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A Multiple Regression Analysis Of Selected Variables Effecting The Transmission Of Video Over Internet Protocol Networks

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A MULTIPLE REGRESSION ANALYSIS OF SELECTED VARIABLES EFFECTING THE TRANSMISSION OF VIDEO OVER INTERNET PROTOCOL NETWORKS

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ABSTRACT

The goal of this study was to conduct multiple regression analyses of selected variables effecting the transmission of video over Internet Protocol (IP) networks, utilizing data from Peak Signal-to-Noise Ratio (PSNR) and Picture Quality Rating (PQR) measurement metrics. The variables selected for this research were congestion-specific in nature and multiple regression analyses were performed to identify the amount of variance in video signal quality attributable to their combined effects.

The foundational research hypothesis constructed for this study proposed that increases in CODEC bit rates and network bandwidth would have no significant effect on the PSNR and PQR levels of transmitted video sequences that were devoid of packet loss, packet delay, or jitter.

The packet-specific impairments and limitations on bandwidth were introduced into the test network by way of commercial off-the-shelf (COTS) impairment emulation software, and the CODEC bit rates were controlled at the device level. The testing outcomes indicated a high level of significance in the variance of the CODEC bit rate and network bandwidth for non-impaired video sequences to the PSNR and PQR levels achieved when bit rate and bandwidth were maximized. However, only a moderately significant amount of variance in the overall video quality was attributable to the combined effects of packet loss (drop), packet delay (latency), jitter, and the selected combinations of CODEC bit rate and bandwidth availability.

The results of this research validated that with or without the presence of selected packet-specific impairments, increases in bandwidth and CODEC bit rates do in fact improve the quality of video transmitted over IP networks. For technology managers, this study highlights the importance of recognizing video quality degradation as a by-product of packet-specific impairments and network traffic dynamics. The accompanying documentation includes a video quality improvement decision flowchart emphasizing the need for a continuous process of output monitoring and quality improvement.

Keywords: streaming video, objective video quality, Picture Quality Rating (PQR), Peak Signal-to-Noise Ratio (PSNR), CODEC, IP impairments

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CHAPTER 1

INTRODUCTION

The rapid growth of the Internet has advanced the proliferation of multimedia applications and created new opportunities for organizations to leverage their existing packet-switching networks. Network engineers and administrators from every industry have recognized the need to add value to their technology infrastructures and have found ways to take advantage of the standardization and proven reliability found within technologies such as Transmission Control Protocol/Internet Protocol (TCP/IP) and Asynchronous Transfer Mode (ATM) protocol. Recent innovations in packet switching architecture have been especially significant for organizations in which the ability to utilize effective streaming video and/or videoconferencing solutions is tantamount to operational readiness. Medical schools, hospitals, and telemedicine centers that utilize nomadic and real-time interactive computing for rural patient care diagnosis and treatment can now improve clinical outcomes while simultaneously enhancing the performance and scalability of their legacy network infrastructures.

It is evident that many advances and innovations have come to fruition furthering the efficacy of real-time multimedia transmitted over IP networks, but there still exists a number of technical issues that continue to negatively impact the quality and delivery of such applications.

Resource intensive and bandwidth intensive applications such as video streaming and real-time videoconferencing are often hampered by the vulnerabilities inherent within IP networks specifically related to congestion such as packet loss, packet delay (latency), and jitter. Video file compression accomplished by COmpession/DECompression (CODEC) devices along with insufficiently allocated network bandwidth are also factors whose effects can degrade the quality of the video stream. Moreover, bandwidth and compression issues are not specifically confined or limited to packet-switching networks by virtue of their common application across various technology infrastructures.

The same can be said of data encryption and decryption processes. The security of information transmitted over IP networks is an important consideration within the design of network infrastructures, and the increases in Internet malfeasance and computer crime have made the implementation of effective technical security services and mechanisms paramount for all organizations. Again, this is especially important in situations where local and wide area IP networks are commonly used for the transmission of interactive and/or streaming video, such as in telemedicine centers and rural health clinics.

Encryption and decryption, which transforms data to disguise or conceal its content and then reverses that process to reclaim the original content, has become a common component within IP networks regardless of the type of data being transmitted. Depending on the capabilities and configuration of application security mechanisms, encryption devices will normally introduce a fixed, nominal amount of packet delay irrespective of the type of encryption algorithm employed. However, when combined with the effects of propagation or queuing delays, latency, or jitter, the additive degradation of a video stream may increase to the point that the video received becomes unviewable and subsequently unusable. This too creates a packet switching vulnerability that has not yet been satisfactorily addressed by network engineers or the oversight committees tasked with developing and enhancing encryption and protocol standardization.

Many organizations are unable or unwilling to allocate capital funds for new Information Technology (IT) networking infrastructures, so the need to utilize existing technology in more effective and efficient ways becomes a strategic imperative. The continued advances in network traffic management, error handling, and improved Quality of Service (QoS) options have many organizations now looking to parsimonious IT strategies. These should be based upon recognized economies of scale and the efficient utilization of existing or legacy IP network infrastructures for the delivery of high-quality multimedia and streaming video applications.

Statement of Research

The goal of this study was to conduct multiple regression analyses of selected variables effecting the transmission of video over Internet Protocol (IP) networks. The variables used in this research were congestion-specific in nature and the regression analyses performed within identified the amount of variance in video signal quality that was attributable to their combined effect. The variables selected for analysis were as follows:

- Packet loss (drop)
- Packet delay (latency)

- Jitter
- Bandwidth limitations

Research Hypotheses

The multiple regression analyses that were conducted to quantify the impacts of selected impairments on the transmission of streaming video over IP, were predicated upon the establishment of a foundational research hypothesis (H_0F):

- H_0F: Increases in CODEC bit rates and network bandwidth will have no significant effect on the PSNR and PQR levels of transmitted video sequences that are devoid of packet loss, packet delay, or jitter.

H_0F provided the fundamental basis for several subsequent "impairment-specific" hypotheses (H_01 – H_03), which were developed to guide the research experiments:

- H_01: Dropping 1 out every 500 packets in a periodic distribution will have no significant effect on video stream PSNR and PQR levels.
- H_02: Delaying packets 1 – 10ms in a Gaussian distribution will have no significant effect on video stream PSNR and PQR levels.
- H_03: Random jitter of 1 to 10ms will have no significant effect on video stream PSNR and PQR levels.

Assumptions

This study proceeded under the following assumptions:

- Compression and decompression of composite video signals will degrade the quality of the video stream by some measurable factor, but such processes are accepted requirements for transmitting video over IP networks and were not included as quality-degrading variables within this study.

- The video sequences used within this study were pre-recorded, live-action sequences, which conform to International Radio Consultative Committee (CCIR) standards. The sequences did not contain animations or computer-generated images.
- The network environment used for this study was configured for actual Telemedicine research initiatives, and contained transmission media and network equipment similar to that found in other medical research facilities.

Limitations

The testing of video sequences transmitted over IP networks was accomplished within the context of the following limitations:

- All video quality tests accomplished in this study were performed within a controlled test environment and were therefore unexposed to the impairments that may be present when data is transmitted over other available communication links or the Internet.
- Based upon the review of applicable research literature and the scope of this study, the congestion-specific video quality impairments introduced into the video stream under test were limited to packet loss, packet delay (latency), jitter, and variations in selected bandwidth.

- Due to the operational parameters of the video test equipment used within this study, the video sequences transmitted over the test network were limited to 5 seconds in length and differed only in motion and color complexity, and individual scene characteristics.

Procedure

Impairment variables were introduced by way of impairment emulation software, and bandwidth limitations were applied by varying CODEC bit rates and network bandwidth capacity. MPEG-2 compression was utilized and CODEC bit rates were set at 1.5Mbps, 3.5Mbps, and 10Mbps respectively. Available network bandwidth was set at 10Mbps and 100Mbps respectively.

Analysis of data gathered on video packet loss and subsequent recoverability enabled Feamster and Balakrishnan (2002) to create a *general packet loss model* that explained the quality degradation of MPEG-4 video streamed over the Internet, in the face of variations in bandwidth and delay. Remediation techniques such as selective retransmission of essential data and Application Level Framing (ALF) (Feamster & Balakrishnan; Cark & Tennenhouse, 1990) were proposed within that study as ways to adaptively deliver MPEG-4 video and effectively recover from packet loss, alleviating the propagation of errors.

The work done by Feamster and Balakrishnan provided the impetus for the basic design of this research initiative, however their remediation efforts as well as the remediation efforts of others were neither proposed nor evaluated within the context of this study.

MPEG-2 compression was employed in this study because of its universal acceptability and the availability of applicable CODEC devices. The measurement metrics employed were Peak Signal to Noise Ratio (PSNR) and Picture Quality Rating (PQR) levels, which provided objective picture quality measurements. It is important to note that while objective measurements of video quality may serve to validate levels of signal degradation that can be considered at or below the industry-defined standards governing the transmission of video over IP networks (e.g. RTP, H.323, etc.), the *subjective* quality of the video stream may still be acceptable to those viewing it.

Definition of Terms

Algorithm – A well-defined set of rules for processing data to arrive at a result in a finite number of steps.

Application Level Framing (ALF) – A protocol architecture that encourages application control over mechanisms that traditionally falls within the ``transport layer", e.g., loss detection and recovery.

Asynchronous Transfer Mode (ATM) - A dedicated-connection switching technology that organizes digital data into 53-byte cell units and transmits them over a physical medium using digital signal technology.

Artifact – A term used to describe a change from the original or a defect that results from processing a video signal.

Chroma (Chrominance) – The color information used to created the colors seen in the video display. The RGB (red, green, and blue) color information is mathematically mixed with the luminance (Y) information to produce the R-Y and B-Y signals sent along with the Y signal in a video signal.

CODEC – An acronym that stands for "COmpression/DECompression." An algorithm, or specialized computer program, that reduces the number of bytes consumed by large video files and programs.

Composite Video Signal – A video signal that has the chrominance (RGB) and luminance (Y) of a video image combined with the horizontal and vertical synchronization to provide the complete picture information in a single channel.

Congestion – Degradation in network performance that exists when messages or packets transmitted by an application must wait to utilize necessary resources while traveling from sender to receiver.

Constant Bit Rate (CBR) – An encoding method that varies the quality level in order to ensure a consistent bit rate throughout an encoded file.

Decibel (dB) – A logarithmic expression of the ratio between two signal, power, voltage, or current levels.

