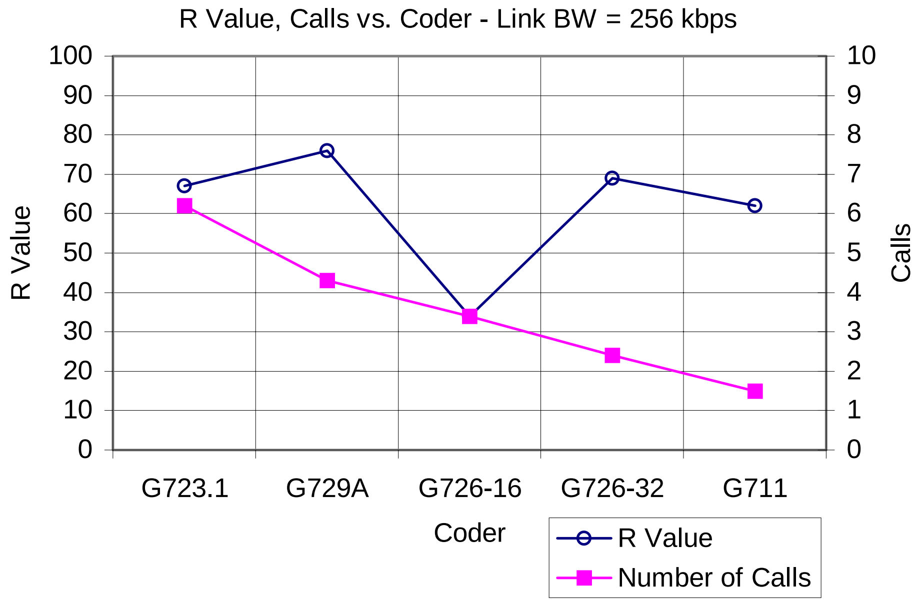

Figure 5.1 – Case 1, Link Rate = 256 kbps

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 Bitrate	256000
Link Utilization	0.5
Packet Loss Level	0.005

Table 5.2 - Case 1, Link Rate = 256 kbps

For Case 1 with a link speed of 1.544 Mbps, G.723.1 was selected as the optimum coder by the optimization routine. The objective function found 37.1 calls. Figure 5.2 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 a more efficient but lower quality of voice. The significant decline in the *R* value of G726-16 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).

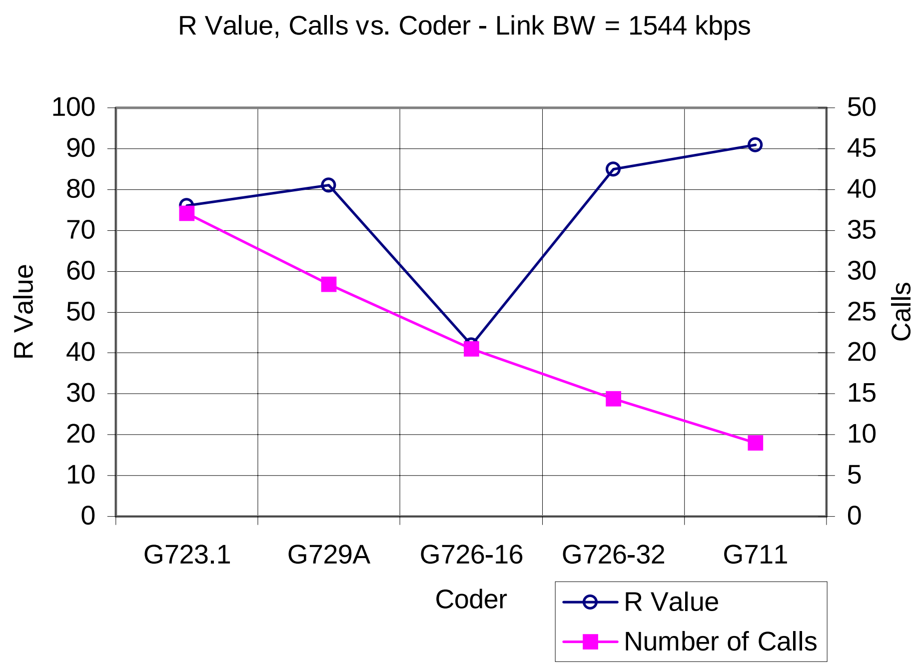

Figure 5.2 – Case 1, Link Rate = 1.544 Mbps

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 Bitrate	1544000
Link Utilization	0.5
Packet Loss Level	0.005

Table 5.3 - Case 1, Link Rate = 1.544 Mbps

An interesting observation is that the results generated for both the *R* value and number of calls is similar except for the scale. Even though the results are similar, the G.723.1 coder came into a feasible position on the 1544 kbps test.

5.2.2 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 2.4). A polynomial fit was completed for each of the three coders. For G.711, the following polynomial was generated, where *x* represents the level of packet loss and *y* represents the level of impairment (*Ie*).

$$y = 0.0037x^3 - 0.1376x^2 + 3.4978x + 0.6663 \qquad 5.6$$

Figure 6.5 shows a graph of the observed versus the curve fit. The estimated data was not more than .5 points from the observed data at any point.

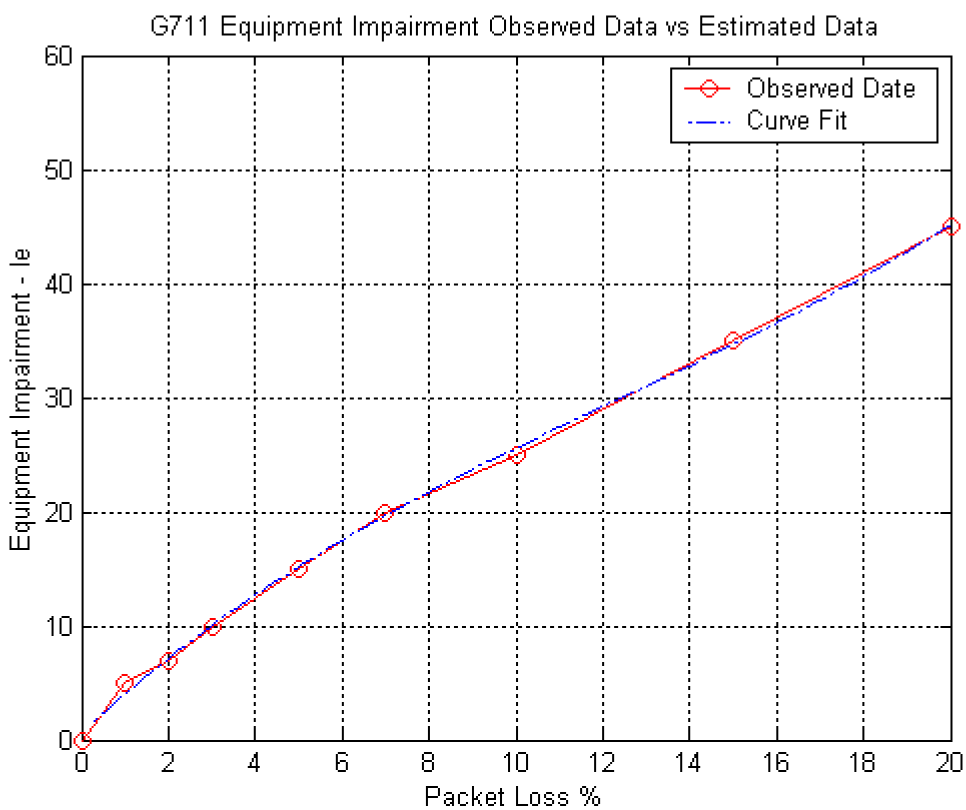

Figure 5.3 – G.711 Polynomial Fit

For G.729A, the following polynomial was generated:

$$y = 0.0051x^3 - 0.2197x^2 + 4.5869x + 10.7880 \qquad 5.7$$

Figure 6.6 shows a graph of the observed versus the curve fit. The estimated data was not more than .5 points from the observed data at any point.

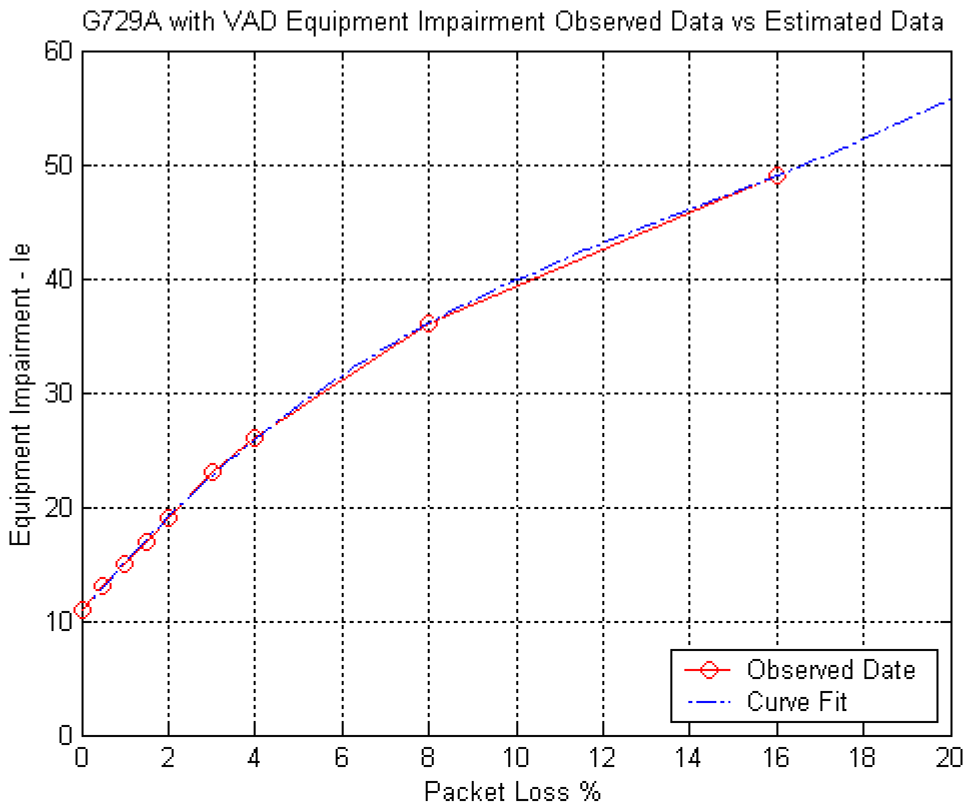

Figure 5.4 – G.729A Polynomial Fit

For G.723.1, the following polynomial was generated:

$$y = 0.0087x^3 - 0.3094x^2 + 5.2348x + 14.5982 \qquad 5.8$$

Figure 5.5 shows a graph of the observed data versus the curve fit. The estimated data was not more than 1.0 points from the observed data at any point.

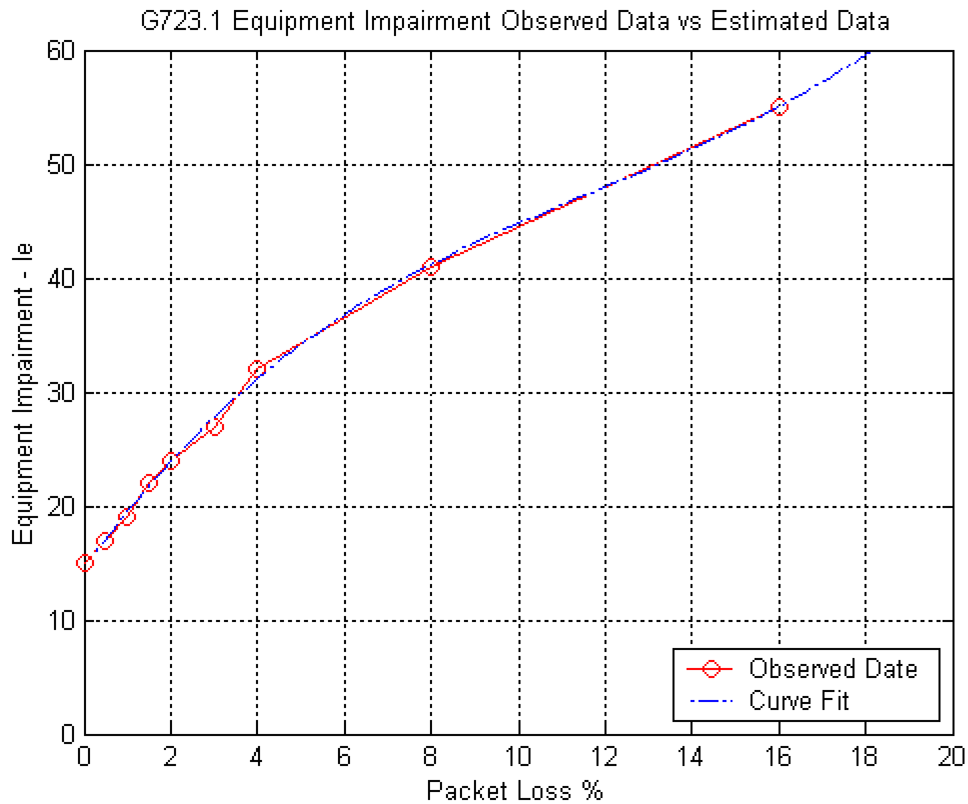

Figure 5.5 – G.723.1 Polynomial Fit

The curve fit was sampled at 1, 2, 3, ...9 %. and used as a *set* parameter in the data file. 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 1.544 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.

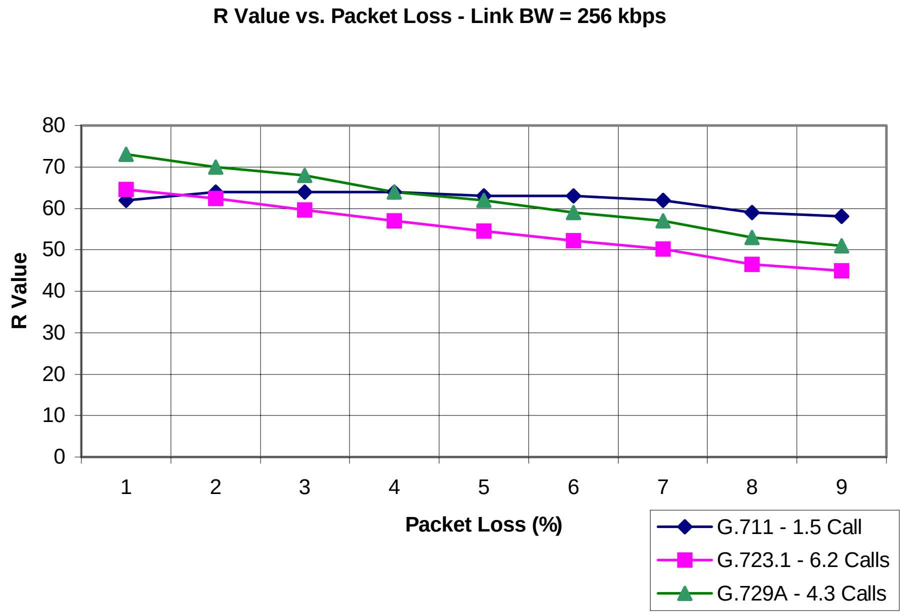

Figure 5.6 – Case 2, Link Rate = 256 kbps

	R - Value		
Packet Loss %	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	1.5	6.15	4.3
Link Bitrate	256000		
Link Utilization Level	0.5		

Table 5.4 - Case 2, Link Rate = 256 kbps

Next, the test was run with a link speed of 1544 Mbps. The objective returned was 37.1. G.723.1 with packet loss of 1% was the combination chosen. Looking at Figure 5.7, we can see that G.711 and G.729A were feasible at least a portion of the packet loss spectrum.

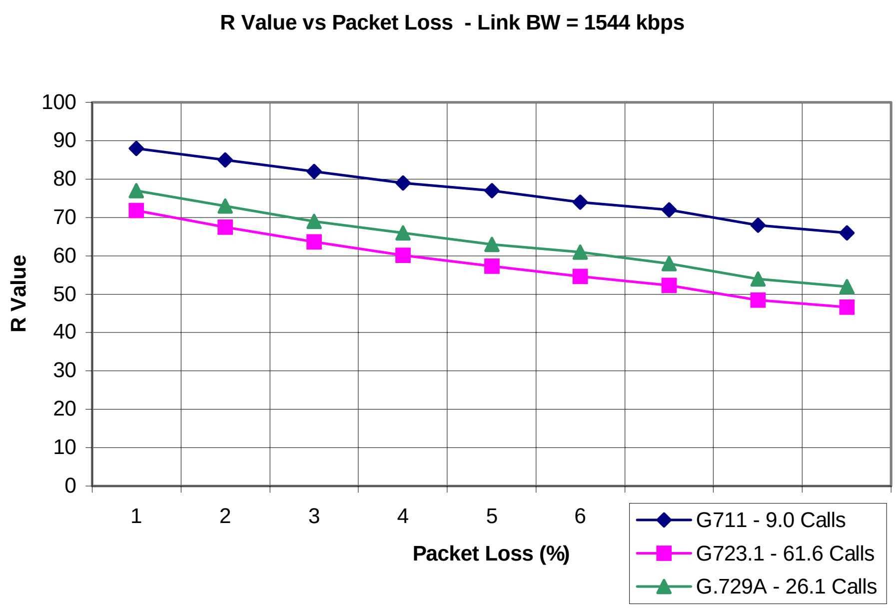

Figure 5.7 – Case 2, Link Rate = 1.544 Mbps

	R - Value		
Packet Loss %	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	9.0	37.1	26.1
Link Bitrate	1544000		
Link Utilization Level	0.5		

Table 5.5 - Case 2, Link Rate = 1.544 Mbps

The algorithm for Case 2 is interesting because it has a modification that allows it to favor higher packet losses (given the choice). The modification is accomplished by adding a small benefit to the objective function (+*portion[i,j]**PacketLoss[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.

5.2.3 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 5.8, we can see that the G.729A coder was in a feasible range till link utilization reached approximately 62%.

