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Performance Study of Voice Over Frame Relay

A thesis presented in partial fulfilment of the
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in
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Abstract

Frame Relay (FR) represents an important paradigm shift in modern telecommunication. This technology is beginning to evolve from data only application to broad spectrum of multimedia users and potential to provide end users with cost effective transport of voice traffic for intra office communication. In this project the recent development in voice communication over Frame relay is investigated. Simulations were carried out using OPNET, a powerful simulation software. Following the simulation model, a practical design of the LAN-to-LAN connectivity experiment was also done in the Net Lab. From the results of the simulation, Performance measures such as delay, jitter, and throughput are reported. It is evident from the results that real-time voice or video across a frame relay network can provide acceptable performance.

Acknowledgement

I, at the completion of this thesis want to state that my efforts have come to function chiefly because of the help and advice received from teachers, friends and well wishers. This is the opportune moment to express my indebtedness to all.

I wish to express my deep sense of gratitude to Dr. M. A. Rashid for his invaluable advice and constant guidance throughout the duration of this project.

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Chapter 1

Introduction

New applications are rapidly evolving that allows multiple users to communicate real time voice and video along with traditional file and screen sharing and other data applications. Changing from the current multiple networks into a single multi service network would usually take place in phases. Frame Relay (FR) is a high-speed communication technology that is used to connect LAN, Internet and voice applications. Frame Relay is a way of sending information over a wide area network (WAN) that divides the information into frames or packets. The frames travel through a series of switches within the FR network and arrive at their destination. Frame Relay employs a simple form of packet switching that is well suited to powerful PCs, workstations and servers that operate with intelligent protocols. As a result, FR offers high throughput and reliability that is perfect for a variety of today's business applications. FR can be used around the edges of the WAN to connect remote offices into the Intranet, or can be used as the entire backbone technology for the WAN. The inherent characteristics of Frame Relay networks can lead to substantial savings in transmitting costs, as well as increased network flexibility. The integration of packetised voice leads to design challenges such as controlling network delay, prioritizing traffic types and choosing a suitable compression algorithm.

Voice quality on a packet network is affected by delay and jitter. The objective of Quality of service (QoS) features in a packet network is to minimize delay and jitter for voice traffic.

Due to the increased number of networks in existence and their greater complexity, designing new systems and improving the performance of existing ones has become more difficult and time consuming. It is therefore more practical to use modeling and simulation tools to deal with this complexity. OPNET (Optimized Network Engineering Tools) is a comprehensive engineering system, capable of simulating small and large communication networks with detailed protocol modeling and performance analysis. It provides an opportunity to examine Frame Relay network in a simple model, which can provide the same performance as if it is the complex one.

In this research project our approach is to set up special Permanent Virtual Circuits (PVCs) for voice, compression technique is applied for two different types of voice applications to see the various parameters that affect the FR network for voice application. Several network models has been developed and simulation has been done using OPNET modeler 8.1. The results were used to evaluate the performance of voice over Frame Relay networks.

The structure of the thesis is as follows.

Chapter 2 introduces the Frame Relay technology. Some of the key aspects of FR are discussed. It explains the relationship between ISDN, OSI, and ANSI layers. It also describes the structure and operation of the Frame Relay. Error control and recovery, congestion control and management is also studied in this chapter. The internetworking with other networks and effect of different service parameters are explained.

Chapter 3 describes a detail discussion on Voice Over Frame Relay. Voice compression techniques, Voice quality of service (QoS), delay and delay variation, frame loss, fixed and variable delay components and different options of voice packet transports.

Chapter 4 gives review of literatures. In this chapter detail discussion of previous and present work has been elaborated. Different real time applications regarding Voice Over Frame Relay are also discussed.

Chapter 5 presents the simulation of the Frame Relay network model using the OPNET simulation package. This chapter discusses briefly about the OPNET programming package and provides a quick overview of various steps for building a network.

Chapter 6 explains FR network design for voice application. The procedures for the simulation of network models, implementation procedure and the selection of attributes are discussed. A description is given of devices and tools within the OPNET program that have been used in this research. It also shows different network scenario and model used. The selected attributes are also shown in the figures obtained from print screen options. This also gives the hardware experimental setup description at the IE-Net lab.

Chapter 7 gives performance results obtained from the simulation procedure of the Frame Relay networks in graphical forms. Statistical results of different global parameters of the networks are also included in tabular form.

Finally in chapter 8 an overall conclusion on the work is added with some suggestion about further work.

Chapter 2

An Introduction to Frame Relay Technology

Frame Relay(FR) represents an important paradigm shift in modern telecommunication. A shift away from network based intelligence and complexity towards a simplified, streamline network infrastructure that provide cost effective, multiplexed information path between intelligent devices.

In the Information Age, corporate intranets, extranets, branch offices via wide area networks (WANs) or local area networks (LANs) are the channels for everyday business. Frame Relay, ATM and TCP/ IP are the principal networking techniques used to build wide area networks, among them frame relay is a good choice for transferring data and voice, especially across multiple private or public networks. The telecom industry is trending toward network convergence and favoring open system solutions that reduce networking and operating costs.

Frame Relay evolves with interoperability as a principal goal through the international standards bodies, such as ITU-T, Frame Relay Forum, and ANSI. These organizations define specifications to assure internetworking among the various vendors in large networks.

Frame Relay evolved from X.25 and ISDN in the mid-1980s. The technology promised to minimize network functions to provide a simple and flexible service to users. From X.25, Frame Relay borrowed the concept of virtual connections, which decoupled the idea that one physical interface mapped to only one connection. Many virtual circuits (VC) can exist simultaneously over the same Frame Relay link. In addition, virtual circuits may be either permanent virtual circuits (PVCs) or switched virtual circuits (SVCs). PVCs are set up administratively for dedicated point-to-point connections. SVCs are set up dynamically on a per call basis. From ISDN, Frame Relay adopted the idea of separation of functionality based on control, data, and management planes.

Initially, Frame Relay served mainly as a cheaper alternative to leased lines for establishing wide-area networks with LAN-to-LAN connections. These access connections remain the largest application for FR. Vendors have developed other

applications deployed over Frame Relay, such as IP, SNA, X.25. Frame Relay is also used for high-speed access to the Internet. FR can commonly be found in corporate LANs, wireless infrastructure, travel reservations infrastructure, banking (ATM teller machines), and mainframe-based SNA systems.

Frame Relay also embodied some pioneering concepts: it removed hop-by-hop flow control and error control, moving them to the network boundaries, it specified quality of service (QoS) parameters for virtual connections, and it specified aspects of traffic management to handle congestion caused by bursty use of virtual connections. FR was ideal for data traffic and made modern high-speed WANs possible. The vast majority of connections are permanent virtual circuits (PVCs) although switched virtual circuit (SVC) service is also commercially obtainable.

FR became one of the fastest growing market segments in telecommunications. The reasons include bandwidth efficiency which means virtual circuits use bandwidth only when transporting packets, and the protocol reduces error-handling processing, improved reliability of network circuits, an avid demand for bandwidth from users, and the continued advancement of FR standards.

FR is an Integrated Services Digital Network (ISDN), packet-mode, and carrier service designed for the transport of variable length data units over a connection-oriented path. The FR protocol eliminates protocol processing in the network nodes above layer 2 of the OSI model [1, 2]. Study of voice transmission over FR networks is motivated by the flexibility of bandwidth management offered by integration of voice and data in a single network. FR networks were originally intended for data communications alone. Integration of voice and data presents technical difficulties arising from the conflicting QoS requirements for both these traffic types; data is extremely loss-sensitive but can tolerate long delays. In direct contrast, voice traffic can accommodate limited amounts of lost or corrupted information but there exists a delay constraint for acceptable voice quality. The principal challenge in supporting real-time voice applications over any packet-switching network is the need to obtain synchronous playout of voice packets in the presence of network delays. Compensation of variable delays is typically achieved by buffering received packets and voice playout algorithms differ with respect to the techniques used to estimate the delay to be introduced prior to playout.

Before describing the issues surrounding the transport of Voice over Frame Relay (VoFR), it is necessary to briefly review the concept of FR since many of the obstacles to transporting voice calls in a public network are inherent in Frame Relay's technology.

2.1 Frame Relay Protocol

Frame Relay is a standardized packet interface protocol and does not define the backbone switching protocols or topologies. American National Standards Institute (ANSI) and the International Telecommunications Union-Telecom sector (ITU-T) define the standards.

The standards defining it specifically as an interface between the FR network and the attached devices like LAN router, front-end processor. Any type of packets can be carried over FR, but it should inter-networked with Link Access Protocols for ISDN D channel at layer two. The fundamental assumption to success the FR networks is the reliability of the transmission medium; the devices using the network generating bursty traffic and the end systems should be intelligent.

FR standards implemented this technology in three different ways. Frame Relay as an interface, as a signaling protocol and as a network service, these are the key aspects which will improve the wide area networking problems.

2.2 Frame Relay as an Interface

Frame Relay standards are developed by ITU and ANSI specifying as an access interface that provides the statistical multiplexing capability, allowing many logical connections over a physical link. These connections were maintained via PVCs.

FR standards evolved from ISDN interface standards. Which is the only interface can operate over basic rate access (BRI); primary rate access integrated digital access networks (PRI ISDN), V-series, digital data systems (DDS) and DDN. Manufacturers of FR access devices (FRADs), Routers and other equipment's used in FR networks are given some of the facilities like traffic priorities, methods for congestion control and

congestion recovery procedures have been implemented in their own ways. Some of the benefits of using FR as an interface are:

- Multiple users per physical access line
- Reduce the network access hardware costs
- Wide industry support
- Can maintain the interface standards
- High speed of access due to low packet demand
- Provide higher throughput

2.3 Frame Relay as a signaling protocol

Frame Relay is a high-performance WAN protocol that operates at the physical and data link layers of the OSI reference model. FR comes under fast packet technologies. This is an extension of X-25. Some of the benefits of FR using as a protocol are:

- Variability of the frame size.
- Improved performance over older packet protocols
- In band customer service and link management
- Discards error frames and excess data
- Protocol Flexibility
- Transparent to the higher layer protocols

2.4 Frame Relay as a network service

Nowadays Frame Relays network services are offering in both public and private networks. Because of its virtual networking capability and pay only for the actual bandwidth consumed by the user and this can be implemented over a surplus bandwidth of T1/E1 line, which is already using TDM multiplexing [3]. FR as a network service offers some benefits as follows

- Higher network availability
- Single network with multiple protocols
- Dynamic network architecture
- Fills the need for high speed LAN to LAN connectivity

2.5 Frame Relay layers and its relation ship to ISDN, OSI and ANSI layers

It uses variable length frames and routing of the packets will be done through data link layer by binding the data link connection identifiers (DLCI), some of the operations of upper sub layer of the data link layer like error recovery, flow control will be done on users end to end basis. Some of the core functions are added to the data link lower sub layer such as frame delimiting, multiplexing. The capability supports the internetworking of different type of networks like LANs and WANs. These variable data units will generate variable delays, because of this reason the delay sensitive data (voice and video) was not suitable to this type of networks. But nowadays the transport technology used in frame relay network is based on cell relay technology. The Frame Relay frames are splits in to cells and transported through backbone and reassembled as frames at the other end.

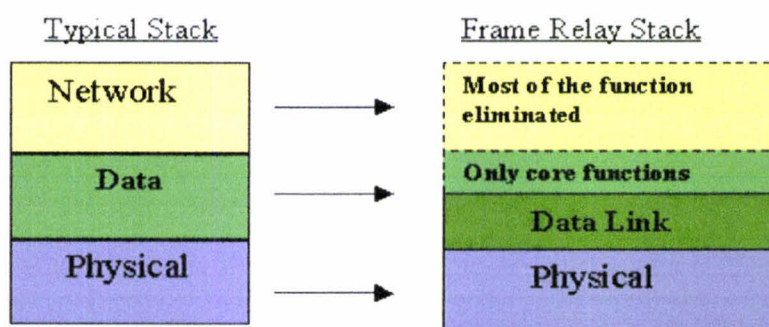


Figure 2.1 Comparison of typical and Frame Relay stack

The figure 2.1 explains the comparison between a typical and FR stack. The left side of the block diagram represents a typical data communication protocol stack. It contains physical, data link and network layers. The physical layer is responsible for modulation, encoding and decoding data etc. The data link layer is responsible for the error checking and retransmission of error traffic, and proper framing at the receiver. The network layer is responsible for routing and managing the traffic within the network and establishes a virtual connection etc.

The right side of the figure explains the frame relay stack. It eliminates most of the network layer and several functions of the data link layer.

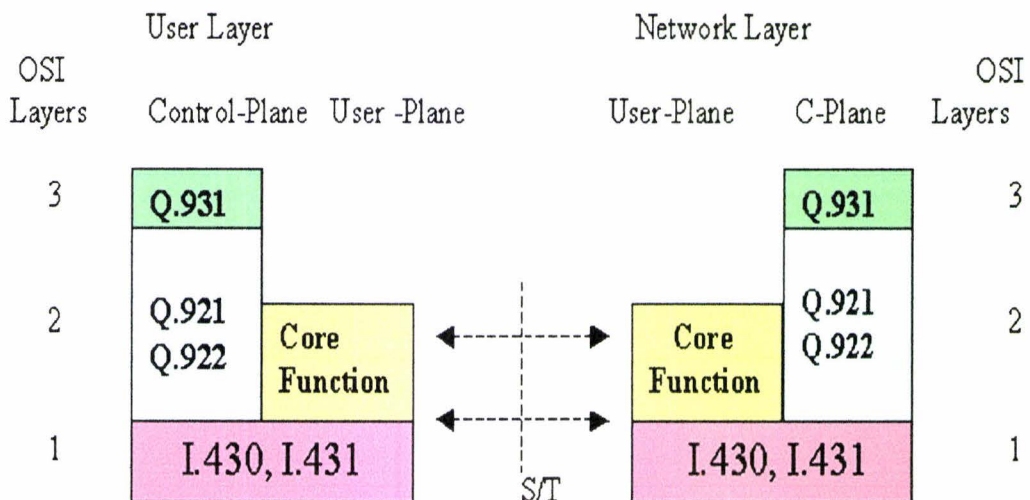


Figure 2.2 The User –to- Network Element interfaces (CCITT)

The FR service model has been shown as OSI and ISDN layer architecture in figure 2.2. The FR network only supports the core functions of the Q.922 layer 2 protocol. Which is an extended version of Q.921. This is the specification for the frame mode bearer service at data link layer.

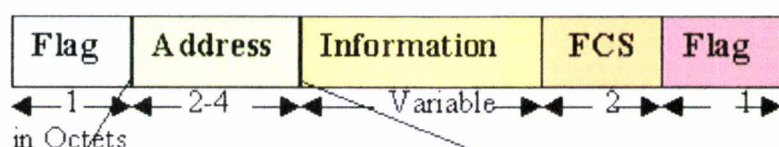
Some of the core functions are

- Frame delimiting& bit level transparency
- Virtual circuit multiplexing and demultiplexing
- Octet alignment of the traffic
- Check the maximum and minimum frame sizes
- Error detection and congestion management

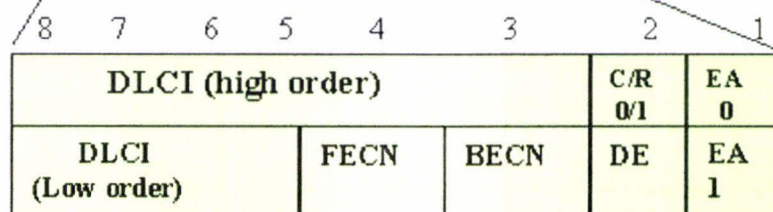
The remaining operations are implemented on an end-to-end basis. The Q.931 is a user network interface layer 3 specifications for call control. I.430 is an ISDN user network Interface layer 1 recommendation.

2.6 Frame Structure

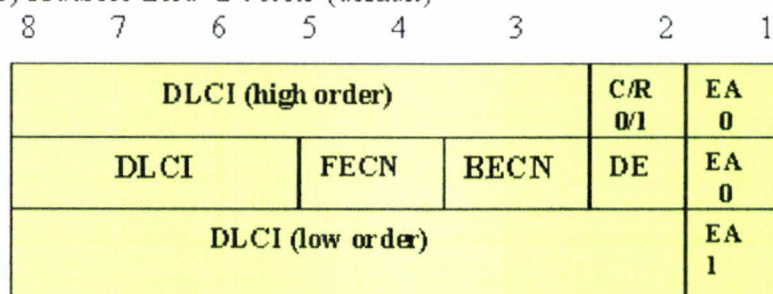
The frame relay frame derived from the HDLC (High level data link control) and it is another form of LAPD protocol. There is one major difference is that there is no control field. Frame structure is shown in the figure 2.3.



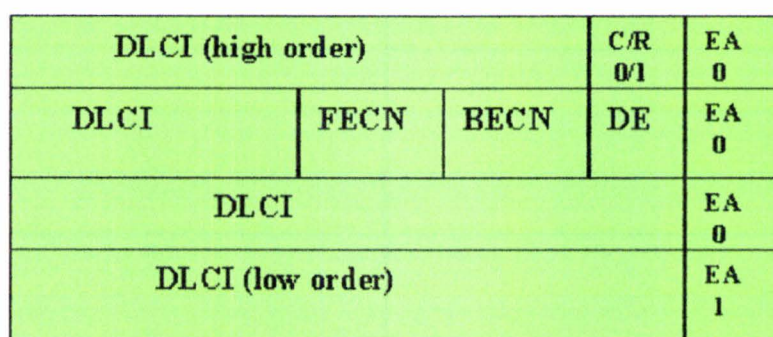
a) Frame format



b) Address field- 2 octets (default)



c) Address field- 3 octets



d) Address Field – 4 octets

C/A- Command/Response indicator

EA- Extended Address

F- Flag

FCS- Frame Check Sequence

DE- Discard Eligibility

DLCI- Data Link Connection Identifier

FECN- Forward Explicit Congestion Notification

BECN- Backward Explicit Congestion Notification

Figure 2.3 Frame Relay frame structure

This was defined in ANSI Standard T1.618. The flag field is a unique string of bits, which indicates the start and end of a frame. There are two octets of address field. The address field can vary from 2 to 4 octets in size. The primary function of the address field is to carry the DLCI. This allows multiple connections to be carried over a single channel (multiplexed). The Data Link Control Indicator field indicates that octet is an extended or not extended DLCI. The data field can vary in size up to 4096 octets long. Frame Relay is able to support multiple higher level protocols by encapsulating the other protocol data units (PDUs) inside the data field of FR frame. This allows the FR network to be transparent to these upper layer protocols. This size of the data field is limited by the integrity of the frame check sequence (FCS). If any error was found at the receiving end, that frame will be discarded. Discard Eligibility (DE) is set to indicate that a certain frame should be dropped in preference to other frames without the bit set when and if the link becomes congested. The frame also contains three bits, which are used for congestion notification.

2.7 Frame Relay Operation

Frame Relay uses the 10-bit DLCI to identify the destination end user address. These bits can be extended as shown earlier. DLCIs are pre-mapped to the destination node. So total process was simplified at the router. Now the nodes only need to see their routing table, for this DLCI and route the traffic to the proper output port based on the address. This translation of user address to DLCI is not defined in the FR standards. Some of the Switch vendors strip-off the FR header and trailer at the sending User Network Interface (UNI) and add their control fields to transfer the data to the receiving UNI. At the receiving end, FR header will replaces the internal network header.

2.8 Error Control and Recovery

One of the basic assumptions under the FR is the transmission media should be free from error. However, current digital transmission technologies, in particular optical fiber, have very low error rates. In FR the end user equipment does correction and retransmission. The network can detect the errors in the frame by using cyclic

redundancy check (CRC) operation only on the header field. If it detects any error in the address field it will just discard the frame.

2.9 Congestion control and Management

All of the users of FR network are sharing the same medium capacity. It might happen that all the users may try to access the network simultaneously. As a result congestion will happen in the network. In FR, congestion identification and control will be done in different ways, and it all depends on the vendor of the switch.

Mainly these congestion mechanisms will come under two categories. One is implicit congestion control. Usually in this type of networks the top layers will control this function. By seeing the delays of the acknowledgement packets it will control the speed of the packets, which are sending, into the frame relay port. Second one is explicit congestion control. This will use Forward Explicit Congestion Notification (FECN) and Backward Explicit Congestion Notification (BECN) bits of the FR frame. Suppose the congestion occurs at one Nodal processor the FECN will be set to 1 and it will be forwarded to the next processor. The traffic coming from the destination side knows about the congestion. The congested processor will set same way the BECN bit, which are going in the opposite side. Now the both ends of the networks know where the congestion happened. This mechanism implementation is depends on the user equipment and provider of the network [4]. This has been explained in the figure 2.4.

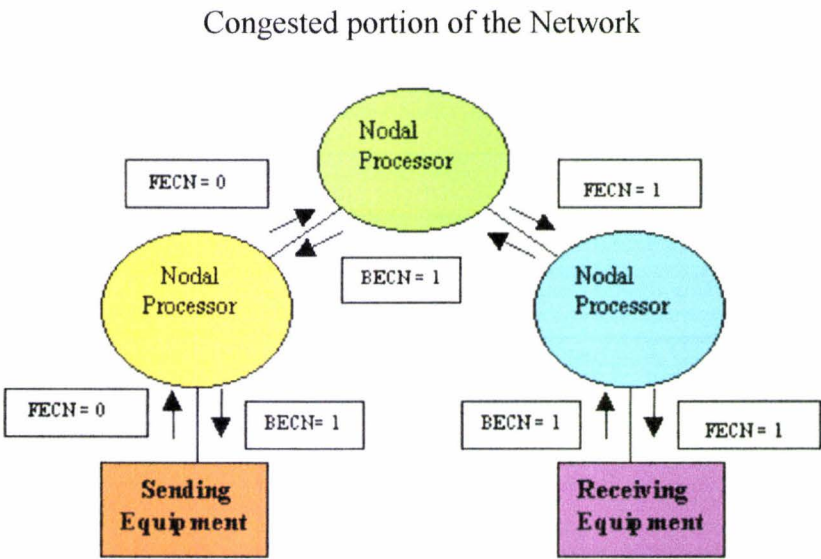


Figure 2.4 FECN and BECN action

The vendor has to fix the time interval T_c and it was not defined by the standards. CIR can be calculated by B_c/T_c . This was implemented by Leaky bucket algorithm [5] and the values of the different parameters are adjusted. Some of the vendors will use the rate control mechanism. By decreasing and increasing the transmission speeds at a fixed step, it will control the congestion. This has been implemented by British Telecom Networks [6]. Differences between the 2 types of congestion control methods concern response time, data flows and switch processing capabilities.

2.10 Internetworking with Other Networks

The interconnection with FR can be implemented in different ways. Some of the strategies are

- Over a point-to-point line, connected two routers directly.
- Using Single or multiple mode mixed media processors.
- Data only service for any type of networks through Public Networks.
- Hybrid Network implementation.

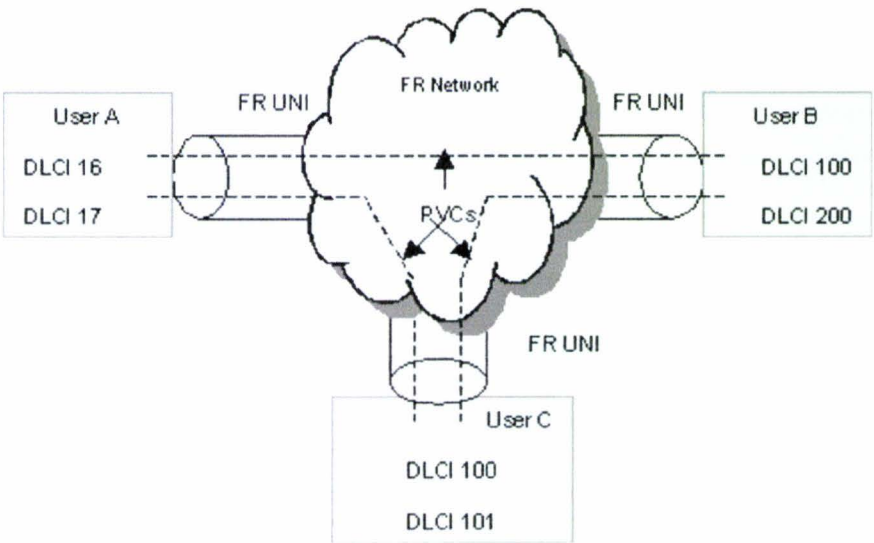


Figure 2.5 Frame Relay UNI and PVC

The procedure of the interconnection between user equipment and network is called user network interface (UNI). The PVC, which is passing more than one network, is

called multi-network PVC as shown in the figure 2.5 and 2.6. The procedure for implementing UNI and NNI was published in ANSI T1.617 Annex D.

Figure 2.5 illustrates the FR network environment, with three users accessing the FR network through a UNI, each with a PVC to the other site, and each PVC end point identified by a DLCI.