Delay - A synonym for *latency*, delay describes the amount of time a packet of data must wait to get from one designated point to another.

H.261 – A video coding standard published by the ITU (International Telecom Union) designed for data rates which are multiples of 64Kbit/s, and is sometimes called p x 64Kbit/s (p is in the range 1-30).

H.263 – A video coding standard published by the ITU (International Telecom Union) designed for low bit rate communication of data rates less than 64 Kbits/s, however this limitation has now been removed. It is expected that the standard will be used for a wide range of bit rates, not just low bit rate applications and may replace H.261 in many applications.

Integrated Services Digital Network (ISDN) – A set of standards for the digital transmission of data over ordinary telephone copper wire as well as over other media.

Internet Protocol (IP) – A connectionless networking protocol in which there is no continuing connection between the end points that are communicating. Each data packet that travels across an IP network is treated as an independent unit of data without any relation to any other unit of data.

Jitter – The deviation in or displacement of some aspect of the pulses in a high-frequency digital signal.

Just Noticeable Difference (JND) – A perceptually based unit of measure for the magnitude of difference between two stimuli utilizing the JNDmetrix™ human-vision algorithm.

Latency – A synonym for *delay*, latency is an expression of how much time it takes for a packet of data to get from one designated point to another.

Luma (Luminance) – The part of the video signal that carries the brightness information used to create the video display.

Moving Picture Experts Group (MPEG) – A working group of the ISO/IEC in charge of the development of standards for coded representation of digital audio and video.

MPEG-1 - A video coding standard on which such products as Video CD and MP3 are based.

MPEG -2 – A video coding standard on which such products as Digital Television set top boxes and DVD are based.

MPEG-4 – A video coding standard that made interactive video on CD-ROM, DVD and Digital Television possible.

Packet - The unit of data that is routed between an origin and a destination on the Internet or any other packet-switched network.

Peak Signal to Noise Ratio (PSNR) – The root mean square (RMS) ratio between the peak signal amplitude of a signal and the noise accompanying the signal.

Picture Quality Rating (PQR) – A picture quality rating number that corresponds to a subjective viewer's rating of picture quality. It is derived from the JND numbers obtained from the measurements made in comparing a test reference video sequence to a captured copy of the same video sequence.

Quality-of-Service (QoS) – The idea that transmission rates, error rates, and other characteristics can be measured, improved, and, to some extent, guaranteed in advance.

Real-Time Transport Protocol (RTP) – An Internet protocol standard that specifies a way for programs to manage the real-time transmission of multimedia data over either unicast or multicast network services.

TCP (Transmission Control Protocol) – A set of rules (protocol) used along with the Internet Protocol (IP) to send data in the form of message units between computers over the disparate networks.

Threshold – A level or point that divides one condition of operation or position from another and perceptually different condition of operation or position.

Transcoding – A process used to reformat data that would otherwise have to be developed separately for use on different platforms. Working like an interpreter, the technology translates content to suitable formats for various platforms, regardless of protocol, application, and language used.

Variable Bit Rate (VBR) – An encoding method that ensures consistent high audio quality throughout an encoded file by making intelligent bit-allocation decisions during the encoding process. VBR encoding produces an overall higher and more consistent quality level than Constant Bit Rate encoding.

Chapter Synopsis

This chapter provided an introduction to the growing popularity and utilization of packet switching networks for the transmission of streaming video from one location to another. Identified advantages include the inherent reliability and stability of the Internet Protocol (IP) architecture, as well as the ability of organizations to effectively leverage their existing or legacy IP networks for use with today's multimedia applications. However, vulnerabilities such as packet loss, latency, and jitter continue to plague packet switching networks when transmitting streaming or real-time video, further propagating resource contention and creating higher bandwidth requirements.

CHAPTER 2

REVIEW OF LITERATURE

The changing dynamics of the business world have facilitated many of the advances in communications and networking technology, and the transmission of continuous or streaming media across Internet Protocol (IP) networks has now become a strategic goal for some organizations as well as an operational requirement for others. Perhaps the availability and affordability of streaming media applications are merely indicative of the increasing interest in leveraging video and audio over IP. More to the point, organizations are being forced by slumping economic conditions to utilize existing network infrastructures and resources to meet the changing demands of the marketplace.

Shi and Sun (2000) suggested that the increased interest in the digital transmission of video signals has come about because of the rapid growth of digital transmission services. They also proposed that the affordable bit-rate reduction brought about by video compression algorithms has become a necessity when utilizing digitized video source signals that require very high bit rates, such as broadcast video (100 Mbps) and High Definition Television (HDTV) (1Gbps) signals.

Rzeszewski (1995) recognized the changing landscape of digital video technology evidenced by the many different formats and standards that have emerged within all market segments and industries. According to Rzeszewski, "the concept of a single 'one-size-fits-all' format for television and video applications no longer makes sense, now that we can customize image systems to particular applications without the risk of being stranded when connections across applications boundaries are needed."

Circuit-switched infrastructures such as digital telephone networks are based on Constant Bit Rate (CBR) communications and do not effectively compensate for the data loss, but have traditionally been the recommended solution for applications requiring video storage and transmission. This is because of the stringent latency requirements found within the H.261 videoconferencing standard (Blake, 1995), and the ease with which media is transported over conventional communications channels (Huang, 1999).

Although circuit-switching networks have the capability to provide sufficient levels of digital video Quality-of-Service (QoS), the network capacity necessary to handle bursty traffic cannot be allocated efficiently enough to provide high signal quality. Conversely, packet-switched networks are designed to enable Variable Bit Rate (VBR) communications and to facilitate the transmission of bursty traffic. VBR coding attempts to encode video content to a constant quality if possible, such that the number of bits used for coding each frame can be defined as a function of the frame content complexity, spatially as well as temporally (Huang, 1999).

Packet switching networks have a number of advantages over circuit switching networks (Stallings, 1997), including greater line efficiency, the ability to carry out data-rate conversion, packet acceptance even during heavy traffic, and the use of priorities to help control transmission delay. Therefore, the migration from CBR coding to VBR encoding achieves sizable bit savings without forfeiting encoded video quality, and ultimately improves the information throughput of digital broadcasting networks (Huang, 1999).

Unfortunately, the advantages of transmitting video over packet switched networks are often tempered by the numerous challenges facing packet video applications. These include frame rate inconsistency under varying network conditions, and jittery frame rates brought about by a lack of detection and adaptation to the constantly changing environment (Chakrabarti & Wang, 1994). Zhang (2001), offered a more comprehensive view suggesting that the fidelity of video transmitted over IP networks could be considered a function of several "end-to-end" factors directly effecting the quality of the stream, such as the video source, the network status, channel coding schemes and the decoder's recovery capabilities.

By design, the transmission of video over IP requires specific flow control mechanisms but as Charabarti and Wang (1994) pointed out, overly conservative control techniques risk wasting network resources while overly aggressive flow control mechanisms can lead to high packet losses.

Research initiatives that address the operational and support issues associated with local area networks (LANs) continue to be valid and necessary, but many organizations still prefer to leverage their wide area communications infrastructure with smaller, more robust internetworks that connect multiple buildings, facilities and campuses. Consequently, the practice of using conventional LANs as the "last mile" to the desktop will most likely continue for many years to come (Stone & Jeffay, 1995; Jeffay, Parris, Smith, & Talley, 1996).

Congestion

Network congestion occurs when any transmitted message or packet (resulting from a message) must wait for the use of any resource while traversing the network from sender to receiver. The degree of congestion may vary from one extreme to the other, such as in the case of a complete absence of congestion where no packet ever waits to use resources (a dedicated network path), to a situation where the network path is totally blocked and all packets wait to use resources indefinitely.

Figure 1 illustrates the basic concept of network congestion. In this example, the source CODEC receives an analog video signal from the camera and then digitizes and transmits it on to the 100 Mbps network at a compression bit rate of 2 Mbps, which will exceed the 1.544 Mbps capacity of the T1 transmission path between Router "A" and Router "B". A queue of data will form at Router "A" due to the discrepancy between its incoming and outgoing bit rates and the limited size of its buffers, resulting in some level of network congestion and a measurable loss of video stream data.

It has been shown that congestion can be mitigated within packet-switched networks by implementing intuitive gateways within an environment where the transport protocol can respond to congestion indications from the network, such as with the Random Early Detection (RED) gateways proposed by Floyd and Jacobson (1993) and further developed by Parris, Jeffay, Smith, and Borgersen (1998). RED gateways are designed to accompany transport-layer protocols such as TCP, and work by notifying network connections of congestion either by setting a bit in packet headers or by dropping specific packets as they arrive at the gateway. The gateway may drop or *mark* arriving packets with a certain probability once the average queue size exceeds a preset threshold, but the gateway has no bias against bursty traffic entering the queue and avoids the possibility that many connections will decrease their windows simultaneously via global synchronization. It is important to note that the increased burstiness and subsequent buffer loss in LANs are often attributable to the fast window openings brought about by the typically short round trip times in such environments (Noureddine & Tobagi, 1999).

During times of congestion, the RED gateway will notify a particular connection to reduce its window based on a probability that is a function of the average queue size, and roughly proportional to that connection's share of bandwidth through the gateway.

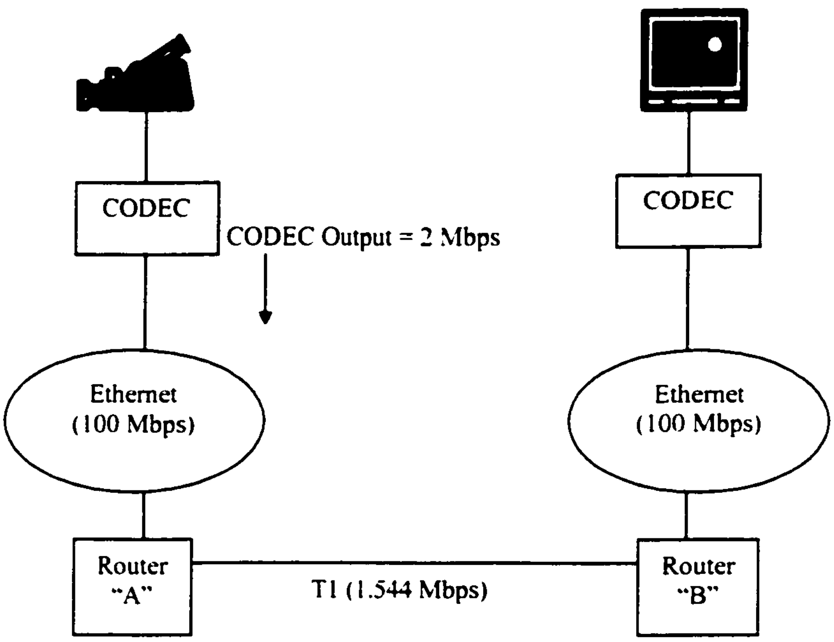

Figure 1 - Basic Concept of Network Congestion

Floyd and Jacobson found that because Random Early Detection (RED) gateways provide the opportunity to control the average queue size before the gateway queue overflows, they could be particularly useful in networks where dropped packets at the gateway are undesirable, such as when running TCP over ATM.