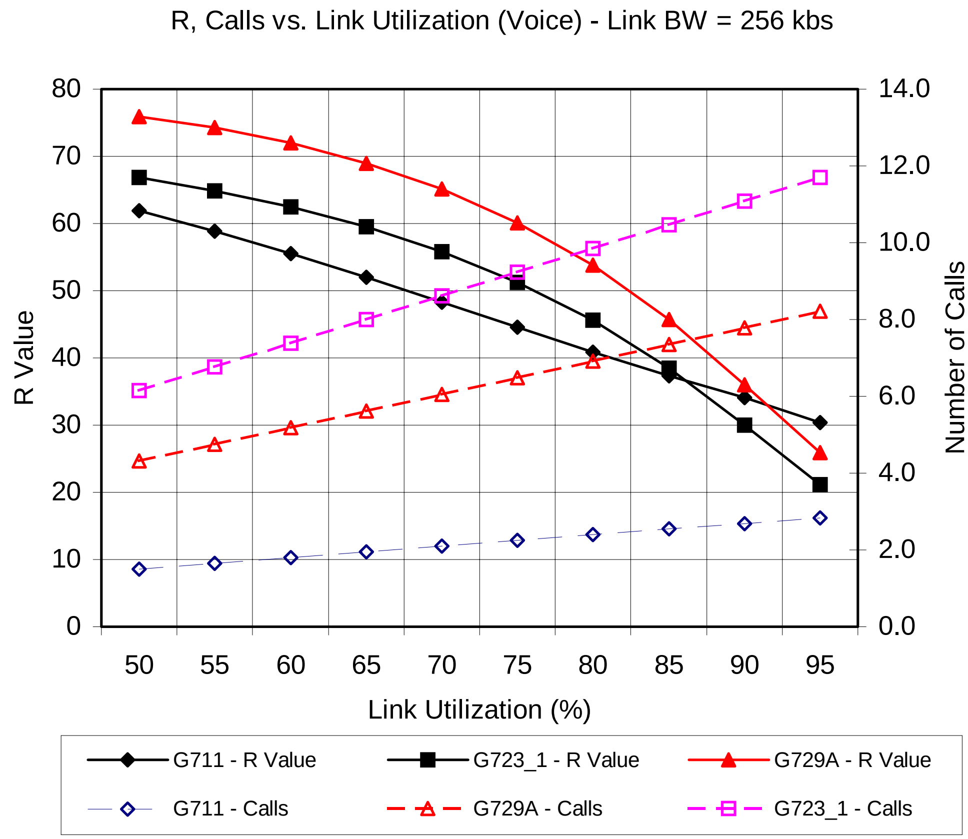

Figure 5.8 – Case 3, Link Rate = 256 kbps

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 Bitrate	256000
Packet Loss Level	0.005

Table 5.6 - Case 3, Link Rate = 256 kbps

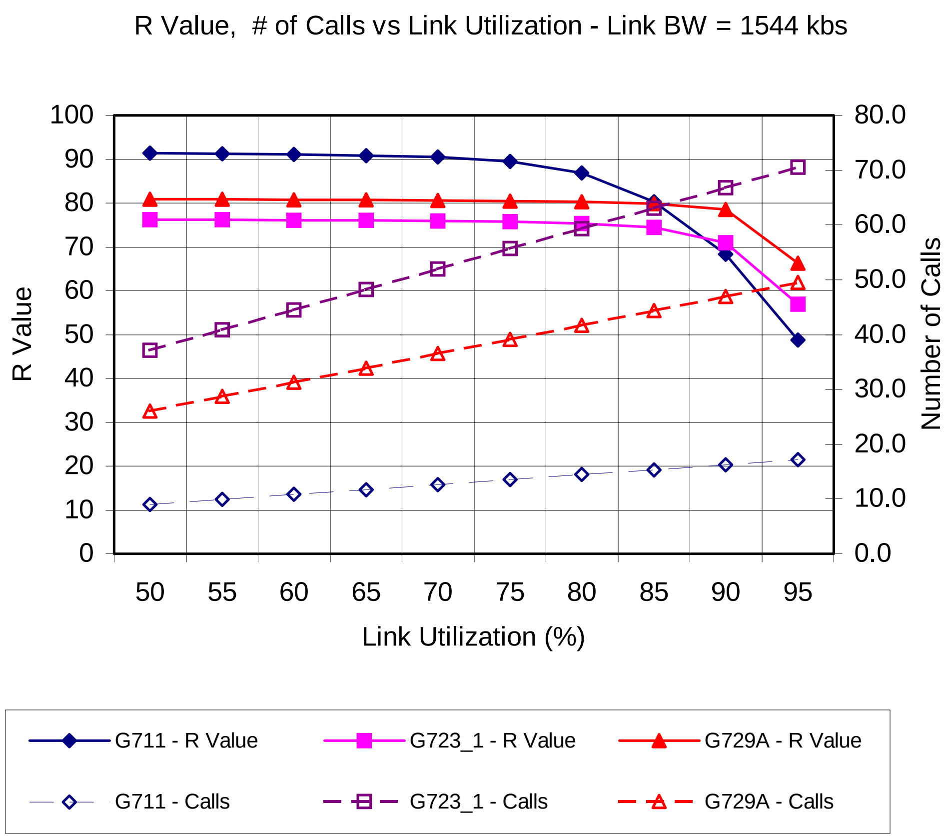

Figure 5.9 – Test 3, Link Rate = 1.544 Mbps

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 Bitrate	1544000
Packet Loss Level	0.005

Table 5.7 - Case 3, Link Rate = 1.544 Mbps

For Case 3 with a link bitrate of 1.544 Mbps, the objective returned was 66.8. G.723.1 running with 90% link utilization was the combination chosen. Looking at Figure 5.9, 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.

5.3 Discussion of E-Model Optimization Results

This chapter utilized the E-Model to assist with the selection of parameters important to mission critical networks. These include the voice coder, allowable packet loss, and allowable latency. 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 5.8 reviews some of the results of the optimization problem.

Case #	Variables	Link Bitrate (b/s)	Optimum Solution
1	Coder	256000	G.729A
1	Coder	1544000	G.723.1
2	Coder, Packet Loss %	256000	G.729A with 2% PL
2	Coder, Packet Loss %	1544000	G.723.1 with 1% PL
3	Coder, Link Utilization	256000	G.729A with 60% Load
3	Coder, Link Utilization	1544000	G.723.1 with 90% Load

Table 5.8 - 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 favorable 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 bitrate was 256 kbps. When the link bitrate was 1.544 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 (using M/M/1 assumptions).

Case 3 with a link bitrate 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 1544 kbps, 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. It would be interesting to run the same optimization using a priority queue instead of an M/M/1 queue for assumption about delay. This may possibly allow users to maximize lower priority traffic bandwidth available in addition to maximizing calls.

In all tests, a delay was assumed (based on the M/M/1 estimations). 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. If higher delay bounds were derived using different models (see Chapter 3), those delay bounds would need to be used in the optimization. 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 mode.

Although this model has potential to be very useful in helping to select coders and allowable parameters like utilization and packet loss, it is not without weaknesses. In all three tests, G.729A and G.723.1 were near enough to an *R* Value of 70 that any error in the estimates for delay or packet loss may cause an error in the optimization. In these situations, the optimization is considered to be sensitive to changes in those parameters.

The ability to analyze various coders, delay and packets loss are vital to assessing voice quality in mission critical voice networks. The optimization of the E-Model provides a tool that is useful for this purpose. Chapter 6 looks at latency issues in mission critical networks.

Chapter 6

Latency in Voice over IP networks, Simulation Results

This section looks at latency associated with mission critical voice over IP networks in two situations. The Remote Network model simulates the edge of the network where the voice traffic encounters a router connected to a link with limited bandwidth. In the Air Traffic Control environment, this would occur at the edge of the network in a remote facility that only has a low bit rate tail circuit serving the facility. The Core Network model simulates the core of the network. The goal for this simulation is to determine if it is possible to meet the criteria for delay in a typical mission critical voice network as shown in Figure 6.1. This model simulates over 2000 voice sources.

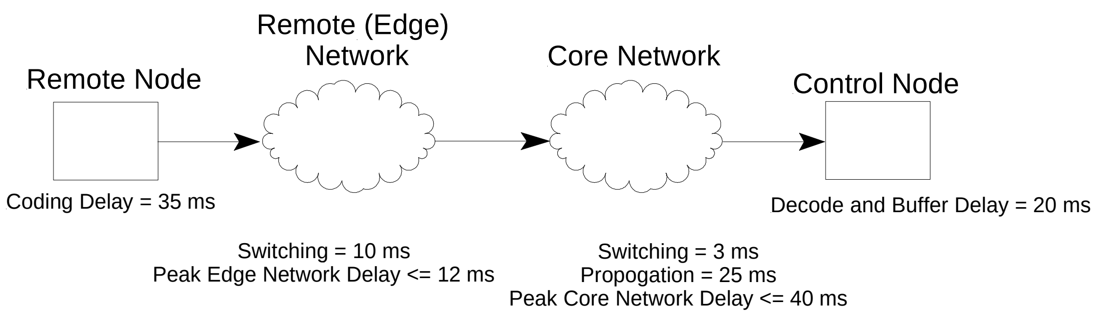

Figure 6.1 - Delay Analysis of Mission Critical Voice Network

Figure 6.1 also shows typical delay times for various parts of the example network. The data used for typical delay times was derived from recommendations in [TIA116] for a network using G.729 coding. The maximum edge network delay and the maximum core network delay was determined as follows. For networks carrying Air Traffic Control voice, [NADLER] stated that a "step on" (see Chapter 2), can occur with delays above 150 ms. Other sources discussed in Chapter 2, link maximum voice delays to approximately 200 ms [HERSENT][TIA116]. This simulation was designed with a 145 ms peak delay as the goal for a single hop edge network and a three hop (core) network. As Figure 6.1 shows, 12 ms and 40 ms are the goals for the peak delays in the remote and core networks respectively.

The model that was used to collect data about latency was a discrete event simulation at the IP network layer. This was accomplished by generating and queuing IP packets that were various sizes and had varying interarrival times depending on the type of traffic. The software that was used to create the models and execute the simulation is OPNET Modeler by OPNET Technologies, Inc [OPNET].

6.1 Model Description

Two models were created to simulate the two previously described situations. The Remote Network model simulates an edge VOIP router that has 6 voice inputs and 2 data inputs. This is a possible network design for a mission critical network that has remote facilities. The Core Network model is a three hop model that has varying link speeds and a large quantity of voice sources. A detailed description of the model components and validation is included in Appendix 1.

Differentiated Services 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 packets. For data, the packets are marked as best effort. These details are included in the descriptions of the modules.

The overall load, load of voice versus data, maximum packet sizes, queuing mechanism, and data source distribution are varied to provide the different scenarios. The next section describes the Remote (Edge) Network Model.

6.1.1 Remote (Edge) Network Model

Figure 6.2 shows the network diagram that was used to simulate a VOIP implementation of a remote site. In the Remote Network model, 12 ms is considered the maximum delay that is allowable. This model is at the IP network layer and focuses on delays as the primary criteria for success. The source model generates IP packets. The network (where it is required) uses static routing and does not focus on dynamics associated with routing. The model does, however, focus on scheduling mechanisms, shaping and policing, and network utilization. As stated earlier, the router uses the DSCP to decide how to schedule the packets.

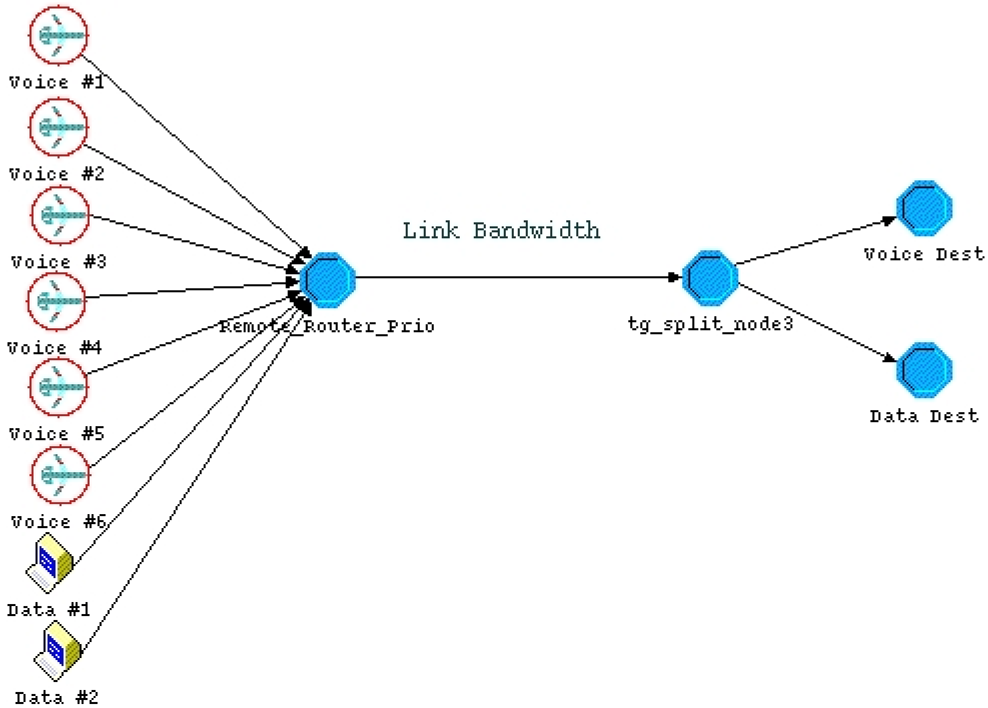

Figure 6.2 - Remote Site (Edge) VOIP Model

(Shown for a Priority Queue)

The Remote Site model focuses on delays associated with the low bit rate link leaving the edge router. Scheduling is accomplished by queuing the packets in a priority queue according the DSCP mark for the packets. The source marks the packets. The Remote Network model is run for 5 scenarios. The default is:

- 6 voice sources, 2 best effort data sources (with exponentially distributed interarrival times and file sizes), priority queue, constant link data rate, constant limit on data MTU size.

The details of each module is in following sections in this chapter. The scenarios listed below show the item that was varied for each scenario:

1. Varied limits of packet MTU size.
2. Varied link data rates, 1 kb limit on data MTU size.
3. Varied proportion of voice load versus best effort load, 1 kb limit on data MTU size.
4. Deficit round robin (DRR) queue instead of priority queue.
5. Scenario 3 with Pareto distribution (file sizes) on best effort data sources

6. Scenario 4 with Pareto distribution (file sizes) on best effort data sources

The intent of the Remote Network simulation is to analyze delays associated with slow links that connect the remote network with the core network. The following section looks at the core of the network and delays associated higher speed links and large quanities of sources.

6.1.2 Core Network Model

The goal of the Core Network simulation is to analyze delays associated with the core of the network and determine if it is possible to meet the 40 ms peak delay criteria through the core of the network, as shown in Figure 6.1. The Core Network model include many voice and data sources (over 2000) and high speed links that egress on low speed links. Figure 6.3 shows the network diagram that was used to simulate a VOIP implementation through the core of a network.

To simulate a core network, routers were modeled that can pass traffic out of an second output port based on the destination address of the IP packet. This is to allow a large amount of traffic to enter a node, which could cause burst accumulation. This network had a total of 2048 voice sources and 32 best effort data sources. There are shapers and meters on the input ports of the Ingress Routers. There are also meters on the outputs of each of the routers. The voice and data sources are the same as in the Remote Network simulation. The average interarrival times of the BE data sources and link speeds are varied to create the necessary loads. The interarrival times and file sizes are exponentially distributed unless otherwise specified. The DSCP is used by the routers to determine the scheduling for that packet.

The Core Network model has three types of routers: ingress, core, and egress. The ingress router implements shaping/policing as well as scheduling. The second output port (pass-thru) allows packets to be sent out without passing through the output queue on the primary link. This simulates normal pass through traffic with output queuing only, but still allows accumulation of burst on input links.

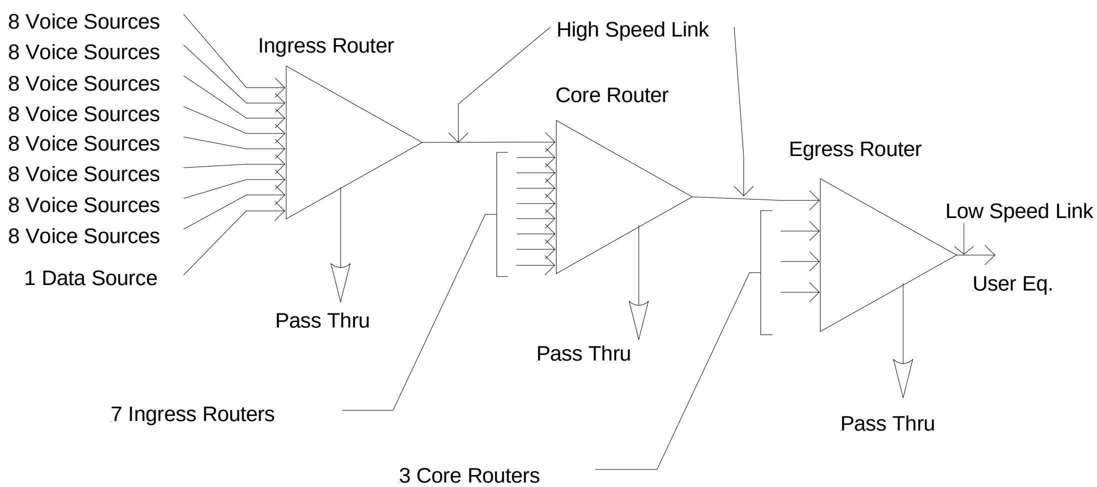

Figure 6.3- Core Network Model

(Variable Link Speeds and Data Sources)

This simulation was set up so that the voice sources have a set destination throughout the duration of the simulation. This is to simulate a network that has a given number of conversations that are "pinned" to a particular path. The data sources generate packets that have random destinations subject to certain probabilities for certain destinations. The loads are controlled by varying the average data interarrival times and the link rates.