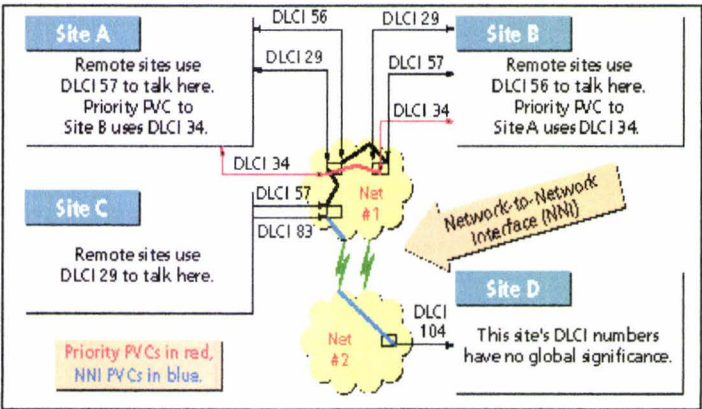


Figure 2.6 Logical layout of a Frame Relay Network (Cisco press)

The FR network to network Interface (NNI) is defined by the standards as a method for two FR networks to interconnect, pass FR traffic, and manage the logical connections (PVCs) which originate on one FR network and terminate on another. Figure 2.6 shows an example of utilization of priority PVCs. Frame Relay Data link Connection Identifiers (DLCIs) are mapped by carrier switch into permanent virtual circuits. Here in figure 2.6 Site A, B and C use DLCIs to identify sites. Sites A and B use a common DLCI number to identify a circuit whereas site D does not assign any meaning to DLCIs.

2.11 Effect of different Service Parameters

Figure 2.7 explains different FR network service parameters affecting the virtual circuit.

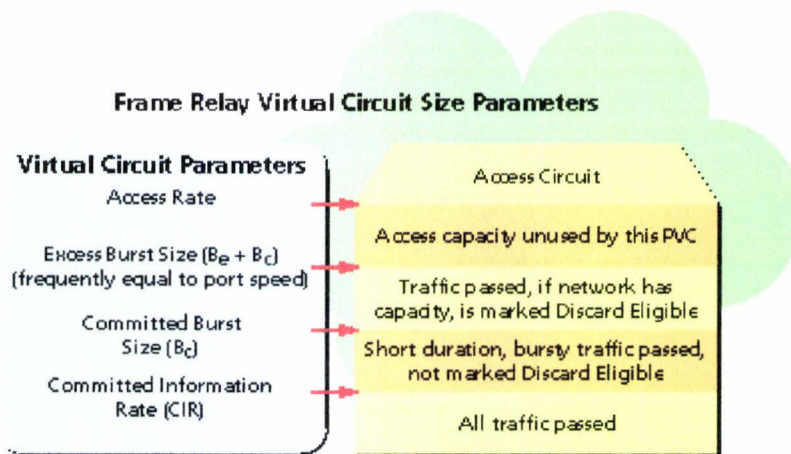


Figure 2.7 Important parameters for virtual circuits (Cisco press)

Access rate is the rate at which the access circuits join the FR network and are typically at 56 Kbps, T1 (1.536 Mbps) or Fractional T1 (a multiple of 56 Kbps or 64 Kbps). It is not possible to send data at higher than access rate.

Port speed is the rate at which the port on the FR switch is clocked. It is not possible to send data at higher than port speed.

Committed burst size (B_c) is the maximum amount of data that the network will transfer in a burst, defined over a (short) interval. All frames received at this level will be passed.

Committed information rate (CIR) is the amount of data that the network will receive from the access circuit. All frames received at this level will be passed. CIR can be computed by dividing the committed burst size by the measurement interval as $CIR = B_c / T_c$

Excess burst size (B_e) is the amount of data above B_c that the network will try to deliver. Frames submitted at this level may be marked as "Discard Eligible," indicating that they may be dropped if there is not capacity in the cloud.

A Data-Link Connection Identifier (DLCI) identifies a virtual circuit to user equipment. This is an end point for a PVC. Based on the standard, this number doesn't describe anything more than that, and has no significance beyond the single

link. In practice, however, many designers assign some significance to it. It often represents either a site or a circuit.

When using Frame Relay, overhead is little and overhead decreases as the frame size increases. Larger the frame size accepted by the network, the greater the throughput. [7] Throughput can be calculated based upon number of data bit successfully transferred from one boundary to another within a set period of time. From the FR throughput graphical representation of figure 2.8, the relationship between CIR, Bc and access rate can be explained. As explained earlier, all data below the CIR line will be received. Frames above CIR and below Maximum Burst Rate will be tagged Discard Eligible (DE). Frames above MBR will be discarded.

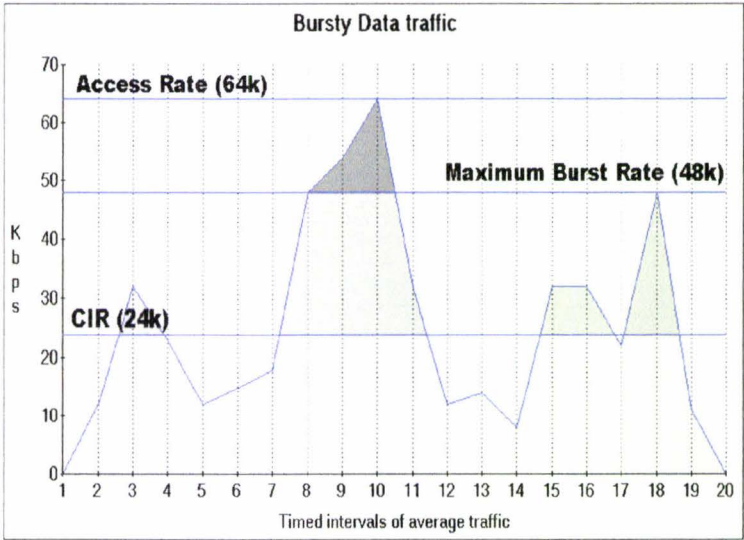


Figure 2.8 FR Throughput graph (PVC / Port level)

Chapter 3

Voice Over Frame Relay

In May 1997, the Frame Relay Forum ratified its Voice Over Frame Relay (VoFR) Implementation Agreement (IA) FRF.11. This IA defines standards for the transmission of voice over a public FR network. Included in the IA is the method by which compressed voice should be placed within the payload of a FR Frame, how DTEs can signal each other concerning the use of a specific voice compression method, and a method for multiplexing up to 255 voice sub-channels on a single DLCI. As vendors develop FRF.11 compliant products, it is eventual that use of FR can be in a similar manner like public switched telephone network with employees of one company contacting employees of another company via the use of Frame Relay. [8]

The block diagram of reference model [IA] for Voice over Frame Relay (VoFR) is shown in figure 3.1. Using the VoFR service, it is possible for any of VFRAD on the left hand side of figure 1 to exchange voice and signaling with any type of VFRAD on the right hand side of figure 1. End System devices could be a PC with Fax or telephony application software using a FR network port for connectivity to other VFRAD devices.

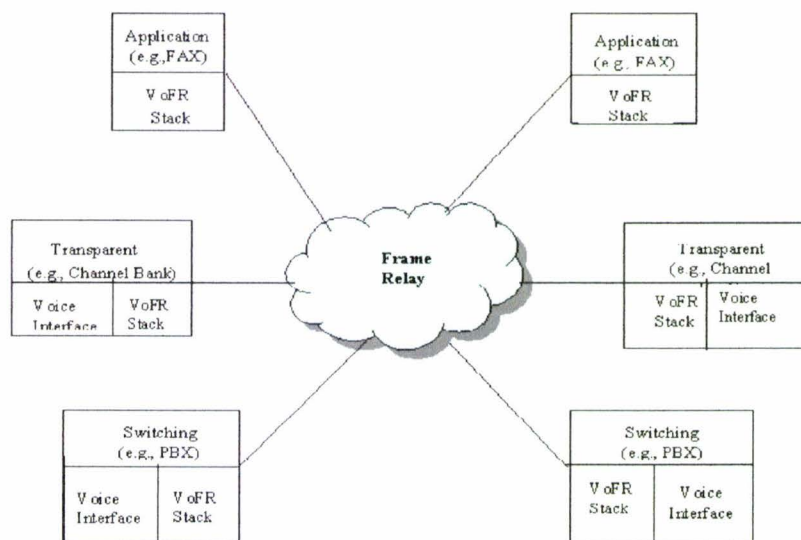


Figure 3.1 VoFR Reference Model

FRF.12 is the Frame Relay Fragmentation Implementation Agreement (IA). Fragmentation queuing reduces both delay and delay variation in FR networks by

dividing large data packets into smaller packets and then reassembling the data into the original frames at the destination. This ability is particularly relevant to combine voice and other time-sensitive applications, with non-time-sensitive applications or other data on single PVC. The main benefit of fragmentation is the ability to utilize common User to Network Interface (UNI) access lines or Network-to-Network Interface (NNI) lines and/or PVCs for communications combining large data frames and real-time protocols.

Fragmenting frames enhances the utility and uniformity of FR networks, reducing delay and delay variation while upgrading application responsiveness, quality and reliability. As a result, multiple types of traffic, such as voice, fax and data, can be transparently combined on a single UNI, NNI and/or PVC.

The Fragmentation Implementation Agreement provides for transmission of FR Data Terminal Equipment (DTE) and Data Communications Equipment (DCE) with the ability to fragment long data frames into a sequence of shorter frames that are then reassembled into the original frame by the receiving peer DTE or DCE. Frame fragmentation is necessary to control delay and delay variation when real-time traffic, such as voice, is carried across the same interfaces as data traffic. Fragmentation enables the interleaving of delay-sensitive traffic on one PVC with fragments of long data frames on another PVC using the same interface.

Through a fragmentation process, an even flow of minimized-length frames will be created, such that frames transporting digitized voice do not have too long behind frames transporting data. Figure 3.2 shows a long data a FRAD or Router that is particularly programmed to intersperse frames transporting voice and data divides frame into a sequence of shorter length frames.

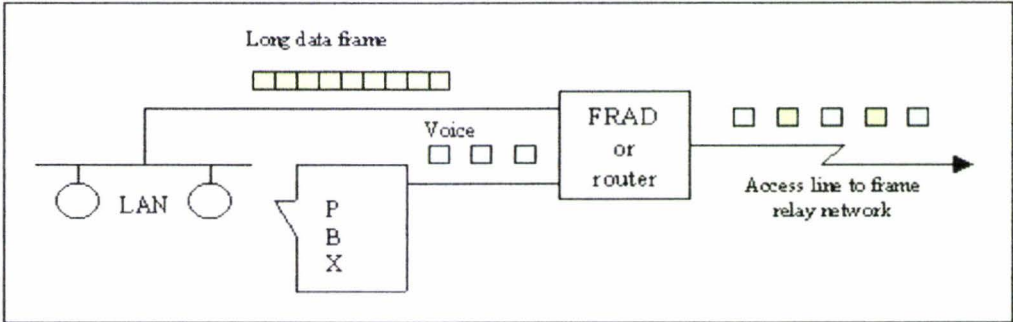


Figure 3.2 Fragmentation of frame

Digital technology offers a lot of advantages, including greater bandwidth, better error performance and enhanced management and control. Voice is analog in nature. Therefore analog voice has to be coded into a digital format at some point prior to entering the WAN and decoded back into analog format as or after it exits the WAN. Those conversion processes are accomplished by a matching pair of codecs, with a conventional method being Pulse Code Modulation (PCM), standardized by ITU-T as G.711. PCM specifies that the amplitude of the analog signal be sampled 8000 times per second, at precise intervals of 125ms. Each sample is coded into 8 bit digital value. The resulting eight-bit bytes are interleaved by multiplexers and sent across channelized digital circuits to be directed and redirected by circuit switches, and decoded on the receiving end of the transmission. The decoded signal is in analog form, an approximation of the original signal. Unlike most data communications, which can tolerate delay, voice communications must be performed in near real-time. This means that transmission and network delays must be kept small to remain hardly noticeable to the user.

Packetized voice is now possible; low bit rates are attained by analyzing and processing only the essential components of a voice sample, rather than attempting to digitize the entire voice sample (with all the associated pauses and repetitive patterns). Current speech processing technology takes the voice digitizing process several steps further than conventional encoding methods.

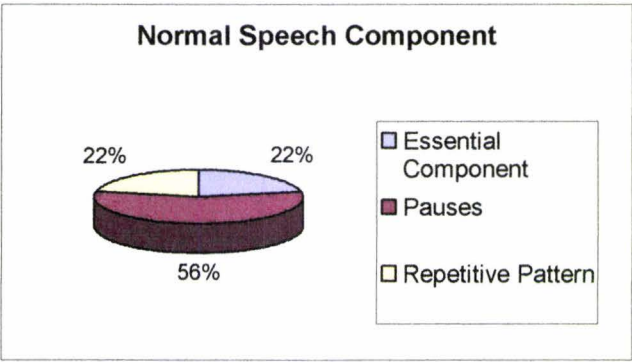


Figure 3. 3 Speech components for high-quality voice communication

Analysis of a representative voice sample shows that only 22 percent of a typical conversation consists of essential speech components that need to be transmitted for

complete voice clarity (Figure3.3). The balance is made up of pauses, background noise, and repetitive patterns.

Transmission of these identical sounds is not necessary. Removal of Repetitive Speech Sounds can increase bandwidth efficiency. The suppression and removal of silent periods can also significantly improve bandwidth utilization.

The removal of silent periods and redundant information through advanced techniques enables voice to be efficiently “compressed”. After the removal of repetitive patterns and silent periods, the remaining speech information may then be digitized and placed into voice packets suitable for transmission over a FR network. These packets or frames (both terms are often used interchangeably) also tend to be smaller than the average data frame. The use of smaller packets helps to reduce transmission delay across a FR network. The concepts introduced above provide the basis for efficiently using the smallest amount of bandwidth possible for voice transmission over a FR network. Figure 3.4 describes the transport of voice through multiple networks as reference architecture.

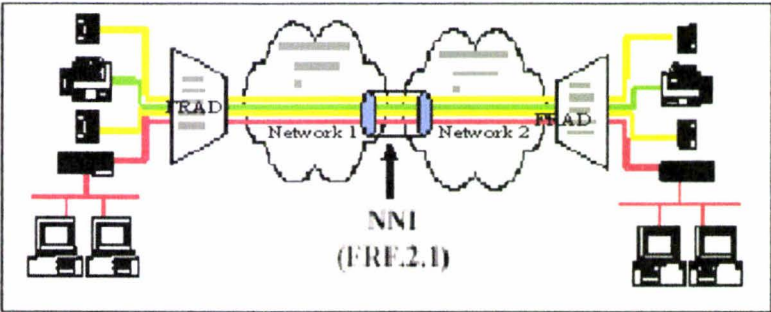


Figure 3.4 Transport of compressed voice in FR frames over multiple networks (Reference architecture from FR Forum)

Voice compression is the method of reducing the amount of digital information below the traditional 64-kbps. The available bit rate is reduced from 64 Kbps to 32, 16, 8, and 4 Kbps or below, the strategies for redundancy removal and bit allocation need to be more sophisticated. Advances in technology have greatly improved the quality of compressed voice and have resulted in a spectrum of ITU standards for compression algorithms. The most common compression methods, and their associated quality scores, are listed in Table 3.1

Table 3.1: Compression Methods

| Compression Method | ITU Standard | Payload Bandwidth | MOS Score | Delay |
|---------------------------|---------------------|--------------------------|------------------|--------------|
| PCM | G.711 | 64-kbps | 4.1 | 0.75 ms |
| MP-MLQ/ACELP | G.723.1 | 6.3-kbps/5.3-kbps | 3.8/3.75 | 30 ms |
| ADPCM | G.726 | 32-kbps | 3.85 | 1 ms |
| LD-CELP | G.728 | 16-kbps | 3.61 | 3-5 ms |
| CS-ACELP | G.729 | 8-kbps | 3.9 | 10 ms |
| CS-ACELP | G.729a | 8-kbps | 3.85 | 10 ms |

Digital Signal Processing (DSP) is a computer discipline specializing in applications such as voice coding and compression. DSP chips perform these functions on streams of digital traffic, and are commonly referred to as codecs (a contraction of coding or decoding).

MOS, or Mean Opinion Score, is a widely quoted subjective measure of voice quality. Scores of 4 to 5 are deemed toll-quality, 3 to 4 communication quality, and less than 3, synthetic quality. Table 3.1 shows sample MOS scores for the commonly used codecs. MOS scores are not standard; they are based on the results of individual studies and although the ITU guides how studies should be conducted (P.800 series), they do not publish standardized MOS scores.

The contents of Table 3.1 make clear the opportunity that exists to integrate voice and data networks while maintaining high voice quality. The trade-off is in between MOS scores, bandwidth use and delay. When designing networks, these factors must be balanced to ensure overall voice quality as well as meeting network objectives.

CIR Enforcement is another important factor to consider. In a FR service; the backbone network recognizes only the main FR address. The sub frame structure of VoFR is hidden from the switches. By definition, any bandwidth parameter can be assigned only to the DLCI or PVC as a whole. It is up to the user's equipment, the VFRAD, to enforce the bandwidth constraints when and where the frames originate. After that it is very difficult to distinguish between voice and data channels with the same DLCI. Network designers recommend that any FR virtual circuit that carries voice be assigned a CIR at least as large as the sum of the encoding rates for all voice channels on the Virtual circuit [9]

3.1 Voice Quality of Service (QoS)

3.1.1 Echo Cancellation

Echo occurs when the transmitted voice is reflected back to the point from which it was transmitted. In voice networks, echo cancellation devices are used within a carrier's network when the propagation delay increases to the point where echo results. The longer the distance, the more the delay, and the more likely that echo will result. Voice transmitted over a FR network will also face propagation delays. As the end-to-end delay increases, the echo will become noticeable to the end-user if it is not canceled. Standards-based echo cancellation is used to compensate for this.

3.1.2 Delay and Delay Variation

Two common network characteristics that affect quality are delay and jitter. Delay can cause two potential impairments to speech. First, long absolute delays cause both speakers to tend to begin to talk at the same time. Second, delay exacerbates echo, which is the reflection of the original signal back to the sender. Echo is indiscernible under low delay.

The bursty nature and variable frame sizes of FR may result in variable delays between consecutive packets. The variation in the time difference between each arriving packet is called "jitter". Since voice is inherently a continuous waveform, a large gap between the regenerated voice packets will result in distorted sound. Employing fragmentation of data packets in order to transmit uniform packet sizes into the network can mitigate

jitter across the network. FRF.12 provides an interoperable technique for fragmenting data prior to its transmission into a FR network.

To avoid dropping speech samples, data can be buffered sufficiently at the speech decoder to account for the worst-case delay jitter through the network.

Table 3.2 shows a summary of the ITU recommendations for voice delay guidelines. One-way delays below 150 milliseconds (ms) are considered acceptable for most applications. [10] Furthermore, higher delays may be acceptable if cost savings are taken into account. Generally, voice networks should be designed to achieve less than 200ms one-way-delay.

Table 3. 2: ITU Delay Recommendations

| One-Way Delay | Description |
|---------------|--|
| 0 to 150 ms | Acceptable for most user applications |
| 150 to 400 ms | Acceptable provided that administrators are aware of the transmission time impact on the transmission quality of user applications |
| 400+ ms | Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases this limit will be exceeded for example, with satellite connections |

3.1.3 Frame Loss

Packet loss is a phenomenon on packet networks. Loss can be caused by many different reasons: overloaded links, congestion, excessive collisions on a LAN, and physical media errors etc. Transport layers such as TCP account for loss and allow packet recovery under reasonable loss conditions. Network devices, like switches and routers, sometimes have to hold data packets in buffered queues when a link gets congested. If

the link remains congested for too long, the buffered queues will overflow and data will be lost.

In current IP networks all voice frames are treated like data. Under peak loads and congestion, voice frames will be dropped equally with data frames. Data frames are not time sensitive like voice frames and there is no point in retransmission of lost frames as in voice transmission. To compensate for this, the network must be designed to selectively drop packets only from traffic types that can tolerate it. The codecs have auto-fill algorithms that will cover the loss of the occasional packet to the end user's ear.

3.2 Fixed Delay Components

The components of fixed delay are propagation delay, Serialization Delay and Processing Delay. Some fixed delay components can be controlled by the network designer, while others cannot.

Propagation Delay—is based on the distance between source and destination. As a planning number, 6 microseconds per kilometers can be used. Propagation delay can be a significant factor in inter-continental connections

Serialization Delay—is the process of placing bits on the circuit. So, the higher the speed, the less serialization delays. For example, it takes 125 ns to place one byte on a 64-kb circuit. Serialization delay is a significant factor on link speeds slower than 768-kbps. Serialization delay can be control by adjusting packet sizes, or by deploying packet fragmentation QoS techniques

Processing Delay—includes both packetization and algorithmic delay. Algorithmic delay is the time it takes the codec to produce a sample of voice. This varies between 1 and 30 ms for typical codecs. This function is often performed in Digital signal processor (DSP) hardware and firmware, although it can be done in software as well. Packetization delay is the process of holding the digital voice samples for placement into the packet until enough samples are collected to fill the packet payload.

3.3 Variable Delay Components

Queuing Delay—Queuing delay is only incurred when there is contention on the interface, when there is more traffic waiting to go out than the link speed can accommodate. Queuing delays are therefore more likely and more severe on slower speed interfaces. Proper configuration of routers is important to minimize this type of delay.

Dejitter Buffers—are used at the receiving end to smooth out the delay variability of the arriving voice packets. A dejitter buffer reduces or eliminates delay variation by converting it to constant delay.

3.4 Packet Voice Transport Options

From the previous section on voice compression, it is clear that coding, delay, and packet loss are the most serious problems in voice networks. While all packet voice networks must address these issues, the sensitivity of the packet voice network options to each issue is different, and packet voice application design must consider the transport technology before deciding on how to deal with specific delay or loss concerns.

Packet voice can be transported over any of the following wide-area network connection types as follows

- Leased-line networks. These networks are often based on T1/E1 leased from carriers, providing fixed synchronous bandwidth. Protocol encapsulations such as Higher Level Data Link Control (HDLC), Point-to-Point (PPP) and Multi link Point to Point (MLPPP) are typically used on leased lines.
- Public IP networks, including the Internet.
- ATM constant-bit-rate (CBR) or circuit emulation connections, Variable Bit Rate (VBR), Available Bit Rate (ABR), or Unspecified Bit Rate (UBR) classes of service.

- Frame Relay networks, both those provided by public service providers and those built by corporations as private networks. These provide public data services in many international locations and are also widely used as network access technologies from smaller branch office locations.

Figure 3.5 shows Frame Relay Service on ATM Backbone

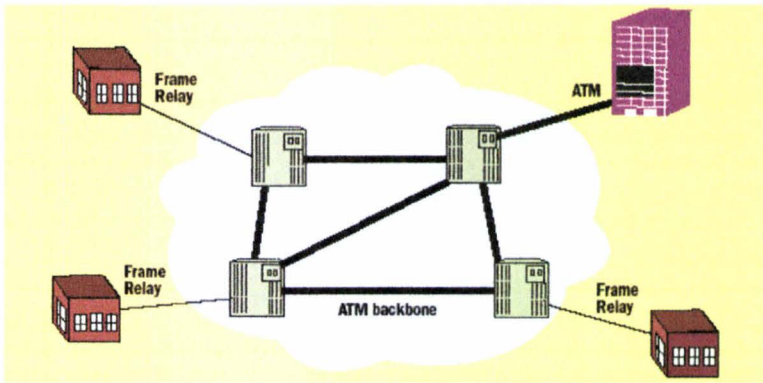


Figure 3.5 Frame Relay Service on ATM Backbone

Chapter 4

Review of Literature

4.1 Real time applications

Multimedia traffic is becoming more common. Real-time interactive video conferencing and IP telephony, non-interactive traffic such as video and audio on demand, real time streaming of data such as stock quotes, and other new and developing uses have placed demands on the Internet that the system was not originally designed to meet. There is more to transmitting audio and video over a network than just providing sufficient bandwidth.

We refer to applications that are sensitive to the timeliness of data as real time applications. The characteristics of real-time applications are that they need some sort of assurance from the network that data is likely to arrive on time. Non-real-time applications focus more on the correctness of the data that are transmitted. This means retransmission when data arrives too late or is corrupted. Retransmission means increased latency, but no harm is done as long as the data arrives within reasonable time limits.

The Best Effort model tries to deliver data, but makes no promises neither for timeliness nor guaranteed delivery. This is not sufficient for real-time applications. A summary of different kinds of applications can be made in order to better understand how complex the needs for QoS guarantees are. Quality of Service (QoS) refers to the capability of a network to provide better service to selected network traffic over various technologies, including FR, Asynchronous Transfer Mode (ATM), Ethernet, SONET, and IP-routed networks that may use any or all of these underlying technologies. The primary goal of QoS is to provide guarantees including dedicated bandwidth, controlled latency and jitter, and improved loss characteristics. Dedicating bandwidth allows some of the clock rate to be reserved for certain traffic. Several solutions were found for the delay sensitive type of traffic in the earlier research. [11,12].

We can divide applications in two types: non-real-time and real-time.