Talley (1997) claimed that low-latency, high fidelity videoconferences could be sustained over current packet-switched networks, even in the presence of high levels of network congestion, if the transmission of the audio and video streams is carefully managed at the endpoints of the conference. When a network has constraints related to capacity (limited bit rate), access (limited message rate) or some combination thereof, complementary transmission adaptations such as *scaling* and *packaging* can be used to preserve the conference quality. Scaling processes modify the media stream bit rate by controlling the stream's generation and compression, and packaging modifies the conference message rate by controlling the number and type of frames placed into each message.

It has been suggested that extreme traffic variability may be one of the most prominent characteristics of LANs because of the way it impacts traffic congestion (Fowler and Leland, 1991). Such variability exists on time scales ranging from milliseconds to months, and by applying realistic data to simple LAN interconnection models, Fowler and Leland observed four specific congestion paradigms:

- Periods of persistent congestion can lead to significant losses.
- Modest increases in buffer capacity will not prevent congestion losses.
- Mis-engineering is a costly dilemma and the consequences can be severe.
- Congestion is not instantaneous and is normally preceded by warning signs.

Noureddine and Tobagi (1999) suggested that the reason LAN congestion control has traditionally been given less consideration than that of wide area network environments is because LANs are normally smaller in size and number of users, and have been typically over-provisioned with regard to congestion avoidance schemes. However, the large-scale LAN deployments of today often contain mixtures of different link speeds (i.e. 10, 100 and 1000 Mbps) that can introduce mismatch considerations, and when combined with aggregate traffic from many different sources can lead to congestion between links.

The presence of congestion along the transmission medium is a significant factor in the degradation of streaming multimedia applications, but the effects of congestion are attributable to and magnified by impairments that require mitigation, such as latency, jitter, packet loss and bandwidth limitations. An examination of each is presented in the following sections.

Delay (Latency)

Ideally the transmission of a video stream across an IP network should occur at or near line speed unabated as it travels the medium. Such "optimal" conditions rarely exist due to the presence of network congestion that can delay the stream as it proceeds from sender to receiver. These incidents of delay or "latency" occur when the amount of time needed for a transmitted video stream to reach its destination exceeds an acceptable or "reasonable" period of time. Latency can be attributable to network impediments, which reside along and within the transmission medium, or as the result of the acquisition-to-display processes, which encompass digitization, compression, and synchronization.

Some amount of so-called "display" latency will inevitably occur as a result of the delays that are present within the transmission media itself regardless of the network conditions (Stone & Jeffay, 1995; Talley, 1997). In situations where the video acquisition and display processes are synchronized, there still exists latency between the start of the digitization process and the start of the display process, due to delay inherent within the media pipeline. Moreover, in unsynchronized situations such as is found in most video conferencing systems, video frames may be queued up and waiting for the display clock to increment before being allowed to enter the decompression process at the CODEC, and the resulting delay, called synchronization delay, will also add to the stream's overall display latency (Talley).

Transmission or "propagation" delay occurs as the video frames traverse the originating network, subsequent gateways, routers, and other subnetworks en route to the destination host for decompression and display. Variables such as router queuing delays, packet processing intervals, and subnetwork medium control processes will add to the overall latency of the video stream. Figure 2 depicts how transmission latency can result from a video stream traveling the physical medium from its source to its destination. As the stream moves across the network path from subnetwork "A" to subnetwork "C", queuing and/or packet processing delays may be encountered at some or all of the router "hops"(R1 – R3), contributing to the network congestion and resulting in additive latency.

Talley (1997) identified previous studies that suggested an acceptable audio latency level to be between 200 and 250 milliseconds for video conferencing and between 250 and 400 milliseconds for one-way audio, but conceded that there was a lack of empirical data addressing "reasonable" latency bounds for video. However, the fact that the video stream should be approximately synchronized with the audio when transmitted implies the acceptance of similar latency levels for the video component.

The effects of latency and delay have challenged researchers in many different disciplines and areas of specialization and have been of particular interest to those concerned with the effective transmission of resource-intensive, variable-bandwidth applications such as those found within healthcare environments.

Studies focusing on the transmission of two dimensional and three dimensional medical images (Salous, Pycock, & Cruickshank, 2001), radiology image archival and retrieval (Ho et al., 1995), image compression within telerobotics applications (Rovetta, Bejczy, & Sala, 1997; Wells, 2000), and telemedicine applications (Hariprasad, Shin, & Berger, 1999; Lemaire, Boudrias, & Greene, 2000) have documented the quality degradation that is apparent when latency impedes such applications.

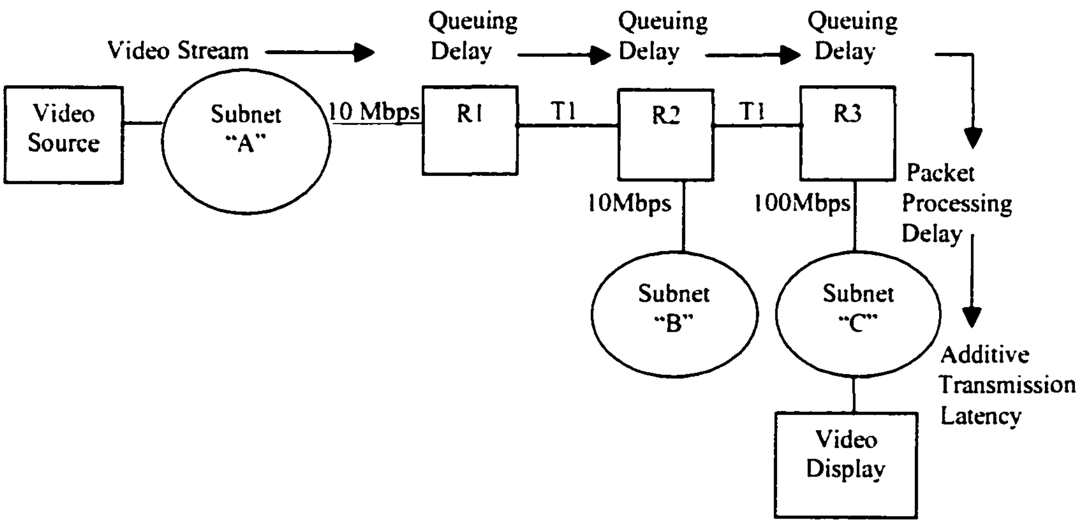

Figure 2 - Transmission or "Propagation" Delay

The ability of a network to cope with the traffic loads that applications may generate is a fundamental requirement for optimal performance, and often effected by factors such as network latency and throughput. In addition to network topology, factors such as routing algorithms, switching techniques, application traffic patterns and implementation technology may also play important roles in determining overall network performance (Mackenzie & Ould-Khaoua, 2000).

Describing a general outline of research into the characteristics of multicomputer architectures, Mackenzie and Ould-Khaoua (2000) discussed the performance impacts brought about by the following:

- Routing algorithms - which prevent the possibility of message deadlock that occurs when queues fill blocking the advancement of messages toward their destinations.
- Switching techniques - which define the methods in which messages are passed to intermediate routers.
- Traffic characteristics - that can cause networks to perform differently based on the execution of the particular application mix and the handling of that mix by the system.
- Implementation technologies - that restrict bandwidth on particular network channels in order to determine the best methods of utilizing the network topology.

In addition, issues such as bandwidth limitations and switching delays were addressed by a combination of discrete event simulation and mathematical modeling.

The control of packet delays and the mitigation of latency within the transmission of streaming media across packet-switching networks will continue to be the focus of technologists and engineers in both the public and private sectors. As the use of streaming media applications continues to grow, it is hoped that the demand for improved levels of quality and fidelity will help to guide future research initiatives.

The effects of quality degrading factors such as those mentioned in this section were controlled within this study by the use of commercially available impairment emulation software, which enabled the introduction of specific packet impairments at varying distributions and within predetermined bandwidth levels. Traffic loads were limited by minimizing the use of unnecessary devices, peripherals, and workstations, and the network topology was designed in a manner that enabled an acceptable amount of flexibility and infrastructure scalability.

Jitter

The variance in delay that occurs as video frames pass through a series of successive pipeline stages is referred to as "jitter" or "delay jitter" (Talley, 1997; Stone & Jeffay, 1995), which is a transmission impairment that results in disruptive "gaps" along the video stream. Gaps can best be described as periods of minimal or zero activity within the transmission process that occur because a video frame is not available for display at a time specified by the display clock. Stated another way, gaps occur when frames arrive with end-to-end delay that is greater than that of the previous frame (Stone & Jeffay).

Talley (1997) referred to the duration of the gap as the "display period" and notes that the gap interval may involve the replaying of the previous frame or may be absent of activity all together. Regardless, it is suggested that the perceived effect of the gap depends on factors such as the transmission media and the output devices.

Nezhad (1999) examined the presence of jitter in a video-on-demand (VOD) system with the goal of identifying the traffic arrival rate boundaries at the video client, such that buffer overflow or underflow conditions might be avoided. Nezhad found constraints on traffic burstiness at the client but was able to establish the client's minimum required buffer size by way of the observed overflow conditions. Conversely, the buffer underflow conditions revealed a relationship between the minimum frame rate maintained by the network and the user's maximum amount of acceptable jitter.

Figure 3 depicts delay jitter as a combination of temporal delays within transmission of a video stream that occur among and between the key processes of acquisition and display.

Preventing delay jitter is the goal of any video conferencing or video streaming system and a number of approaches have been developed to address this issue, such as transmitting streams with constant delay levels or reserving network resources to provide specific bounds or guarantees on delay and delay jitter (Stone & Jeffay, 1995). One scheme presented by Ferrari (as cited in Stone & Jeffay) forces clients requesting real-time communication services to identify their performance requirements and specific traffic characteristics, so that the system can manage the resources at each network node and allocate the required buffer space and processor capacity to guarantee the performance needs of that client.