Referring to Figure 6.3, we can see that each ingress router has 64 voice sources and one data source that feeds that router. No data is "passed out" of the ingress router. At each core router, there is a total of 8 X 64 = 512 voice sources feeding each core router. Half of the traffic is passed out of the network at that core router. The voice sources chosen to leave the core router, make up 4 of each group of 8 voice sources. The data source randomly assigns a destination to each packet. An average of 50% of the data packets leave the network at each core router. This leaves 1024 voice sources and half of the data traffic entering the egress router. Of these 1024 voice sources, 32 are sent to the main destination. The remaining 996 voice sources are passed out of the network at the egress router. To accomplish this, one voice source in each group of 64 is sent to the main destination. An average of 1.56% of the data packets are marked for the main destination. The remaining data packet are passed out of the network. The scenarios listed below show the item that was varied for each scenario:

1. Varied link data rates, 4 kb limit on data MTU size.
2. Varied proportion of voice data rate versus best effort data rate, 4 kb limit on data MTU size.
3. Deficit round robin (DRR) queue instead of priority queue, 4 kb limit on data MTU size.
4. Scenario 2 with the use of Pareto distribution (file sizes) on best effort data sources.
5. Scenario 3 with the use of Pareto distribution (file sizes) on best effort data sources.

Appendix 1 contains more information about the components of the Core Network and Remote Network model in addition to validation data about the important components of the model. The next section contains the results of the simulation.

6.2 Remote (Edge) Network Latency

Section 6.1 described the goals of the remote model. The goal on this set of scenarios is to determine if the voice traffic can be transmitted across the remote network with less than 12 ms of delay. All scenarios have some common configuration settings. These include:

- All scenarios include 6 voice sources (also referred to as Expedited Forwarding, EF) which are on-off packet generators that generate a 60 byte packet every 20 ms when "on". The bandwidth when "on" is 24 kbps. Since the average "on" time is .4 seconds and the average "off" time is .6 seconds, the average bit rate is 9600 bps. The voice sources are always considered to be higher priority except when a DRR router is used.

- All scenarios include 2 data sources (also referred to as Best Effort, BE). In all scenarios except for 5 and 6, the are file generators that generate files that have exponentially distributed interarrival times and file sizes. In scenarios 5 and 6, the distributions are Pareto with a shape parameter of 1.06 and minimum possible size of 453 bits. The average data file size is 8 kb for all scenarios. The interarrival times are dependant on the data load required.
- The primary parameters that can be set to create the various scenarios are: data MTU size, link speed, data packet interarrival time, and DRR weight for each type of traffic.

All simulations were run for 120 seconds except for the simulations that involved a Pareto generator (5 and 6). Scenarios 5 and 6 were run for 600 sec. Scenario 1 looks at the effect of data MTU differences on latency.

6.2.1 Remote Network Scenario 1 - Data MTU Size

This scenario varies the data MTU size to study the effect on delay The default parameters used for all for this set of scenarios are:

- Link speed of 288 kbps.
- Data interarrival time of .101 sec.
- Data load of .64, voice load of .20
- Data MTU settings of 64 kb, 5 kb, 3 kb, and 1 kb.

Figure 6.4 shows the results of Scenario 1. We can make some important observations from Figure 6.4 . As the data MTU size grows, the latency experienced by the data and voice begin to converge. This is due to the fact that as data MTU sizes grow on a low bit rate link, they begin to dominate the delays experienced on the link. The theoretical delay is defined as the maximum delay that can be experienced by a voice packet. It is defined as:

*(# Voice channels * Voice Packet Size (b) + Data Packet Size (b))/Link Speed (b/s)* *6.1*

Equation 6.1 holds as long as the (*# Voice Channels * Voice Packet Size)/Link Speed* does not exceed the interarrival rate for the voice packets. Equation 6.1 is applicable to all scenarios in the Remote Network model.

It is also interesting to note that the theoretical delay did not deviate from the peak voice delay by a significant amount. This is due to the relatively high probability that a scenario will occur that will cause maximum delay over a course of several minutes. This scenario includes a large data MTU with all six voice sources in the "on" mode. In this scenario, the worst case delay for voice is to wait for the data MTU and 5 voice packets to be serviced before the worst case voice packet is serviced.

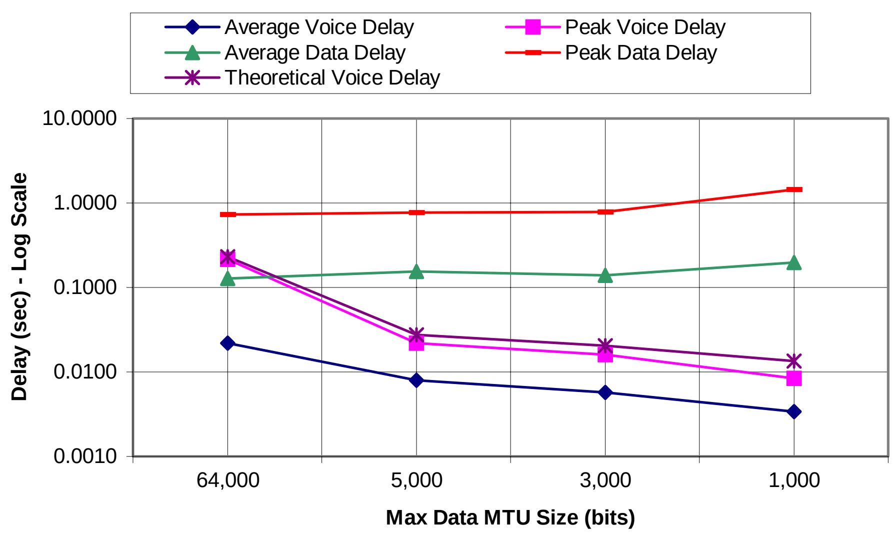

Figure 6.4 - Remote Network Scenario 1 Results

Finally it is interesting to note that the delay for data tends to increase as MTU size decreases. As the data MTU size decreases, two things occur. The first thing that happens is the amount of overhead per file increases, which increases the amount of time needed to traverse link. The second thing that happens is that voice packets are allowed in the stream more often during the transfer of a data file. This reduces the delay for the voice packets but increases the delay in the data packet. Figure 6.5 and Figure 6.6 is a trace of the delay for voice and data for with the data MTU size of 3 kb.

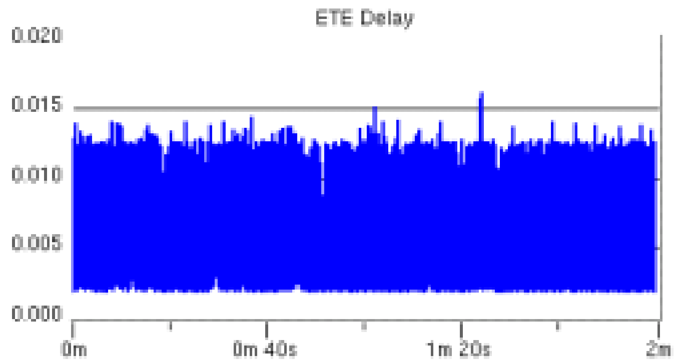

Figure 6.5 - Voice Delay with MTU = 3000 bits

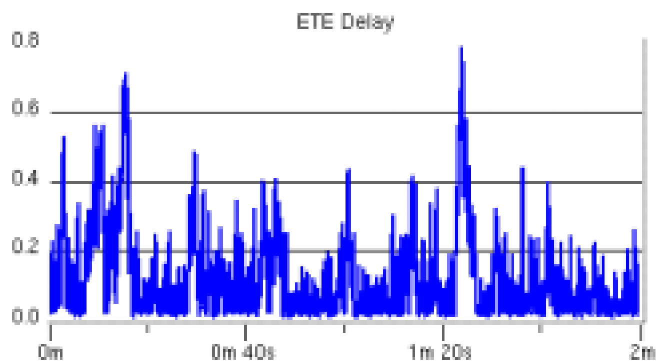

Figure 6.6 - Data Delay with MTU = 3000 bits

The most interesting item to note from Figure 6.5 and 6.6 is that the large delay experienced by the data source at approximately 1m 25s was also experienced in a lesser amount by the voice source. This was probably due to a large data file that produced many maximum sized data packets. The lesson from this scenario is that small data MTU sizes are very important and should be kept small to minimize voice delay.

6.2.2 Remote Network Scenario 2 - Link Rate

This scenario varies the rate of the link from 235,840 bps to 314,453 bps. This had the effect of varying the total load on the link from 67% to 89%. This also has the effect of varying the load of the voice traffic as well as the data traffic. Table 6.2 shows the voice and data load with varying link rates.

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 6.1 - Remote Scenario 2 Utilization Levels

Using the results of Scenario 1 to choose an MTU size, the max MTU for the remainder of the Remote Network scenarios is 1 kb. Figure 6.7 shows the results from Scenario 2. The interesting observation from this scenario is that the voice delay was relatively constant across the all load levels, even when there was obvious congestion on the link. Congestion occurred when the the overall load was nearing .89, as evidences by the rapid increase in the data delay.

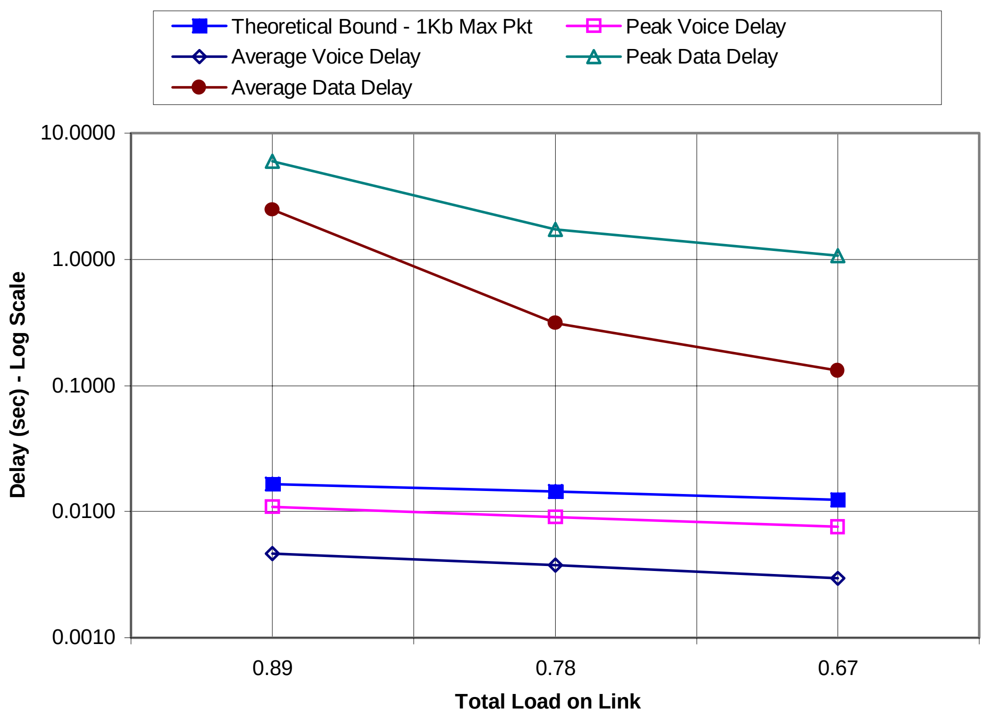

Figure 6.7 - Remote Network Scenario 2 Results

The remainder of the Remote Network model will use a total load of approximately .85. In this scenario, this is the highest utilization level before the voice delay exceeded approximately 10 ms. While the average delay of the data packets will likely be over 1.0 seconds for this load, it is important to stress the network to see the effect on the voice traffic. In reality, a much lower load would probably be required to reduce the delay of the data packets. The next scenario varies the data load with respect to the voice load.

6.2.3 Remote Network Scenario 3 - Voice/Data Ratio

This scenario examines the latency caused by different proportions of data load to voice load. The total load was approximately .84 for all cases. Table 6.2 shows the data loads, voice loads, and total load for all cases.

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 6.2 - Remote Scenario 3 Utilization Levels

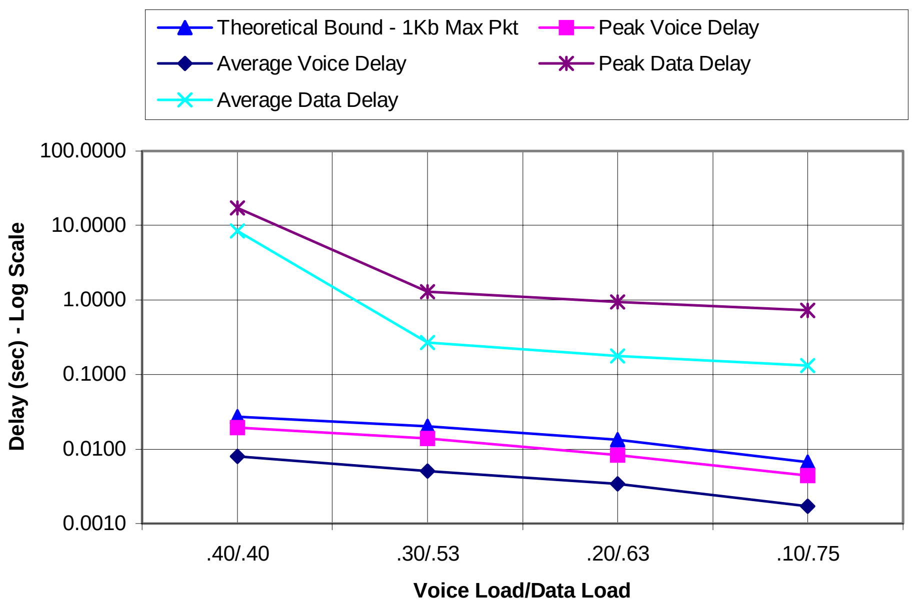

Figure 6.8 - Remote Network Scenario 3 Results

Figure 6.8 shows the results of Scenario 3. An interesting observation is that when the voice utilization climbed above 30%, the data delay suffered by a factor of 10. 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 24,000 = 144,000 bps). This is logical since the voice sources are on/off models with an "on" percentage of 40%.

For the remaining Remote Network Scenarios, a 20% voice load is assumed to be the standard percentage of the load for voice. This represents the need to remain under 12 ms for delay in the Remote Network. The next scenario looks at the differences caused by using a DRR queue instead of a Priority Queue.

6.2.4 Remote Network Scenario 4 - Deficit Round Robin Queue

This scenario looks at the latency in the Remote Network 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 that each class is given each round. Since this scenario has a total quantum of 100 per round, the quantum can also be considered its weight. However if a class does not have enough traffic to use its quantum, that quantum is given to the other classes. Although, it is noted that if the link is not full, the quantum is not technically its weight, this thesis will refer to the quantum and weight interchangeably. Appendix 1 contains a more detailed description of the DRR queue as implemented in this scenario. For example, 95/5, indicates that the voice quantum (weight) is 95 bits and the data quantum (weight) is 5 bits. The following weight combinations were tested: 95/5, 80/20, 65/35, 50/50, 35/65, 20/80, and 5/95. The link speed was 288,000 bps for all cases. The voice load is .20 and the data load is .65. Figure 6.9 shows the results of this scenario.

The voice delay begins to climb very rapidly at approximately the 50/50 weight setting. This can be explained by the fact that with a 288,000 bps link speed, 50 percent of the link is 144,000 bps, which is the exact amount of 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 .65 of 288,000 bps or 187,200 bps.

The data delay is relatively flat through all weighting scenarios. This can be explained by the fact that the voice traffic can only use 144,000 bps (all sources "on"). This is equivalent to the 50/50 setting. So, any bandwidth that the voice traffic has allocated above 50/50 will only be given back to the data traffic and essentially turning the link back into a 50/50 link. This is almost equivalent to a priority queue at this point.

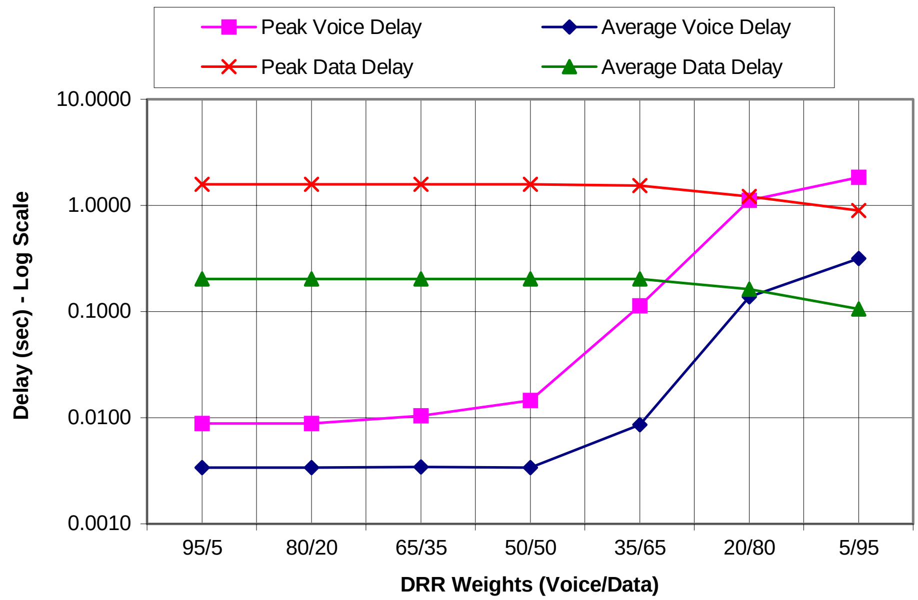

Figure 6.9 - Remote Network Scenario 4 Results

Using a DRR queue with adequate bandwidth given to the voice, is a good way to guarantee the voice latency achieved using a priority queue, while deriving the other benefits of using a DRR queue. These benefits include protection of the data traffic.

6.2.5 Remote Network Scenario 5 - Pareto Distribution with Priority Queue

This scenario is a repeat of Scenario 1 using a Pareto distribution for the file sizes instead of the exponential distribution. The shape parameter used is 1.06 and the k parameter used is 453. This leads to average file sizes of 8000 bits, which is the same as the exponential distribution. Appendix 1 describes the rational behind using 1.06 and 453 for the parameters of the Pareto distribution. Figures 6.10 and 6.11 compare the data collected for the exponential (called M/M/1) distribution (in Scenario 1) compared to the Pareto distribution for file sizes.