Non-real-time applications are also called elastic and include common applications like Telnet, File Transfer Protocol (FTP), email, Web browsing, and so on. They are often bursty, i.e. they have unpredictable delivery of “blocks” of data at a variable bit rate (VBR). All of these applications can work without the guarantees of timely deliver of data, but the delay requirements may vary from interactive applications like Telnet to more asynchronous ones like email.

Real-time applications can be divided into two groups, interactive applications and one-way streaming applications. Both have predictable delivery at a relatively constant bit rate (CBR). Two or more people that talk together on the Internet typically use an interactive application. They have strict demands to delay and the amount of data that are transferred is small. Such data gets delayed by other traffic on the Internet and may arrive too late at the receiver. One-way streaming services are less delay sensitive, since the data is sent only in one direction. Streaming usually aims at giving a live audio or video experience at the receiver. The service uses an adaptive playback buffer to limit variations in delay. Table 4.1 summarizes Quality of services (QoS) requirements for some common application types.

Table 4.1: QoS requirements for common application types.

| Application Types | QoS requirements | | | |
|--------------------------|----------------------------|-----------------------|----------------------|---------------------------|
| | <i>Bandwidth</i> | <i>Latency</i> | <i>Jitter</i> | <i>Packet Loss</i> |
| E Mail | Low to-Moderate | - | - | - |
| File Transfer | Bursty High | - | - | - |
| Telnet | Bursty Low | Moderate | - | - |
| Streaming Media | Sustained Moderate to High | Sensitive | Sensitive | Sensitive |
| Videoconferencing | Sustained High | Critical | Critical | Sensitive |
| Voice over IP | Sustained Moderate | Critical | Critical | Sensitive |

Playback time is the point in time at which the data from the sender is needed at the receiving host. Recommendations from ITU-T show that a playback time less than 150 ms is acceptable for most user applications. Data that arrives after the playback time is completely worthless. Delay does not affect voice quality directly but instead affects the character of a conversation. Below 100 ms, most users will not notice the delay. Between 100 ms and 300 ms, users will notice a slight hesitation in their partner's response. Beyond 300 ms, the delay is obvious to the users. Obviously, shorted delay results in better conversation quality and in better-perceived overall voice quality. [13]

Usually, a playback buffer is used to make sure that arrived data is played back at a steady rate in an application. It is used to minimize jitter introduced when packets are traversing a network, and as long as the playback time is after packet arrival and within acceptable time limits, jitter is never noticed by the application.

If the packet loss varies with time, the playback point may be shifted to play out samples at an increased or decreased rate for some period of time. With a voice application, this can be done in a way that is barely perceptible, simply by shortening or increasing the silences between words. Applications that can adjust their playback point are called delay-adaptive. There are also rate-adaptive applications, which are used in videoconferencing. Many video-coding algorithms can trade-off bit rate versus quality, so if the network only supports a certain bandwidth, the picture is compressed harder. If more bandwidth becomes available the compression is lowered to increase the quality.

Real-time applications are used in many different areas. An example of a network that is adjusting to the IP standard is the global mobile telephone network. [14]. Third generation mobile networks will have their own IP backbones connected to the Internet, and there will be a demand for QoS guarantees for real-time applications in the same way as in fixed networks. The demands for throughput and delay are difficult to fulfill, and it is important that resources are shared in the best possible way.

4.2 Frame Relay Networks

A Frame Relay network consists of endpoints (e.g., PCs, servers, host computers), FR access equipment (e.g., bridges, routers, hosts, Frame Relay access devices) and

network devices (e.g., switches, network routers, T1/E1 multiplexers). Accessing the network using a standard FR interface, the FR access equipment is responsible for delivering frames to the network in the prescribed format. The job of the network device is to switch or route the frame through the network to the proper destination user device. A Physical layout of a FR network is shown in the figure 4.1.

A Frame Relay network is often depicted as a network cloud, because the FR network is not a single physical connection between one endpoint and the other. Instead, a logical path called a virtual circuit is defined within the network. Bandwidth is allocated to the path until actual data needs to be transmitted. Then, the bandwidth within the network is allocated on a packet-by-packet basis.

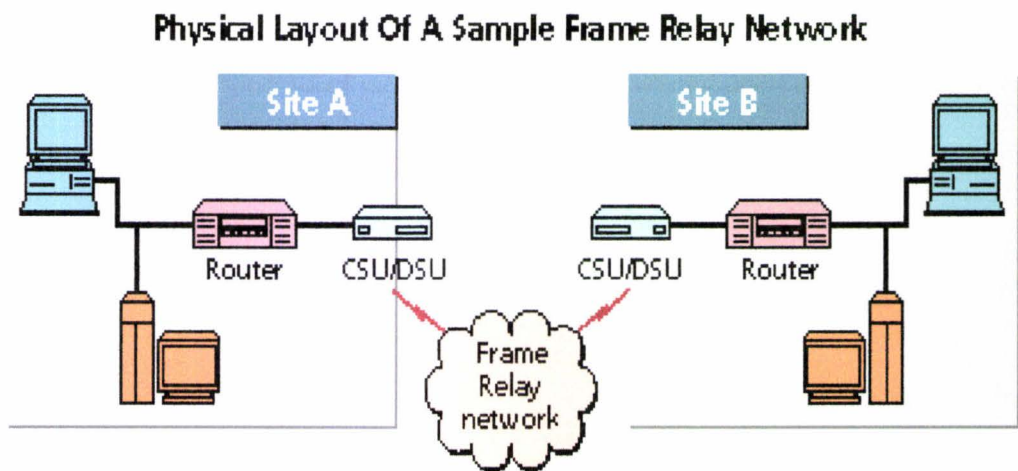


Figure 4.1: Physical layout of a FR network

The Frame Relay industry has moved away from point to point based line connections to FR virtual network, because the virtual network provides for a robust connectivity solution at a less expensive cost. FR was initially around a packet switching data communications need, especially for LAN-to-LAN traffic, stated to look for a means of carrying voice communications on a FR network. FR is easy to setup and allows users to build virtual connections from just about anywhere to build anywhere.

Frame Relay was good for bursty traffic as opposed to the sustained information rates of voice. What we traditionally think of as separate voice and data networks will converge and run over an IP infrastructure. VoFR is a logical development for offices already running data over Frame Relay.

Voice Frame Relay access devices (VFRADs), integrate voice into the data network by connecting the router SNA controller and the PBX at each site in the corporate network to the FR network employ sophisticated techniques to overcome the limitations of transporting voice over the Frame Relay network without the need to add costly bandwidth.

A paper found regarding voice and multimedia over ATM discusses the simplification of networks. [15] Regardless of whether voice is sent over packet networks (ATM, Frame Relay or IP) or traditional TDM-based networking, there are two fundamental models for the transport of voice: "voice trunking" and "voice switching". Voice trunking typically involves the tunneling of voice traffic across a network between two fixed end-points and is widely used for connecting voice switches and PABXs. With this methodology, there is no requirement for the network to be able to process or terminate voice signaling, other than the opportunity to use the signaling to detect idle channels. By contrast, voice switching involves the packet or TDM network in the interpretation of voice signaling information and the routing of calls across the network. Network design is simplified when network can interpret signaling and route calls accordingly. This method also reduces the number of access ports and virtual circuits required

Frame Relay is implemented in both public carrier-provided networks and in private enterprise networks. Service providers that intend to offer transmission services to customers provision the majority of Frame Relay networks deployed today. This is often referred to as a public FR service. Public FR's main application is to build WANs to interconnect multiple remotely located local area networks (LANs) using routers and bridges. Products like as shown in figure 4.2 by MICOM can integrate simultaneous voice, data, fax and LAN traffic over a single WAN connection, significantly reducing the communications costs. MICOM's integration products use voice, fax and data compression, combined with silence suppression and dynamic bandwidth allocation, to fully utilize a FR connection. More than 100 Kbps of LAN-WAN throughput, together with two voice/fax connections and a data channel, can travel over a 56 Kbps FR access line is working nowadays.

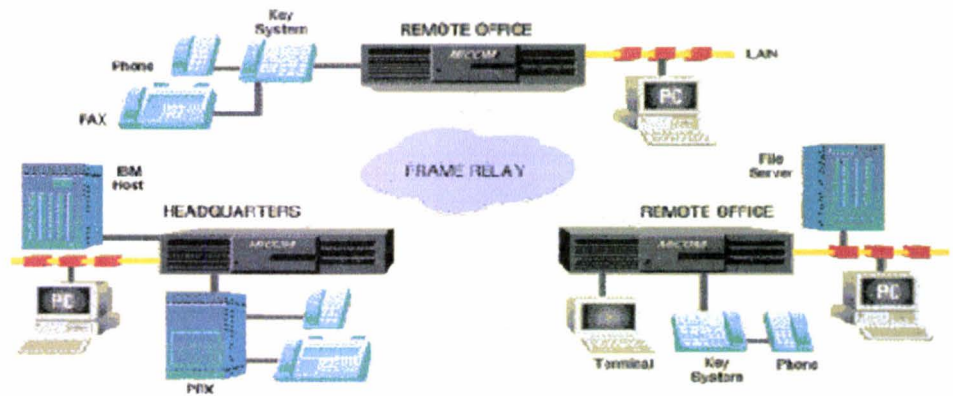


Figure 4.2 Public Frame Relay network (MICOM)

Users with many distributed sites and large volumes of traffic shuttling from these sites to headquarters are starting to benefit by integrating their voice and data networks on a single packet network. A considerable number of VoFR ports are in use, though they represent a small percentage of the overall total of public FR ports. According to Vertical Systems Group, [16] a network research firm in Dedham, MA, about 14,000 VoFR ports are in use in the U.S., 18,000 throughout the rest of the world

Public FR services offer the user a cost-effective alternative to leased lines services, often saving as much as 30 - 40%. [17]. The public FR connection the user was going to use for data or LAN traffic to a remote site can now carry simultaneous voice and fax to the same site. In effect, voice and fax ride for free over an already-paid-for remote data or LAN connection. In most applications, the bandwidth used by voice and fax traffic to a branch office is only 2 - 5% of the access bandwidth, leaving the remaining 95 - 98% to carry the data and LAN-WAN traffic [17]. Now a hybrid combination of Private and FR networks are going to be used. With a hybrid combination, the communications between branch offices are faultless and transparent and, as connection costs change, changing connection types without modifying the equipment can optimize the network.

Most carriers are now offering QoS (Quality of Service) or SLA (Service Level Agreement) guarantees, which categorically provide an end user with confidence. A discussion of the current status of combining Voice with Fax, Data and LAN Communications using Frame Relay over Public and Private Lines shows that VoFR is fairly well accepted as being economical when used in international applications. [18]

Previous research has shown that transmission of compressed voice over both FR and packet networks are possible by using an adaptive build out delays experienced in the network. [19]

Packetized voice synchronization has received considerable attention in the literature [20- 23] in the last decade.

An article found which discusses Voice Synchronization over Compressed Voice in Integrated Services Frame Relay Networks [24]. This paper reevaluates the original playout algorithm in the context of inter-LAN Ethernet traffic and presents a Modified Adaptive Buildout Delay algorithm for improved voice performance in the presence of bursty data traffic. This paper presents a design to absorb network delay jitter by adaptive variation of the holding time of voice packets at the playout buffer. The algorithm requires no common time-reference between sender and receiver or special processing in the FR switches and has been optimized to operate in a mixed voice-data FR environment characterized by bursty inter LAN Ethernet traffic. It meets the quality of service (QoS) requirements of conversational voice applications.

An early paper was found [25] where Modeling & control of Variable bit rate (VBR) M.261 transmission over integrated services FR network is investigated. Here the methodology is introduced to permit simulations of VBR codec in which the codec quantization step size is adjusted in response to prevailing network condition. It shows that QoS requirements of VBR video can be met in the presence of inter-LAN data traffic by making use of the Frame Relay BECN facility; in conjunction with a modified H.261 codec whose rate is controlled by the congestion notification.

There are research going on relating class based traffic management for voice and video. A recent paper published [26] shown that per-flow delay statistics, including the average and the percentile delay, can be very different from the corresponding class delay statistics, depending on flow burstiness, overall traffic load, as well as the queue discipline considering that the same type of applications can generate traffic having very different statistical behaviors such as video traffic with different activity levels, or voice traffic with different compression schemes They also propose a solution to reduce the mean delay variance experienced by flows in the same class.

If physical security is maintained, FR is as secure as connecting network end points with dedicated private lines. A study [27] comparing IP VPN and FR shows that Frame Relay offers security to the user automatically, while the IP VPN user must take additional measures to add security. Inherent in its technology, Frame Relay's predefined layer 2 PVC paths between the FRADs cannot be breached through FR user access.

A comparison of performances of voice over IP and voice over ATM [28] shows that the major factor affecting the performance of a consolidated voice and data network are QoS parameters. VOIP was found to show better performance compared to VoATM.

VoFR offered quality and reliability significantly better than voice over the Internet for real time interactive calls. But, as Steve Taylor, president of Distributed Networking Associates, Inc. and publisher, editor of Webtorials.com, regretted, "The best technology doesn't always win." Voice-over-Frame was marketed to the telecom community, who wanted excellent reliability. Most users favor Frame Relay as their service interface. Recent Research direction shows that Frame Relay Forum is working with the MPLS Forum on Frame-to-MPLS inter working and soliciting contributions on Frame Relay -to-Ethernet and Voice over Frame Relay inter networking [29].

Chapter 5

Optimized Network Engineering Tools (OPNET)

This chapter discusses briefly about the OPNET programming package and provides a quick overview of various tools of the OPNET that have been used in this project. By the end of this chapter, the reader should gain a basic knowledge of the network modelling process used in this study, for better understanding of the overall research.

Optimized Network Engineering Tools (OPNET) is a powerful comprehensive engineering system that can be used to model communication systems and predict network performance. It is capable of simulating large communications networks with detailed protocol modeling and performance analysis. Its accuracy and ease-of-use make it a valuable tool for network planners and administrators. OPNET supports modeling efforts with a system of interrelated programs, model libraries, and data files. The key features of this program include object orientation, graphical specification, automated model creation, an extensive model suite, integrated analysis tools, and animation support. The OPNET system consists of a set of good application programs, object code libraries, and data files. The OPNET components interact with the operating system and the user in a predictable manner that is common to other programs. For instance, OPNET programs run as regular user processes and do not interfere with the operating system's control of the computer. OPNET provides an opportunity for students, developers, and researchers of network communications to develop and foresee what is happening in complex network environments by changing parameters and seeing the corresponding impact through performance statistics.

5.1 Steps for Building a Network Topology

The initial step towards performance analysis of a network is to baseline the network topology. The methods for building the network topology vary depending on what information is to be obtained from the performance analysis. For example, a backbone

analysis may require the backbone portion to be modeled in detail but allow the remaining portions to be abstracted, a client-server performance analysis may require modeling of a single path between the client and the server. The software supports many different techniques for building network topologies. Topologies can be created manually or imported from tools that perform auto-discovery. The model library that comes with the software provides models of various devices used in current day networks. Devices that aggregate sections of the network (such as LAN segments, ATM/Frame Relay clouds) are provided in order to simplify the network topology and improve simulation performance.

Listed below are recommended steps that a user should follow during the process of building a network topology. There are four basic steps outlined for building a network topology as shown below in figure 5.1.

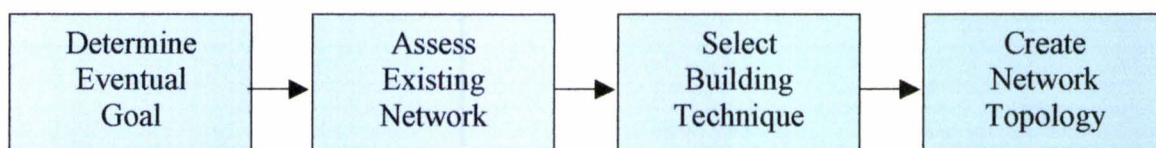


Figure 5.1 Basic steps for building network topology

5.1.1 Determine Eventual Goal

The choice of devices for constructing a network topology and the method is very much a function of the goal of a simulation study. Depending upon the context in which the software is being used, the topology may be represented as a single path between two devices, a partial topology with portions of the network abstracted or a complete topology with every device explicitly modelled.

Single Path: In this case, only the infrastructure supporting traffic between two devices of interest is represented. For example, if the objective of using the software is to analyse a client-server application, it is only necessary to identify the devices and networks constituting the single path between the client and the server. The effect of the remaining portions of the network will be taken into account by representing the traffic that crosses, and therefore affects, the path of interest. The advantage of this topology is that it enhances the focus of the study to the main objects of interest, i.e. the client and the server. Since

other devices are not explicitly modelled, the simulation is very efficient. There is no loss of accuracy since the effect of the remaining network on the client-server traffic is taken into account. A number of what-if scenarios can be easily and quickly performed by changing client/server parameters or the amount of traffic in the intermediate sections of the network.

Partial Topology: In certain situations, it may be important to represent portions of the network in detail while abstracting other sections. If the objective is to study the utilization of a backbone, the backbone portion must be completely represented. Other sections of the network may be abstracted. However their effects should be captured when traffic information is entered.

Complete Topology: A complete topology is necessary when a particular problem is scaled across the network and it is important to identify the impact of the problem on all devices involved. Table 5.1 listed some examples of goals and recommended topologies.

Table5.1 List of some examples of goals and recommended topologies

| Goals of The Simulation | Recommended Topology |
|---|----------------------|
| Performance of a database application | Single Path |
| Performance of FTP with different TCP settings | Single Path |
| Performance of applications for remote users | Single Path |
| Performance of a new application in the network | Single Path |
| Effect of a new application deployment on the network | Complete |
| Backbone analysis | Partial |
| Moving from shared media to a switched network | Partial |

5.1.2 Assess the existing network

Before beginning construction the final topology, it is important to have a complete picture of the existing network in the form of a drawing or a map. It is important to identify all

existing devices, their role, and the protocols that are running in the existing network. It is also important to identify the flows and traffic patterns in the existing network.

Aggregation: Before performing the final steps in generating a topology, it is important to determine if portions of the network can be aggregated. Aggregation can be performed at the segment level (LAN segment) or at the subnet level (IP subnet). Portions of the network, which are outside the corporate control such as the Internet or the carriers, can also be represented as simple "cloud" objects with the appropriate latencies. Examples of the above are illustrated graphically in figure 5.2, 5.3 and 5.4 .

A LAN segment can be aggregated into a shared LAN object. The number of workstations that the shared LAN represents and the different applications generating traffic can be specified. The shared LAN automatically scales the traffic based on the number of stations. Figure 5.2 describe the shared LAN aggregation for 5 nodes connected to a hub.

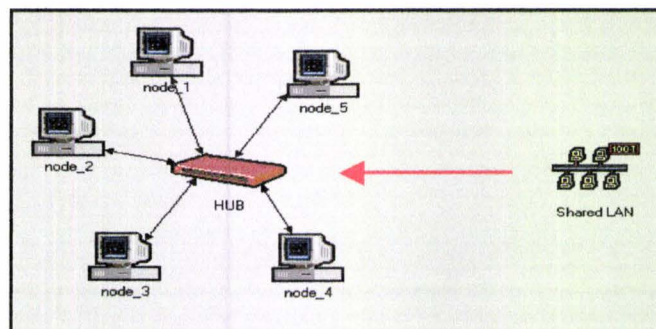


Figure 5.2 Shared LAN Aggregation (OPNET)

Shared LAN segments connected to a switch may be aggregated into a switched LAN object. Figure 5.3 shows a switched LAN aggregation of 5 shared LANs connected with a switch. The number of workstations that the entire switched LAN segment represents can be specified and would be set to the total number of users on all the shared LANs that are

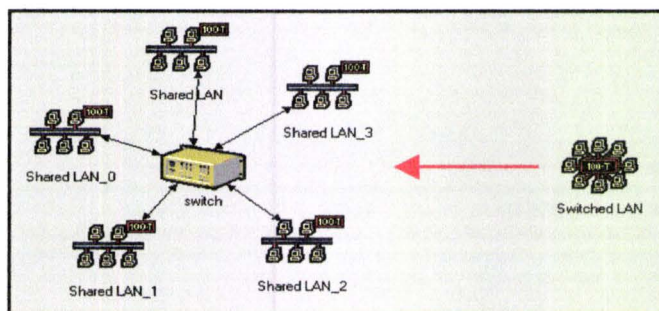


Figure 5.3 Switched LAN Aggregation (OPNET)

utilizing the switch. Applications can be configured on the switched LAN and the model automatically scales the total traffic based on the number of workstations.

The Frame Relay Cloud can be used to represent a section of the core FR network as shown in figure 5.4. Typically, a company uses a Frame Relay carrier service for long haul communications and the carrier infrastructure can be represented as the FR cloud. Packet latency and discard probabilities can be modeled on the FR cloud.

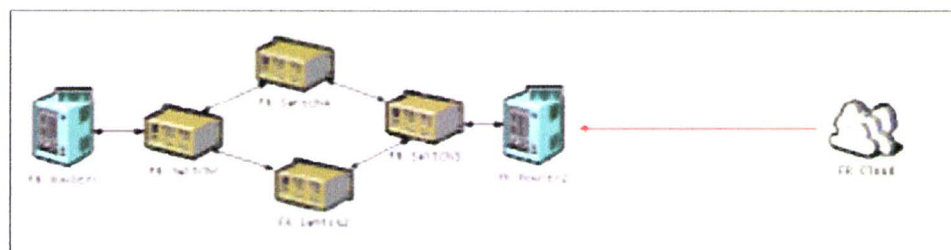


Figure 5.4: Frame Relay Aggregation

5.1.3 Select a technique for creating the topology

Topologies can be created manually or automatically through a process of importing from tools. Each one of these techniques is discussed in detail.

Direct Import: The software supports importing topologies directly from tools.

The import preserves the network layout and hierarchy. The relative positioning of objects is preserved. If objects are within subnets, the software will create subnets and place objects within them. Devices are mapped accurately to the model library. The software maintains a large database of device models (e.g. routers, switches, servers etc.) and their characteristics. During import, devices are identified based on their function and vendor.

Manual Construction: Manual construction can be used when the topology is simple (in terms of number of objects and complexity of interconnection) or when supporting tools that provide topology information for direct import are not available.

To build topologies manually, the software provides a number of object palettes that contain common network devices and links used for interconnection. The software also provides a technique called "Rapid Configuration" that allows to quickly create standard topologies (star, tree, bus, mesh, etc.) containing many devices with a few clicks. Once a topology has been created by hand, the software provides features that allow the user to select a large number of objects and apply attribute values in one operation. Consistency checking is provided to ensure that links are accurate and the interconnected devices are compatible.

5.1.4 Create the topology

Once the steps above have been completed, one can build the network topology. The link consistency check must be executed to ensure that the topology is accurate and there are no disconnected links. When the topology construction is complete, one can specify attribute values on devices and run simulations. Some features that are useful for this purpose are listed below.

- ❖ Find node: Allows the user to locate any node by name across the network. LAN objects or any objects that represent many devices can be configured with the names

of individual members that they represent (aliases). The find node utility also locates a device by its alias.

- ❖ Logical object selection: Objects can be selected based on their type or any of their attribute settings. Selection sets can be retained for further selection.
- ❖ Applying changes to selected objects: Changes to attribute values for objects selected manually or using the find logical object selection command can be applied with a single click
- ❖ Configuring attributes: Attributes can be configured to represent characteristics of the device. Protocol parameters can also be tuned.
- ❖ Selecting advanced models that contain many attributes: Models are divided into a three-tier structure: advanced, intermediate and final. Advanced models contain numerous attributes that can be tuned based on the different protocols they contain. Intermediate models contain a subset of the advanced model attributes and Final models contain only very basic attributes that have to be configured by the user.

Once the process of building a topology is complete, one can proceed to the documents on modelling network traffic and characterizing applications.

5.2 Examples of Various Topologies Used for Different Studies

Figure 5.5 is an example network of a complete topology. The example topology selected consists of users on shared LAN segments. LAN segments on the same floor of an office are connected to each other via a switch. All switches interface to a building router. A backbone spans various campuses and connects the building routers together. Servers are located on a server farm in one of the buildings.

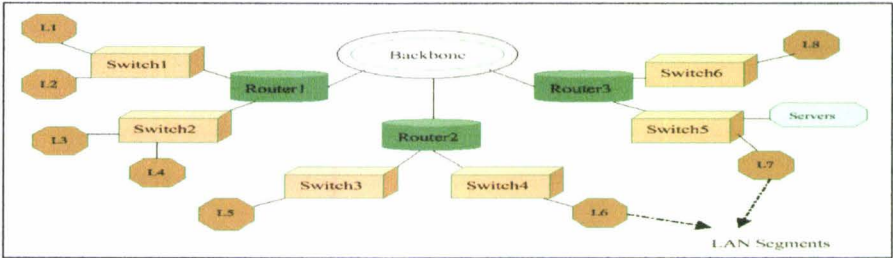


Figure 5.5 Complete Network Topology

Topology for Client/Server Performance Analysis: If the network is being used to study the performance of an existing application in the presence of regular day-to-day network traffic, a single path from the client to the server may be used as shown in figure 5.6

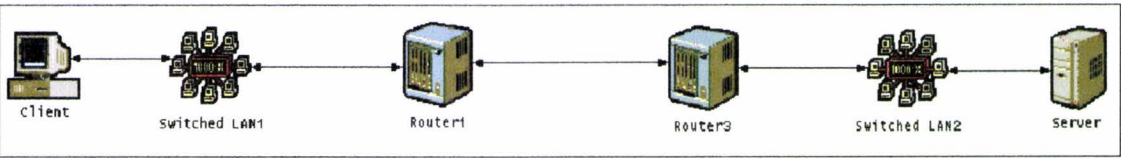


Figure 5.6 Single Path Topology

In a single path topology, one can isolate the model and the infrastructure between two devices of interest (e.g. client and server). The effect of the remaining portions of the network will be taken into account by representing the traffic that crosses, and therefore affects, the path of interest. The advantage of this topology is that it enhances the focus of the study to the main objects of interest. Since other devices are not explicitly modeled, the simulation is very efficient. There is no loss of accuracy since the effect of the remaining network on the client-server traffic is taken into account.