Jeffay et al. (1992) discussed the effects and possible remediation techniques for short-term and long-term jitter, which can commonly occur within packet-switching networks, and emphasized the importance of adequately managing the transport and display queues within the stream.

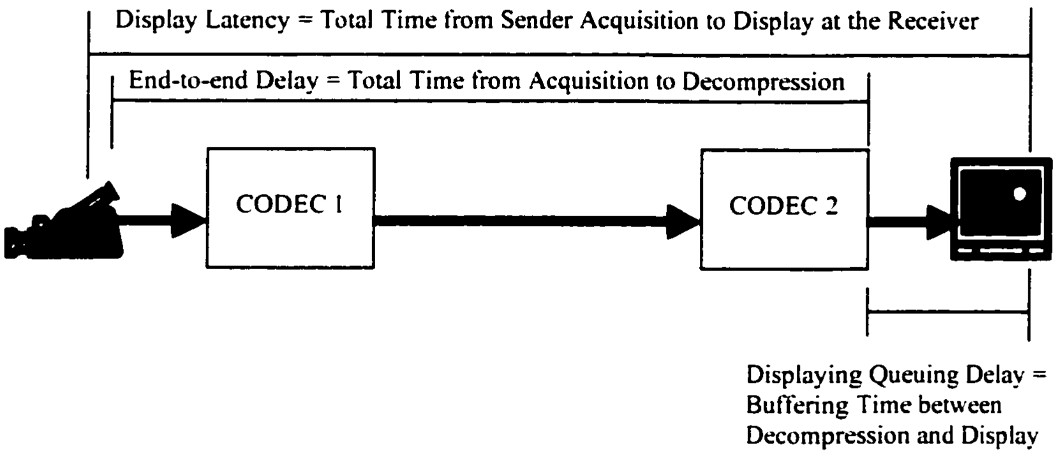

Figure 3 – Temporal Delays Resulting in Delay Jitter

Short-term jitter can be caused by packet bursts or other short-term increases in network load that result in late arriving frames. This type of jitter can be addressed by buffering the audio frames at the display and then playing them ahead of the corresponding video frames, out of synchronization. Conversely, long-term increases in network load my result from decreases in available bandwidth and can cause jitter that must be ameliorated by adapting the frame rate to a level that is sustainable by the network.

Packet Loss

Decreases in video stream fidelity caused by high latency levels or periods of jitter will fluctuate in both severity and duration, with the packets eventually arriving at the receiver having been delayed at various stages along the transmission pipeline. If packets are lost (dropped) the message and all frames carried by the message are unrecoverable, which may create gaps within the video stream.

Packet loss is an indication that the network, or some component thereof, cannot meet the aggregate demand for resources. Moreover, Gerla and Kleinrock (as cited in

Talley, 1997) suggested that reduced levels of network throughput experienced during times of congestion, are directly attributable to the wasted network resources used to transmit the packets that are eventually dropped before reaching their destination. And as with delay jitter, lost packets can disrupt communications by creating gaps or periods of video and audio inactivity within the media stream (Jeffay, Parris, Smith, & Talley, 1996).

In Boyce and Gaglianello (1998) IP-specific metrics such as absolute packet loss, conditional packet loss, and packet loss rate over time were used to describe network loss/error characteristics that can affect the quality of received MPEG compressed streams. Of particular interest was the researcher's decision to utilize the User Datagram Protocol (UDP) as the preferred packet transportation protocol rather than the more reliable Transmission Control Protocol (TCP). This was based on the UDP's suitability for real-time interactive video applications and the fact that it has lower delay levels and lower overhead than TCP. However, the unreliability of UDP requires that errors introduced by packet loss must be concealed.

The use of TCP guarantees the delivery of all packets in order, which UDP cannot provide. However, the retransmission nature of TCP will magnify the effects of packet loss and can be very problematic for streaming media applications, especially when utilizing MPEG compression.

Research efforts such as those completed by Mathis, Semke, Mahdavi and Ott (as cited in Shanableh & Ghanbari, 2000) have addressed the so-called "TCP-friendly" transmission of video over IP by utilizing a TCP-throughput model that enables more effective transmission control and quality of service than is possible when employing unreliable transport protocols such as UDP and Real Time Transport Protocol (RTP).

Packet losses commonly occur at high rates over IP when the network transmission capacity is exceeded, and the retransmission of lost packets is the commonly implemented means of re-establishing network reliability. However, the temporal dependencies associated with MPEG and other motion-compensated video coding can lead to degraded video quality when packet loss rates are high (Boyce & Gaglianello, 1998; Goshi, Mohr, Riskin, Ladner, & Lippman, 2001). Moreover, video quality may be degraded over a large number of frames with compression algorithms such as H.261 and MPEG. The impacts of the lost packets on the quality of the video delivered to a destination can be effected by the coding scheme used at the video source as well as the characteristics of the loss process in the network (Bolot & Turletti, 1998).

Lost packets within a transmitted video stream will propagate content loss downstream because of the spatial and/or temporal losses that occur as a result of the inter-block and inter-frame correlation within the stream (Shin, Kim, & Kuo, 2000). This situation may be more damaging than expected if the potential packet loss has not been accounted for within the stream. For example, an MPEG video stream subjected to a 3% packet loss may in fact equate to a 30% frame error rate (Shin et al.; Boyce & Gaglianello, 1998).

Motion estimation compression is used in many different CODECs including MPEG-1, MPEG-2, MPEG-4, H.261, and H.263. It removes the temporal redundancies within successive video frames by encoding the pixel value differences between the current image and its motion-predicted image, which has been reconstructed from a previous reference frame. However, one of the most common issues identified in studies addressing motion estimation compression is the damaging effect of packet loss and error propagation on video quality (Bolot & Truletti, 1996, 1998; Boyce & Gaglianello, 1998; Feamster & Balakrishnan, 2002; Hemy et al., 1999; Joshi & Rhee 2000; Rhee, 1998). If packets belonging to a video frame are lost or discarded, the quality of that video frame is degraded and the distortion or error propagates to successive frames.

Composite or analog video signals must be digitized and compressed in order to enable transmission across communications networks. Compression is the process by which video files are decreased (compressed) in size for faster more efficient delivery across the transmission medium. A number of standards exist upon which the development and performance of compression technology is founded, including MPEG, H.261, and H.263. MPEG-specific compression techniques have been widely used for video and image transmission, and the dynamics that surround packet loss and recovery continue to be of interest to researchers (Boyce & Gaglianello, 1998; Feamster & Balakrishnan, 2002; Hemy, Hengartner, Steenkiste, & Gross, 1999; Joshi & Rhee, 2000).

Bandwidth Limitations

One of the major traffic management challenges faced when transmitting video streams at constant or near-constant frame rates is providing adequate network support for the video traffic without underutilizing bandwidth resources (Krunz & Tripathi, 1997). In such cases the video will exhibit a variable bit rate (VBR) depending on the compression technique and scene dynamics. Research by Krunz and Tripathi focused on attempting to determine efficient bandwidth allocation levels for delivering pre-recorded video, and noted that other issues such as operating system support, media synchronization, and storage can also have an impact. Allocation efficiency was of particular interest in this regard and was measured by determining the effective bandwidth level per stream. This in effect, represented the maximum number of connections that could be simultaneously transported using some amount of fixed total bandwidth.

Chimienti, Conti, Gregori, Lucenteforte, & Picco (2000) investigated efficient bandwidth allocation schemes for the transmission of MPEG-2 video traffic over high speed networks and were able to identify rare high-rate periods within the CODEC bit stream, which were responsible for low bandwidth utilization. Efficient bandwidth utilization and a "quasi-constant" quality transmission were achieved by way of video source scalability, specifically by defining a Markovian model for an MPEG-2 scalable source.

Chakrabarti and Wang (1994), however, recognized that advances in network technology, better operating systems, and more effective network software support might one day obviate the need for applications that can intuitively adjust to changes in the network environment.

Proper bandwidth management is often a determining factor in maintaining the quality and fidelity of streaming applications and can be used as a congestion control mechanism in localized network environments. So-called "best-effort" approaches such as spatial and temporal scaling mechanisms are techniques that can be used to adapt media streams to current network conditions by scaling (increasing or decreasing) the bandwidth requirements of the transmitted multimedia stream such that it approximates a network connection that is sustainable (Jeffay et al., 1996).

The challenges associated with transmitting multimedia traffic over non-QoS networks are many, but can be lessened by proactive planning and proper resource management. It is more desirable to have the performance guarantees offered by measured QoS solutions dynamically allocate resources and bandwidth when and where they are needed. Unfortunately, guaranteed QoS for multimedia traffic is difficult to accomplish (Fulp & Reeves, 1997). This is because of the unpredictability of the future behavior of real-time or interactive applications, the potential resource contention caused by allocation renegotiations (changes), and the long-range dependencies exhibited by compressed video. However, bandwidth allocation schemes such as the *off-line* and *on-line* approaches described by Fulp and Reeves can provide the opportunity to mitigate the aforementioned challenges. Off-line methods such as peak-rate allocation rely on the pre-transmission availability of the traffic to perform analysis and make allocation decisions, whereas on-line methods utilize the predicted future traffic behavior to renegotiate the resource allocation.

Video Quality Degradation

The effects of scene complexity and motion can cause the rate of a video sequence to vary rapidly with time. In order to send a variable rate video sequence into a constant bit rate channel, in effect creating a constant data stream from a variable rate sequence, the variable rate stream must first be buffered and then released at a constant rate (Bolot & Turletti, 1998). Rate control mechanisms that prohibit buffer overflow or underflow have been in use for more than 25 years, and their main purpose has been to maintain a constant output rate by increasing the video quality for scenes of lower complexity and decreasing the quality for scenes of higher complexity (Bolot & Turletti, 1998).

The use of IP networks to deliver video conferencing sessions can be found in many different industries and environments and because of the functionality of IP, there are many design and configuration options available including those that specifically utilize the Internet. Edwards (1999) described the expansion of one company's IP videoconferencing solutions with gateway technology connecting ISDN and IP videoconferencing system, enabling multiple site participation in the same conference session. Luther and Inglis (1999) identified various Quality of Service (QoS) factors that can adversely impact video signal transmissions over the Internet including data rate, error rate, and delay, leading them to suggest that present-day Internet connections may not provide adequate QoS for real-time video delivery.