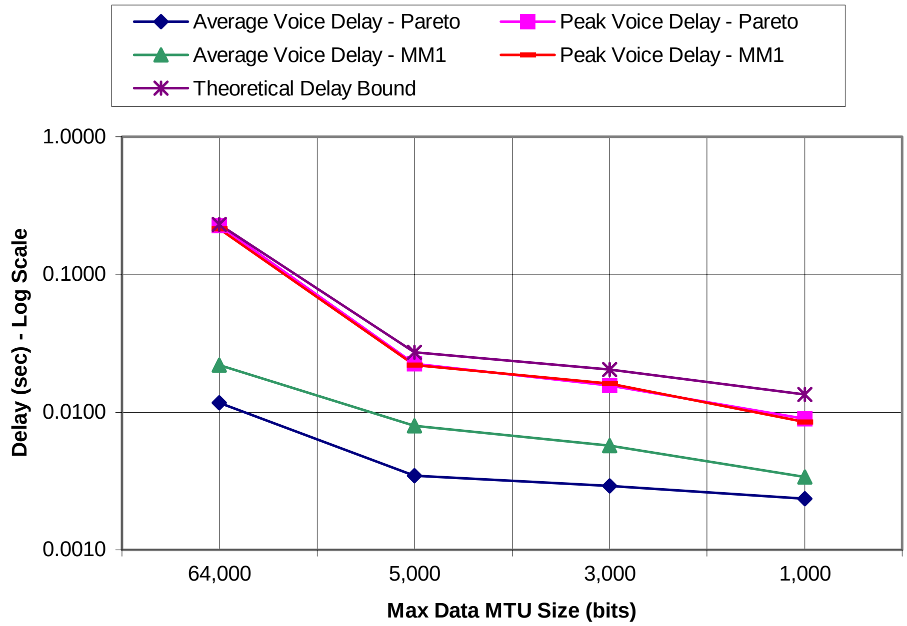

Figure 6.10 - Remote Network Scenario 5 Results (a)

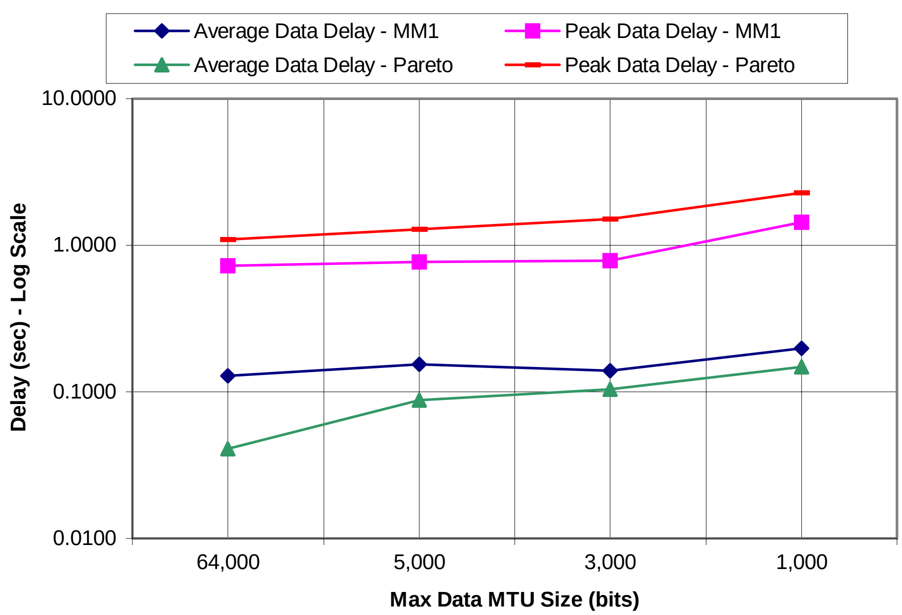

Figure 6.11 - Remote Network Scenario 5 Results (b)

The interesting observations that Figure 6.10 and Figure 6.11 illustrate are as follows:

- The peak delay for voice did not change significantly between the Pareto case and exponential case. The peak delay is based on a worst case situation that involves all voice sources being "on", and having a large data MTU to wait behind. The distribution of data file sizes would not affect this greatly.
- The average delay for voice was less with the Pareto case than the exponential case. This is because the Pareto distribution generally generates many very small file sizes (as compared to the exponential case) and a few very large file sizes (so that the average is the same). These small file sizes tend to make the data load look smaller on the average (except during the transmission of large files).
- The peak delay for data for the Pareto case was between 50% and 200% greater than for the exponential case. This is reasonable considering the potential to generate very large files with the Pareto case. One problem with this simulation is that the largest possible file generated by the data generator was 316,640 bits in this case. This is based on the maximum delay and the link speed. Much larger files sizes are certainly possible. With a longer simulation, it is possible that the results would differ for the voice delay. The result would certainly differ for the peak data delay.
- The average delay for data was less for the Pareto case than the exponential case. This would not have been the case if a very large file was processed. The average delay for data will depend on the number of very large file sizes that are received during the measurement period.

6.2.6 Remote Network Scenario 6 - Pareto Distribution with DRR Queue

This scenario is a repeat of Scenario 4 using a Pareto distribution for the data file sizes instead of the exponential distribution. The shape parameter used is 1.06 and the k parameter used is 453. This leads to average file sizes of 8000 bits, which is the same as the exponential distribution. Figures 6.12 and 6.13 compare the data collected for the exponential (called M/M/1) case compared to the Pareto case for different file MTU sizes.

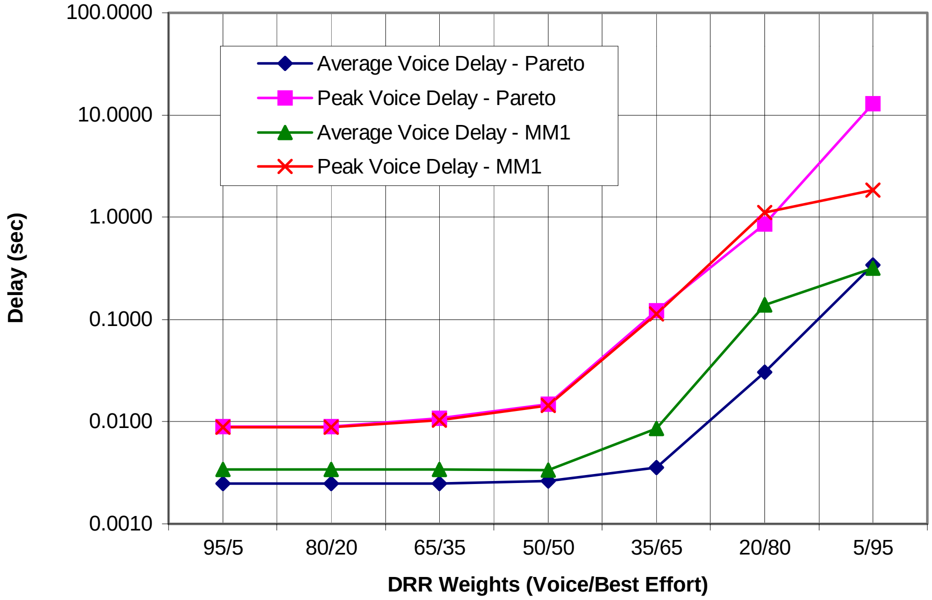

Figure 6.12 - Remote Network Scenario 6 Results (a)

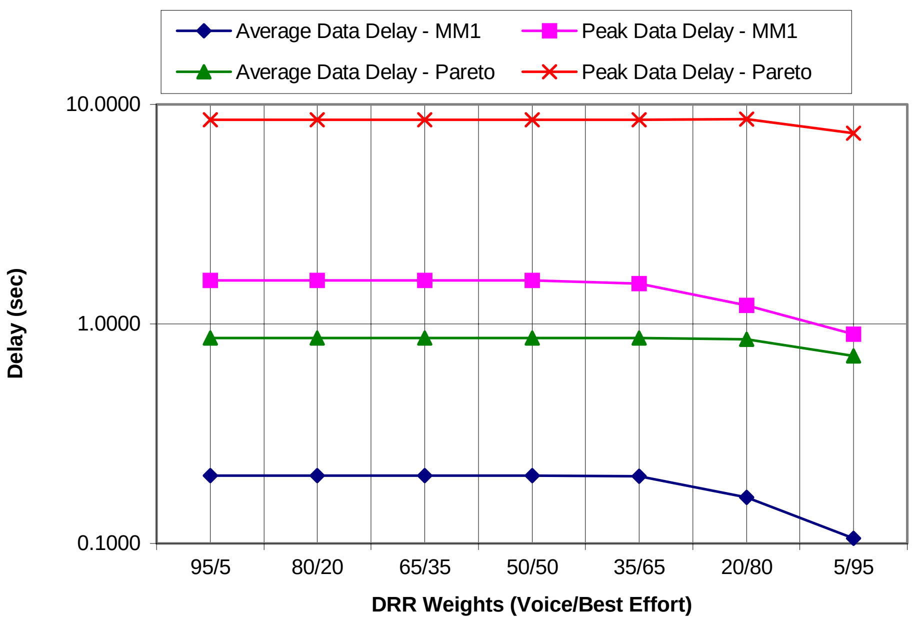

Figure 6.13 - Remote Network Scenario 6 Results (b)

The interesting observations that Figure 6.12 and Figure 6.13 illustrate are as follows:

- The peak delay for voice did not change significantly between the Pareto case and exponential case except for the 5/95 case. This is because it is based on a worst case situation which is: all voice sources being "on", and data MTU to wait behind. The distribution of data file sizes would not affect this greatly. 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. This is because the Pareto distribution generally generates many very small file sizes (as compared to the exponential case) and a few very large file sizes (so that the average is the same). These small file sizes tend to make the data load look smaller on the average (except during the transmission of large files).
- It is obvious that at least one large file was received during this simulation as opposed to Scenario 5. The peak data delay and the average data delay were almost a magnitude greater with the Pareto case as opposed to the exponential case. This is considerably different than Scenario 5.

The next section looks at the latency associated with the core network.

6.3 Core Network Latency

The goal of this portion of the simulation is to look at networks that have the following qualities:

- Mutiple hop (in this case, 3 hops was used).
- Routers that have multiple inputs and outputs, with the potential of having multiple packets arrive at once on an output port.
- Low speed to high speed links.
- High speed to low speed links.

Section 6.1.2 describes the inner workings of the core network in detail. It also lists the scenarios that are created. The first scenario keeps the voice load constant and gradually increases the data load.

6.3.1 Core Network Scenario 1 Results - Link Rate

This scenario maintains the voice load on all links at .5 while modifying the data loads from approximately .16 to .47. The overall load on the links varies from approximately .66 to .97. Figure 6.14 shows the delay observed in this scenario.

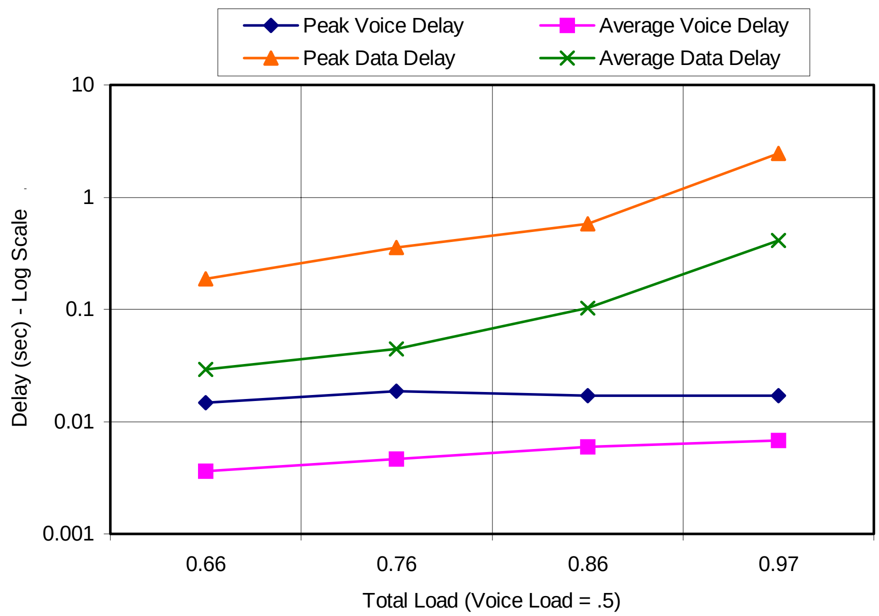

Figure 6.14 - Core Network Scenario 1 Results

The results from this scenario seem fairly intuitive. The data delay climbs with increased data (and overall) load. One interesting point is that the peak voice delay levels at approximately .76 load and maintains that level. The reason for this is because the peak delay does not rely on the data load, only the maximum data MTU size. The average voice delay does continue to climb. This is logically explained by example, if there are more data packets, it is more probably 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.

6.3.2 Core Network Scenario 2 Results - Voice/Data Ratio

This scenario maintains a relatively constant total load (.85 to .88) and varies the portion of the load that is voice to the portion of the load that is data. Table 6.3 shows the load configuration for each part of this scenario.

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

Table 6.3 - Core Scenario 3 Utilization Levels

Figure 6.15 shows the results from this scenario. Some interesting observations can be made from Figure 6.15.

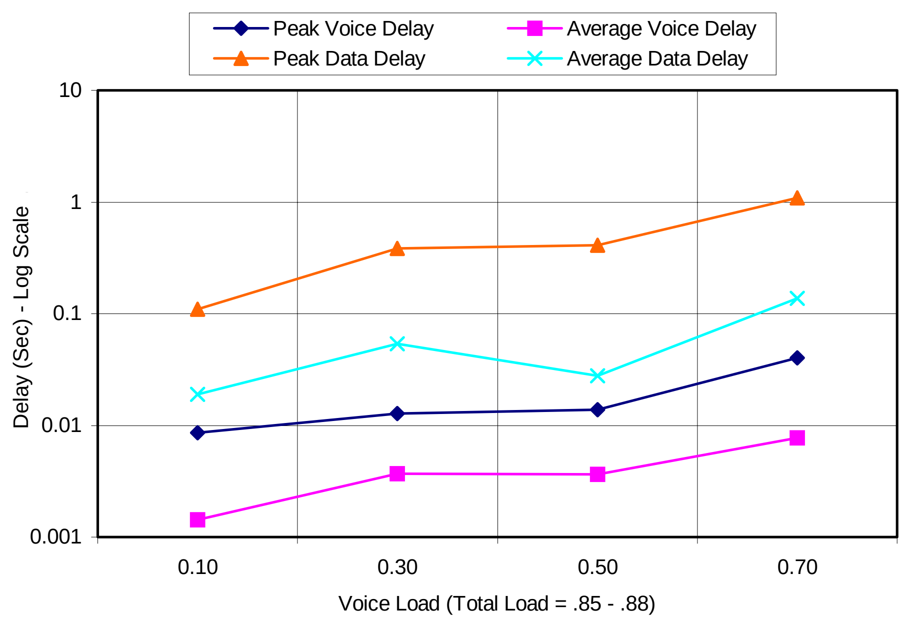

Figure 6.15 - Core Network Scenario 2 Results

- The data packet delays tend to track with the voice packet delays. This is logical because the voice packets have priority, so if they are suffering from queuing delays, then the data packets will be impacted from the same delays.

- The delays for voice and data either leveled or actually decreased at when the voice load was 50% and the data load was 36%. This is an interesting result. A possible reason for the decrease at this point is an "optimization" effect. Since the data load is decreasing and the voice load is increasing, this point has low enough voice load that the voice traffic is not starving the data and therefore we see a decrease in the data delay.

Since the voice sources generate packets every 20 ms, if the voice sources cannot utilize all of the bandwidth, a data packet will begin service every 20 ms. What this does is create a guaranteed bandwidth amount for data as long as the voice traffic below a certain load. Since the voice sources are "on" off type sources and are "on" an average of 40% of the time, the network will typically have a slot for data every 20 ms until the voice load is at least 40%. Note that the slot would need to be able to completely service the data MTU or queuing delays would impact the next set of voice packets. The next scenario uses a DRR queue instead of a priority queue.

6.3.3 Core Network Scenario 3 - Deficit Round Robin Queue

This scenario looks at a constant voice load and data load queued with a Deficit Round Robin (DRR) Queue. The DRR weights that are used for voice/data are 95/5, 80/20, 65/35, 50/50, and 35/65. The voice load for all tests is .5 and the data load is .36 for a total load of .86. Figure 6.16 shows the results of this scenario.

In Figure 6.16, the voice delays increase sharply between DRR voice weights of 65 and 50. The reason is intuitive. Since voice traffic is 50% of the traffic on the link, if the bandwidth available to voice is reduced to a point near 50%, delays should rise sharply.

The next two observation are complicated. First, the data and voice delay (both peak and average) are flat below the 50/50 quantum. Second, the peak voice delays are significantly higher than was seen in the equivalent priority queue model. The explanation of these two observations is related to the fact that the DRR queue in these cases gave the data traffic at least 5% of the queue all the time. The voice traffic waited on data traffic even if there were voice packets waiting. This explains why peak voice

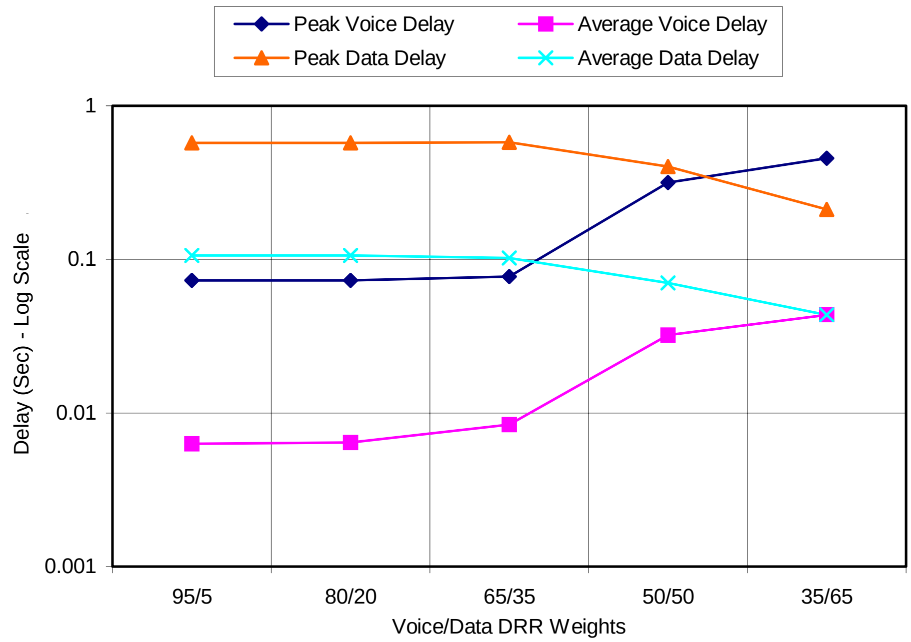

Figure 6.16 - Core Network Scenario 3 Results

delay is higher with the DRR queue than with the priority queue and why the data delays are bounded. The flatness of the delays below the 65/35 DRR weight level is related to the fact that there is only a limited amount of voice traffic possible. Therefore, any bandwidth above the level that corresponds to the weight given to the voice is given to the data traffic which gives both the voice and data the same amount of bandwidth below the 65/35 weight level. The next two scenarios look at the Pareto distribution used for data file sizes.