With this topology, one can study the effect of QoS on application response time, TCP effects (retransmissions, windowing) due to network delays and application protocol behavior.

OPNET’s application architecture allows modeling a variety of applications explicitly. Example application models include FTP, HTTP, and Email etc. One can configure these applications to generate traffic or import application traces using OPNET’s ACE module. OPNET also provides a “custom” application model, which can be used if none of the standard application models suit the purpose.

Run Initial Simulations: Once a topology has been selected and traffic is configured, all the statistics of interest can be selected for QoS studies including link utilizations, router interface queuing delays, packet drops, application response times and application end-to-end delays. Simulation can be run for a selected time.

Chapter 6

FR Network Design for Voice Implementation

Frame Relay network is designed and built with several devices: routers, workstations and links. The characteristics of traffic are defined by configuring application and profiles. In this project our approach is to set up special PVCs for voice, compression technique is applied for two different types of voice applications to see the various parameters that affect the FR network for voice application. To see the actual end-to-end process a practical Frame Relay network is also set up. After simulation, performance of selected parameters, device status and utilisation are obtained as function of time and in graphics. End of simulation reports are created that are viewed and printed out as documentation. The results are analyzed to complete the study.

6.1 Network Design and Implementation

In order to simulate the Frame Relay networks for voice application, a number of steps needed to be taken. The steps taken to complete the network can be summarised as follows.

- **Choose Topology:** Identify what type of network topology was going to be used in the simulations and outlining the types of links that are going to be connected to the network.
- **Build Networks:** Each network needs to be built including interconnections from one network to another. The networks consist of a number of devices including, workstations, routers, and links.
- **Configure Applications and Profiles:** The traffic characteristics for each network need to be defined and configured. Each node needs to be assigned a profile in order to define the application that would be supported on that node.

- **Choose Statistical Results:** As there is a large selection of statistical results to choose from, it must be decided upon as to which statistics will produce more appropriate results so that a study of chosen performance can be made for the network.
- **Simulate and Examine Results:** The networks simulation properties such as duration need to be defined. Duplicating the network and changing the particular attribute as needed makes several different scenarios for different study. Once simulation is complete, detailed analysis of the networks can take place by making use of the statistics chosen in the above step.

6.2 Devices and Tools used to build the Network

6.2.1 Application Configuration



Figure 6.1 Application Configuration node

The "Application Configuration" node can be used for the following specifications:

"ACE Tier Information": Specifies the different tier names used in the network model. This attribute will be automatically populated when the model is created using the "Network->Import Topology->Create from ACE..." option.

"Application Specification": Specifies applications using available application types. A name can be specified and the corresponding description in the process of creating new applications. The specified application name will be used while creating user profiles on the "Profile Config" object.

"Voice Encoder Schemes": Specifies encoder Parameters for each of the encoder schemes used for generating Voice traffic in the network.

6.2.2 Profile Configuration



Figure 6.2 Profile Configuration

The "Profile Configuration" node can be used to create user profiles. These user profiles can then be specified on different nodes in the network to generate application layer traffic. The applications defined in the "Application Config" objects are used by this object to configure profiles. Therefore, applications must be created before using this object. The traffic patterns can be specified followed by the applications as well as the configured profiles on this object.

6.2.3 PVC Configuration



Figure 6.3 PVC Configuration

The "FR PVC Config" node model represents a Frame Relay node used to specify PVC configurations. They can be defined at the interface read from an OPNET gdf format file.

Protocol is Frame Relay. “PVC configuration”: specifies the PVC connections for any portion of the entire Frame Relay network.

6.2.4 Quality Of Service (QoS) Configuration

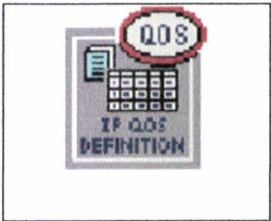


Figure 6.4 IP QoS Configuration

QoS attribute configuration defines attribute configuration details for protocols supported at the IP layer. The individual nodes using symbolic names can reference these specifications. “Queuing Profile” defines different queuing profiles such as FIFO WFQ, Priority Queuing, Custom Queuing, MWRR, MDRR and DWRR. “Car Profiles”: defines different CAR profiles that can be used in the network.

6.2.5 IP Router



Figure 6.5 IP Router

The “fr4_ethernet2_gtwy” node model represents an IP based gateway supporting up to 2 Ethernet interface and 4 Frame Relay interfaces. IP packet arriving on any interface are

routed to the appropriate output interface based on their IP address. The RIP, OSPF may be used to dynamically and automatically create the gateways routing tables and select routes in adaptive manner. Supported protocols are FR, IP, UDP, RIP, Ethernet, Gigabit Ethernet, and OSPF.

6.2.6 Frame Relay Workstation

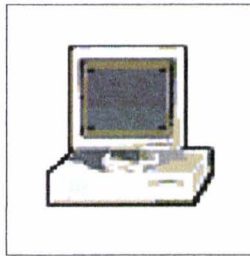


Figure 6.6 Frame Relay Workstation

The “fr4_wkstn” node model represents a Frame Relay node, which has a client server application running over TCP/IP and UDP/IP. The RIP or the OSPF protocol may be used to dynamically and automatically create the gateways routing tables and select routes in an adaptive manner. The supported protocols are: FR, RIP, UDP, IP, and TCP.

Inter-connection: 1 FR input 1 FR output at selectable data rate.

“Transport address”: This attribute allows for the specification of the node address. It has 4 FR connections at selectable data rate as port interface

6.2.7 Cisco Router



Figure 6.7 Router 3640

The Router “cs_3640_4s_e5_fe1_tr1_sl6_adv” node model represents a router with specific configuration of an IP based router model. Packets arriving on an interface are routed to the appropriate output interface based on their destination address.

The supported protocols are: RIP, UDP, IP, TCP, BGP, IGRP, EIGRP, IEEE 802.3 (Ethernet, Fast Ethernet, Gigabit Ethernet), 802.5(Token ring) and OSPF. It allows a Ethernet connection at 100 Mbps, 5 Ethernet hub connections at 10 Mbps, one Token ring at selectable data rate, 6 IP serial connections consisting of 2 WAN connections.

6.2.8 Ethernet Server



Figure 6.8 Ethernet Server

The “Ethernet_server” model represents a server node with server applications running over TCP/IP and UDP/IP. This node supports one underlying Ethernet connection at 10 Mbps, 100 Mbps, or 1 Gbps. The operational speed is determined by the connected link's data rate. The Ethernet MAC in this node can be made to operate either in full-duplex or half-duplex mode. A fixed amount of time is required to route each packet, as determined by the "IP Forwarding Rate" attribute of the node. Packets are routed on a FCFS basis and may encounter queuing at the lower protocol layers, depending on the transmission rates of the corresponding output interface.

The supported protocols are: RIP, UDP, IP, TCP, Ethernet, Fast Ethernet, Gigabit Ethernet, and OSPF.

“Supported Services”: This attribute allows for the specification for various services supported on the node.

“Server address”: this attribute allows for the specification of the address of the node.

6.2.9 Ethernet Workstation



Figure 6.9 Ethernet Workstation

The “ethernet_wkstn” node model represents a workstation with client-server applications running over TCP/IP and UDP/IP. The workstation supports one underlying Ethernet connection at 10 Mbps, 100 Mbps, or 1000 Mbps. This workstation requires a fixed amount of time to route each packet, as determined by the "IP Forwarding Rate" attribute of the node. Packets are routed on a first-come-first-serve basis and may encounter queuing at the lower protocol layers, depending on the transmission rates of the corresponding output interfaces.

The supported protocols are: RIP, UDP, IP, TCP, IEEE 802.3 (Ethernet, Fast Ethernet, Gigabit Ethernet), and OSPF. It allows a single Ethernet connection at 10/100 or 1000 Mbps.

6.2.10 Cisco Router



Figure 6.10 Router 1601

The cs_1601_1s_e1_fr1_sl1 node model represents a router, which implements VP and VC routing capabilities in a network. This device is created using OPNET device creator utility. It is capable of routing through 3 ports each supports the Ethernet, Frame Relay and slip connections available at different link speeds.

6.2.11 Frame Relay Cloud

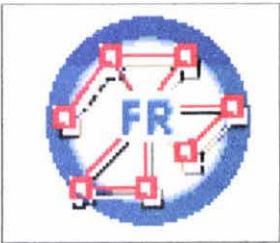


Figure 6.11 Frame Relay Cloud

The “fr 32_cloud” node model represents a cloud through which Frame Relay traffic can be modelled using 32 input/output physical links. Support FR protocol .It has 64 FR serial line connections at selectable data rates. “Packet Discard Ratio” attribute specifies the percentile of dropped packet. “Packet Latency” specifies the distribution parameters and the distribution type for modelling packet latency through this network.

6.2.12 Frame Relay (FR) T1 Link

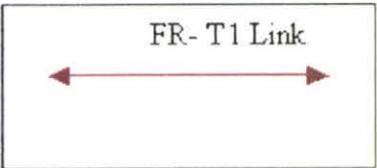


Figure 6.12 Frame Relay T1

FR-T1-link connects Frame Relay station nodes and switches at selectable data rates.
Data rates: T1 (1.544 Mbps)

Model attributes:

“DLC identifier”: specifies the most recently assigned Data link connection identifier (DLCI) value on the link.

“Propagation speed”: specifies the propagation speed (in m/ sec.) for the medium. If the delay attribute of the link is set to “distance Based”, the speed is used to calculate the propagation delay based on the distance between 2 nodes.

6.2.13 PPP-DS1 Link

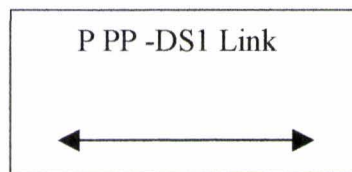


Figure 6.13 PPP -DS1 Link

The PPP-DS1 link connects 2 nodes with serial interfaces (e.g. routers with ppp ports) at a selectable data rate.

Data rates: (e.g. DSO, S, DSB, T1, T3, OC3, OC12, OC36, OC48).

6.2.14 100BaseT Duplex Link

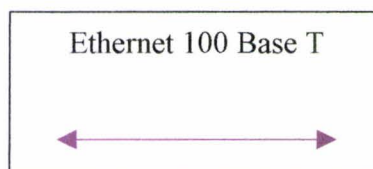


Figure 6.14 100 BaseT duplex links

The 100BaseT duplex link represents an Ethernet connection operating at 100 Mbps. It uses the Ethernet_v2 packet format. This link cannot be used to connect two Ethernet hubs.

6.3 PVC Configuration

The PVC configuration is an important aspect for any FR network. Now this is discussed in details in the following section.

The PVCs in a Frame Relay network can be specified in one of two ways, either by using the Configure PVC between Selected Nodes menu operation, from the Frame Relay PVC Config object, or the Frame Relay Network PVC Configuration compound attribute of any node that contains a FRAD module. The PVC specification can be distributed across multiple nodes, if desired. The PVCs specified in a particular node are not limited to the PVCs that originate or terminate at that node. For instance, all the PVCs of an entire Frame Relay network may be specified at one node.

The FR PVC Config node's PVC Characteristics Definition compound attribute allows defining a traffic pattern for a PVC. Here one can set traffic characteristics based on the TOS (Type of Service) field in the IP datagram, IP address, transport protocol, and application port numbers. The Delay (seconds) and Packet Discount Ratio attributes can be set in conjunction with a FR "cloud" object that uses the PVC configuration, a setting of Cloud Based uses the cloud settings to specify the delay or packet loss as packets traverse the cloud. Once traffic pattern have been defined one can assign it to a PVC by entering the traffic pattern name in the FR Network PVC Configuration table.

Multiple PVCs can be assign for a given source-destination pair. This allows characterizing different PVCs to support specific traffic patterns, and thereby provide varying QoS for different higher-layer traffic flows.

The following section describes the attributes of the Network PVC Configuration compound attribute, which is used to specify PVCs.

6.3.1 Network PVC Configuration Attributes and their description

- ❖ **Source FRAD** The name of the source node of the PVC. If this attribute is set to "Auto Assigned", then this attribute will be set to the local node. To prevent

naming conflicts, all FRADs in a Frame Relay network must have unique names.

- ❖ **Source Application** The name of the application process using the services of the source FRAD. The source application generates traffic destined for a destination application at a different node.
- ❖ **Destination FRAD** The name of the destination node of the PVC. All FRADs in a Frame Relay network must have unique names.
- ❖ **Destination Application** The name of the application process using the services of the destination FRAD. The name of the destination application and the source application can be different.
- ❖ **Committed Information Rate** Committed Information Rate (CIR) is the rate at which the Frame Relay network agrees to transfer information under normal conditions. CIR is expressed in bits/sec.
- ❖ **Committed Burst Size** The Committed Burst Size (**B c**) is the maximum amount of data that a network agrees to transfer under normal conditions over an agreed upon measurement interval. **B c** is expressed in bits, the data may appear in one frame or several frames.
- ❖ **Excess Burst Size** The Excess Burst Size (**B e**) is the maximum amount of uncommitted data that the Frame Relay network will attempt to deliver over the measurement interval. **B e** is expressed in bits.
- ❖ **Reverse CIR** Committed Information Rate for traffic flow from the destination end of the PVC connection. By default, it is set to be the “same” as the forward CIR.
- ❖ **Reverse Bc** Committed Burst Size that the network agrees to deliver from the destination end of this PVC under normal conditions. By default, it is the “same” as the forward Bc.
- ❖ **Reverse Be** Excess (uncommitted) Burst Size that the network attempts to deliver from the destination end of this PVC under normal conditions. By default, its value is the “same” as the forward Be.

6.4 Configuring Quality of Service (QoS)

OPNET provides a variety of QoS schemes. Figure 6.14 shows a QoS configuration table defined for a Router.

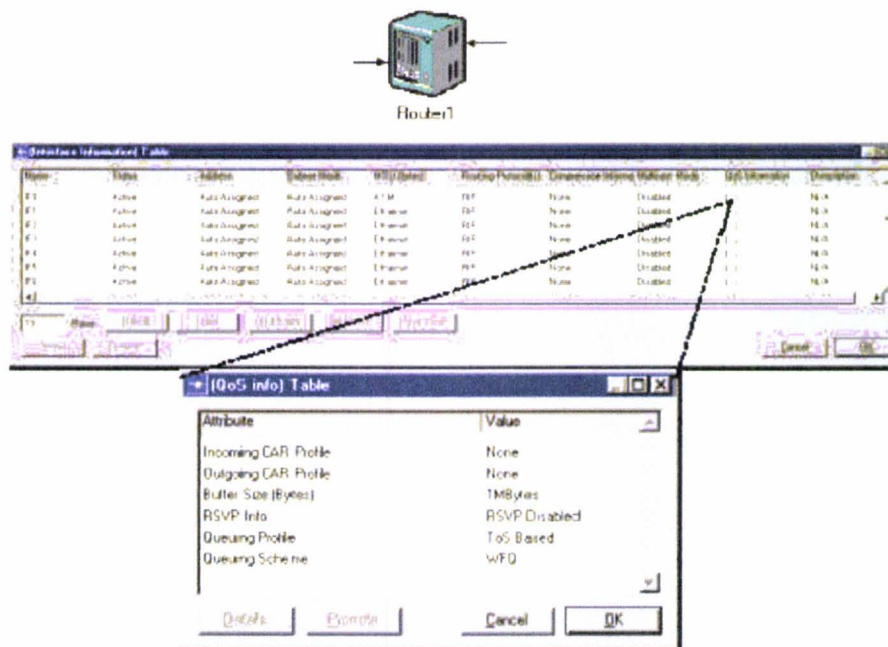


Figure 6.15 QoS configuration table (OPNET)

IP-based networking devices provide support for implementing “Quality of Service,” or QoS, policies. These are useful when traffic with different service requirements like voice and data is mixed in the same network. If no QoS policies are defined for a router, it processes packets in a first-in-first-out (FIFO) manner.

For convenience, profiles can be created of recurring QoS configurations using a “QoS Attribute Configuration” utility object, which is in the “Utilities” object palette. Then same profile can be assign to multiple interfaces using the QoS Information attribute (available in the IP Address Information compound attribute of advanced router models). Because all configurations reside in a single location, this can reduce the work required to configure QoS policies in the network. Profiles can be configured for following types of QoS policies:

- Committed Access Rate (CAR)
- RSVP
- Custom Queuing
- Priority Queuing
- Weighted Fair Queuing (WFQ)

Following the steps below the QoS policies are configured and assigned.

1. From the “Utilities” object palette “QoS Attribute Configuration” object can be inserted into the network. Any number of configuration objects can be included and placed at any network level, however, placing configuration objects at the top level allow to access them more easily when to define new policies or edit existing ones.

2. One of the QoS compound attributes can be opened from: CAR Profiles, Custom Queuing Profiles, Priority Queuing Profiles, or WFQ Profiles. A new row can be created and a name is assigned for that profile.

3. The attributes are set in the profile. Profile configurations vary, depending on the type of QoS profile need to be. Queuing-based profiles (WFQ, Priority Queuing, Custom Queuing) contain a separate set of attributes for each queue. When a router receives a packet, it selects a destination queue by searching each queue’s Classification Scheme attribute for settings that match those of the incoming packet.

4. QoS profiles are assigned to specific router interfaces. QoS Information attribute is opened in a node’s IP Address Information compound attribute. Here CAR profiles for incoming and outgoing traffic, a maximum buffer size, a queuing profile, and a queuing scheme can be assigned. A router model uses the profiles indicated by the latter two attributes to determine the next queue to process (Queuing Scheme) and the correct queue for an incoming packet (Queuing Profile). RSVP can be enabled or disabled for each interface, and maximum bandwidth threshold for each flow and for the entire interface can be set.

6.5 Configuring ToS in Application Models

Each application model (ftp, voice, etc.) contains a Type of Service attribute. With this attribute set, the application generates packets that request a certain ToS from the network. Application models can be configured from the Supported Services.

6.6 Network Scenarios

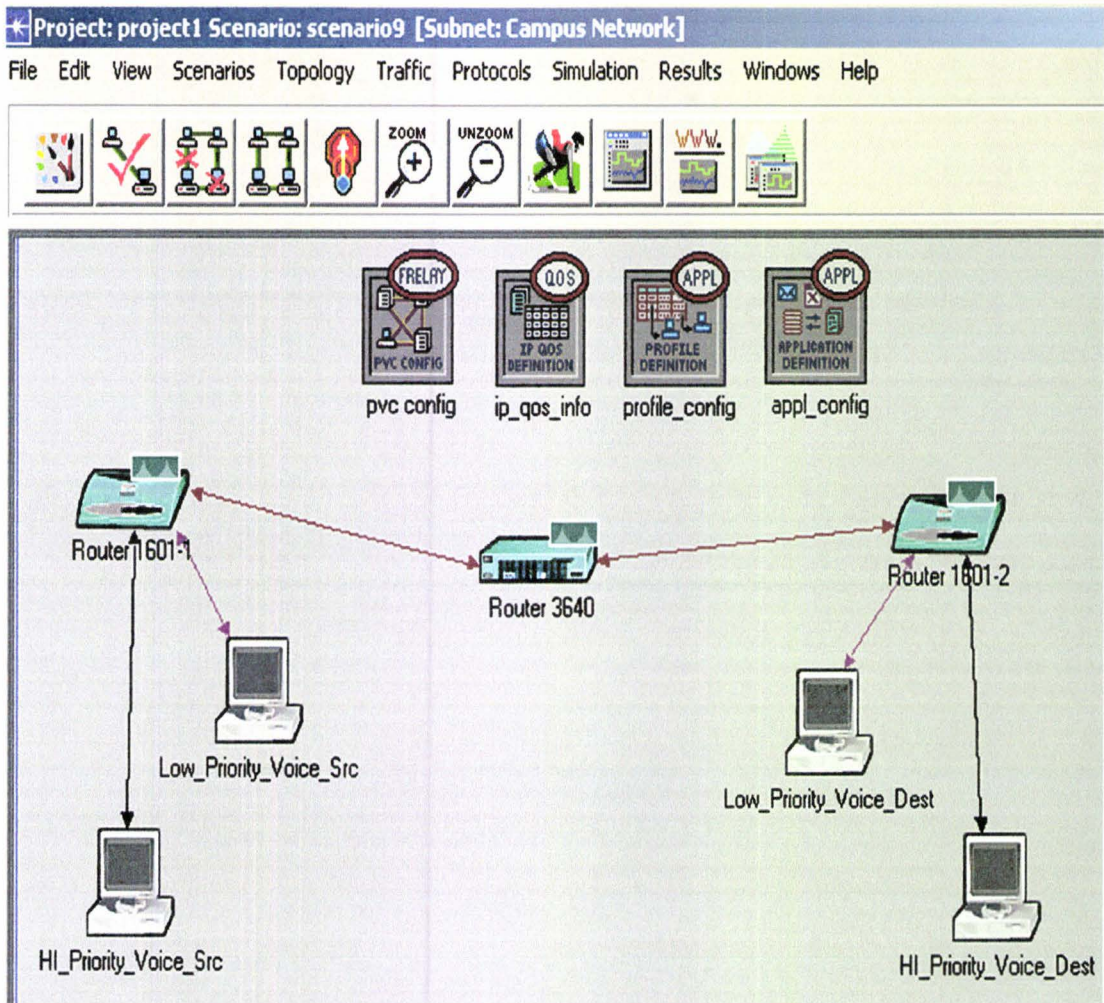


Figure 6.16 Final Scenario of the network 1

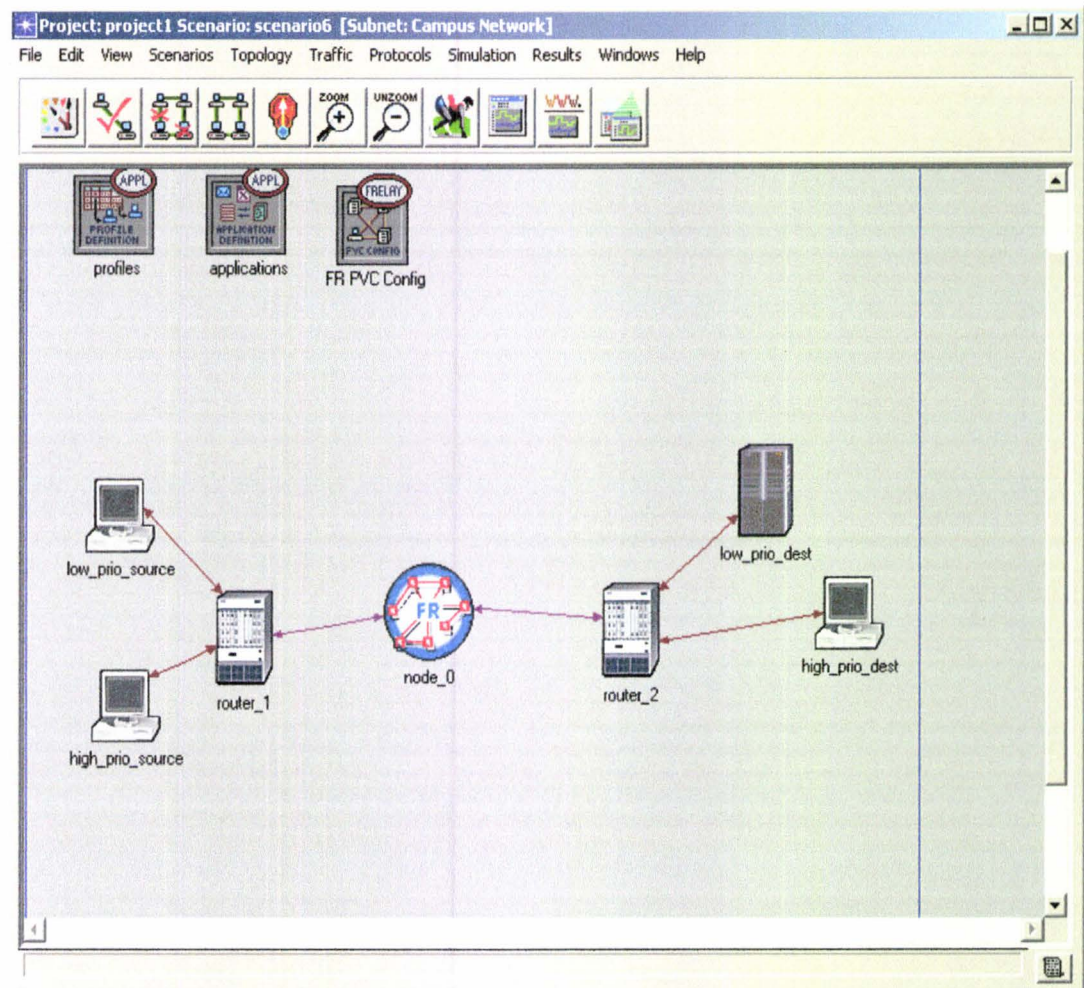


Figure 6.17 Network 2 scenario using FR cloud.