Composite video signal compression is accomplished by way of CODECs (COmpression/DECompression), which take the form of either hardware or software. They function both temporally (time) and spatially (space), and most often utilize complex algorithms to discard redundant or unnecessary (irretrievable) information within the compression process while decompressing only that information that is required within the video file, and this can lead to measurable amounts of video quality degradation. Advances in video compression techniques continue to take place, including efforts to incorporate security and encryption capabilities within the compression technology.

While the basic constructs associated with MPEG compression and the adaptive streaming of MPEG video over packet switched networks can be found in Blake (1995), Izquierdo (1998), and Ramanujan et al. (1997), other studies have chosen to address the framework of error remediation of MPEG-specific issues. Employing error and TCP-friendly congestion control mechanisms (Bansal & Balakrishnan, 2000; Bolot & Turletti, 1996, 1998; Rhee, 1998), error resilient video compression techniques (Hasimoto-Beltran, 2001; Tan & Zakhor, 1999), and forward error correction methods (Goshi et al., 2001) are some of the ways to mitigate the video quality degradation brought about by packet loss.

Composite video compression and decompression processes often result in the development of visual errors or anomalies called "artifacts," which degrade the appearance and quality of the transmitted image. Whitaker and Benson (2000) concluded that compression algorithms found within specific MPEG encoding devices determine in large part the artifacts that result from the compression processes. These include block effects (a blocky grid that appears to remain fixed as objects move beneath it), dirty window (streaking or noise that remains stationary as object move beneath it), mosquito noise (high frequency DCT terms which may be seen at the edges of text, logos, or other sharply defined objects), and wavy noise (coarsely quantized high-frequency terms which are often seen during slow pans across highly detailed scenes).

In research accomplished by Basso, Cash, and Civanlar (1997), a manipulation of Real-time Transport Protocol (RTP) payloads was presented as a way to sustain the playback of MPEG-2 based video without losing multicast functionality on desktop systems subjected to high packet losses. In Stone (1995), a combination of real-time operating system and formal modeling and analysis techniques were used to ameliorate the effects of display latency and delay jitter on continuous media frames at endpoint workstations.

Chapter Synopsis

This focus of this chapter was to provide a review of some of the previous studies and applied research initiatives that have addressed the developments and innovations associated with the transmission of video over packet switching networks. The literature reviewed within offers insight into the characteristics and limitations of bursty traffic and the efforts made to mitigate video quality degradation by varying software or protocol architectures.

Network congestion is indigenous to IP environments and some of the packet specific impairments that contribute significantly to congestive conditions were discussed within this chapter including packet loss, packet delay (latency) and jitter. In addition, previous studies examining the dynamics of network bandwidth as a factor in video quality degradation were also reviewed.

CHAPTER 3

METHODOLOGY

The identification of the threshold levels (baseline values) at which certain quality-degrading variables render streaming video unusable or unviewable was key to this study. The testing design required that video stream performance baselines were established first, from which a series of subsequent video stream tests could then be compared.

The measurement of the PSNR and PQR values of unimpaired video streams provided the necessary quality baseline values. Each of the subsequent video stream tests included one of the performance-degrading variables (packet loss, packet delay, and jitter), transmitted within the confines of predetermined bandwidth levels and CODEC bit rate settings.

Video Stream Impairments

The congestion-specific impairments of packet loss, packet delay (latency) and jitter, as well as bandwidth limitations were introduced by way of commercial off-the-shelf (COTS) impairment emulation software, and CODEC bit rates were controlled at the device. The impairments were applied at progressive levels of severity, to ensure that the transmitted video streams were subjected to a comprehensive range of impairment levels within each impairment category.

Video Quality Measurements

Objective video quality was measured based upon two metrics: Peak Signal to Noise Ratio (PSNR) and Picture Quality Rating (PQR). PSNR is a raw, non-human vision system picture differencing measurement (Tektronix, n.d.), derived from the ratio of the peak signal to the Root Mean Square (RMS) noise observed between a reference composite video signal and a captured test signal. The output of the PSNR measurement is presented in decibels (dB), and the metric utilizes the luminance component of the video stream to determine field-by-field differences between reference and test video streams, even if those differences are not detectible by a human viewer. The recommended methods to obtain PSNR of a video signal are outlined in ANSI T1.801.03-1995 (NTIA, 1995) and the utility of this objective measurement of video quality is well documented (Feamster & Balakrishnan, 2002; Goshi, Mohr, Riskin, Ladner, & Lippman, 2001; Joshi & Rhee, 2000; Kuhn et al., 1998; Rhee, 1998; Yang, 2000).

The PQR value for a video field is a non-standard video quality metric based on the JNDmetrix™ human-vision algorithm (Tektronix, n.d.), which accumulates "just noticeable difference" (JND) values for image blocks of 32 pixels by 32 lines by 4 fields deep. JND values are perceptually based units of measure that identify the magnitude of difference between two stimuli. The aggregate of the JND values of all image blocks in four fields creates a single field PQR value, which when further aggregated with JND values from other fields provides a single number, the Picture Quality Rating (PQR) for the scene. The scene PQR correlates with human video fidelity ratings when the scene is viewed at a distance of four times the height of the viewing screen.

Although PQR measurements can be made on both the luminance and chrominance components of the video signal, only luminance measurements were used for this study. Luminance provides the basis for an adequate comparison of video quality differences, whereas chrominance impacts the overall PQR level only slightly and requires much more time to derive. It is important to note that the PQR metric was also used as a measure video quality by Huang (1999), in which an earlier version of the test equipment used in this study was employed, along with a number of the same CCIR standardized video sequences described below.

Testing Dynamics

Nine different video sequences of varying characteristics and motion were used as sources for the video quality tests within this study. Each sequence was five seconds in length and conformed to International Radio Consultative Committee (CCIR) standards. The CCIR is a predecessor organization of the International Telecommunication Union (ITU), which is responsible for studying technical issues related to radio-communications and working toward the goal of standardizing telecommunications worldwide.

Table 1 details the scene characteristics, motion, and CCIR source of each video sequence used within this study.

Test Configuration "A"

An impairment-free test configuration was required for the purpose of capturing baseline PSNR and PQR values from all nine video sequences prior to the introduction of

IP impairments. This was accomplished with test configuration "A", in which two CODECs were configured for MPEG-2 composite (analog) video compression at a bit rate of 1.5Mbps, and assigned individual IP addresses that reflected a common subnetwork ID and subnet mask (e.g. 172.40.1.10/16 and 172.40.1.15/16). The CODECs were connected in a "back-to-back" fashion using a crossover Ethernet cable between both units (Figure 4).

Table 1 - Test Video Sequences

Scene Name	Scene Characteristics	Motion	Video Source
Cheer	Fast complex sports, rich background	Sports	CCIR 39
Flower	Color details, landscape	Slow Pan	CCIR 15
Football1	Sports, busy, large objects	Rapid motion	CCIR 38
Kiel	Luminance detail, landscape	Zoom	CCIR 26
Mobile	Random motion of objects	Slow	CCIR 30
Popple	Moving colors	Pan, rotate	CCIR 28
Susie	Skin tone, talking head	Slow	CCIR 16
Tempete	Horizontal, vertical, luminance, color detail	Random motion	CCIR 44

The Picture Quality Analysis (PQA-300) system used for measuring PSNR and PQR was configured to generate and capture analog video sequences, measure the captured sequences against stored reference sequences, and display the measurement results on a SVGA monitor connected to the PQA unit. The PQA-300 system was connected to the "Video IN" port of CODEC 1 and the "Video OUT" port of CODEC 2 using coax cables with BNC connectors. This enabled the transmission of the video sequences through the CODECs in a manner free from impairments and the potential effects of the transmission medium or fixed network infrastructure.

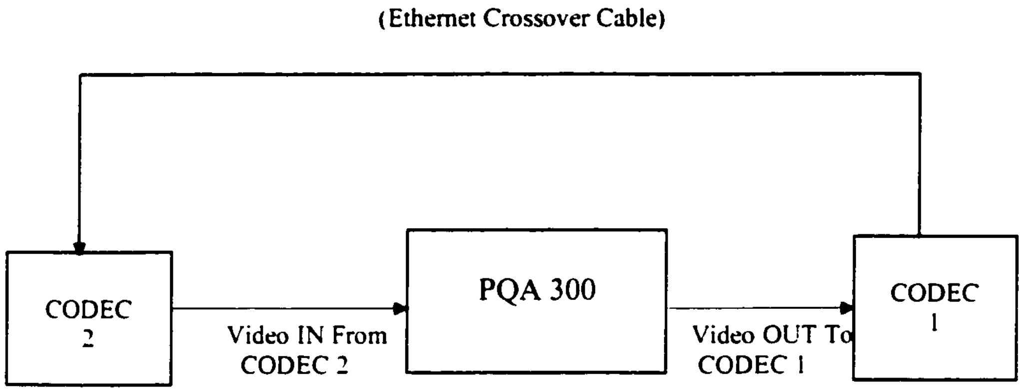

Figure 4 – Configuration "A": CODECs Back-To-Back

Test Configuration "B"

Test configuration "B" was implemented to examine the video sequences as they traversed a limited network infrastructure, which was comprised of the two CODECs connected together using straight-through Ethernet cables, in an "extended" back-to-back configuration via a network switch (Figure 5). The CODEC IP addressing scheme as well as the PQA-300 configuration remained the same as that used in test configuration "A".

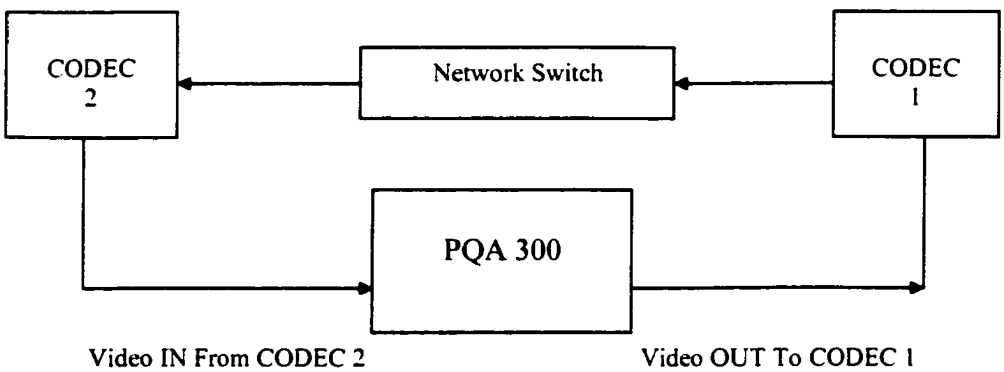

Figure 5 – Configuration "B": CODEC Back-To-Back Via a Network Switch

Test Configuration "C"

A two-subnet test network was implemented with test configuration "C" to facilitate the introduction of IP packet impairments. This configuration consisted of two network switches separated by the impairment emulation system, which was installed on an NT workstation and configured to function as a router, via two onboard Ethernet Network Interface Card (NIC) ports.