6.3.4 Core Network Scenario 4 - Pareto Distribution with Priority Queue

This scenario looks at using a Pareto distribution for the file size of the data source with a priority queue. Two maintain consistency, Core Network, Scenario 2 is repeated with the only difference being the use of the Pareto distribution. Pareto shape parameter of 1.06 and *k* parameter of 453 are used. Figure 6.17 and Figure 6.18 contains the results of this scenario compared with the results of Scenario 2.

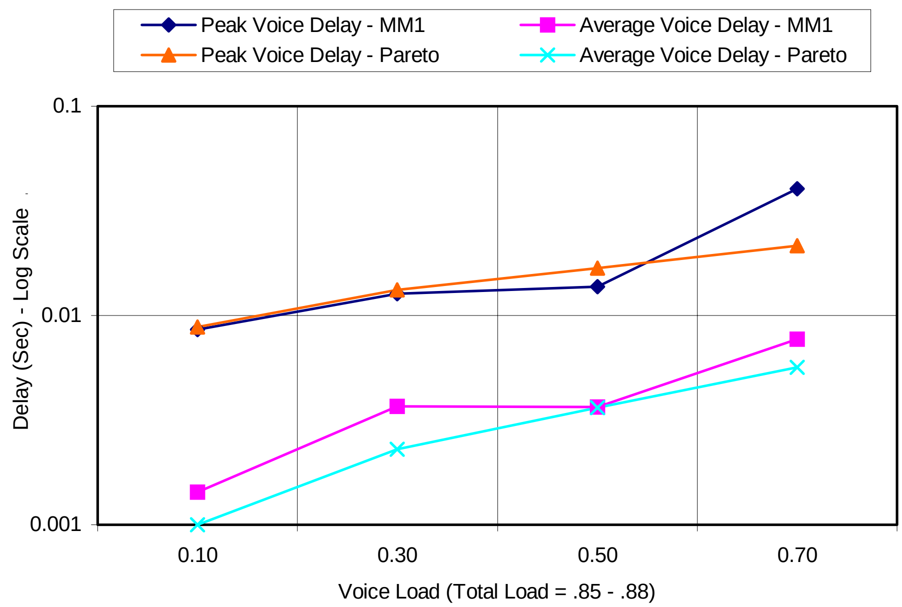

Figure 6.17 - Core Network Scenario 4 Results (a)

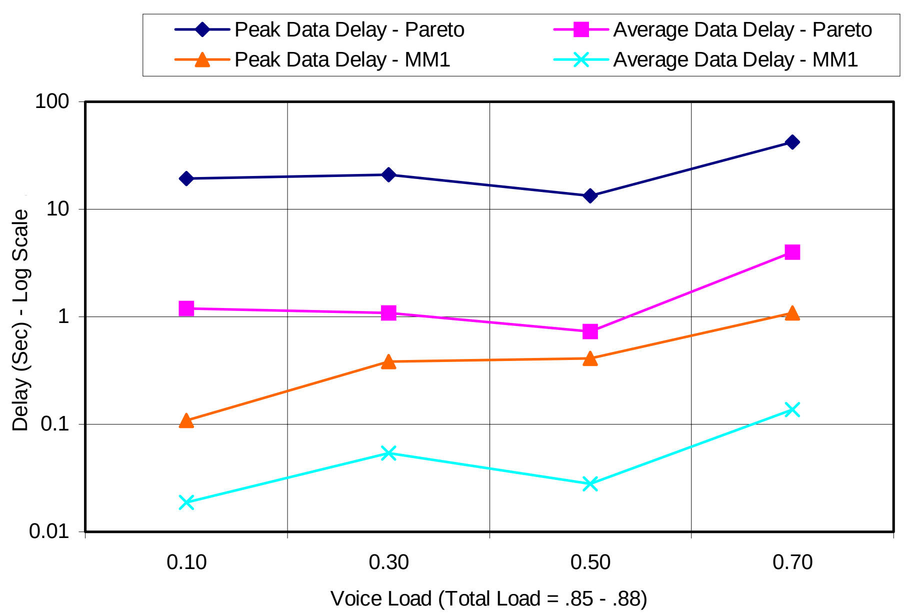

Figure 6.18 - Core Network Scenario 4 Results (b)

The interesting observations that Figure 6.17 and Figure 6.18 illustrate are as follows:

- The peak delay for voice did not change significantly between the Pareto case and exponential case except when the voice load was .7. If the voice load is less than .4, the voice packets cannot use their allotted bandwidth and the data will be given the extra bandwidth. When the voice load rises above .4 (especially as high as .7), the Pareto distribution will make a difference by generating more MTU size packets at once than the M/M/1 case. These would tend to have an additive effect.
- For the data delays, the differences between M/M/1 and Pareto were substantial. Most of the time the difference was an order of magnitude. This is consistent with the behavior of the Pareto distribution. Very large file sizes that are broken into many MTU sized packets would tend to saturate queues, and have long play out times.

6.3.5 Core Network Scenario 5 - Pareto Distribution with DRR Queue

This final scenario looks at the effect that a Pareto distribution for data file sizes has on the DRR queue. Scenario 5 is the same simulation as Scenario 3 with the Pareto distribution for data file sizes. The results of Scenario 3 and Scenario 5 are compared. The Pareto shape parameter used is 1.06 and k parameter used is 453. Figure 6.19 and Figure 6.20 give the results of this scenario.

The interesting observations that can be made here are:

- The similarities between Figures 6.19 and 6.20 versus Figures 6.12 and 6.13 (Remote Network model with DRR queue and Pareto data file size distribution) are very interesting. The general shape of the corresponding graphs are similar. In both cases, the rapid rise in voice delay appears to be associated with the point that voice traffic begins to encounter congestion.

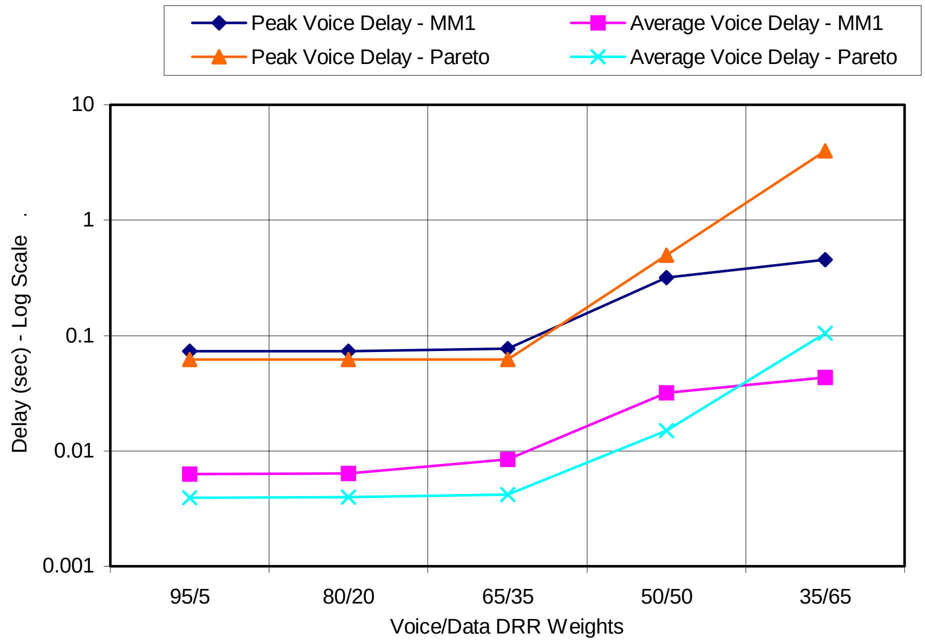

Figure 6.19 - Core Network Scenario 5 Results (a)

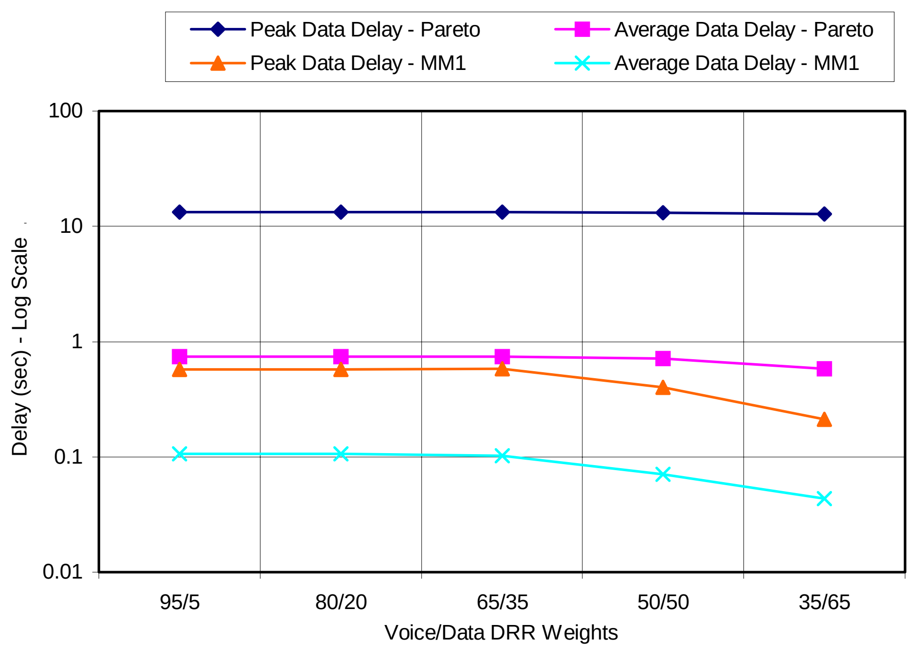

Figure 6.20 - Core Network Scenario 5 Results (b)

- As the DRR weight goes from 95/5 to 35/65 (Voice/Data), the data delays when using the Pareto data traffic generator are flat compared to the M/M/1 data traffic generator. However, the voice delay, when the Pareto data traffic generator is being used, continues to rise at a rapid rate when the voice weight is being reduced as opposed to the M/M/1 data traffic generator, which moderates slightly. One possible cause of this effect is that the Pareto distribution for file sizes has the ability to generate very large files. Since delay is measured from the time the file started transmitting, it would tend to be dominated more by serialization and packetization delays than by queuing delays (as smaller file sizes would).

6.4 Discussion of Latency in Voice over IP networks

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

- The interarrival time between voice packets (20 ms for this simulation) is extremely important in the determination of load and the calculation of delay. In both the Remote Network and Core Network cases, we saw that if enough voice source are transmitting to completely fill the time slot (represented by the voice interarrival time), the peak voice delays can rise rapidly.
- The assumption that was made on the voice generator of 40% "on" time also played an important role. This number typically helped to determine when delays were likely to be bounded. This was seen in simulation not derived mathematically.
- Deficit Round Robin (DRR) routers have a place in VOIP networks because they can protect the data, as was shown in Figure 6.9 and Figure 6.16. However, it was also shown that DRR routers can compromise the voice service if the router is not provisioned properly. Typical use may be to give most of the quantum to the voice source and leave enough quantum for the data that it does not starve and suffer high delays in the event that a voice source misbehaves.
- DRR routers had a potentially prohibitive effect on voice delay when used in the Core Network model. This was shown in Figure 6.19 and Figure 6.16. In both of those figures, maximum delays of more than 60 ms were seen regardless of the

DRR weighting. This is probably due to the fact that the DRR router will let a data packet start if that "data queue" has built up enough weight (quantum). This occurs even if the of the delay in the "voice queue" is high. This effect was not seen when using priority router (as Figure 6.15 shows) or when using DRR routers in the Remote Network model.

- Maximum MTU size is very important in the Remote Network (Edge) configuration. Figure 6.4 shows the impact that large packet and MTU sizes have on delay. Of course, the link rate is equally important. A 20 kb packet will have less effect on the delay with a link speed of 1.5 Mbps as opposed to a link speed of 256 kbps.
- As the work that was done by [CHARNY] and reviewed in Chapter 3, showed that with multiple hop counts, analytical calculation of maximum possible delay can lead to very high delay times for relatively low link utilization levels. This study did not see the delay times in simulation for the same levels of utilization noted in [CHARNY]. This was probably for a number of reasons.
  1. The networks in this simulation were tightly controlled with respect to load and data MTU size. Also, the voice sources were all the same and were also tightly controlled.
  2. In order to see high latency levels in this simulation, many of the voice sources would have to be "on" at once. With a 40% chance of being on, the probability of a very high percentage of the sources being "on" is very small. Certainly in the Remote case, where there is only 6 voices, the probability of all sources being "on" is more likely (4.1%). But, as the network grows, the probably of all sources being on decreases rapidly.
  3. The analysis in [CHARNY] relied upon many packets arriving at once destined for a particular output (essentially flooding that output). In this simulation, this was possible however unlikely due to the fact that each source is only "on" 40% of the time.

In Figure 6.1, a set of criteria to achieve acceptable voice over IP service was shown. The two main components of that diagram that this study was concerned with were maximum network delays in the edge (Remote) network (12 ms) and the maximum network delays in the core network (40 ms). This study showed that is possible to achieve that criteria but there are some significant issues.

First, Figure 6.1 allowed 12 ms for maximum network delay in the Remote Network. In Figure 6.12, we can see that with 65/35 DRR (voice/data) weighting or higher voice weighting, maximum delays did not exceed 12 ms (even if Pareto distributed data file sizes are used). But, in Figure 6.8, it is evident that if voice utilization rises above .2, voice delays can climb beyond 12 ms. It was also shown in Figure 6.10 that with .2 voice utilization and a Pareto distribution for data file sizes, maximum voice delays were too high (>16 ms).

Second, Figure 6.1 allowed 40 ms for maximum network delay in the Core Network. In Figure 6.17, we saw that 40 ms was achievable even with voice utilization as high as .50 (at least with a 3 hop network). This is using a strict priority queue that gives the voice the top priority. When a DRR queue was used with .50 voice utilization, maximum delays were greater than 60 ms for all DRR weighting levels and any distribution for data file sizes (Pareto or exponential). This raises concerns about the applicability of the DRR queue in this particular situation.

Overall, it appears that it is possible to use IP to transport voice over IP. But, it is reliant on strict control of the network and voice coding parameters (voice packet interarrival times, overall utilization, maximum MTU size, voice utilization, and file size distributions). Scheduling in the core of the network is also an issue. Priority type queues are satisfactory for the top priority traffic, but allow starvation of lower priority traffic which may or may not be acceptable. The impact on lower priority traffic would need to be researched. Chapter 7 contains a discussion about the interlocking roles of the various types of analysis that this research has addressed.

Chapter 7

Architectural Issues for Mission Critical Voice over IP Networks

This thesis has looked at various components of Mission Critical Voice over IP. This has included latency, survivability, and voice quality. What this chapter attempts to convey to the reader is the importance of an integrated approach with respect to these components.

As we saw in Chapter 4, survivable network design depends partially on the integration of survivability design for the various network layers in order to work efficiently and correctly. This Chapter argues that this is also a requirement with respect to the various components of mission critical voice, latency, survivability, and voice quality.

Section 7.1 and 7.2 look at the issues with integrating the various component of mission critical voice over IP.

7.1 Big Picture Approach - Integrating Layers for Latency, Survivability, and Voice Quality

The primary contribution of the E-Model (discussed in Chapter 4 and 5), is that it clearly shows the relationship between latency, packet loss and voice quality. Increased latency and packet loss cause lower voice quality. While this was always intuitive, the E-Model provides a way to quantify this opinion.

Chapter 3 reviewed the contribution that [TIPPER][ROGERS] made by demonstrating the relationship between network failure and latency (or available bandwidth). When link failures happened, the availability of bandwidth decreased and delays increased. Based on the work on the E-Model, a relationship between network failure and voice quality could be derived.

Chapter 6 studied sample remote and core networks. Simulation was used in this portion of the research to establish that maximum delays are related to network configuration. This is important because latency causes a decline in voice quality. These contributions make one point fairly clear:

Latency, survivability, and voice quality are interrelated and should be analyzed together when designing mission critical voice over IP networks.

This research has looked at several tools used to analyze IP networks and used them to analyze example VOIP networks. The next section presents a methodology for analyzing mission critical VOIP networks.

7.2 Proposed Mission Critical VOIP Design Philosophy

This research is proposing a design philosophy that uses a multiple stage approach to designing mission critical VOIP networks. It encompasses three main areas:

1. Voice Quality Analysis. This is accomplished in this research by the use of the E-Model optimization in Chapter 5.
2. Latency Analysis. This is used to determine delay bounds on the network. This is important because it sets bounds on what can be accomplished with a given network. Chapter 6 looks at a simulation to determine delay bounds.
3. Survivability Analysis. This research did not address a survivability analysis. It is a topic for future research.

It should be noted that each of these analysis require inputs from the other two analysis. It would be reasonable to expect that once an analysis has been completed (like the latency analysis), that that data may not agree with the data used in the voice quality analysis. Then the voice quality analysis may have to be completed again with the new data. While this may be tedious, it should produce high quality results. Chapter 5 and 6 look at an implementation of a voice quality analysis and a latency analysis. The conclusions from this research are in Chapter 8.