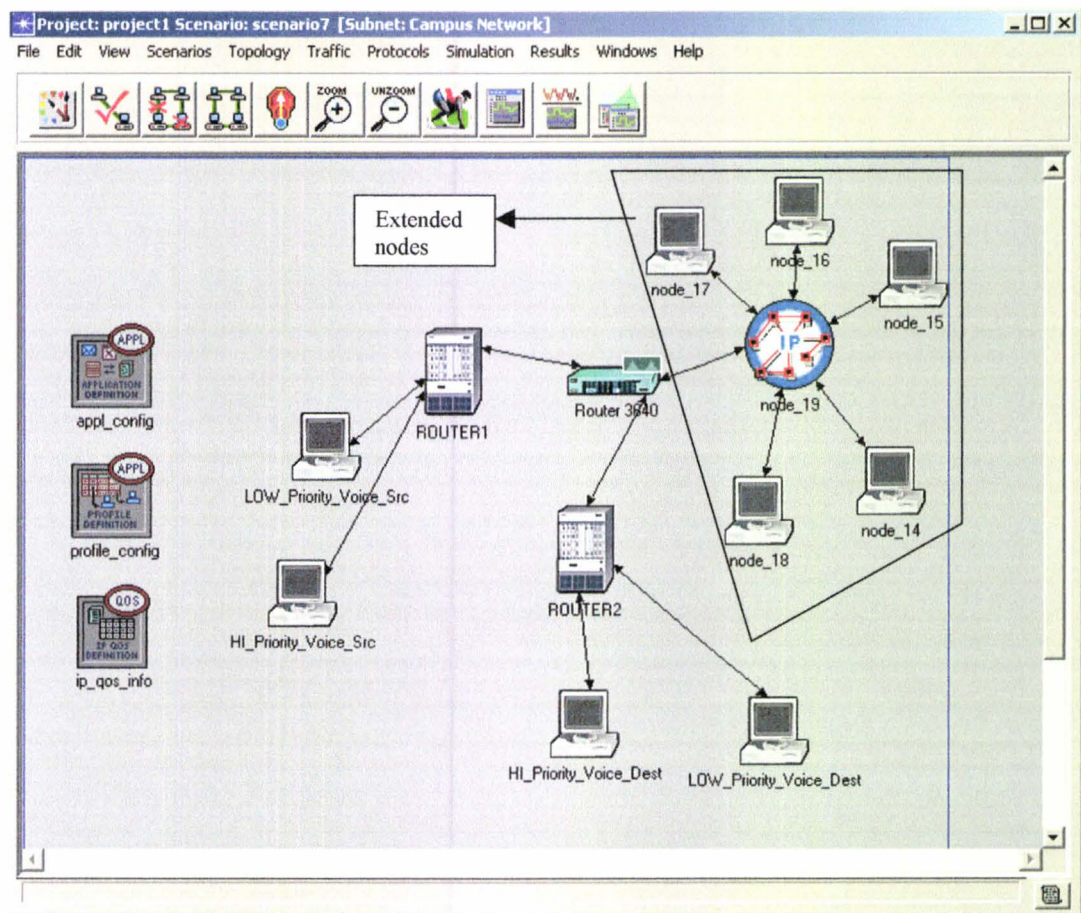


Figure 6.18 Network 1 scenario with extended nodes.

6.6.1 Choose Topology

As we were going to simulate a Frame Relay network for the performance study, instead of choosing large networks, it was decided to simulate a simple network with voice source and destination workstation. This was a Campus Network and it was then duplicated, as we needed several identical Networks for comparison of various study.

6.6.2 Build Networks

Cisco routers and workstations are chosen and selected from the device selection menu. After they have been put into place, they all needed to be linked together with their respective link types. Frame Relay T1 link is used for the connection between the workstation and the router. Fast Ethernet links are used between routers. Also DS1 link is used for the connection between Router and workstation. The scenario is duplicated and some of the attributes are changed for other observation. Another network is also built using Frame Relay cloud object and different router is used for that case for observing the difference. The routers are configured to be almost same with the previous one. The final scenarios of the different network will show on following sections.

6.6.3 Configure Applications and Profiles

Four different tools are set after selecting them from the device selection menu. These are Application config, Profile config, PVC config and IP QoS config. By making use of the Application Configuration attributes, a set of applications that are defined, would be running on the networks. In application configuration Interactive voice and excellent effort voice are chosen. By making use of the Profile Definition attributes a number of Profile users those would like to use in the networks is chosen. Each different profile supports a number of different applications. In this network, voice profile is supported. Among the voice encoder scheme, three different encoder schemes are chosen for different study as G.711, G.729 and G.723.1. TOS based service is selected.

Two PVCs are defined between the routers with 2 different types of service (TOS) set for interactive and excellent effort voice.

In PVC config tools all the attributes are configured and Committed Information Rate (CIR) set to Zero as all frames will be marked discard eligible (DE).

Other CIR values are also set for the different scenario.

FIFO and WFQ are selected for the QoS configuration tool for different scenario.

6.6.4 Choose Statistical Results

The results that are chosen mostly time related response results, as the aim of the study is to see the important performance characteristics. Statistics for voice applications are selected. The Following statistical results were chosen:

- Voice Packet end-to-end Delay (Global)
- Voice Packet Delay Variation
- Voice Traffic Received
- Voice Traffic Sent
- Throughput (Node)
- Point to point Utilization (Link)
- Queuing Delay
- PVC status

6.6.5 Simulate and Study Results

Finally the networks were simulated. The configuration time was set to 5 minutes. No additional configuration was necessary. After simulation, if errorless, the results were brought up and a report is also obtained giving both graphical and tabular form.

Comparison between the networks was also made. These results may be seen in the Results and Discussion Section.

6.6.5.1 Selecting and viewing the results

The approach of examining application statistics is provided as follows.

1. From Simulation menu, “Choose Individual Statistics” operation is used by right clicking on empty-space in the Project Editor to collect statistics on all of the applications of interest. Choose result dialog box gives Global, Node or Link options.
2. In “Global Statistics”, to collect information about how an application is performing in general across all nodes all options of FR and voice are selected. “Node Statistics” is also taken to collect information about how each individual node is performing. In FR and voice case, all of the statistics available for the application of interest are chosen.
3. Under the “Node Statistics” category, required performance is chosen and all of the available statistics are selected.
4. From the “Choose Results” dialog box, “Link Statistics” is selected to collect all information under the “point-to-point” category.
5. From simulation menu, configure simulation is chosen. The duration is set to 5 minutes and simulation is run.
6. From result, view result is chosen, statistics report is selected under result and then generate web report gives the report.

6.7 Configured Attributes

After selecting the parameter of choice we get the following configuration tables, which is shown taken by print screen process.

6.7.1 Attributes of network 1

| Name | Description |
|------------------------|-------------|
| Interactive Voice | (...) |
| Excellent Effort Voice | (...) |

2 Rows Delete Insert Duplicate Move Up Move Down Details Promote Cancel OK

Figure 6.19 Application configuration attribute table for network 1

| Name | Frame Size (secs) | Lookahead Size (sec) | DSP Processing Rate | Coding Rate (bits/sec) | Speech Activity Detection |
|--------|-------------------|----------------------|---------------------|------------------------|---------------------------|
| G.711 | 4 msec | 0.0 | 1.0 | 8 Kbps | Enabled |
| G.711a | 4 msec | 0.0 | 1.0 | 8 Kbps | Enabled |

2 Rows Delete Insert Duplicate Move Up Move Down Details Promote Cancel OK

Figure 6.20 G.711 Voice encoder scheme attribute for network 1

(Voice Encoder Schemes) Table

| Name | Frame Size (secs) | Lookahead Size (sec) | DSP Processing Rat | Coding Rate (bits/se | Speech Activity De |
|-------------------|-------------------|----------------------|--------------------|----------------------|--------------------|
| G.723.1 (silence) | 30 msec | 0.0 | 1.0 | 5.3 Kbps | Enabled |

1 Rows

Figure 6.21 G.723 Voice encoder scheme attribute for network 1

(appl_config) Attributes

| Attribute | Value |
|-------------------------|--------------------|
| name | appl_config |
| model | Application Config |
| ACE Tier Information | None |
| Application Definitions | (...) |
| Voice Encoder Schemes | G.729 |

☐ Apply Changes to Selected Objects ☐ Advanced

(Voice Encoder Schemes) Table

| Name | Frame Size (secs) | Lookahead Size (sec) | DSP Processing Rat | Coding Rate (bits/se | Speech Activity De |
|-------|-------------------|----------------------|--------------------|----------------------|--------------------|
| G.729 | 10 msec | 5 msec | 1.0 | 8 Kbps | Disabled |

1 Rows

Figure 6.22 G.729 Voice encoder scheme attribute for network 1

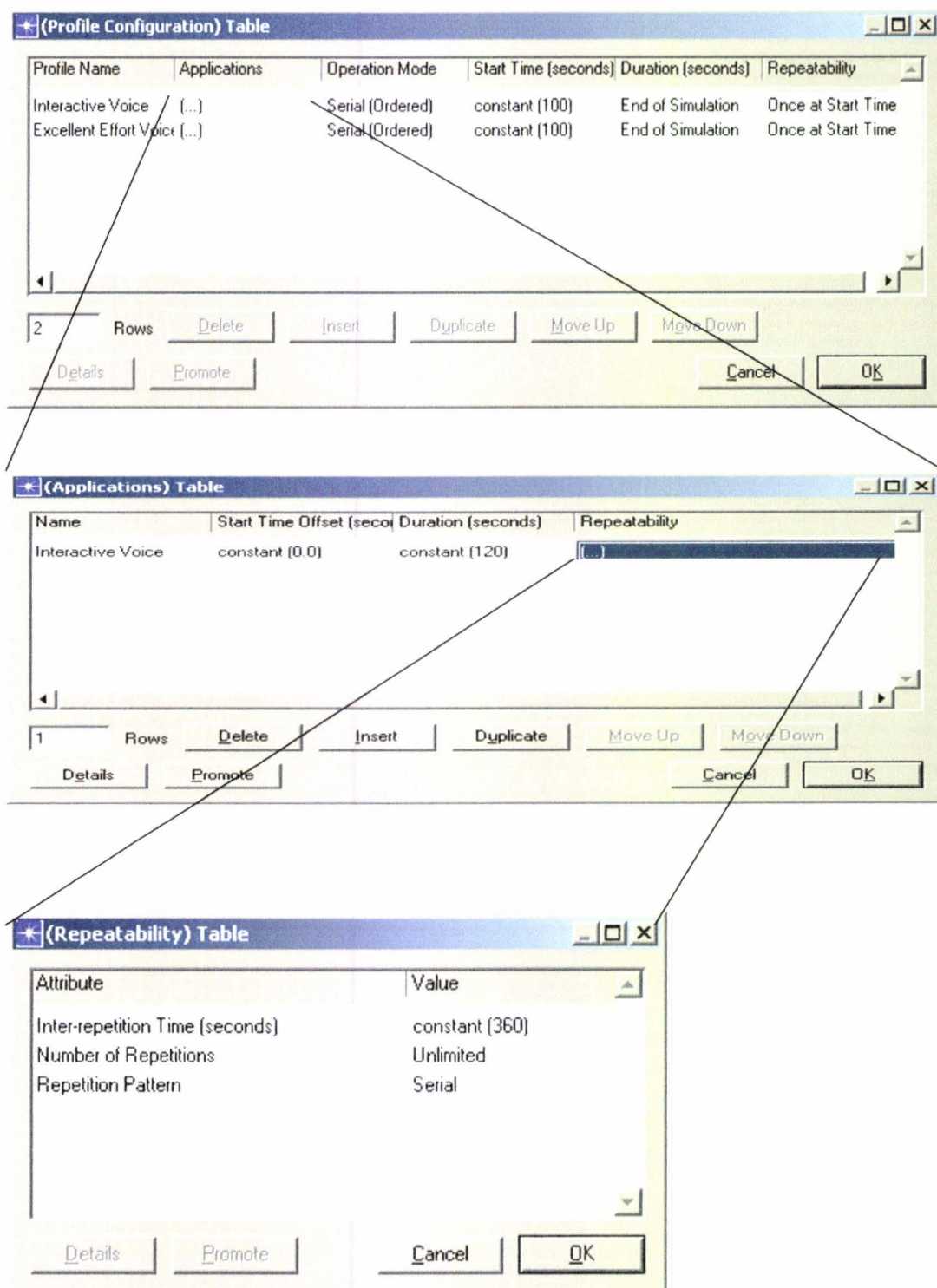


Figure 6.23 Profile configuration attribute for network 1

6.7.2 Additional Attributes of network 2

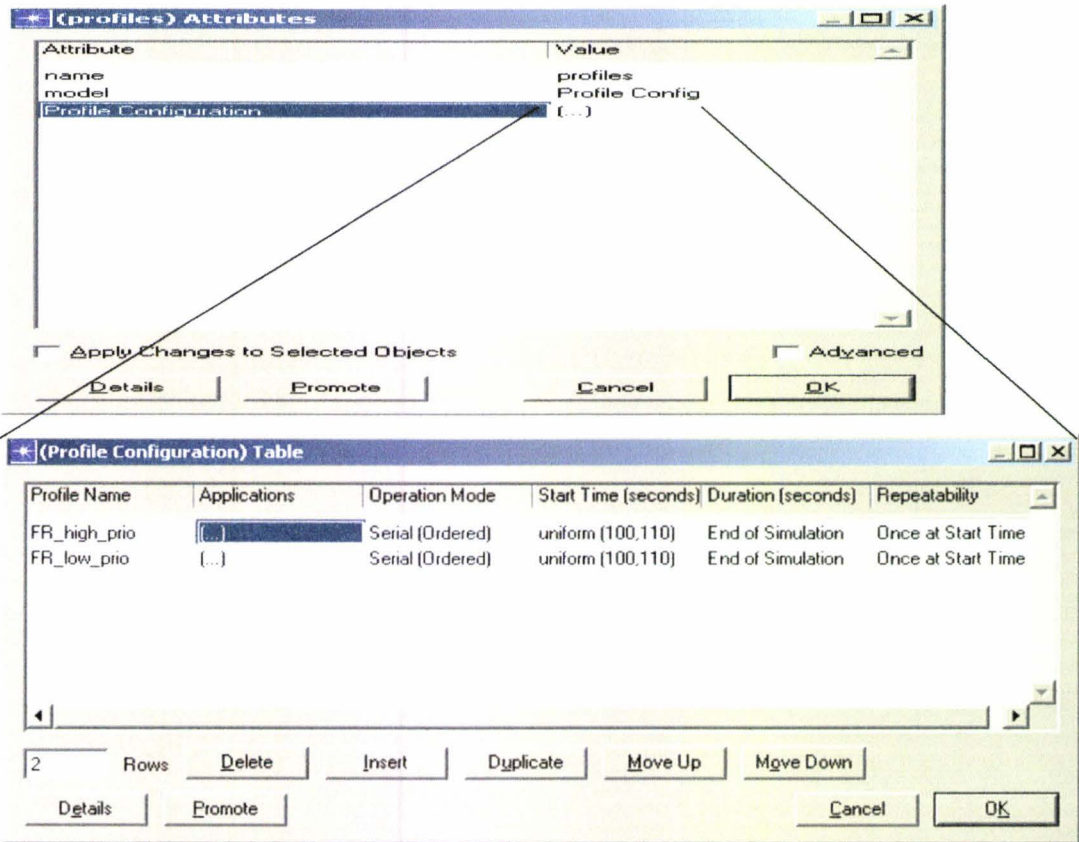


Figure 6.24 Profile configuration attribute for network 2

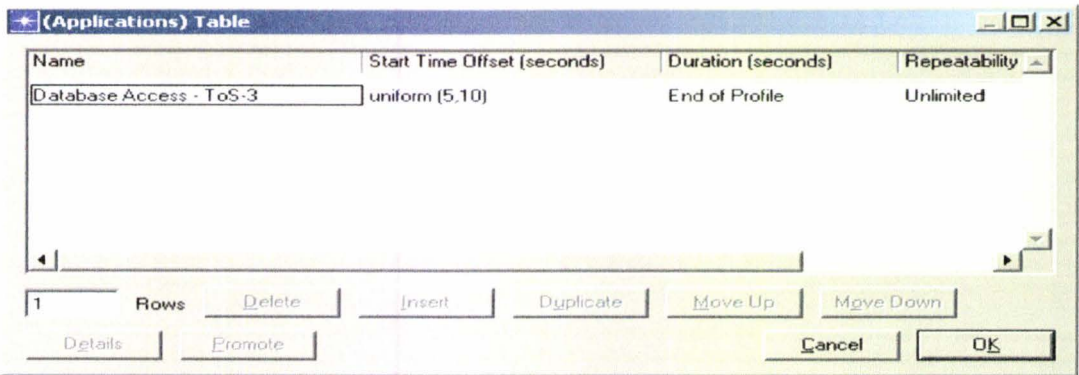


Figure 6.25 Application table (High Priority) of Profile configuration attribute for network 2

| Profile Name | Applications | Operation Mode | Start Time (seconds) | Duration (seconds) | Repeatability |
|--------------|--------------|------------------|----------------------|--------------------|--------------------|
| FR_high_prio | (...) | Serial (Ordered) | uniform (100,110) | End of Simulation | Once at Start Time |
| FR_low_prio | (...) | Serial (Ordered) | uniform (100,110) | End of Simulation | Once at Start Time |

2 Rows Delete Insert Duplicate Move Up Move Down

Details Promote Cancel OK

Figure 6.26 Profile configuration attribute for network 2

| Name | Start Time Offset (seconds) | Duration (seconds) | Repeatability |
|-------------------------|-----------------------------|--------------------|---------------|
| Database Access - ToS-0 | uniform (5,10) | End of Profile | Unlimited |

1 Rows Delete Insert Duplicate Move Up Move Down

Details Promote Cancel OK

Figure 6.27 Application table (Low Priority) of Profile configuration attribute for network 2

| Attribute | Value |
|-------------------------|--------|
| Outgoing CIR (bits/sec) | 64,000 |
| Outgoing Bc (bits) | 64,000 |
| Outgoing Be (bits) | 32,000 |
| Incoming CIR (bits/sec) | Same |
| Incoming Bc (bits) | Same |
| Incoming Be (bits) | Same |

Details Promote Cancel OK

Figure 6.28 Contract Parameter table for network 2

| Attribute | Value |
|-----------|------------------|
| Outgoing | low_prio_pvc |
| Incoming | Same as Outgoing |

Details Promote Cancel OK

Figure 6.29 Characteristics table of PVC configuration attribute for network 2

| Name | Characteristics |
|---------------|-----------------|
| low_prio_pvc | (...) |
| high_prio_pvc | (...) |

2 Rows Delete Insert Duplicate Move Up Move Down

Details Promote Cancel OK

Figure 6.30 Characteristics definition table of PVC configuration attribute for network 2

| Supported TOS |
|-----------------|
| Best Effort (0) |

1 Rows Delete Insert Duplicate Move Up Move Down

Details Promote Cancel OK

Figure 6.31 Type of service table of PVC configuration attribute for network 2

6.8 Hardware Experimental Set-up

A practical Frame Relay network is also setup to see the actual end-to-end process. Cisco 3640 Router is configured as a Frame Relay switch, which is set up with additional WAN interface card. Router 1601 is also configured and connected with back-to-back WAN cable with the router 3640. Frame Relay Network as shown in Figure 6.32 between site A and site B is operated as per design.

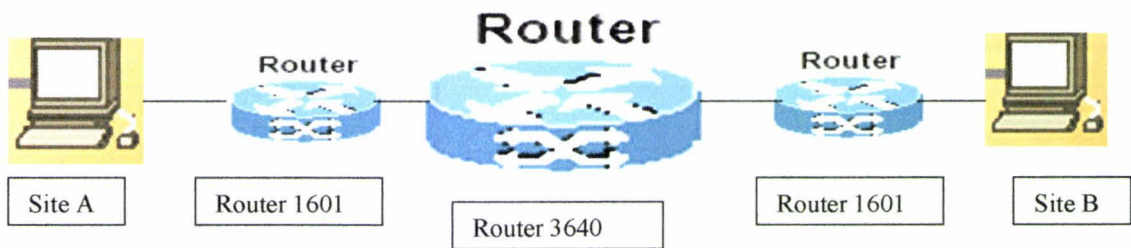


Figure 6.32: Frame Relay network

6.8.1 Description of Devices Used

Router 3640

The Cisco 3600 has multifunction platform with the versatility to support branch or enterprise hybrid dial access applications, LAN-to-LAN or routing applications, and multiservice applications in a single platform. It provides unprecedented modularity options with a broad range of available network modules, enormous flexibility with a variety of configurable options for customer-specific application scenarios, and, above all, high-performance to support any of these applications.

The Cisco 3640 router is equipped with four network module slots. Dial connectivity is supported with a series of network modules offering Integrated Services Digital Network (ISDN), Primary Rate Interface (PRI), ISDN Basic Rate Interface (BRI), integrated digital and analog modems, very high density asynchronous interfaces, two and four port Voice/Fax, 25-Mbps ATM, low-speed asynchronous/synchronous serial, and hardware

compression assisted network modules. LAN and WAN connectivity are provided by a series of mixed-media cards supporting Ethernet, Token Ring, and a variety of WAN technologies. Routing applications are supported with high-density Ethernet and a single-port, auto sensing, Fast Ethernet network module. Voice and Fax multi service applications are also supported with integrated voice network modules and voice interface cards

The Cisco 3600 with its redundant high-speed serial ports provides for backhaul of data to points of aggregation, while its redundant LAN interfaces, including the ability to provide Fast Ethernet and Token Ring, provide the maximum flexibility of deployment in a variety of LAN environments.

The dual-serial port WAN interface cards (WICs) for the Cisco 2600 and 1700 series feature Cisco's new, compact, high-density Smart Serial connector to support a wide variety of electrical interfaces when used with the appropriate transition cable. Two cables are required to support the two ports on the WIC. Each port on a WIC is a different physical interface and can support different protocols such as Point-to-Point protocol (PPP) or Frame Relay and Data Terminal Equipment/Data Communications Equipment (DTE/DCE). The WIC-2T provides two serial ports using the Smart Serial connector.

Asynchronous support with a maximum speed (per port) of 115.2 Kbps, minimum 600 bps. Synchronous support with a maximum speed of 2.048 Mbps per port

Router 1601

Cisco 1600 series routers connect small offices with Ethernet LANs to the Internet and to company intranets using several wide-area-network (WAN) technologies: Integrated Service Digital Network (ISDN), asynchronous serial, and synchronous serial such as Frame Relay, leased lines, Switched 56, Switched Multi megabit Data Service (SMDS), and X.25. The Cisco 1600 series consists of four models, each of which has an Ethernet port, a built-in WAN port, and a slot for an optional second WAN port. Cisco 1601 has one Ethernet, one serial and one WAN interface card slot

Extensive end-to-end security and data privacy features in the Cisco 1600 series ensure the safety of customers' information throughout and across their networks. The Cisco 1600

series routers, with key Cisco Inter network Operating System provide the end-to-end QoS and multimedia support not offered by any other small-office routers today.