The NIC ports were assigned IP addresses that would be "seen" as default gateways by the two CODECs, thereby routing the packets back and forth between the two subnetworks. The CODEC IP addressing schemes were modified to reflect the two-subnet environment and the specific CODEC configurations were changed to assign the impairment workstation NIC ports as the default gateway addresses. The PQA-300 video quality analysis system configuration remained the same as in the two previous test configurations. Figure 6 illustrates the two-subnet configuration.

Project Critical Path

The principles that guide effective project management within the business world are equally applicable to research initiatives, and require the identification of the critical project deliverables that guide task completion and milestone achievement. The following outline identifies the milestones and deliverables, which defined this study's critical path:

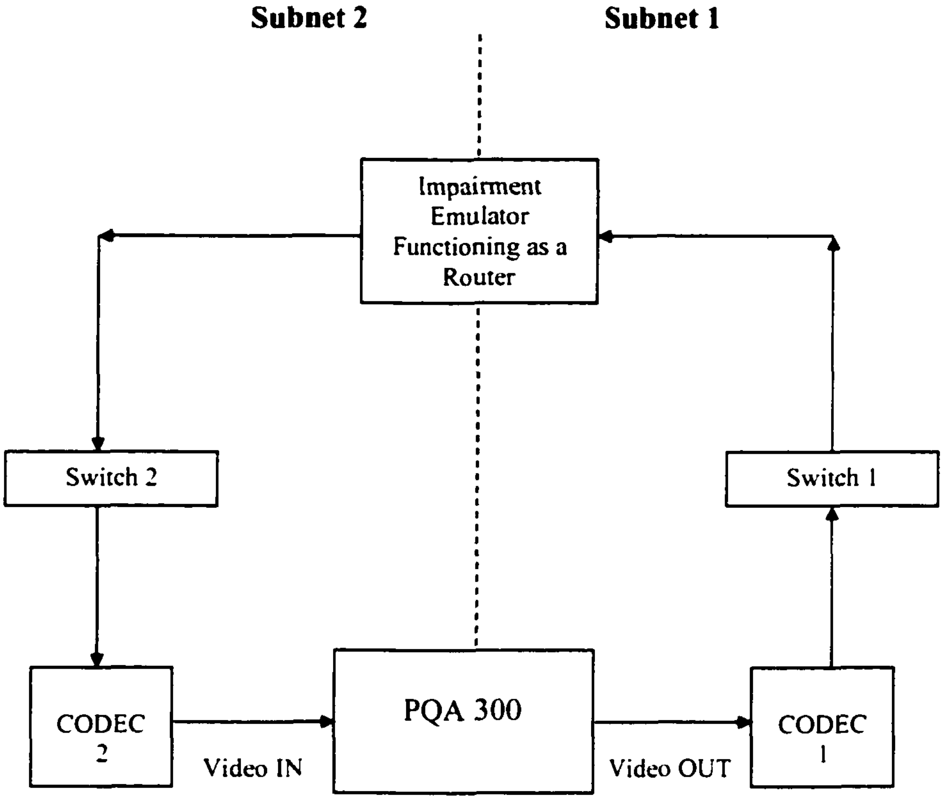

Figure 6 – Configuration "C": Two-Subnet Environment

1. Development/Refinement of the Research Topic

    - Identification of general area of research
    - Review of general topical information
    - Development/refinement of specific research objectives
    - Formation of appropriate research questions
    - Review of relevant topical literature
    - Development of subsequent research hypotheses
    - Selection of specific testing variables
    - Development of specific evaluation criteria
    - Identification of appropriate measurement metrics

2. Research Testing
    - Procurement and configuration of necessary test equipment
    - Configuration of network environment for video testing
    - Performance of baseline video testing
    - Establishment of baseline (non-impairment) PSNR values
    - Establishment of baseline (non-impairment) PQR values
    - Performance of impairment-specific video testing

- Acquisition of impairment PSNR values
- Acquisition of impairment PQR values

3. Analysis Of Test Data
   - Performance of comparative and multiple regression analysis of comprehensive test data
   - Synthesis and communication of test results within research documentation

Methodology of Analysis

Roscoe (1969) suggested that the variation in a given dependent variable is usually a function of a similar variation in a number of independent variables acting simultaneously. Although this statistical construct is normally applied to the study of human behavior, it can be appropriately developed within the context of information technology as in the applied research of video quality. Moreover, the techniques of multiple regression enable the researcher to use the acquired knowledge of multiple independent variables to predict their impacts on a single dependent variable with greater success than is possible with information pertaining to only one of the independent variables (Roscoe; Levin, 1987). The result is an estimating equation that more accurately describes the relationship between the dependent and independent variables.

Therefore, the method for statistically analyzing the video stream test data acquired within this study was the performance of multiple regression analyses of the dependent variable as a function of the independent variables, introduced within the transmission of video streams over an IP network. The multiple regression analyses determined the amount of change in the quality of the transmitted video signal (dependent variable), which could be explained by the combined congestive effects of the independent variables (packet loss, packet delay, jitter, and limited bandwidth and CODEC bit rate). Each of these variables were observed and measured for their contribution to the way in which the regression described the acquired test data.

In symbolic form, the multiple regression equation for this study was as follows:

$\hat{Y} = a + b_1X_1 + b_2X_2 + b_3X_3 + b_4X_4$

where:

- X_1, X_2, X_3, and X_4 = the values of the four independent variables (X_1 = packet delay (latency); X_2 = packet loss (drop); X_3 = jitter; X_4 = bandwidth limitations).
- $\hat{Y}$ = the estimated value corresponding to the dependent variable (video quality).
- a = the Y-intercept
- b_1, b_2, b_3, and b_4 = the slopes associated with X_1, X_2, X_3 and X_4, respectively.

The numerical constants a(Y-intercept) and $b_{1\text{-}4}$ (slopes of the multiple regression line) are also referred to as the *estimated regression coefficients*, such that the constant a is the value of Y if $X_{1\text{-}4}$ were to all equal zero. The coefficients $b_{1\text{-}4}$ describe how changes in $X_{1\text{-}4}$ affect the value of Y.

Multiple regression analyses were performed using the Microsoft Excel Data Analysis Tools for the PSNR and PQR values obtained from each of the test configurations, the results of which are presented and interpreted in RESULTS (chapter 4). The significance of the obtained PSNR and PQR values and the potential relationship between the various network configurations and the applied impairments were evaluated by interpreting the Excel regression analysis summary output, with emphasis placed on the values of the following derived statistics:

- Multiple R – The square root of R^2 and a correlation coefficient that expresses the correlation between the dependent variable and the derived combination of the predictor (independent) variables.
- R^2 – The square of the correlation coefficient, which expresses the proportion of the variance in y associated with the variance in x.
- Adjusted R^2 – The adjusted correlation coefficient that takes into account the number of observations and the number of predictor (independent) variables. In multiple regression analysis, when the number of observations is small, relative to the number of predictor variables, the R^2 tends to be biased upward.
- P-values – Used for analysis with regard to a two-tailed test, in conjunction with the upper and lower limits of a 95% confidence interval around the intercept and around each coefficient. If the P-value is .05 or greater, the 95% confidence interval for that term would span zero. If not, it could be concluded with 95% confidence that the intercept and regression coefficients are non-zero, and therefore the predictor variables add information meaningfully to the regression equation. If the P-values for the t-tests of the independent variables are lower than .05, and the 95% confidence intervals do not span zero, it can be inferred that the independent variables are all significantly related to the dependent variable at the 95% confidence level.

Chapter Synopsis

The research design and video testing methodology discussed in this chapter included the use of nine different International Radio Consultative Committee (CCIR) standard video sequences of varying motion and scene characteristics, which were subjected to IP congestion and variations in network bandwidth and CODEC bit rate.

Peak Signal-to-Noise Ratio (PSNR) and Picture Quality Rating (PQR) metrics were the tools chosen to provide objective evaluations of the quality of each video sequence. The PSNR and PQR values that were recorded for each sequence reflect an objective measurement for the network environment at the time of transmission.

The data acquired by way of PSNR and PQR measurements were collated and the inferential statistics were applied in the form of multiple regression analyses to determine the existence and strength of the relationship between IP impairments, bandwidth and CODEC bit rate, and video sequence quality.

CHAPTER 4

RESULTS

As stated in previous chapter, the video quality metrics utilized in this study were Peak Signal to Noise Ratio (PSNR) and Picture Quality Rating (PQR). The PSNR value is a non-human vision system picture differencing measurement that is observed between a reference composite video signal and a captured test signal. As video quality increases, so too does the dB level of the PSNR measurement.

The PQR for each video sequence is the aggregate value of the field and scene PQR and correlates with observed fidelity ratings when viewed at a distance of four times the screen height. As opposed to PSNR, a *decrease* in the measured PQR level is indicative of an *increase* in video quality. According to Tektronix (n.d.), PQR values can be loosely interpreted using the following guidelines:

- A PQR rating of 1 indicates impairments that have a small perceptual impact.
- A PQR rating of at least 3 indicates impairments that may not be strong, but that are almost always observable.
- A PQR rating of at least 10 indicates clearly observable impairments.

For the purpose of review, three video transmission test configurations were utilized in this study each providing a unique test environment free of introduced impairments, from which quality baseline PQR and PSNR values could be established (see Figures 4, 5 & 6). Configurations "A" and "B" were limited to non-network designs, with the former comprised of the CODEC devices connected in a straight back-to-back fashion, and the latter configured in a similar fashion via a network switch. In both configurations, the CODEC bit rate was held to 1.5Mbps. Configuration "C" however, mirrored a typical IP test network infrastructure and included the implementation of two small subnetworks and a "router" workstation. Moreover, it was the only configuration utilized for measuring video quality based on the introduction of IP impairments into the transmission stream. The CODEC bit rates and available network bandwidth for configuration "C" were set at 3.5 Mbps/10 Mbps and 10Mbps/100Mbps, respectively.