Chapter 8

Conclusions

This research studied several aspects of mission critical communications and in particular mission critical voice over IP. First, mission critical communications was defined and the qualities of mission critical voice were studied. The background that the author has in air traffic control communications was used to provide a basis for this research. The models used in this research simulate a possible air to ground air traffic communication scheme.

The research focuses on two areas of mission critical voice over IP, voice quality and latency. These areas were chosen because they (along with survivability and availability) are the primary qualities associated with mission critical voice over IP.

8.1 Lessons Learned

The lessons learned from this research are numerous, indicating the complexity of transmitting mission critical voice over IP. The primary conclusion from the research is that a mission critical voice over IP network is feasible given that certain stringent requirements are met. First, an E-Model optimization was completed to determine delay, coder type, and other crucial parameters. Second, on the edge of the network (where link bitrates are small), packet size and the number of voice sources must be controlled. Third, the core of the network must be tightly controlled with respect to voice load and scheduling mechanisms.

Latency cannot be studied without looking at voice quality, and with packet networks, availability cannot be studied without looking at latency. This research accomplishes this by trying to use a three phase approach. As was discussed in Chapter

7, the following analysis are required for mission critical voice over IP to be successful:

1. Voice Quality Analysis.
2. Latency Analysis.
3. Survivability Analysis (not studied in this research).

An optimization of the E-Model was created to study the effects of latency, utilization, and coder design on voice quality. This research used this optimization to choose a coder and utilization levels given certain conditions. This thesis demonstrated that an optimization of the E-Model is not only possible, but also has significant application with voice over IP network design. The results illustrate that the optimization solved the problem correctly (as we defined the problem). This is important because this model could be extended to more complex problems that cannot easily be analyzed by hand.

The E-Model was used to form an optimization algorithm, which assists with the selection of parameters important to voice over IP networks. These parameters include the voice coder, allowable packet loss, and link utilization. This research also demonstrates that different coders, packet loss levels, and utilization levels are optimal in different situations and that an optimization can correctly select combinations of these parameters that will allow network resources to be used in an efficient manner without using "trial and error" to select network and coding parameters.

A simulation model was used to study latency in mission critical voice over IP networks. This model has two primary components, a Remote Network component and a Core Network component. The Remote Network component studied some of the issues surrounding remote site communications over low speed links. The Core Network component used a large number of voice and data sources to study delay in the core of the network. Different queuing methods, MTU sizes, data file size distribution schemes were studied in an attempt to analyze the feasibility of transporting mission critical voice over IP. This portion of the research showed that mission critical VOIP networks seem possible with numerous restrictions. The primary restriction is that the network must be tightly controlled with respect to link utilization, voice utilization, scheduling methods, file size distributions, and hop count.

During the Remote Network portion of the simulation study, a requirement for a maximum network delay (in the edge portion) was established to be 12 ms. This was achieved in several scenarios. But, three assumptions proved to be very important. First, maximum data packet size (MTU) is highly correlated with worst case delay for voice. Second, since the number of voice sources is limited (in this model, there were 6 voice sources), the likelihood of all sources being "on" is not remote. It is approximately .41% in this simulation with each source being "on" an average of 40% of the time. The fact that all six sources can be "on" means that worst case delay planning must assume that all sources are "on". This was shown to be true in simulation. Third, voice utilization is very important and becomes even more important when a Pareto distribution is used for data file sizes.

During the Core Network simulation, it was observed that it was possible to limit voice delay through the core of the network to 40 ms, which was the requirement that was established for core network delay. But, this was only observed with strict priority queuing. When DRR queues were used, maximum delays climbed to at least 60 ms. This presents problems with scheduling in the core of the network because priority queues are not "fair" to the classes below the top priority. More research needs to be done to address the issues of scheduling in the core of the network. Section 8.2 documents the contributions that this research has to offer.

8.2 Contributions

The goal of this research is to study the feasibility of using IP to transport mission critical voice traffic. This research has contributed to the study of mission critical voice over IP network in several ways. These are listed here:

- The overall approach for studying mission critical voice over IP was documented. Two facets of this approach were implemented in this research.
- The limitations for designing a feasible mission critical voice over IP network were explored and documented.
- The optimization of the E-Model extends the E-Model to provide a useful tool to choose coding schemes, packet loss tolerances, and utilization levels.

- The model of the remote and core voice over IP networks provide a framework to study the complex queuing issues surrounding VOIP networks. These models look at some of the variables that must be controlled in order to control latency.

8.3 Future Research

This research is literally the "tip of the iceberg" with respect to how much research is needed to effectively and efficiently implement voice over IP. Documented below are some of the key areas of future research that are needed to further the work in this thesis.

- Research is needed that studies the survivability and availability of mission critical voice over IP networks. More research is needed into ways to tie voice quality and latency into availability analysis.
- More accurate methods to model large IP networks are needed. This research modeled a large network, but frequently assumptions were necessary to manage the size of the model.
- More accurate estimates of delay bounds in core networks are needed.
- More research into ways to extend the optimization of the E-Model to include more variables is needed, which will increase its usefulness. In addition, better estimates of network latency will help this model to be more accurate.

There are many other areas of research into mission critical voice communications that are not addressed in this research. Hopefully, this and other research will help to further the potential to someday transport mission critical voice over IP.

Chapter 9

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Appendix 1

OPNET Model Description and Validation

A.1 IP Packet Model

The IP packet that was used for this model is shown in Figure A.1. It implements a standard IPv4 packet. The data field size is set dynamically based on the total size required for that packet. All other fields can be set to a particular value if necessary. All fields can also be queried and modified. This IP packet model was used for all sources, both voice and data. The next section looks at the Voice Traffic Model.

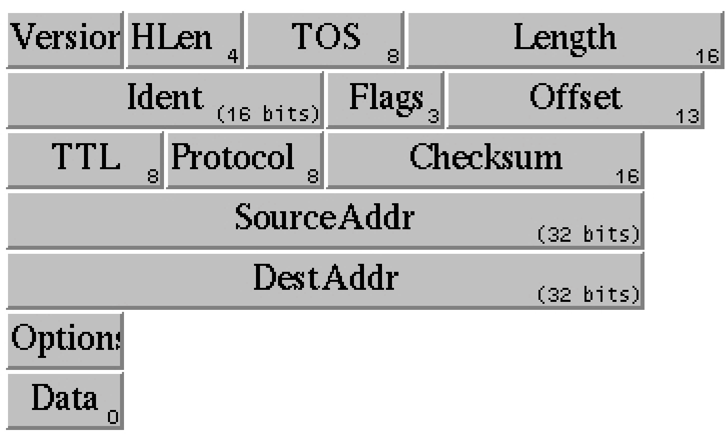

Figure A.1 - IP Packet Model (IP_pkt)

A.2 Voice Traffic Model

The voice traffic source is a EF voice source that is modeled 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 [RFC2598] for EF traffic. The node model is shown in Figure A.2.

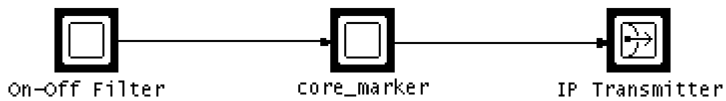

Figure A.2 - Node Model for Voice Packet Generator (VOIP_Gen4)

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. Figure A.3 is the process model for the On-Off source.

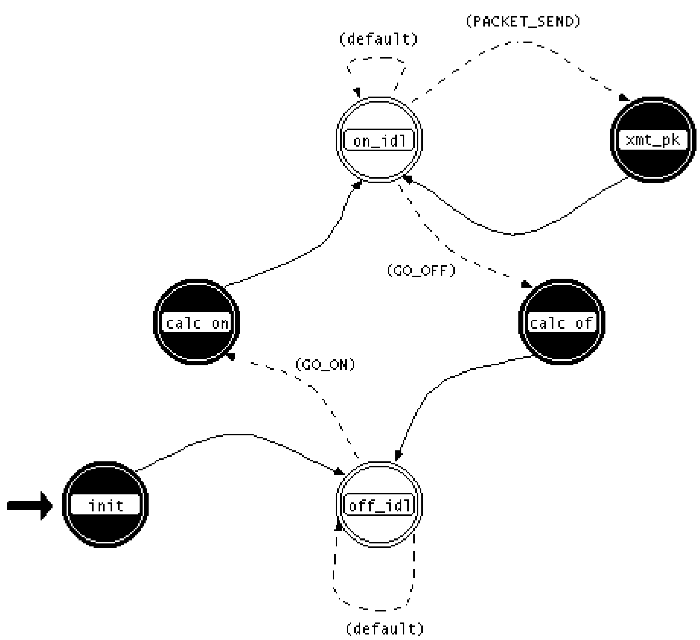

Figure A.3 - On-Off Process Model (On_Off_Process3)

This model was tested as a standalone model and generated packets when "ON". When "OFF", no packets are generated. Figure A.4 shows a graph of "ON" time versus "OFF" time. Value 1.0 indicated "ON" time and value 0.0 indicates "OFF" time.

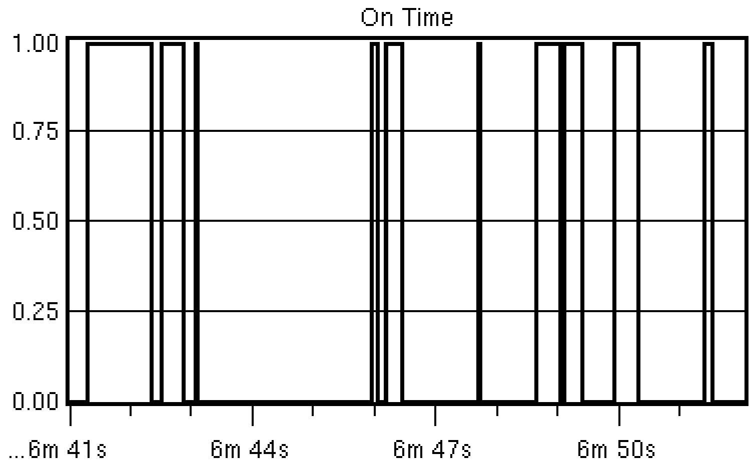

Figure A.4 - ON vs. OFF Times for Voice Source

Figure A.5 shows the distribution of ON times. This is shown to verify that the model is working as intended.

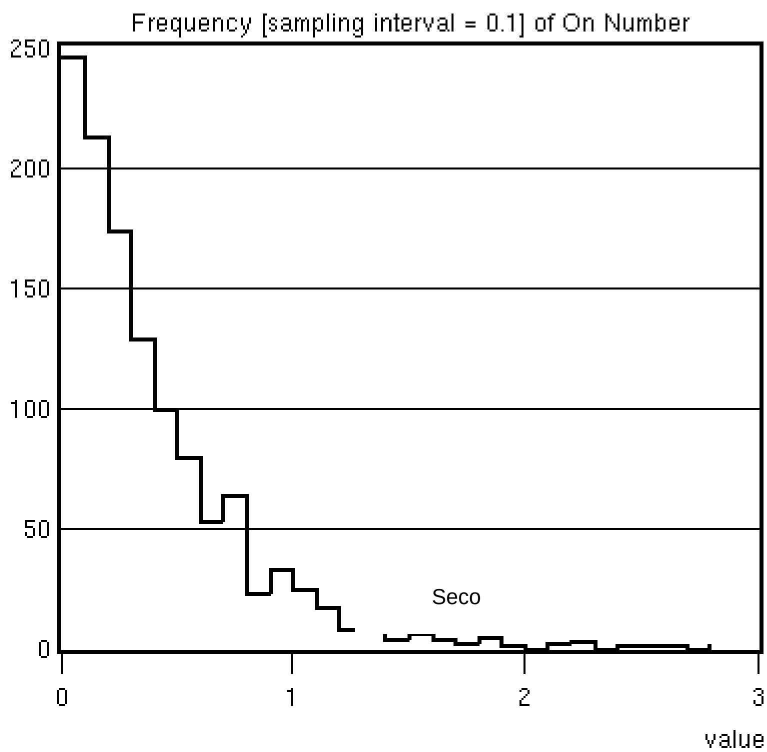

Figure A.5 - Distribution of ON Times for Voice Source

A.3 Best Effort Traffic 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. Settings may be abstracted to the top layer (Network) of the model. The TOS field for BE traffic is marked with a DSCP of 000000 (0).

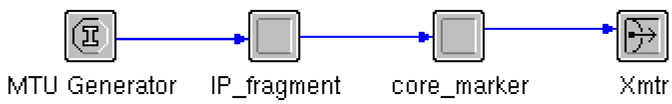

Figure A.6 - Node Model for BE Traffic Generator Model (BE_Frag_Gen)

Figure A.6 shows the node diagram for the BE Traffic Generator. The IP_fragment process is shown in Figure A.7.

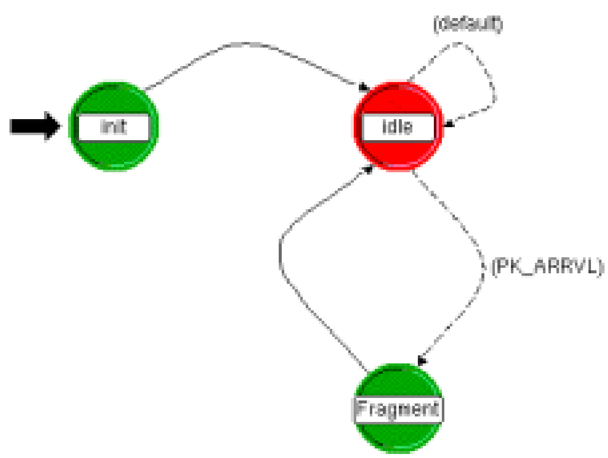

Figure A.7 - Fragment Process Model (IP_Fragment)

After a file arrives, the process state is changed from idle to Fragment. At this point, IP packets are generated with the user specified MTU size plus the IP header. The MTU size is subtracted from the file size. After the file size is less than the MTU size, the final packet is sent of the file size plus the IP header. This was tested by printing the file size and then printing the subsequent packet sizes. The MTU size for the test is 4kb. A portion of this data is shown below.

```
 MTU 3640

Send 3800
MTU 4026
send 4160
Send 186
MTU 1688
Send 1848
MTU 2700
Send 2860
MTU 5765
send 4160
Send 1925
MTU 12600
send 4160
send 4160
send 4160
Send 760
MTU 9097
send 4160
send 4160
Send 1257
MTU 12585
send 4160
send 4160
send 4160
Send 745
MTU 6931
send 4160
Send 3091
MTU 18812
send 4160
send 4160
send 4160
send 4160
Send 2972
```

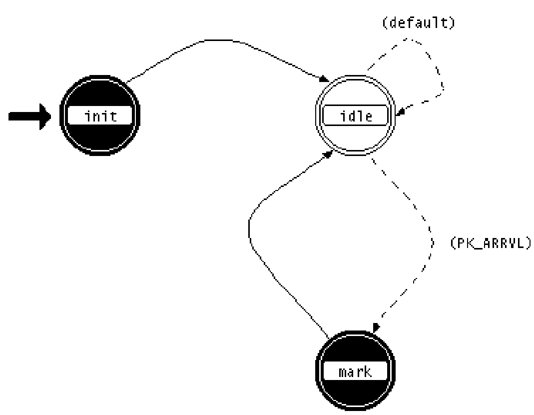

Figure A.8 - Marker Process Model (core_marker)

Figure A.8 shows the process used for the core_marker. This process switches from idle state to mark state upon packet arrival. The mark state accesses user defined parameters to mark the packet. This process has provisions to fill the address fields based on statistical probabilities. For example, if 45% of the packets should have a destination address of 1, then this process uses a uniform distribution to determine if that packet should be marked for 1. This is not used in the Remote Network Model, but it is used in the Core Network Model. .

To look at the Pareto Generator that is included in version OPNET 7.0, a stand alone generator was used to create the data that is shown in Figures A.9 - A.11. 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 [CROVELLA][OPNET]:

$$p(x) = \alpha k^{\alpha} x^{-\alpha-1} \quad \alpha, k > 0, x \geq k \qquad \text{A.}\mathit{1}$$

$$E(x) = k\alpha/(1-\alpha) \qquad \text{A.}\mathit{2}$$

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 [CROVELLA].

The research in [CROVELLA] looked at what factors of *k* and α 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. This research also showed that the WWW favors smaller files (256-512 bytes) sizes than Unix files (1Kbyte - 4 Kbyte). Using this research as a guide, this model used an α 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. First, using a shape parameter of 1.06, 453 bits was necessary to arrive at a mean file size of 8000 bits, which was used when exponentially distributed file sizes were used. Second, very small file sizes are possible in WWW traffic. Figures A.9, A.10, and A.11 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 interarrival time of 1 second.