Serial Interfaces Supported by the Cisco 1601 Router and Serial WAN Interface Cards:

- Asynchronous serial up to 115.2 kbps over analog basic telephone service
Point-to-Point Protocol (PPP), Serial Line Interface Protocol (SLIP) encapsulations
- Synchronous serial up to 2.048 Mbps Digital leased line, Frame Relay, SMDS, Switched 56, X.25, High-Level Data Link Control (HDLC), Link Access Procedure, Balanced (LAPB), and PPP

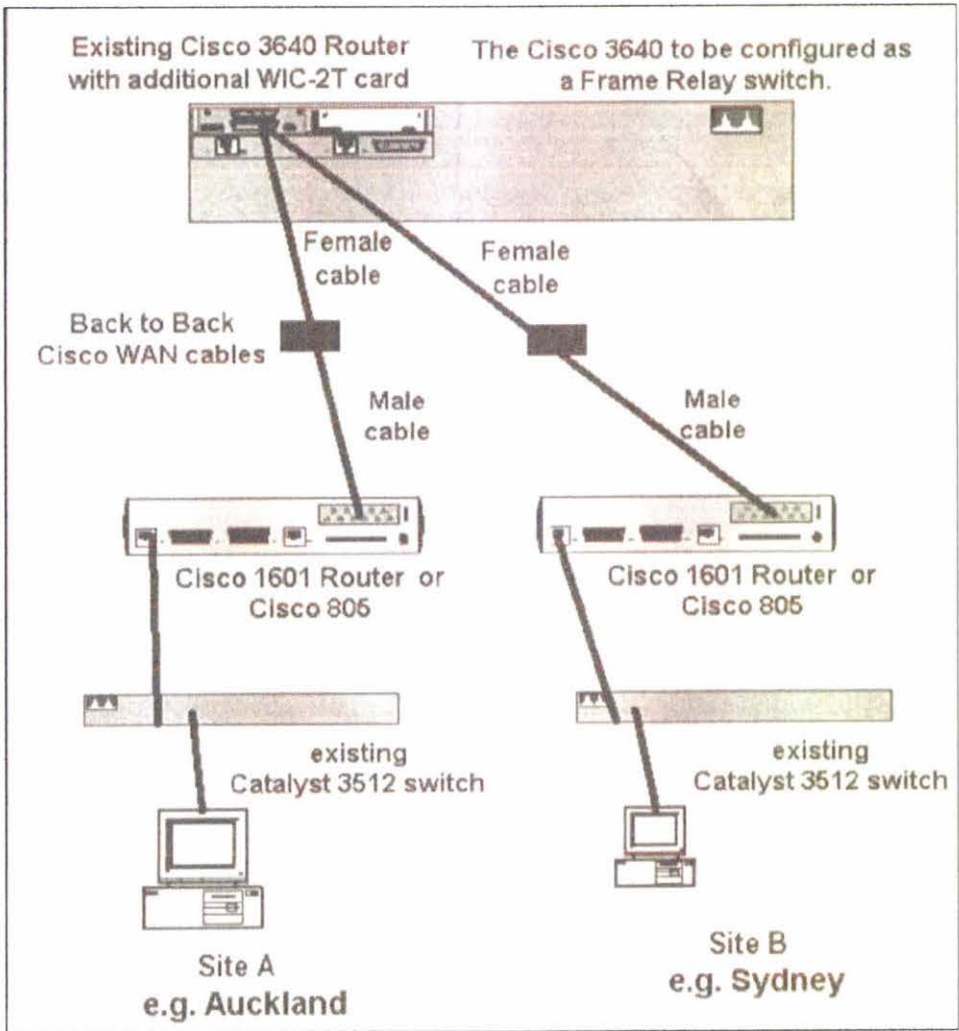


Figure 6.33 Detail connection (block Diagram) of Frame Relay network

6.8.2 Configuration of routers

To configure the router we follow the steps accordingly.

Step 1: A console is connected to the router

Step 2: Power on the router

Messages will appear in the terminal emulation program window.

Step 3 Press enter to accept the default entry.

--- System Configuration Dialog ---

Would you like to enter the initial configuration dialog? [yes/no]: y

At any point you may enter a question mark '?' for help.

Use ctrl-c to abort configuration dialog at any prompt.

Default settings are in square brackets '[]'.

Basic management setup configures only enough connectivity for management of the system, extended setup will ask you to configure each interface on the system

Would you like to enter basic management setup? [yes/no]: y

Configuring global parameters:

Enter host name [Router]: router1

The enable secret is a password used to protect access to privileged EXEC and configuration modes. This password, after entered, becomes encrypted in the configuration.

Enter enable secret: netlab

The enable password is used when you do not specify an enable secret password, with some older software versions, and some boot images.

Enter enable password: netlab

% Please choose a password that is different from the enable secret

Enter enable password: password

The virtual terminal password is used to protect access to the router over a network interface.

Enter virtual terminal password: netlab

Configure SNMP Network Management? [yes]: n

Current interface summary

Any interface listed with OK? value "NO" does not have a valid configuration

| Interface | IP-Address | OK? | Method | Status | Protocol |
|-----------------|------------|-----|--------|--------|----------|
| FastEthernet0/0 | unassigned | NO | unset | up | up |
| FastEthernet0/1 | unassigned | NO | unset | up | up |
| Serial1/0 | unassigned | NO | unset | up | down |
| Serial1/1 | unassigned | NO | unset | up | down |

Router1>

The enable secret is a password used to protect access to privileged EXEC and configuration modes. This password, after entered, becomes encrypted in the configuration.

Enter enable secret: netlab

The enable password is used when you do not specify an enable secret password, with some older software versions, and some boot images.

Enter enable password: password

The virtual terminal password is used to protect access to the router over a network interface.

Enter virtual terminal password: netlab

Configure SNMP Network Management? [yes]: no

Configure IP? [yes]: y

Configure IGRP routing? [yes]:

Your IGRP autonomous system number [1]: 15

Configure bridging? [no]:

Async lines accept incoming modems calls. If you will have users dialing in via modems, configure these lines.

Configure Async lines? [yes]: no

Configuring interface parameters:

Do you want to configure FastEthernet0/0 interface? [yes]:

Use the 100 Base-TX (RJ-45) connector? [yes]:

Operate in full-duplex mode? [no]:

Configure IP on this interface? [yes]:

IP address for this interface: 192.168.1.1

Subnet mask for this interface [255.255.255.0] :

Class C network is 192.168.1.0, 24 subnet bits; mask is /24

Do you want to configure FastEthernet0/1 interface? [yes]:

Use the 100 Base-TX (RJ-45) connector? [yes]:

Operate in full-duplex mode? [no]:

Configure IP on this interface? [yes]:

IP address for this interface: 192.168.3.1

Subnet mask for this interface [255.255.255.0] :

Class C network is 192.168.3.0, 24 subnet bits; mask is /24

The following configuration command script was created:

```
hostname router1
enable secret 5 $1$xmD8$82Nbrh5U9GaoFNO9nNPeF1
enable password password
line vty 0 4
password netlab
no snmp-server
!
ip routing
no bridge 1
!
interface FastEthernet0/0
media-type 100BaseX
half-duplex
ip address 192.168.1.1 255.255.255.0
!
interface FastEthernet0/1
media-type 100BaseX
```

```
half-duplex
ip address 192.168.3.1 255.255.255.0
dialer-list 1 protocol ip permit
dialer-list 1 protocol ipx permit
!
router igrp 15
redistribute connected
network 192.168.1.0
network 192.168.3.0
!
end
```

[0] Go to the IOS command prompt without saving this config.

[1] Return back to the setup without saving this config.

[2] Save this configuration to nvram and exit.

Enter your selection [2]

Router1>

This prompt indicates the basic router configuration completion.

All three routers are configured following the similar process for the Frame Relay connection set-up. The router3640 is further configured as a FR switch following the documentation supplied by Cisco.

The major tasks for the configuration of the router for a basic FR connection involve the following steps.

- Configuring Global Parameters
- Configuring the Ethernet interface
- Configuring the Serial interface for a FR connection.
- Configuring Routing Parameters.

Then the connections are made according to figure 6.33. The site computers are connected with the switch using RJ45 cables. Network component like NIC drivers are configured. The IP addresses for the workstations have been assigned along with the workstations name as site A and Site B. The connection is set up to check the connectivity using ping command.

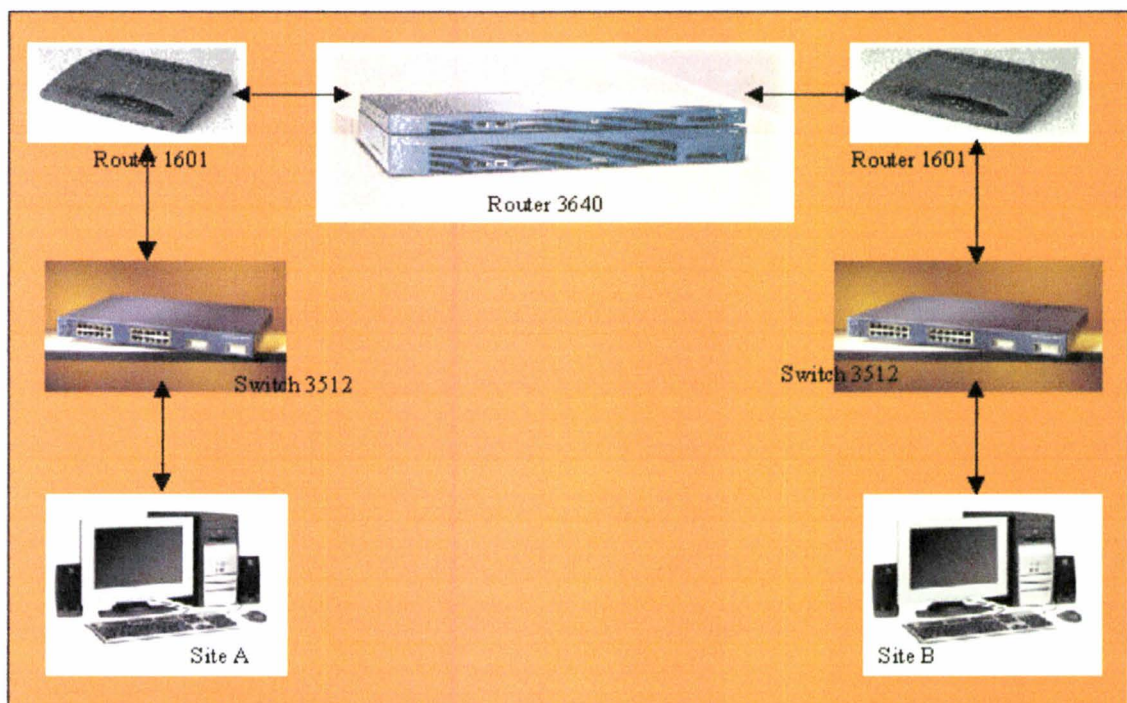


Figure6.34 Experimental set-up

Chapter 7

Results and Discussion

Results from figure 7.1 shows that FR delay and voice packet end to end delay are not more than 2.5ms and around 0.6 ms respectively. In figure 7.2 shows the average of voice packet delay variation which is almost constant nearly 4ns after certain time. Figure 7.3 shows the Frame Relay PVC throughput that is bursty in nature. Packet delay variation or jitter shown in figure 7.4 does not exceed 50ns, which is clearly within the range and shows acceptable performance under typical load condition for voice application. Figure 7.5 and 7.6 are the comparison between the high and low priority voice destination. High priority voice destination has more data received than Low priority node.

Different scenarios are built to see the performance comparison of network 1 for different buffer size. In scenario 3 of network 1 buffer size of attribute WFQ taken as 5000, which is greater than scenario 12 where the buffer size selected as 500. It should be mentioned here that other attributes are the same for simulation. In Figure 7.7, result obtained from the simulation of scenarios with different buffer size attribute using WFQ shows that delay decreases with increasing buffer size.

Different scenarios are taken for the comparison to show the performance of the network with different encoder scheme. Scenario 3 has the voice codec G.729 whereas scenario 11 and scenario 12 are chosen for G.711 and G.723.1 respectively. Figure 7.8 shows that voice packet end to end delays with G.729 and G.723.1 are less compared to the delay occurred for G.711. As shown in figure 7.8 the end to end delay for the codec G.729 and G.723.1 are less than 1 ms. The delay for G.711 is more than 3 ms.

Figure 7.9 describe the Comparison of Voice packet delay variation or jitter for 3 different encoder scheme. Again it shows that G.729 and G.723.1 encoding scheme experienced less than 100 ns and selecting G.711 encoder has the jitter of 1000 ns with respect to time.

From the Graphical representation jitter for the later one shows a large variation whereas earlier two shows almost constant shape.

The results are collected for network 2 scenarios with regard to voice traffic. Figure 7.10 shows the Frame Relay delay within 2.5 ms, which is acceptable. Frame Relay delay variance shown in figure 7.11 is almost constant after certain duration of time. Figure 7.12 and 7.13 shows a huge difference between the voice traffic sent and voice traffic received respectively. This clearly indicates that there is congestion present in the network. There are other possible reason could be CIR value selected was not proper for this particular situation.

From the Voice packet end-to-end delay and Jitter value collected as shown in figure 7.14 and 7.15 respectively gives a large variation delay time which is pointing us to the proper selection of the codec for this Frame Relay network. A delay of more than 190 second has shown which is directly varies with the time. The jitter value is even worst.

Figure 7.16 shows the throughput for high priority destination, which is acceptable as shown that high amount of packets, is received. But for Low priority destination due to congestion as shown in figure 7.17, the station receives negligible amount of packet.

Figure 7.18 and 7.19 describe the BECN and FECN status respectively, both as status “0”. Frame relay PVC status shown in figure 7.20 as the discard eligibility as “1” which is relevant to the low priority voice destination, which indicate a high congestion.

Figure 7.21 shows Point to point Throughput of the Link in both side of high priority destination. Differences in throughput in packets per second are for the asymmetry in traffic load for each direction. About 50% of Link utilisation is shown in Figure 7.22 in both direction of the high priority destination network link.

The results are collected for a comparison between the network1 and network 2 with regards to the voice traffic. The figure 7.23, 7.24, 7.25, and 7.26 shows a comparison between frame relay delay, frame relay delay variance, and voice packet end-to-end delay traffic sent respectively. A global summary of the important statistics is shown in tabular

form in the last section. Frame Relay delay is more for network 2 as well as the sharp increase of delay variance. Voice packet end-to-end delay is certainly less for network1. From the figure 7.26 and figure 7.27, it is indicated that Voice Traffic sent and received for network2 are not acceptable. So network 2 has outperformed in this respect. Figure 7.28 7.29 and figure 7.30 shows the report of global summery of the different performance parameters. This indicates that increasing numbers of node will affect the network and delay increases resulting decrease of Throughput. The statistics obtained are very similar to the graphical representations. Figure 7.31 shows statistics of point-to-point queuing delay from Link Summary for Network 1, which is within the range.

The graphical representations of the results are shown in the following sections.

Hardware experimental set-up is completed following the diagram shown in section 6.8 of chapter 6 to see the connectivity. There are some problems at the stage of configuration. We can communicate with site A and site B without using the switch, implies that the frame relay network will work with some adjustment of attributes.

7.1 Performance parameters for Network 1

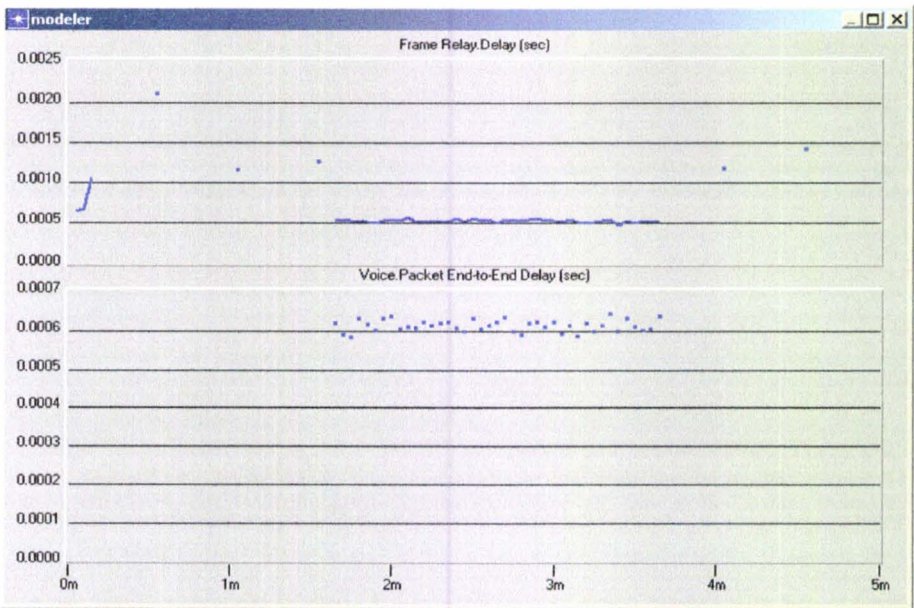


Figure 7.1 FR delay and voice packet End to End delay

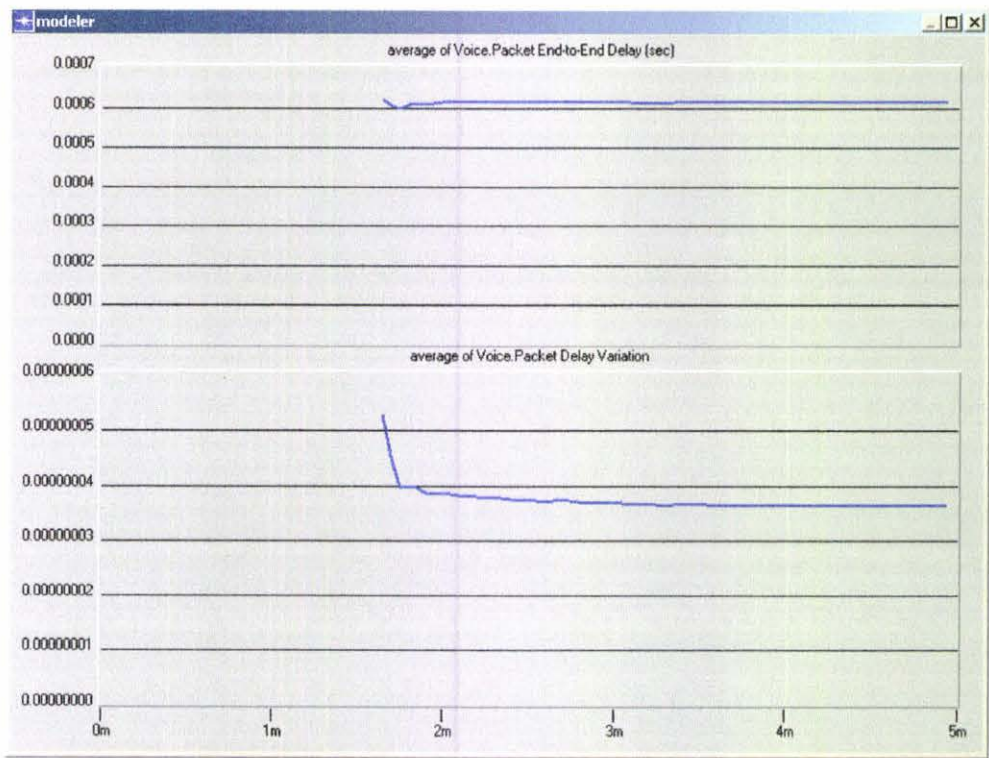


Figure 7.2 Averages of Delays

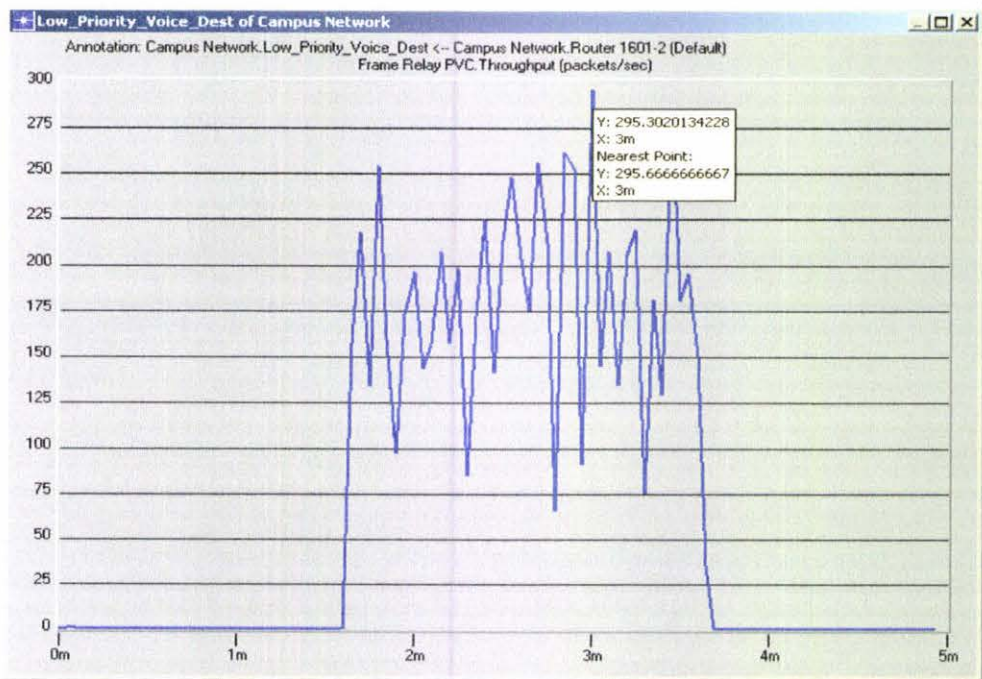


Figure 7.3 Throughput

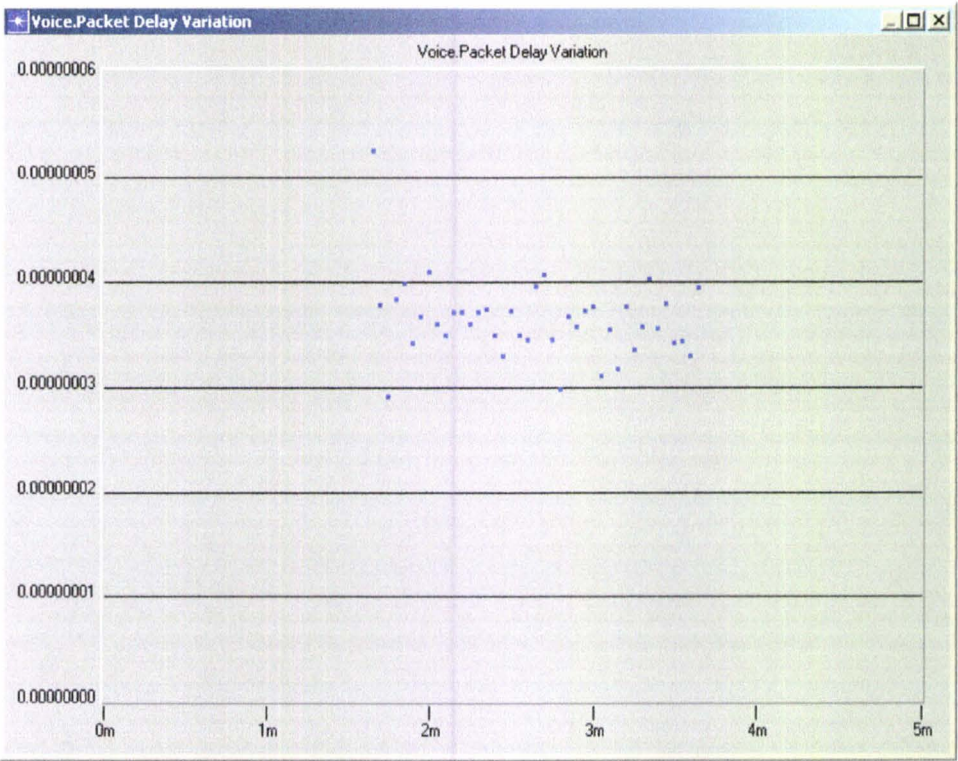


Figure 7.4 Delay variation /Jitter

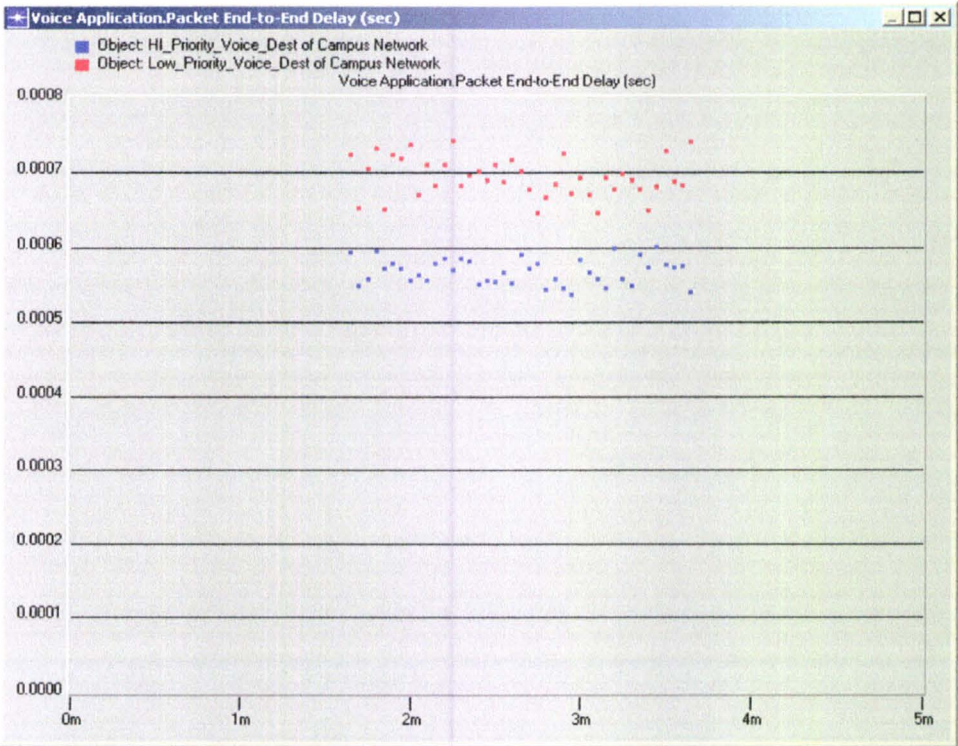


Figure 7.5 Packet End to End Delay at 2 different Destinations with different TOS