Configuration "A" Test Results

The PQR and PSNR values measured for the nine video sequences transmitted without impairments across test configuration "A" ranged from 6.44 to 16.31 and from 21.47 to 34.99 respectively, as shown in Table 2. The PQR output for eight of the nine video sequences exceeded a value of 10, and varied in quality over a range of 5.05 based on each scene's level of motion and complexity. The moving colors sequence "Popple" with its pan and rotating motion yielded the poorest PQR quality measurement of 16.31, while the low-motion, low-complexity "Susie" (talking head) sequence measured an acceptable 6.44, as to be expected for a scene with minimal motion characteristics.

The measured PSNR values for all nine sequences in this configuration were dispersed over a range of 13.52 decibels, and the "Susie" sequence returned the highest objective quality measurement of just less than 35 dB. "Mobile", a slow moving, random motion of objects sequence was measured at 21.47 dB.

Table 2 – CODEC Bit Rate = 1.5Mbps with CODECs Configured Back-to-Back

Sequence	Impairment	PQR	PSNR
Cheer	None	14.89	22.77
Ferris	None	11.55	25.83
Flower	None	11.47	23.67
Football1	None	12.30	26.16
Kiel	None	14.82	22.71
Mobile	None	14.58	21.47
Popple	None	16.31	22.70
Susie	None	6.44	34.99
Tempete	None	11.26	24.69

Across nine video sequences, the average PQR value was 12.63 and PSNR averaged 25.00 dB.

Configuration "B" Test Results

Table 3 provides the acquired video quality measurements based upon the impairment-free environment of configuration "B". Eight of nine sequences returned measurements within one unit of the previous configuration's PQR and PSNR output values (indicating a nominal change in video quality). The values measured for the sequence "Flower", a slow panning landscape scene with many color details, created outliers that were unexpected (a 7.5 unit increase in PQR and a 6.58 dB decrease in PSNR). The nine-sequence averages for PQR and PSNR were 13.44 and 24.24, respectively.

The cause of these results is unclear, however, the effects of the sequence motion, scene characteristics, or the introduction of a network switch into the video sequence transmission path may have had an impact. As with the results documented for configuration "A", the optimal PQR and PSNR values for configuration "B" were also measured from the "talking head" sequence "Susie".

Table 3 - CODEC Bit Rate = 1.5Mbps/CODECs Back-to-Back Via Network Switch

Sequence	Impairment	PQR	PSNR
Cheer	None	14.68	22.90
Ferris	None	11.56	25.85
Flower	None	18.97	17.09
Football1	None	12.32	26.00
Kiel	None	14.74	22.77
Mobile	None	14.57	21.38
Popple	None	16.35	22.63
Susie	None	6.47	34.69
Tempete	None	11.28	24.82

Configuration "C" Test Results

The non-impaired sequences transmitted via configuration "C" with CODEC bit rates set at 3.5 Mbps and network bandwidth set at 10Mbps returned PQR values ranging from 6.44 to 13.79 and PSNR measurements from 20.60 dB to 34.05 dB, which are presented in Table 4. Again, the "talking head" sequence provided the best objective scoring of PQR and PSNR values; however, the poorest quality measurement for both metrics resulted from the "Mobile" sequence, which combines random motion of objects and slow movement. Average PQR for non-impaired sequences in this particular combination of CODEC bit rate and network bandwidth were 10.20 and average PSNR measurements were 25.87.

In addition to non-impaired transmission testing, PQR and PSNR values were captured from the impaired transmission of the nine video sequences using the 3.5Mbps/10Mbps and 10Mbps/100Mbps configurations.

The results of former are also found in Table 4. Packet-specific impairments were applied bi-directionally and consisted of the following:

- Periodic Drops – A periodic distribution impairment that dropped 1 out of every "X" packets.
- Packet Delay (Latency) – Packets delayed using a "Guassian" distribution, which applied a normal distribution between the minimum and maximum delay values creating the bi-directional effects of latency. In a Guassian distribution, the minimum value corresponds to a standard deviation of -3 and the maximum value corresponds to a standard deviation of +3, causing the packets to automatically flow across the network out of sequence.
- Packet Jitter - A packet impairment that applied a normal distribution between the minimum and maximum delay values, causing the packets to automatically flow out of sequence.

Applying impairments to the video stream with CODEC bit rates set at 3.5Mbps and utilizing 10Mbps of available network bandwidth resulted in PQR and PSNR measurements well outside of the baseline values captured as a result of the non-impaired transmissions. Of the impairments introduced, the random application of 1 to 10 milliseconds (ms) of jitter to each video sequence created the highest indication of PQR and PSNR degradation as shown in Table 4.

Table 4 - CODEC Bit Rate = 3.5Mbps/IP Network Bandwidth = 10Mbps

Sequence	Impairment	PQR	PSNR
Cheer	None	10.18	25.83
Ferris	None	8.46	28.03
Flower	None	10.27	23.84
Football1	None	7.84	28.39
Kiel	None	12.49	22.79
Mobile	None	13.79	20.60
Popple	None	12.56	23.90
Susie	None	6.44	34.05
Tempete	None	9.77	25.37
Cheer	Periodic Drop - 1 out of every 500 packets	10.59	20.70
Ferris	Periodic Drop - 1 out of every 500 packets	20.82	12.91
Flower	Periodic Drop - 1 out of every 500 packets	12.23	13.11
Football1	Periodic Drop - 1 out of every 500 packets	17.34	13.36
Kiel	Periodic Drop - 1 out of every 500 packets	20.12	12.60
Mobile	Periodic Drop - 1 out of every 500 packets	20.50	13.39
Popple	Periodic Drop - 1 out of every 500 packets	17.90	17.34
Susie	Periodic Drop - 1 out of every 500 packets	9.86	26.51
Tempete	Periodic Drop - 1 out of every 500 packets	16.11	15.94
Cheer	Delay – Gaussian 1ms to 10ms	33.81	11.95
Ferris	Delay – Gaussian 1ms to 10ms	33.49	11.40
Flower	Delay – Gaussian 1ms to 10ms	34.65	9.70
Football1	Delay – Gaussian 1ms to 10ms	30.97	12.80
Kiel	Delay – Gaussian 1ms to 10ms	34.92	11.23
Mobile	Delay – Gaussian 1ms to 10ms	38.50	10.67
Popple	Delay – Gaussian 1ms to 10ms	32.52	14.48
Susie	Delay – Gaussian 1ms to 10ms	23.47	17.93
Tempete	Delay – Gaussian 1ms to 10ms	32.01	13.07
Cheer	Jitter - 1ms to 10ms	31.26	11.71
Ferris	Jitter - 1ms to 10ms	34.76	11.70
Flower	Jitter - 1ms to 10ms	34.74	3.25
Football1	Jitter - 1ms to 10ms	30.69	12.79
Kiel	Jitter - 1ms to 10ms	33.60	11.24
Mobile	Jitter - 1ms to 10ms	38.64	5.50
Popple	Jitter - 1ms to 10ms	31.13	8.39
Susie	Jitter - 1ms to 10ms	23.43	17.92
Tempete	Jitter - 1ms to 10ms	32.10	12.94

Delaying packets using a Gaussian distribution of 1 to 10ms and periodically dropping 1 out of every 500 transmitted packets, although damaging to the quality of the transmitted sequences, yielded less degradation for both PQR and PSNR.

Table 5 contains the results of both non-impaired and impaired video sequences transmitted at the highest CODEC bit rates and maximum available network bandwidth levels (10Mbps/100Mbps). As shown, an increase in CODEC bit rate and network bandwidth had positive effects on the PQR and PSNR levels for sequences subjected to delay and jitter impairments, but not for those in which periodic packet drops were introduced. Moreover, a tangential examination of average PQR and PSNR values for all sequences within all configurations provided an indication of the change in objective quality based upon the CODEC bit rates, available network bandwidth, and particular impairment applied. The aggregate average values for both metrics are compiled in Table 5a.

Also of interest are the test sequences themselves and whether or not any one sequence performed more consistently than the others with respect to the PQR and PSNR values recorded throughout this study.

Table 5 - CODEC Bit Rate = 10Mbps/IP Network Bandwidth = 100Mbps

Sequence	Impairment	PQR	PSNR
Cheer	None	8.25	27.18
Ferris	None	7.47	28.85
Flower	None	9.58	24.43
Football1	None	5.52	29.34
Kiel	None	11.19	23.21
Mobile	None	13.01	20.98
Popple	None	10.42	24.70
Susie	None	6.23	34.21
Tempete	None	8.84	26.48
Cheer	Periodic Drop - 1 out of every 500 packets	24.34	12.55
Ferris	Periodic Drop - 1 out of every 500 packets	25.74	12.45
Flower	Periodic Drop - 1 out of every 500 packets	25.19	12.87
Football1	Periodic Drop - 1 out of every 500 packets	20.47	13.79
Kiel	Periodic Drop - 1 out of every 500 packets	25.60	12.27
Mobile	Periodic Drop - 1 out of every 500 packets	25.79	12.55
Popple	Periodic Drop - 1 out of every 500 packets	21.00	16.29
Susie	Periodic Drop - 1 out of every 500 packets	16.93	20.92
Tempete	Periodic Drop - 1 out of every 500 packets	18.78	15.49
Cheer	Delay – Gaussian 1ms to 10ms	11.32	13.17
Ferris	Delay – Gaussian 1ms to 10ms	21.62	13.07
Flower	Delay – Gaussian 1ms to 10ms	19.22	13.36
Football1	Delay – Gaussian 1ms to 10ms	19.10	13.93
Kiel	Delay – Gaussian 1ms to 10ms	25.52	12.27
Mobile	Delay – Gaussian 1ms to 10ms	20.16	13.72
Popple	Delay – Gaussian 1ms to 10ms	28.34	14.93
Susie	Delay – Gaussian 1ms to 10ms	7.44	28.23
Tempete	Delay – Gaussian 1ms to 10ms	11.08	19.83
Cheer	Jitter - 1ms to 10ms	25.50	11.78
Ferris	Jitter - 1ms to 10ms	19.36	13.41
Flower	Jitter - 1ms to 10ms	15.86	13.74
Football1	Jitter - 1ms to 10ms	20.85	13.64
Kiel	Jitter - 1ms to 10ms	27.61	12.58
Mobile	Jitter - 1ms to 10ms	27.34	11.97
Popple	Jitter - 1ms to 10ms	25.85	17.00
Susie	Jitter - 1ms to 10ms	6.96	29.80
Tempete	Jitter - 1ms to 10ms	17.59	15.08