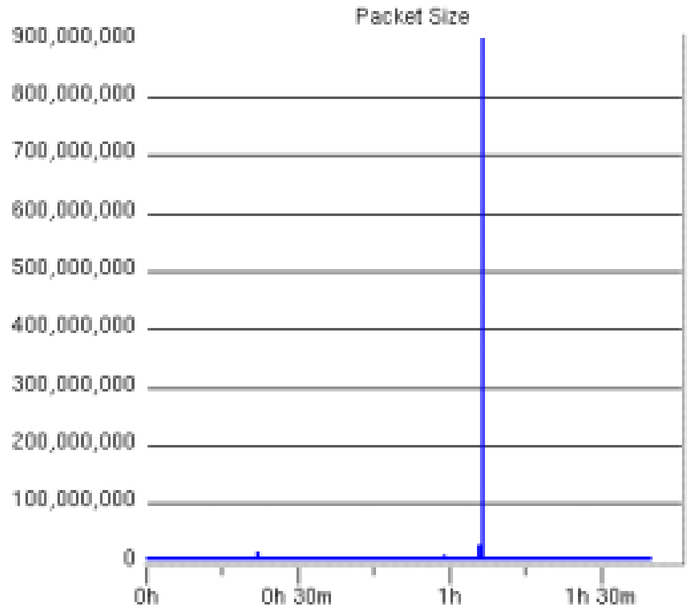

Figure A.9 - Pareto File Size Generator (Seed - 105)

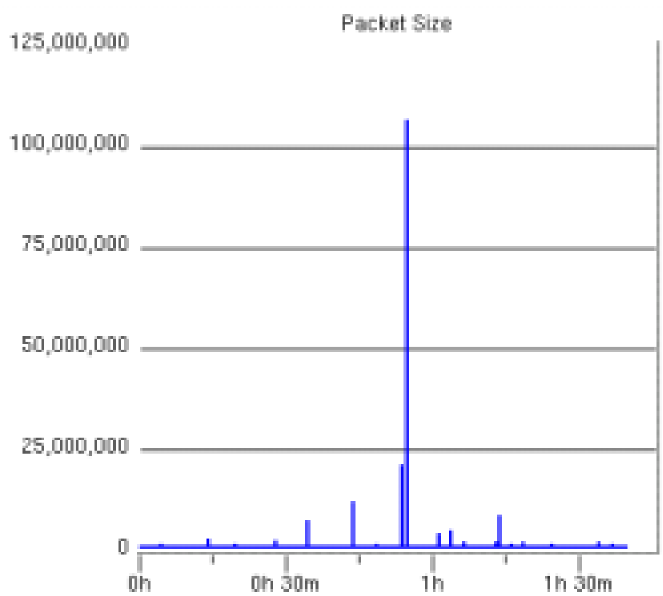

Figure A.10 - Pareto File Size Generator (Seed - 128)

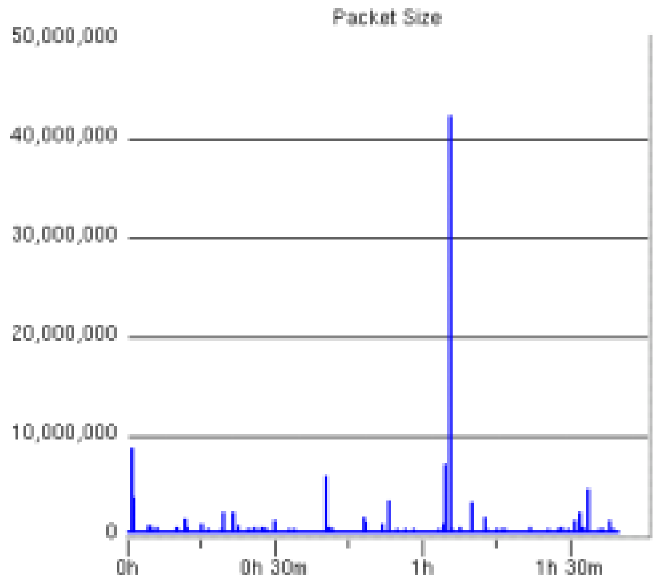

Figure A.11 - Pareto File Size Generator (Seed - 200)

Figures A.9 - A.11 show is 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. Or, to put a different way, when the phantom large file is not generated, the effective load will be considerably lighter.

One important question here is: Why not use the Pareto distribution for all data file size generation in this model? The reason is related to the fact that this is a study of mission critical networks. Given this, it is possible (if not likely) that other traffic sharing the network is also mission critical and may be more stable than traditional WWW or FTP traffic. For example, if the network data that is being sent across the mission critical network is radar data, then the packet sizes are set and sent at regular intervals (typically related to antenna rotation speeds). Other examples of mission critical traffic that may not have a long tail are systems the perform remote maintenance and monitoring. These systems typically pole their sites on regular schedules with constant files sizes. Note that this does not necessarily mean that M/M/1 assumptions are correct, but they may more accurately reflect more stable networks that a Pareto distribution. These different assumptions (M/M/1 vs. Pareto files sizes) also show the problem with making incorrect assumptions on network models. The next sections look at the router models that are used.

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). The scheduler is a priority queue. In OPNET [OPNET], the priority router uses an internal field to determine priority. In this model, the tg_split_dest2 module looks at the DSCP and uses that information to set the priority field. Figure A.12 shows the nodal model for the edge priority router.

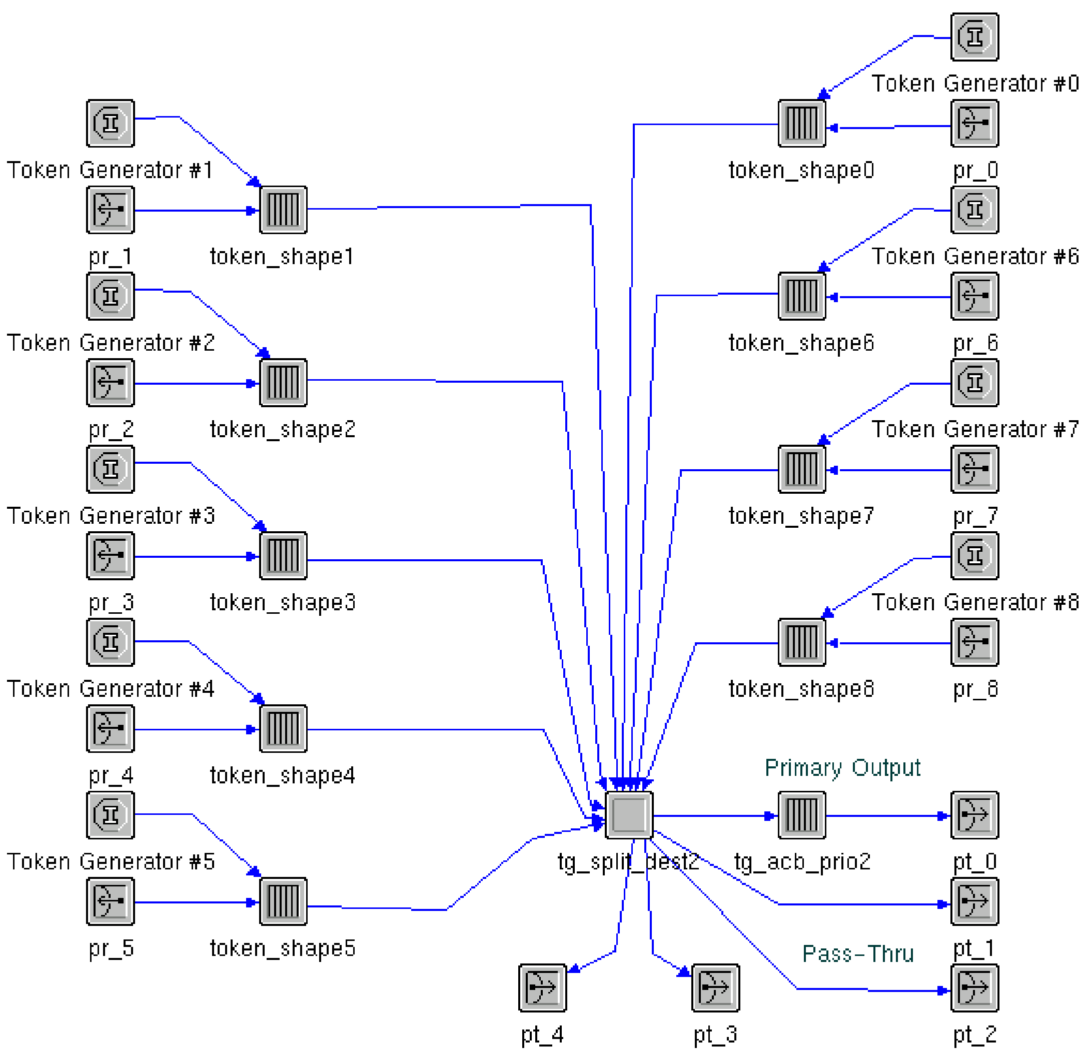

Figure A.12 - Edge Priority Router (DS_Edge_Router)

As we can see from the diagram, this model only incorporates output queuing. All parameters including forwarding addresses, DSCP settings, and token bucket parameters can be set by the user and may be abstracted to the network level to be set for that simulation. The core network does not require the shapers on the inputs and therefore a router model was created that did not include the shapers. The node diagram for tg_core_priority_router is shown in Figure A.13. It works under the same principle as the edge router. 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.

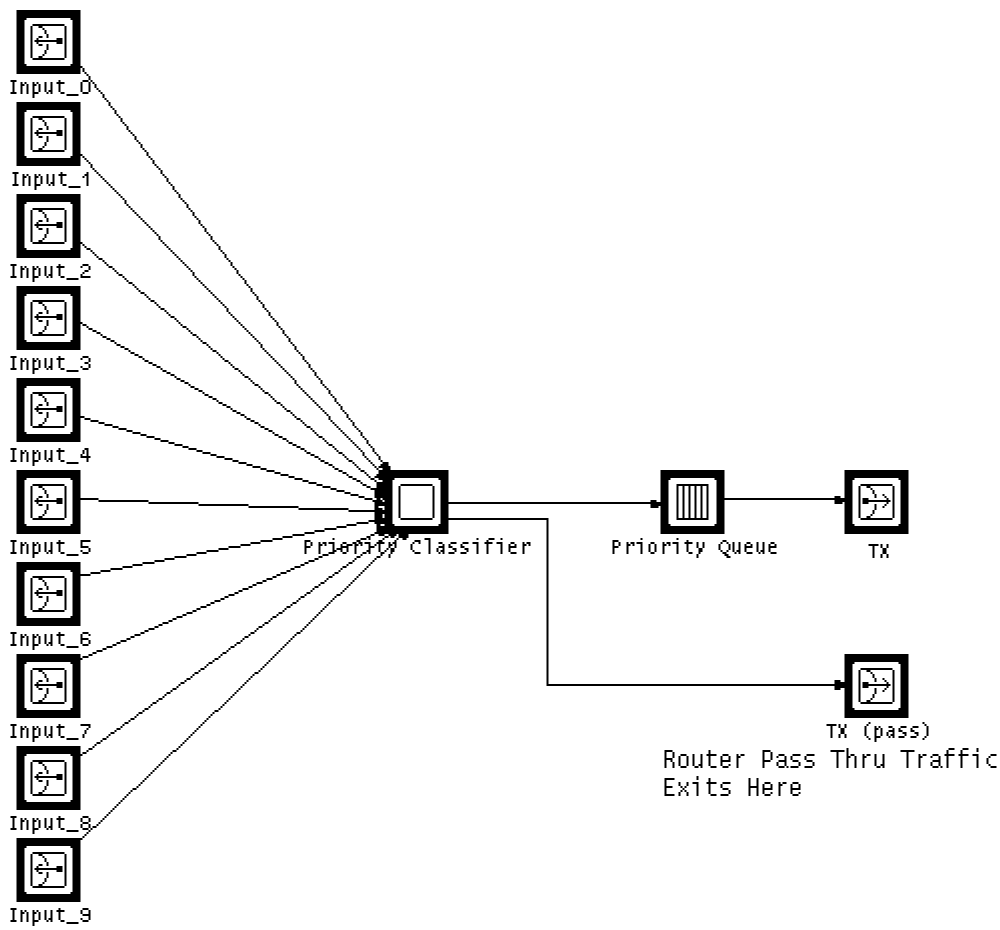

Figure A.13 - Core Priority Queue (tg_core_priority_router)

The token shaper used in the edge routers is implemented using a standard token bucket shaper. As tokens arrive, they dump into a bucket and collect based on their size. As packets arrive, if the bucket has enough tokens, the packet is forwarded unimpeded. If the bucket does not have enough tokens, the packet waits till there is enough in the bucket to cover the size of the packet. All parameters are settable by the user and may be abstracted to higher layers. Figure A.14 shows the process model for the token shaper:

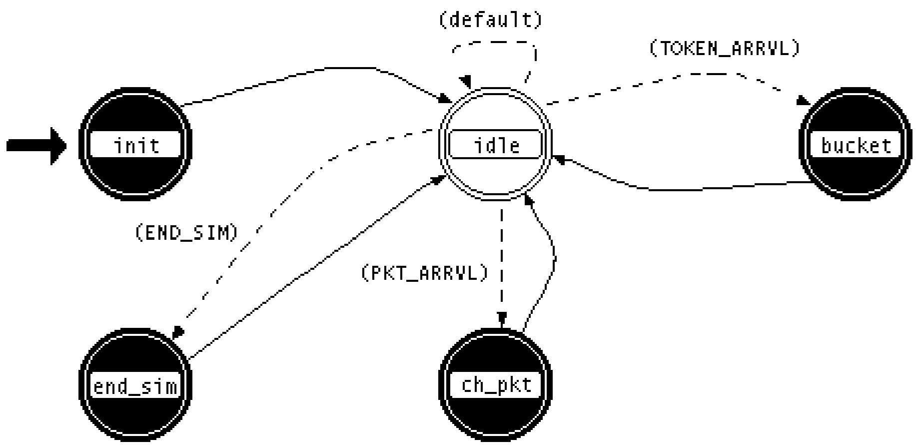

Figure A.14 - Token Shaper Process Model (token_shaper)

To test this model, a simple generator, shaper, and receiver was used. A table was then generated that listed packet and token arrival times, queue depths, bucket depths, and packet sent times. Due to the size of this table, it is not included in the Appendix. It demonstrated that the process works correctly, including the bucket limit feature and the queue full, discard packet feature. Figure A.15 shows the throughput of the test setup before and after the shaper. It is easy to see the shaping effect on the traffic. Figure A.16 shows the bucket fill amount. Since this statistic is written after a token arrival, it would never be less than one token (in this case 1024 bits). Notice that the bucket never contains more than 5000 bits, this is the Bucket Size parameter. It can be set by the user.

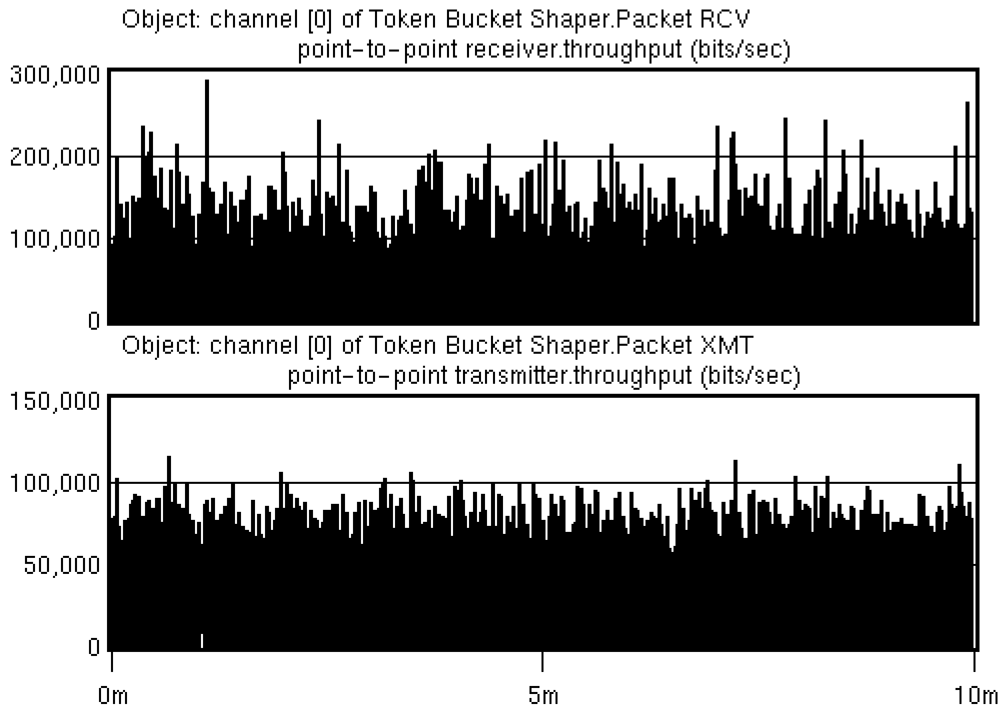

Figure A.15 - Token Bucket Shaper Performance

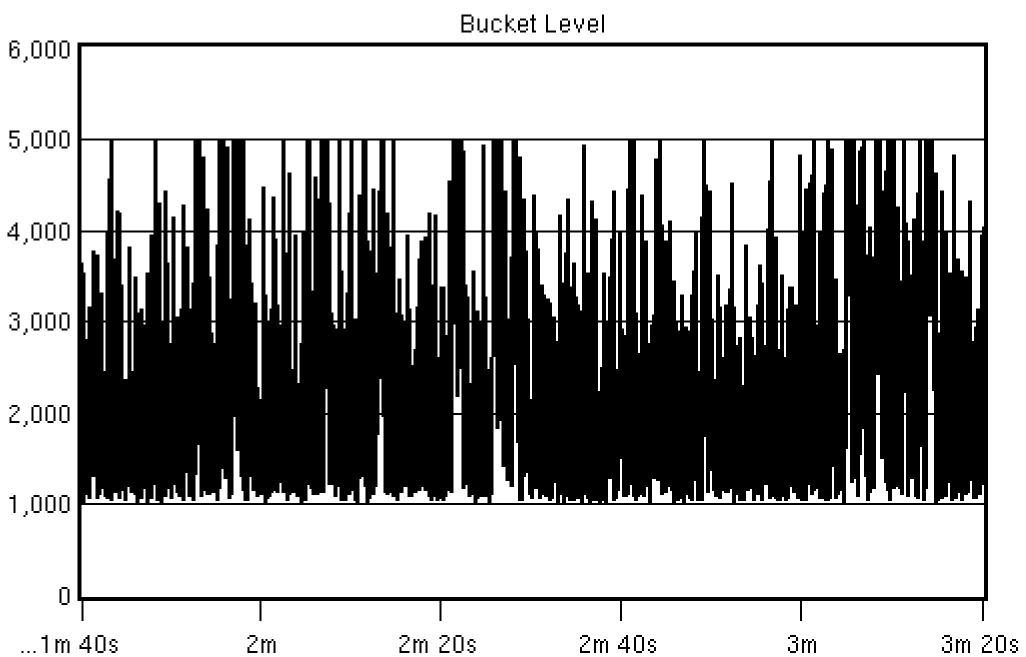

Figure A.16 - Token Bucket Fill Level

The priority scheduling module used in both the core router and the edge router model was created by modifying the OPNET acb_fifo queuing module [OPNET]. The modifications include adding a subqueue called a wait queue. When packets arrive, they are all queued into the wait queue in order based on the priority field (internal OPNET field) in the packet. The other queue is the service queue. It is a FIFO queue. When the service queue is empty, it request a packet from the wait queue. The wait queue takes the first packet in the queue. After the wait queue gives the service queue a packet, it deletes the packets from the wait queue and the service queue serves the packet at the service rate. This implements a non-preemptive priority queue. Figure A.17 shows the priority queuing process model (tg_acb_prio2).