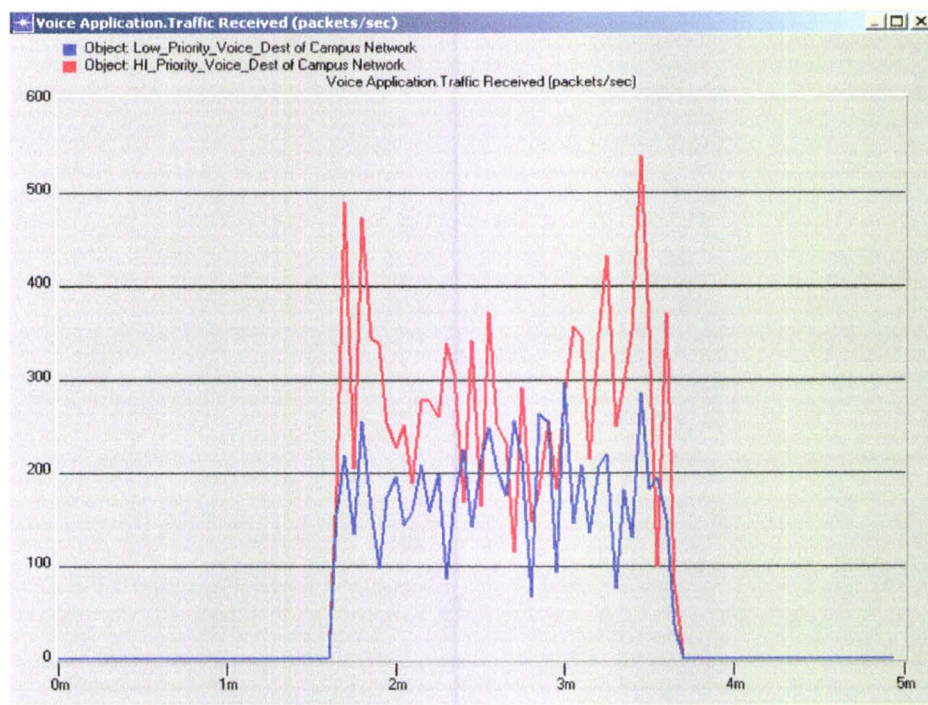


Figure 7.6 Comparison of Traffic received

7.2 Comparison of network 1 for different buffer size

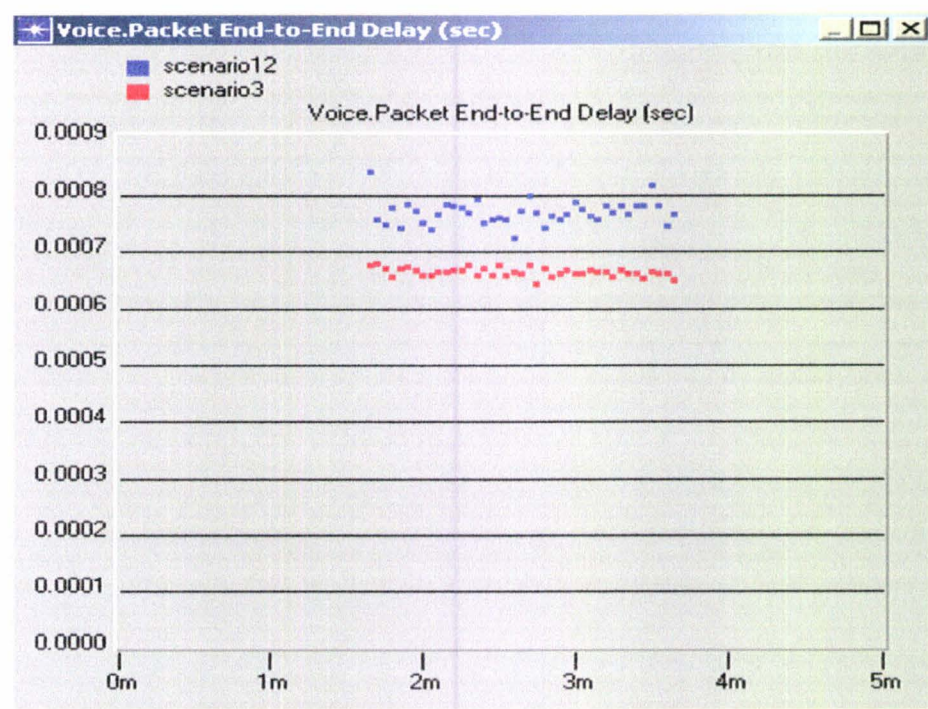


Figure 7.7 Comparison of voice packet end-to-end delay for different buffer size

7.3 Comparison between encoding G.711, G.723.1 and G.729

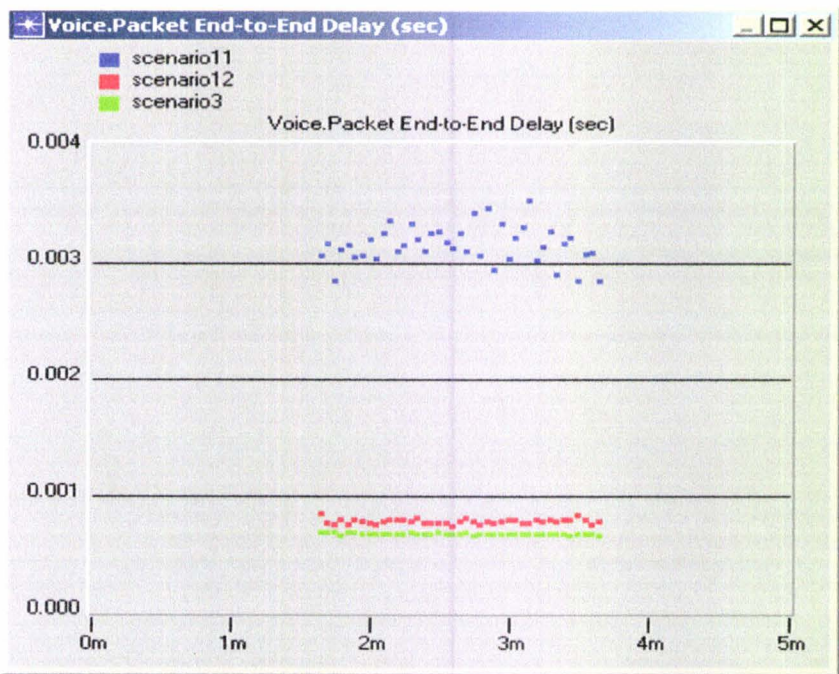


Figure 7.8 Comparison of voice packet End-to-End delay for different encoder scheme.

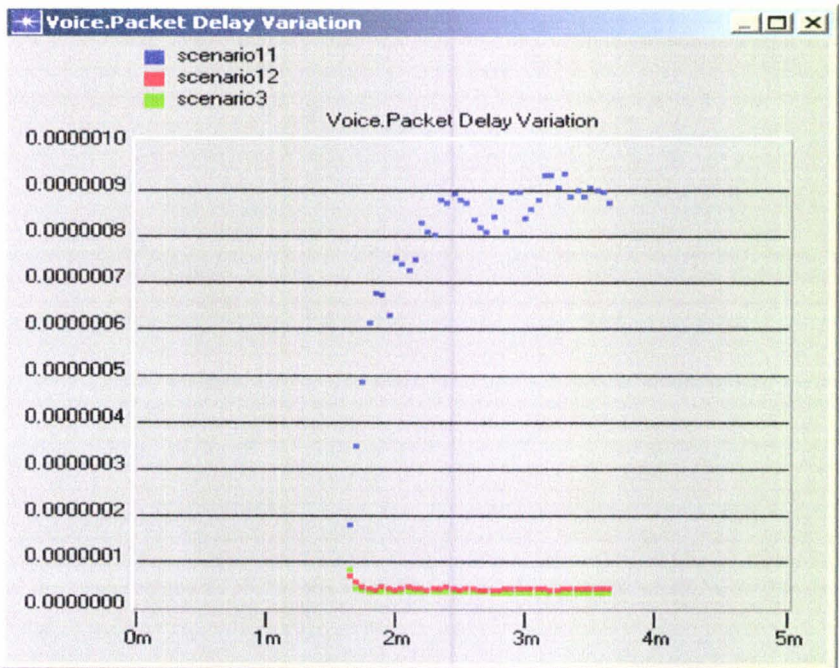


Figure 7.9 Comparison of Voice packet delay variation for 3 different encoder schemes.