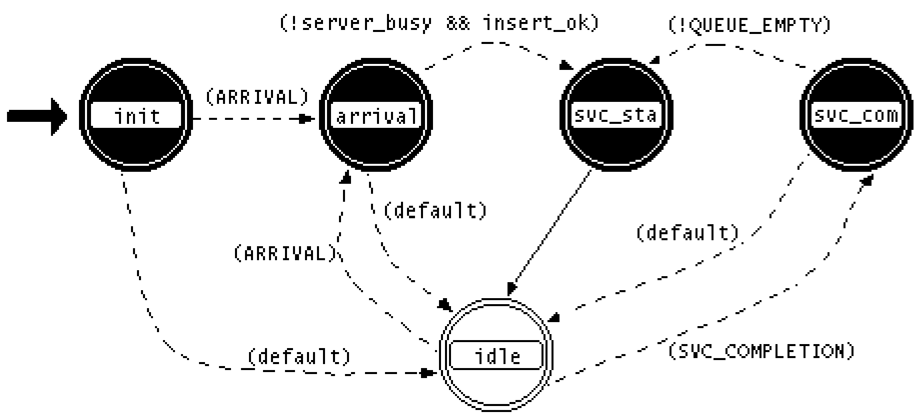

Figure A.17 - Priority Queue Process Model (tg_acb_prio2)

To test the queue, a test was run that sends two M/M/1 packet generators into the priority queue. Each packet is wrapped with a 20 byte IP header prior to entering the queue. One generator marked its packets as priority 1 (low), the other generator marked its packets as priority 2 (high). This test verified that the priority model operates properly. Figure A.18 shows the delay incurred by the high priority and low priority traffic. The low priority traffic had considerably more peak delay than the high priority traffic.

The average delay of the low priority traffic was 67 ms. The average delay for high priority traffic was 12 ms with a link speed of 256 kbps, average interarrival times of .01 sec, and average packet sizes of 1000 bits. Using the classic equations to calculate average wait times for a priority queue with M/M/1 sources [MCDYSAN]:

For Priority 1 $$W_{q1} = \frac{\rho/\mu}{1-\rho_1} \qquad \text{A.3}$$

For Priority 2 $$W_{q2} = \frac{\rho/\mu}{(1-\rho)(1-\rho_1)} \qquad \text{A.4}$$

Where ρ is utilization, μ is service rate, and W is average wait time in the queue. We have a theoretical high priority delay (wait time + service time) of 12.0 ms. This corresponds to our model. For the priority 2, the theoretical delay is 84.5 ms. Our model found average wait times of 67 ms for priority 2 traffic. The difference can be attributed to the fact that we do not have a true M/M/1 queue because the IP headers were attached after the packet was generated. This has the effect of reducing the variance of the packet size distribution compared to a true exponential distribution. This completes the verification of the priority queue. The next section looks at the DRR queue and the routers that use this queue.

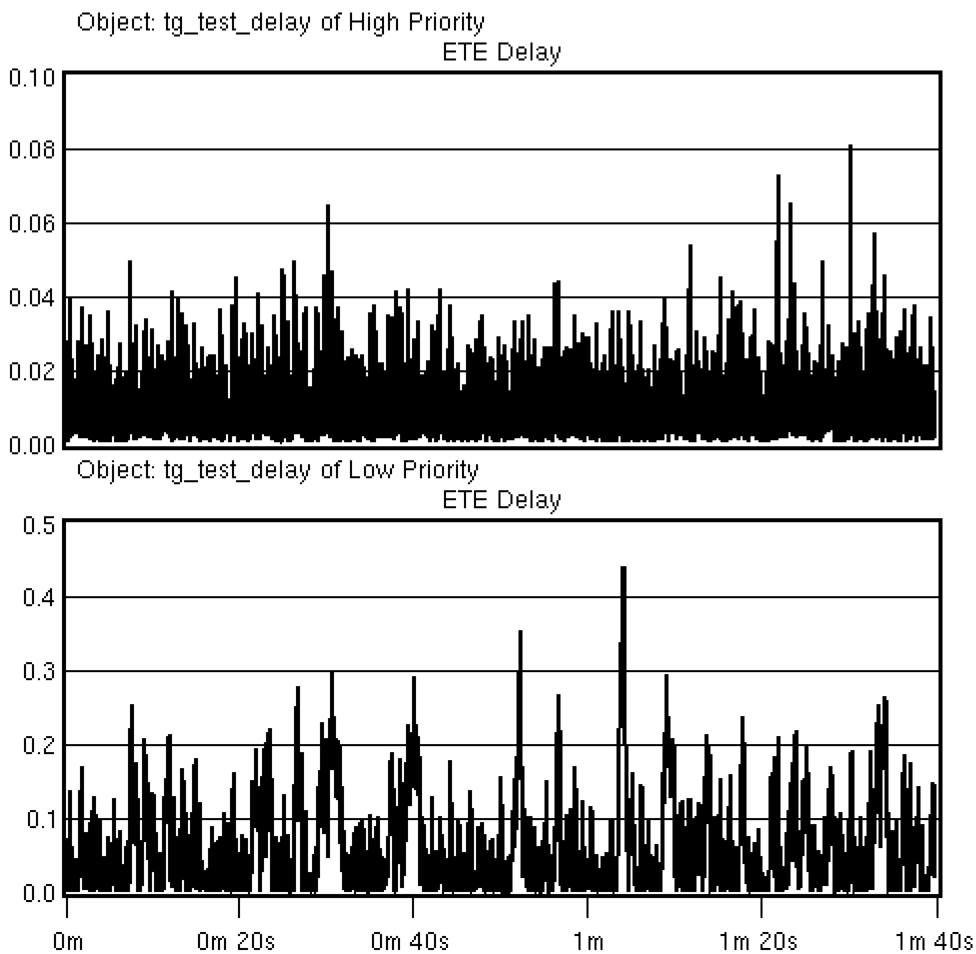

Figure A.18 - Delay times for Priority Queue

A.5 Deficit Round Robin (DRR) Router Models

The implementation of the DRR Router Models is similar to the Priority Router Models with the exception of the scheduling module. Figure A.19 shows the edge DRR router node model (DS_Edge_Drr_Router). The only significance difference between this model and the edge priority router model is the queue and the classifier. They are specific for Deficit Round Robin.

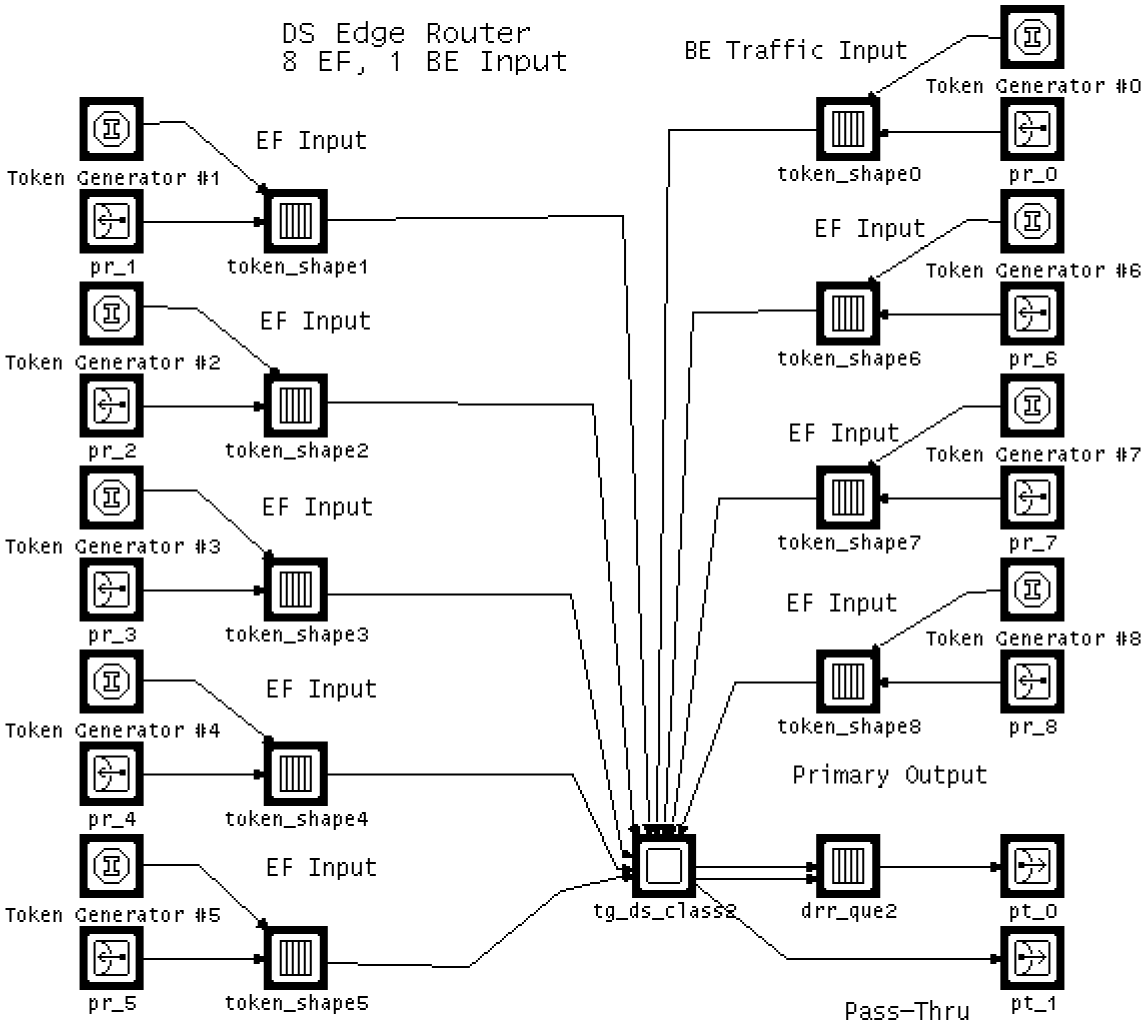

Figure A.19 - Edge DRR Router Model (DS_Edge_Drr_Router)

The core DRR router model is similar to the edge DRR router with the exception that shapers are not included. Figure A.19 shows the node diagram of the core DRR router model (tg_core_drr_router).

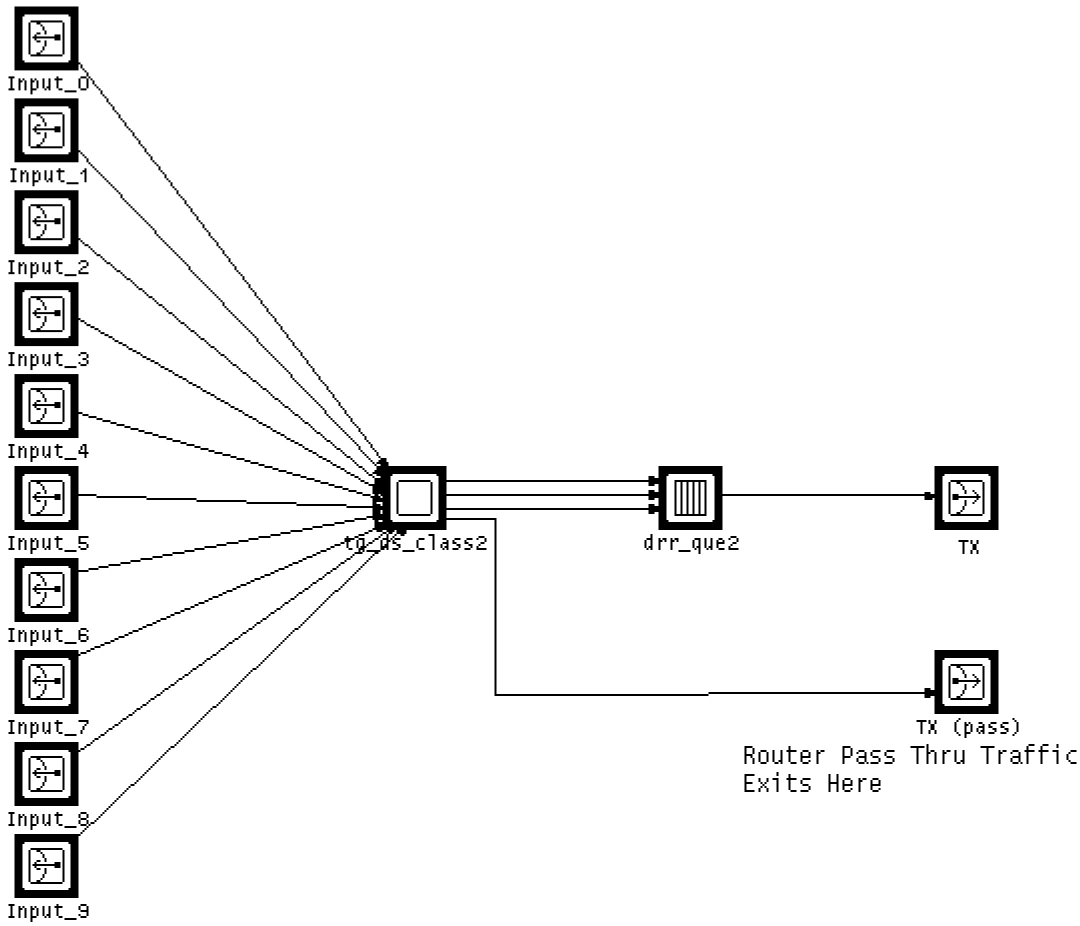

Figure A.20 - Core DRR Router Model (tg_core_drr_router)

When a packet arrives at the router, the classifier looks at the DSCP for the packet. Using mapping set up by the user, the classifier sends packets with like DSCPs to the scheduler on the same input. For example, input one may be for EF traffic (DSCP = 101110 (46)), while input two may be for BE traffic (DSCP = 000000). After the packet arrives at the scheduler it is queued based on the input used to enter the scheduler.

The DRR scheduler uses the Deficit Round Robin algorithm as described in [SHEED]. Briefly, the algorithm works as follows [SHEED]. 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 quanity 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 to 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. Figure A.21 shows the process model (drr_que2) for the DRR queue.

The primary advantage of the DRR queue is its ability to fairly implement round robin queuing with varying packet sizes. In addition, [SHEED] demonstrated that from a processing perspective, DRR is simpler to implement than many fair queuing algorithms. DRR does have its pitfalls. The worst case delay can be as bad as (n-1)*Max MTU size even with a quantum (Q) of 1 per queue. This compares of a worst case delay n-1 bits for a bit by bit round robin queue (BR) [SHEED].

To test this model, two tests were run. First, eight inputs with varying input parameters were inserted into a DRR queue with differing quantum levels. The goal was to force a number of different situations on the queue. All events and the times that the events occurred were checked. In addition, the status of each queue and the DC levels were checked.

The second test that was performed used eight inputs. Each input was M/M/1 with an average interarrival time of .0156 (64 arrivals/sec) and an average packet size of 1024 bits. The service rate is 500 kbps. Table A.1 shows the results of this test.

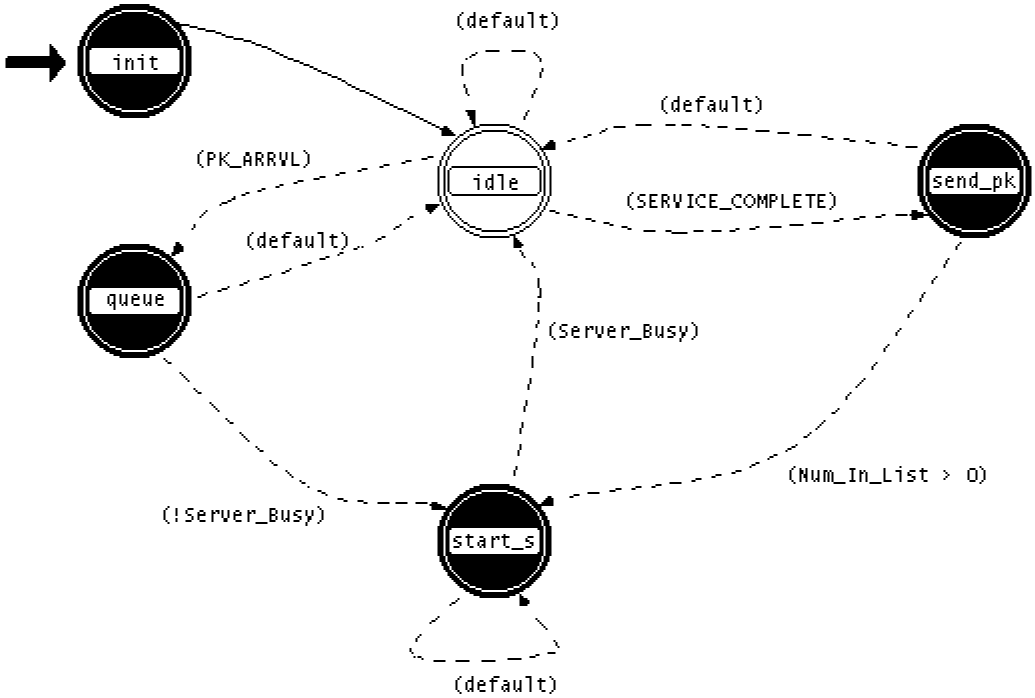

Figure A.21 - DRR Queue Process Model (drr_que2)

Queue #	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	0.0444	0.4027
7	256	27778	17.2423	33.1007
8	256	27778	16.7888	30.6073

Table A.1 - DRR Queue Performance

As we can see from Table A.1, delays got worst when the quantum was 512 from 1024. However, the delays were catastrophic when the quantum was 256. This is for a simple reason. With a quantum of 256 bits out of 4806 bits, the effective bit rate on that queue (if all queues are busy) is 27778 bps. Since 27778 is substantially less than the input rate of 65536 bps, one would expect substantial delays. It is interesting that delays on queue 3 - 6 were not worse. This is because 55556 bps is close enough to 65536 bps that if any of the queues were not actively using the entire bandwidth allocated to them (which 1 & 2 should not be most of the time), there would be adequate bandwidth to serve 3-6. A caution that this research revisits, is the danger of under provisioning the DRR queue as was done on queue 7 and 8.

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.