7.4 Performance parameters of Network 2

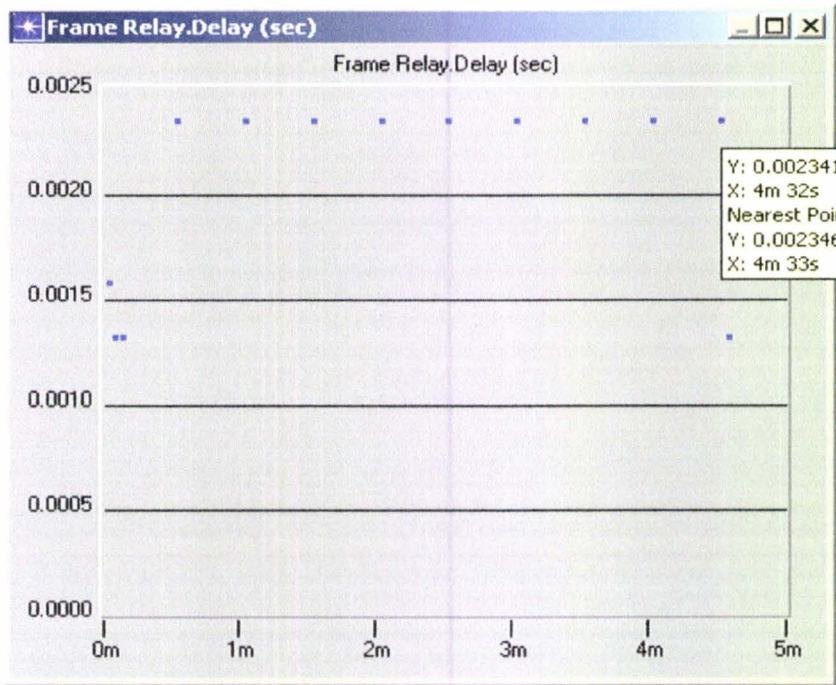


Figure 7.10 Frame Relay Delay

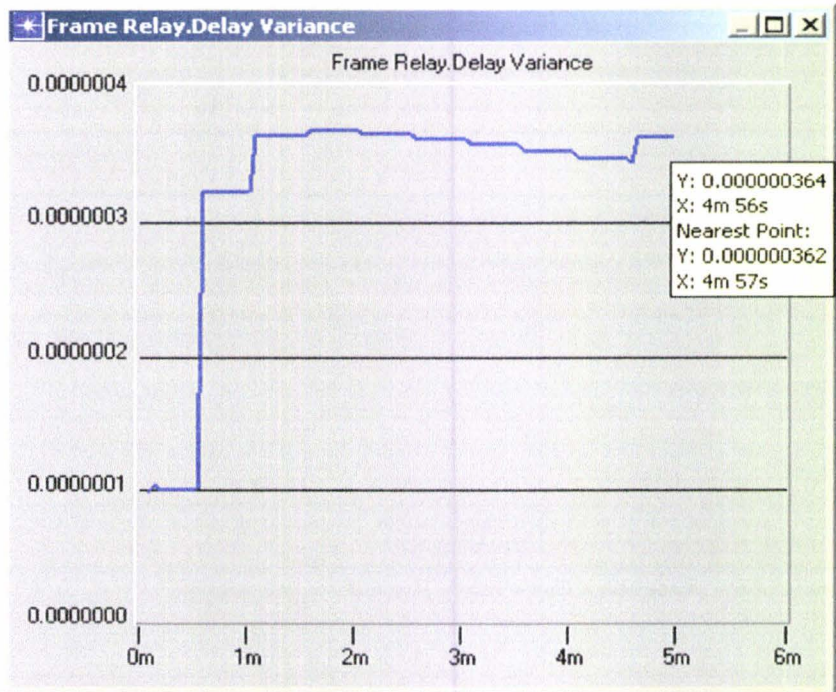


Figure 7.11 Frame Relay Delay variance

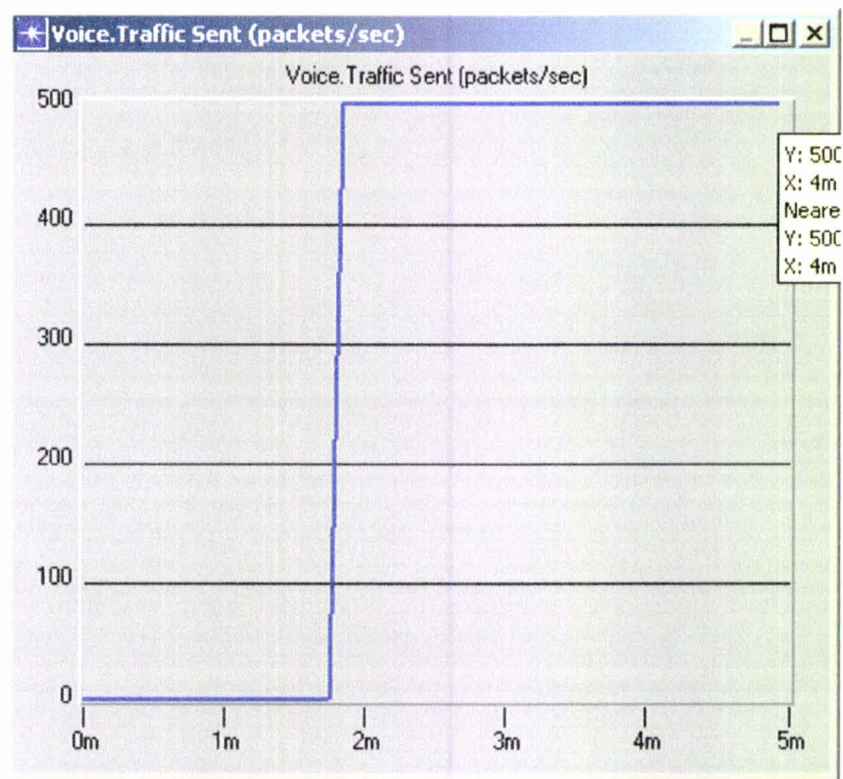


Figure 7.12 Voice Traffic Sent

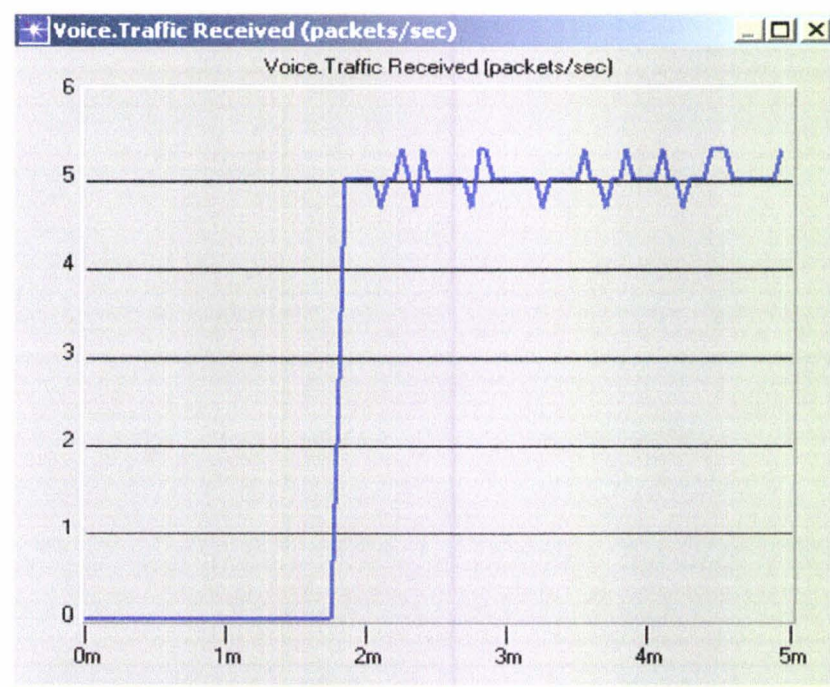


Figure 7.13 Voice Traffic Received

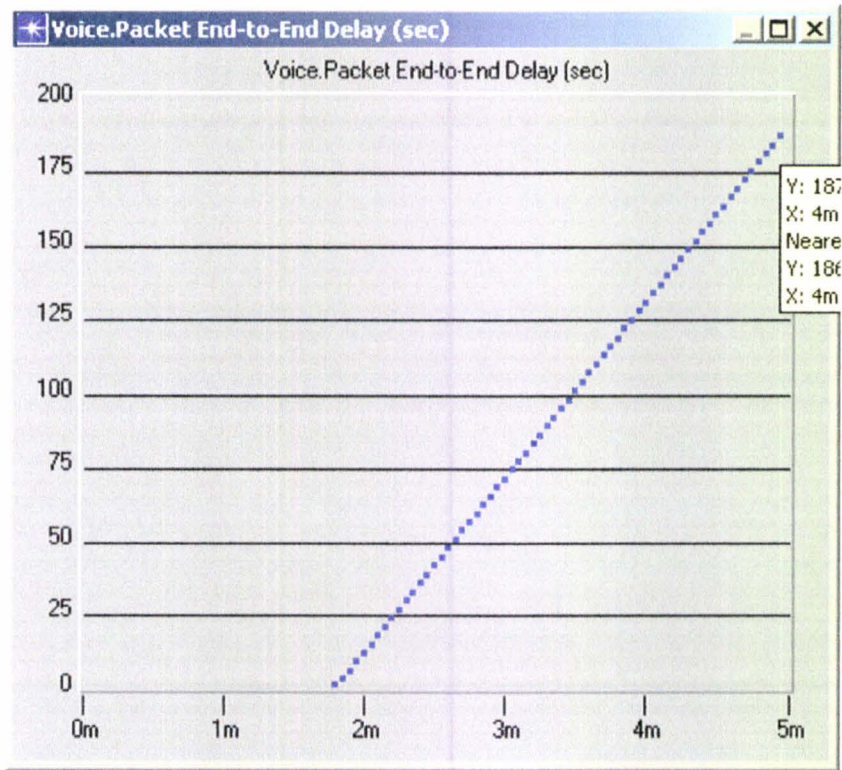


Figure 7.14 Voice Packet End to End Delay

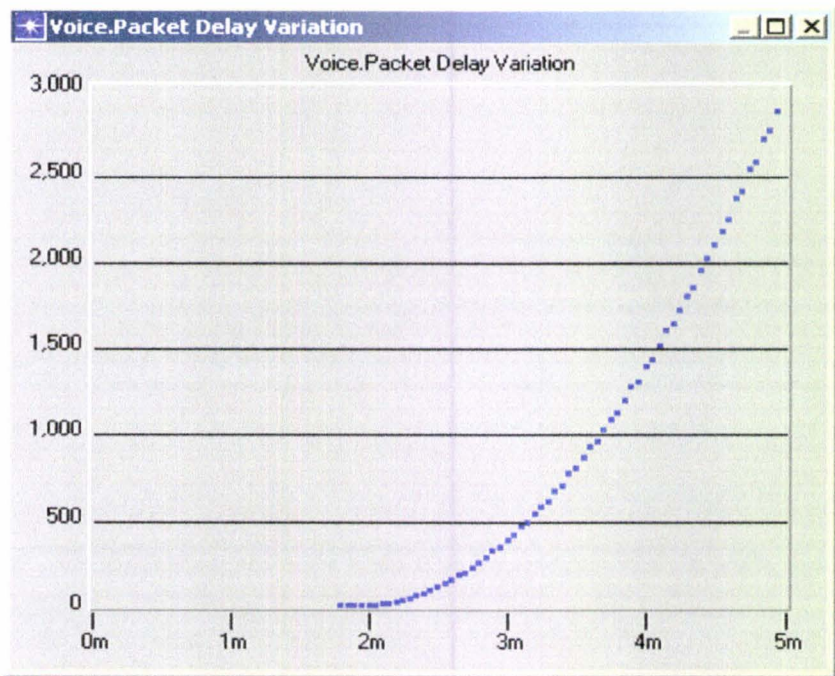


Figure 7.15 Voice packet delay variation/jitter

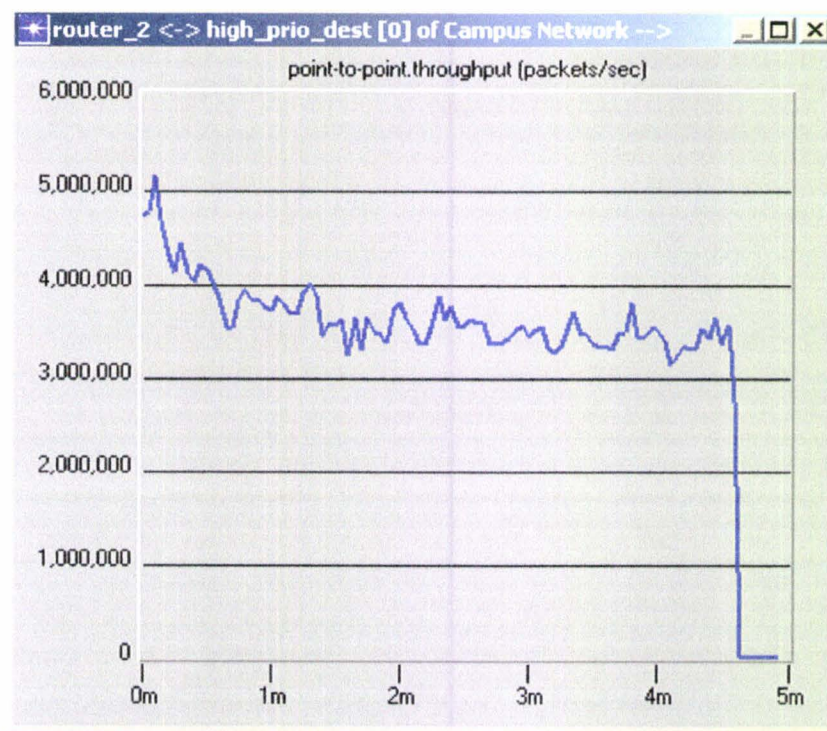


Figure 7.16 Throughput for high priority destination

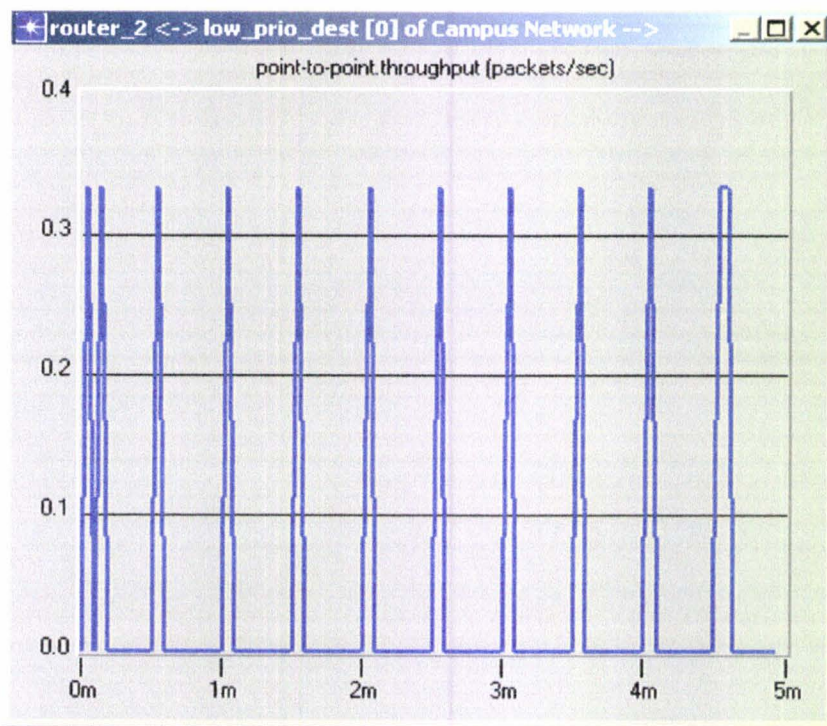


Figure 7.17 Throughput for Low priority destination

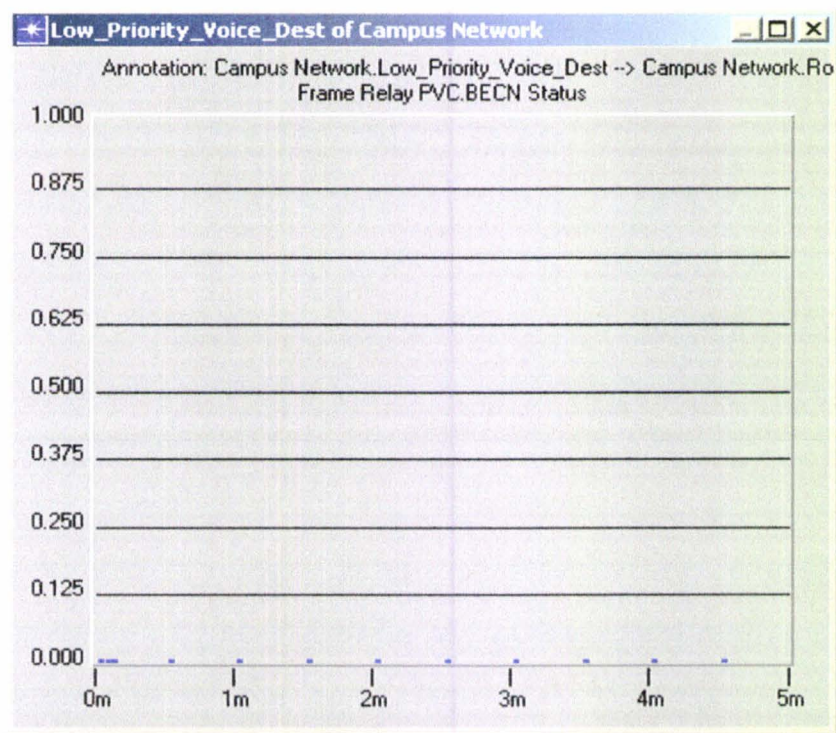


Figure 7.18 Frame Relay PVC.BECN status

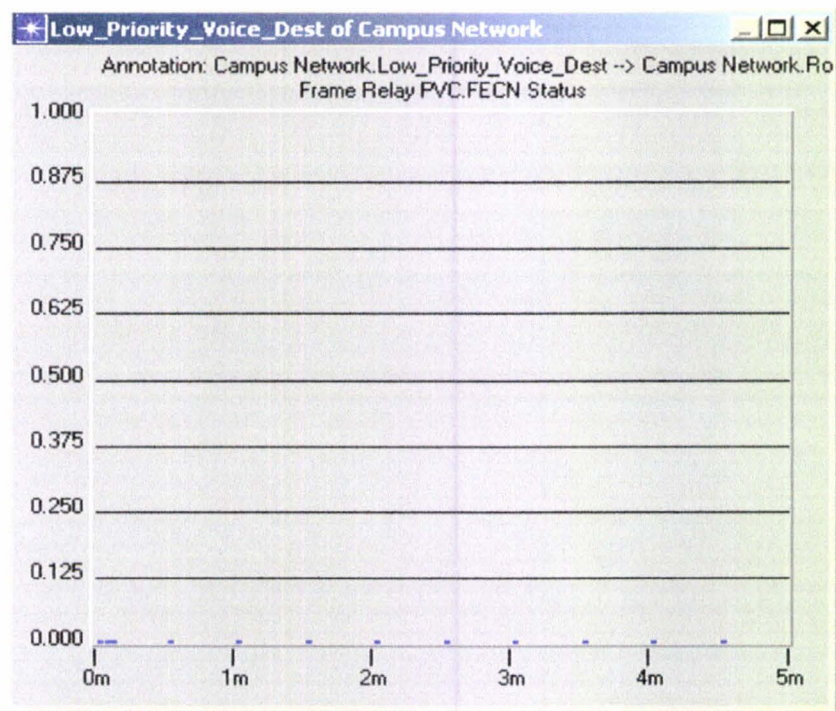


Figure 7.19 Frame Relay PVC.FECN status

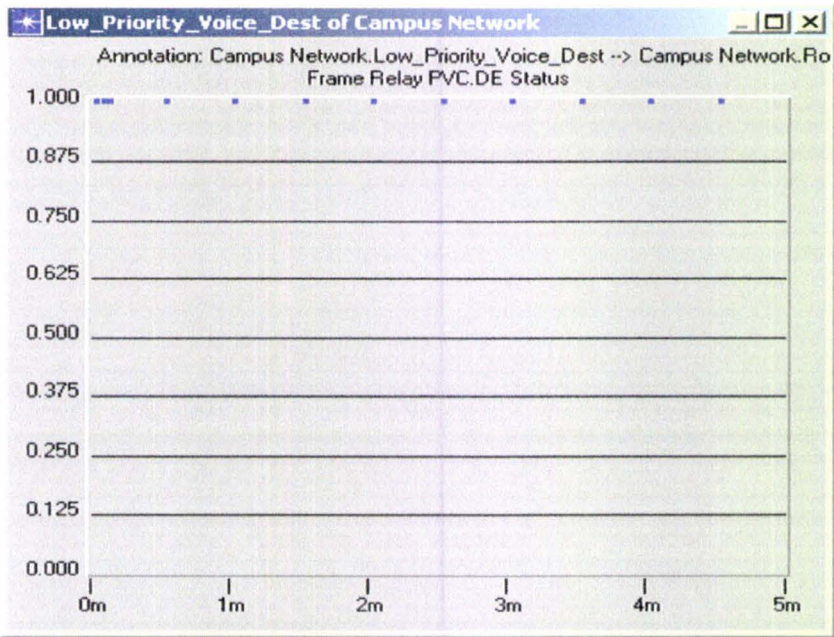


Figure 7.20 Frame Relay PVC.DE status

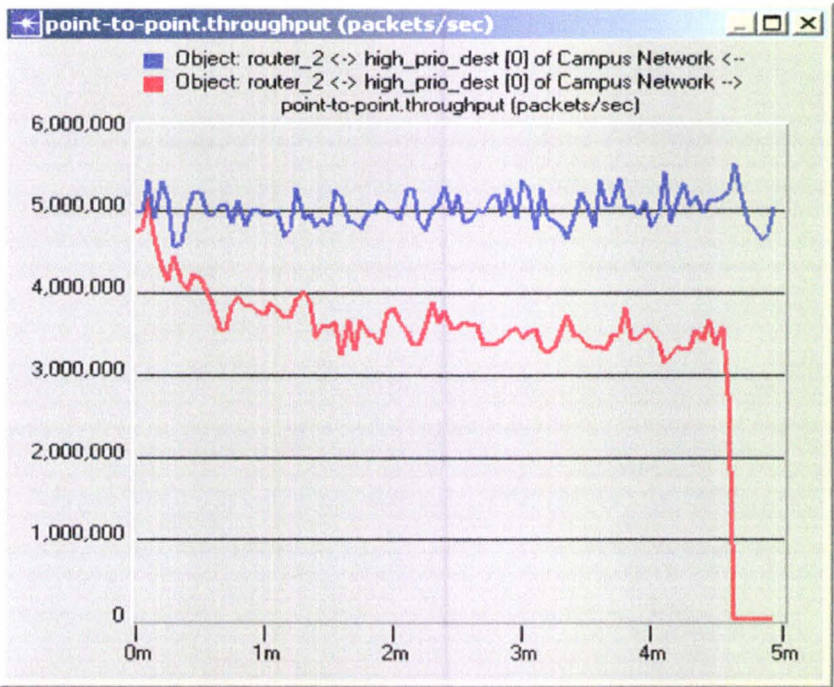


Figure 7.21 Point to Point Throughput of the Link in both sides for high priority destination

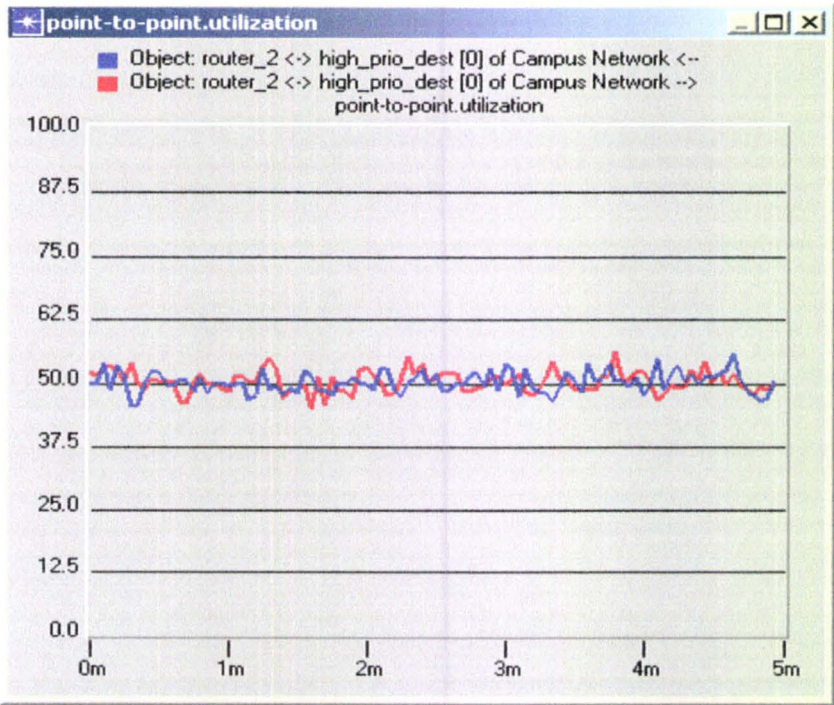


Figure 7.22 Point to point Link Utilisation in both sides for high priority destination

7.5 Comparison between network1 and network2

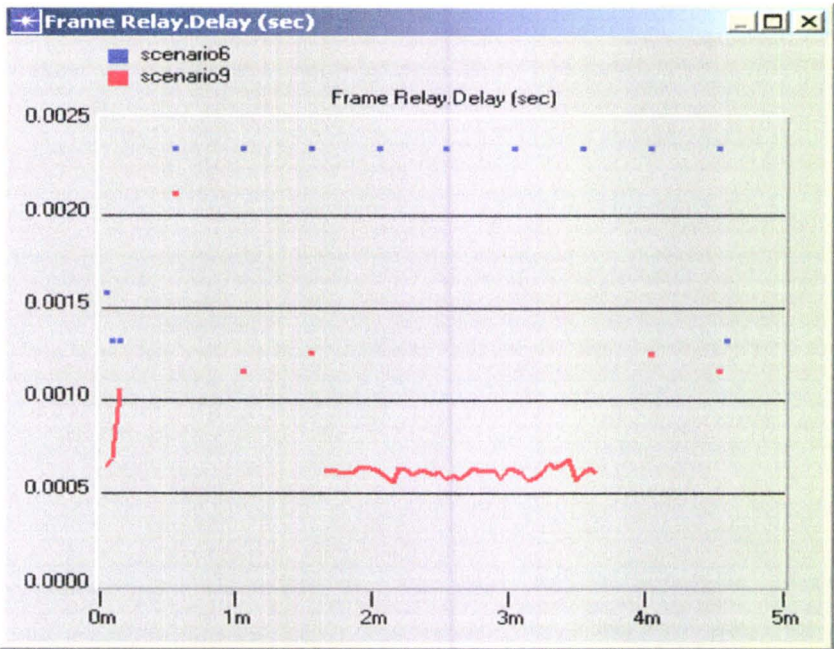


Figure 7.23 Comparison of Frame Relay Delay between network1 and network2

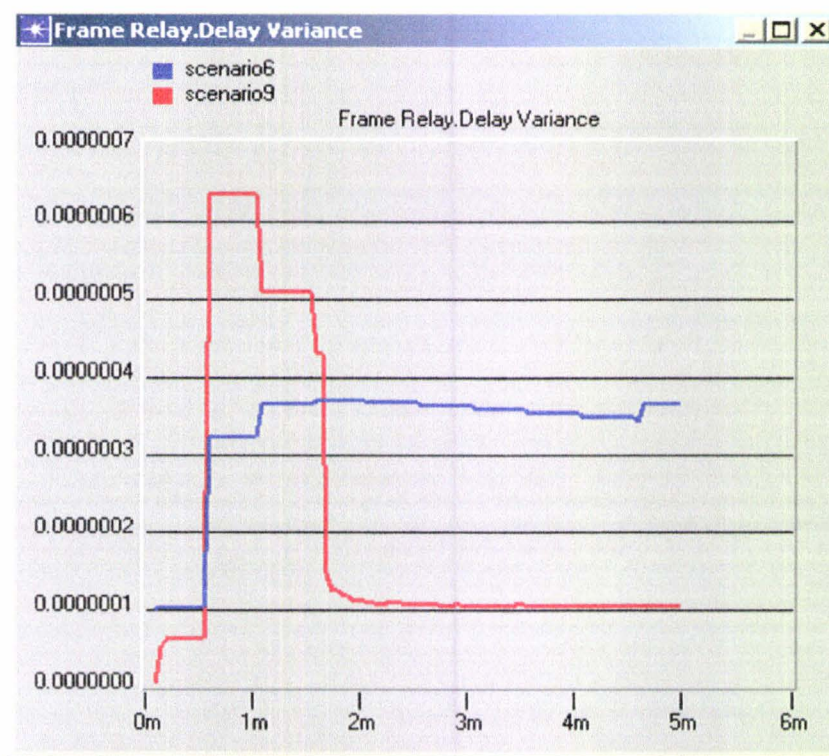


Figure 7.24 Comparison of Frame Relay Delay variance between network1 and network2

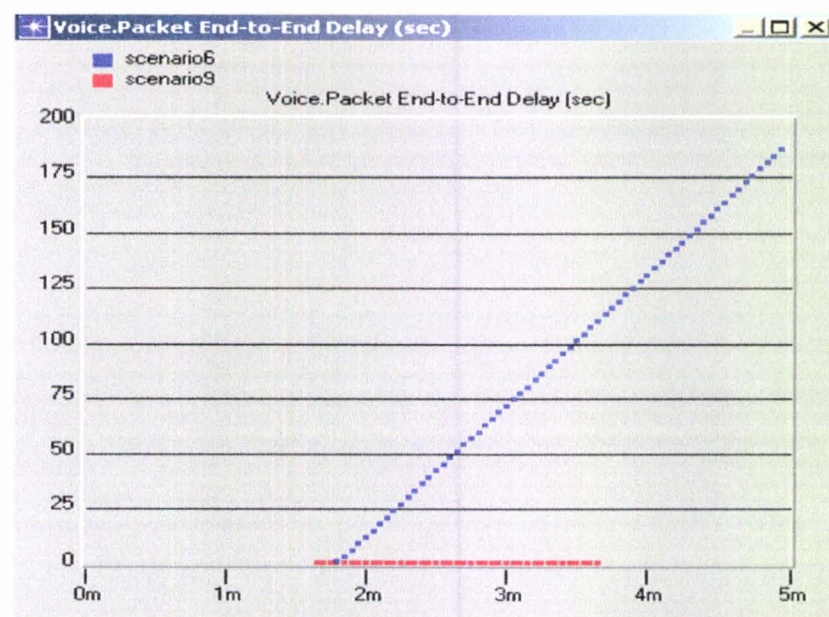


Figure 7.25 Comparison of Voice packet End to End Delay between network1 and network2

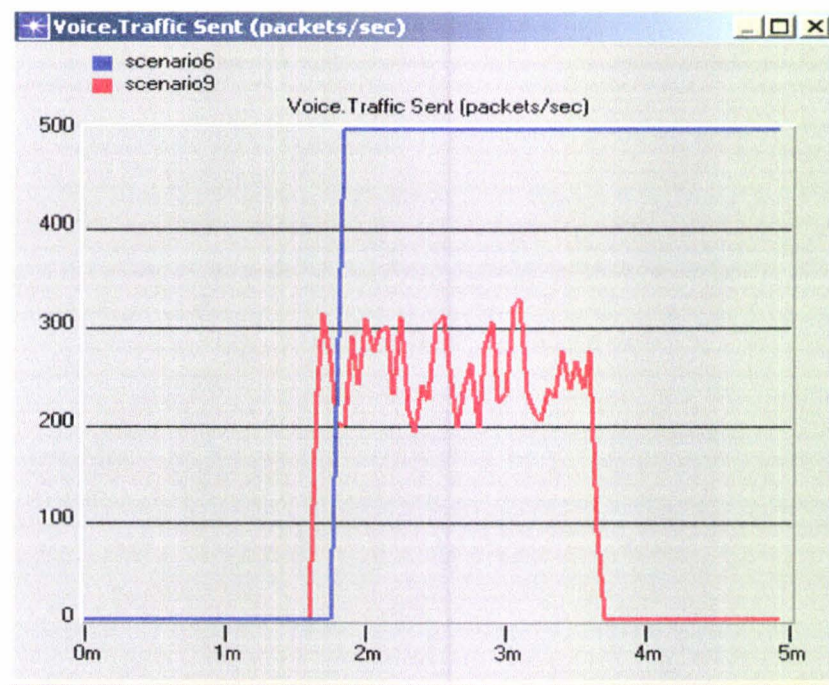


Figure 7.26 Comparison of Voice Traffic sent between network1 and network2

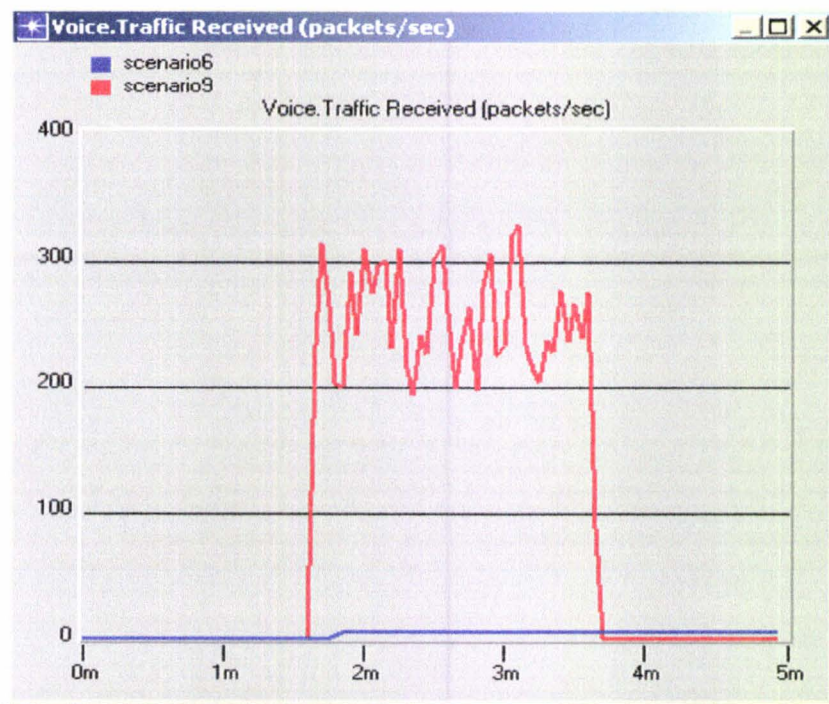


Figure 7.27 Comparison of Voice Traffic Received between network1 and network2

7.6 Report of Global Summary

| | |
|---------------------|----------------------------------|
| Project: project1 | Report: User Selected |
| Scenario: scenario9 | Title: Global Statistics Summary |

Voice

| Statistic | Average | Maximum | Minimum |
|--------------------------------------|--------------|--------------|--------------|
| Voice Packet Delay Variation | 0.0000000521 | 0.0000000990 | 0.0000000435 |
| Voice Packet End-to-End Delay (sec) | 0.000719 | 0.000766 | 0.000677 |
| Voice Traffic Received (bytes/sec) | 1,228 | 3,936 | 0 |
| Voice Traffic Received (packets/sec) | 102 | 328 | 0 |
| Voice Traffic Sent (bytes/sec) | 1,229 | 3,936 | 0 |
| Voice Traffic Sent (packets/sec) | 102 | 328 | 0 |

Figure 7.28 Report of Global Summary for Network 1

| | |
|---------------------|----------------------------------|
| Project: project1 | Report: User Selected |
| Scenario: scenario6 | Title: Global Statistics Summary |

Voice

| Statistic | Average | Maximum | Minimum |
|--------------------------------------|---------|---------|---------|
| Voice Packet Delay Variation | 941 | 2,868 | 0 |
| Voice Packet End-to-End Delay (sec) | 93 | 187 | 1 |
| Voice Traffic Received (bytes/sec) | 102 | 171 | 0 |
| Voice Traffic Received (packets/sec) | 3.19 | 5.33 | 0.00 |
| Voice Traffic Sent (bytes/sec) | 10,150 | 16,000 | 0 |
| Voice Traffic Sent (packets/sec) | 317 | 500 | 0 |

Figure 7.29 Report of Global Summary for Network 2

| Project: project1 | Report: User Selected | | |
|--------------------------------------|----------------------------------|--------------|--------------|
| Scenario: scenario7 | Title: Global Statistics Summary | | |
| Voice | | | |
| Statistic | Average | Maximum | Minimum |
| Voice Packet Delay Variation | 0.0000000194 | 0.0000000545 | 0.0000000170 |
| Voice Packet End-to-End Delay (sec) | 0.000854 | 0.000883 | 0.000842 |
| Voice Traffic Received (bytes/sec) | 552 | 2,003 | 0 |
| Voice Traffic Received (packets/sec) | 138 | 501 | 0 |
| Voice Traffic Sent (bytes/sec) | 553 | 2,000 | 0 |
| Voice Traffic Sent (packets/sec) | 138 | 500 | 0 |

Figure 7.30 Report of Global Summary for Network 1 with extended nodes

| | |
|---------------------|--------------------------|
| Project: project1 | Report: User Selected |
| Scenario: scenario3 | Title: Top Links Summary |

| Link | <u>point-to-point queuing delay</u> |
|------------------------------------|---|
| HI_Priority_Voice_Dest <-> ROUTER2 | 0.000393 <-- |
| HI_Priority_Voice_Src <-> ROUTER1 | 0.000285 <-- |
| LOW_Priority_Voice_Src <-> ROUTER1 | 0.000283 <-- |
| ROUTER1 <-> node_1 | <ul style="list-style-type: none">• 0.000284 -->• 0.000283 <-- |
| node_1 <-> ROUTER2 | 0.000283 --> |

Figure 7.31 Statistics Report of point to point queuing delay from Link Summary for Network 1

Chapter 8

Conclusion And Further Study

When we attempt to characterize system performance, it is generally in comparison to our expectations. Performance is not an absolute. When we seek to validate performance, we validate it against our objectives. Performance reflects application needs.

To reduce the simulation time only two workstations are used to generate traffic load. Each simulation was run for 5 minutes using the advanced simulation menu. Further study could be done using several sites and with more complicated scenario.

For simulation we determine several simulation parameters and network configuration parameters that are used for performance evaluation of the Voice over Frame Relay. Such as Delay, Jitter, Throughput and also codec selection, which are used for real time, voice service. Global end-to-end delay experienced when transmitting voice.

The voice application end-to-end delay does not include the delay of the codecs. To know the delay of the voice stream, a latency of 100 ms has to be taken into account. The delay value obtained from the different network scenario permits a good voice quality.

As shown from the results the network is not overloaded. It has only 50% utilisation. It can carry additional voice traffic. Link utilisation monitoring ensures that there is bandwidth for more traffic. It is a key parameter for integrated voice data and video network.

The type of traffic on the network is changing from the best effort to delay bounded media streams and interactive one. So the networks are investigated whether it would be able to support that type of application.

Buffer size is an important factor, which makes delay and loss. In OPNET, WFQ is classified by ToS based class allocation. There are total 8 classes of type of service (ToS), of which best effort and interactive voice are selected. It should be mentioned that

interactive voice class is the highest class for low delay and loss as compared to best effort class which clearly shown in the results.

Several Voice codec supported by OPNET simulator are G.711, G.729 and G.723 and GSM. In the current environment we can conclude that G.723.1 and G.729 that has small packet size and bandwidth are more appropriate than G.711. Simulation result also indicate that end to end delay for voice packets can be minimized by using efficient codec scheme. So G.729 would be a good choice of Coding scheme in a practical network like ours.

For one network model, the CIR value set equal to 0 for the PVC is investigated, thereby making every packet Discard Eligible. This feature allows user to take advantage of a reduction in operating costs due to the lower cost of uncommitted FR service. With zero CIR level service, the user is not provided with any level of guaranteed bandwidth. Other advantage of this is if there is any congestion in the Frame Relay cloud, voice packets then could be discarded before other data packets.

Packet end-to end delay clearly shows that Low Priority destination node has more delay due to congestion. The table obtained from the OPNET report for a particular model also reveal a large amount of decrease of received packet relative to the sent voice traffic. Here the voice traffic suffers due to lack of guaranteed approach.

To reach more accurate results it would be necessary to perform practical tests for the simulated networks. Frame Relay Network of Net-lab between site A and site B is operated only. Further practical testing could focus on important topics like delay, jitter and other performance parameters as well as improving the network.

The time sensitive applications like multimedia and real time speech need certain delay constraint to be fulfilled by the network. The simulation results indicate that with proper selection of parameters and encoding techniques, FR can support intra office real time voice communication at a satisfactory level. Further study can be done on video Profile where voice conferencing is supported for multimedia application through Frame Relay.

FR technology continues to be a popular access technology. Higher speeds being needed as it continues to grow. The Frame Relay Forum has stepped forward to develop standards considering higher speed, enhanced reliability and cost effectiveness. FRF.14 specifies the standards where the Frame Relay network can run up to 155 Mbps over optical (OC-3) networks. With the increased speed, applications are being built to adjust the new capabilities. Further study on Voice Over Frame Relay is needed to see whether they can cope with the higher speed range than the chosen one. The performance study will reflect the future need of higher speed real time multimedia communication such as Voice and Video Over Frame Relay networks.

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