Table 5a – Average PQR and PSNR Values For All Sequences

Configuration	CODEC Bit Rate	Network Bandwidth	Impairment Applied	Avg PQR	Avg PSNR
A	1.5Mbps	N/A	None	12.63	25.00
B	1.5Mbps	N/A	None	13.44	24.24
C	3.5Mbps	10Mbps	Periodic Drop - 1 out of every 500 packets	16.16	16.21
C	3.5Mbps	10Mbps	Delay - Gaussian 1ms to 10ms	32.70	12.58
C	3.5Mbps	10Mbps	Jitter - 1ms to 10ms	32.26	10.60
C	10Mbps	100Mbps	Periodic Drop - 1 out of every 500 packets	22.65	14.35
C	10Mbps	100Mbps	Delay - Gaussian 1ms to 10ms	18.20	15.83
C	10Mbps	100Mbps	Jitter - 1ms to 10ms	20.77	15.44

Table 5b lists the sequences that resulted in the lowest and highest PQR and PSNR values for each of the separate quality tests, and validates the consistently positive performance of the "talking head" sequence, which was also noted previously in sections 4.1, 4.2, and 4.3 of this chapter. This was not unexpected because of the minimal motion and unremarkable scene characteristics of the "Susie" sequence. The results presented are irrespective of CODEC bit rate, test configuration or available network bandwidth.

Table 5b – Video Sequence Performance Based on Testing Metric

Quality Test #	Low PQR Sequence	High PQR Sequence	Low PSNR Sequence	High PSNR Sequence	Impairments Applied
1	Susie	Popple	Mobile	Susie	N
2	Susie	Flower	Flower	Susie	N
3	Susie	Mobile	Mobile	Susie	N
4	Susie	Ferris	Kiel	Susie	Y
5	Susie	Mobile	Flower	Susie	Y
6	Susie	Mobile	Mobile	Susie	Y
7	Susie	Mobile	Mobile	Susie	N
8	Susie	Mobile	Kiel	Susie	Y
9	Susie	Popple	Kiel	Susie	Y
10	Susie	Kiel	Cheer	Susie	Y

Regression Analysis of Data

To prepare the acquired test data for multiple regression analysis, each set of results was grouped together primarily by testing configuration (CODEC bit rate/network bandwidth) and secondarily by specific testing metric (PSNR/PQR) such that the appropriate regression statistics could be derived. This dichotomy created the following six regression matrices (three for PSNR and three for PQR) from which dependent and independent variables were assigned:

- PSNR - CODEC bit rate 3.5Mbps/Network bandwidth 10Mbps
- PSNR - CODEC bit rate 10Mbps/Network bandwidth 100Mbps
- PSNR - All configurations without impairments
- PQR - CODEC bit rate 3.5Mbps/Network bandwidth 10Mbps
- PQR - CODEC bit rate 10Mbps/Network bandwidth 100Mbps
- PQR - All configurations without impairments

Four of the regression matrices utilized the baseline PSNR and PQR values recorded for non-impaired sequences as the dependent variable Y, and the PSNR and PQR values from sequences subjected to packet drop, delay, and jitter as the predictor or independent variables, labeled X1, X2 and X3. The remaining two matrices were comprised based upon the non-impaired results from all configurations. That is, the PSNR and PQR values recorded from non-impaired sequences with the highest CODEC bit rate and network bandwidth levels (10Mbps/100Mbps) represented the dependent variable Y, and the results from the test utilizing the three non-impaired configurations described in Figures 4, 5, and 6 provided the independent variables X1, X2, and X3.

Tables 6 through 8 represent the PSNR regression matrices with the variables Y and X1 through X3 derived as a result of the transmission of the nine video sequences over the dual subnet configuration "C". The results were collected under varying CODEC bit rates and available network bandwidth, and the regression matrices for the PSNR and PQR metrics were developed as a way to test the validity of the research hypotheses annotated within the introduction to this study:

- H_0F: Increases in CODEC bit rates and network bandwidth will have no significant effect on the PSNR and PQR levels of transmitted video sequences that are devoid of packet loss, packet delay, or jitter.
- H_01: Dropping 1 out every 500 packets in a periodic distribution will have no significant effect on video stream PSNR and PQR levels.
- H_02: Delaying packets 1 – 10ms in a Gaussian distribution will have no significant effect on video stream PSNR and PQR levels.
- H_03: Random jitter of 1 to 10ms will have no significant effect on video stream PSNR and PQR levels.

PSNR Metric Regression Analysis

The following sections provide the derived calculations resulting from the multiple regression analysis performed on each of the six regression matrices. An interpretation of the results is provided, however an application of the interpreted results as it pertains to the overall study will be presented in the next chapter.

CODEC 3.5Mbps/Network 10Mbps With Impairments

Table 6 – PSNR Values for CODEC Bit Rate 3.5Mbps/Network Bandwidth 10Mbps

No Impairment	Periodic Drop (1 out of 500)	Delay (Gaussian Dist. 1 – 10ms)	Jitter (1 - 10ms)
Y	X1	X2	X3
25.83	20.70	11.95	11.71
28.03	12.91	11.40	11.70
23.84	13.11	9.70	3.25
28.39	13.36	12.80	12.79
22.79	12.60	11.23	11.24
20.60	13.39	10.67	5.50
23.90	17.34	14.48	8.39
34.05	26.51	17.93	17.92
25.37	15.94	13.07	12.94
Y = PSNR without impairments X1 = PSNR with Periodic Drop 1 out of 500 X2 = PSNR with Delay Gaussian 1ms to 10ms X3 = PSNR with Jitter 1ms to 10ms			

Table 6a provides the regression summary output for the matrix in Table 6. The Multiple R value of .84 would tend to indicate a high level of correlation between the PSNR values measured from the output of the non-impaired sequences and the derived combination of the impaired sequence PSNR outputs in the 3.5Mbps/10Mbps environment.

Table 6a – 3.5Mbps CODEC/10Mbps Network: Regression Analysis Summary

SUMMARY OUTPUT						
Regression Statistics			Y = PSNR without impairments			
Multiple R	0.842		X1 = PSNR with Periodic Drop 1 out of 500			
R Square	0.710		X2 = PSNR with Delay Gaussian 1ms to 10ms			
Adjusted R Square	0.536		X3 = PSNR with Jitter 1ms to 10ms			
Standard Error	2.677					
Observations	9					
ANOVA						
	df	*SS*	*MS*	*F*	Significance F	
Regression	3	87.605	29.202	4.074	0.082	
Residual	5	35.836	7.167			
Total	8	123.441				
	Coefficients	*Standard Error*	*T Stat*	*P-value*	*Lower 95%*	*Upper 95%*
Intercept	14.515	5.556	2.612	0.048	0.233	28.798
X1	0.147	0.362	0.406	0.701	-0.784	1.078
X2	0.282	0.832	0.339	0.748	-1.856	2.420
X3	0.511	0.333	1.535	0.185	-0.345	1.366

The proportion of variance in the non-impaired PSNR outputs that can be associated with the variance in the PSNR output of the impaired sequences appears to be significant as well, as expressed by the R Square value of .71. However, this may be the result of an upward bias due to the smaller number of observations. The Adjusted R Square value of .54 indicates the potential for a lower level of significance in proportional variance, had there been more than nine observations performed.

The multivariate Analysis of Variance (ANOVA) portion of Table 6a splits the sum of squares value into its components and provides an overall F-test of H0: X1, X2, and X3 = 0, versus H1: at least one does not equal 0. The column labeled "F" has an associated P-value (labeled "Significant F"), and since the value .08 is greater than .05, H0 is not rejected.

The *t* statistic values in Table 6a represent the ratio of each regression coefficient to its standard error (i.e. the estimated standard deviation), and as presented do not indicate statistical significance. The P-values associated with the independent variables provide an analysis of the two-tailed test and are each greater than .05.

Since the upper and lower limits of the 95% confidence interval span zero, it can be concluded with 95% confidence that the PSNR output of the impaired sequences do not add information meaningful to the regression equation and are therefore not significantly related to the PSNR outputs of the non-impaired sequences.

CODEC 10Mbps/Network 100Mbps With Impairments

The regression summary output in Table 7a is similar to that of Table 6a, even though the output PSNR results were based on an increase in CODEC bit rate and available network bandwidth from 3.5Mbps/10Mbps to 10Mbps/100Mbps as shown in Table 7.

The Multiple R value of .74 is indicative of a high degree of correlation between the dependent and independent variables, but the values associated with R Square (.55) and Adjusted R Square (.28) indicate a weak proportion of variance in Y that may be associated with variances in the X1, X2, or X3, even if additional observations had been available.

Table 7 – PSNR Values for CODEC Bit Rate 10Mbps/Network Bandwidth 100Mbps

No Impairment	Periodic Drop (1 out of 500)	Delay (Gaussian Dist. 1 - 10ms)	Jitter (1 – 10ms)
Y	X1	X2	X3
27.18	12.55	13.17	11.78
28.85	12.45	13.07	13.41
24.43	12.87	13.36	13.74
29.34	13.79	13.93	13.64
23.21	12.27	12.27	12.58
20.98	12.55	13.72	11.97
24.70	16.29	14.93	17.00
34.21	20.92	28.23	29.80
26.48	15.49	19.83	15.08

Y = PSNR without impairments

X1 = PSNR with Periodic Drop 1 out of 500
X2 = PSNR with Delay Gaussian 1ms to 10ms
X3 = PSNR with Jitter 1ms to 10ms

The ANOVA F-test validates that none of the independent variables are equal to zero, and the associated P-value of .23 is statistically insignificant.

As was also observed in the previous regression summary, the 95% confidence interval values of each of the regression coefficients span zero. Since the associated *t* stat P-values exceed .05, the independent variables (PSNR values of impaired sequences) are not significantly related to the dependent variable (PSNR values of non-impaired sequences) at a 95% confidence level.

Table 7a – 10 Mbps CODEC/100 Mbps Network: Regression Analysis Summary

SUMMARY OUTPUT

Regression Statistics	
Multiple R	0.740
R Square	0.548
Adjusted R Square	0.277
Standard Error	3.324
Observations	9

Y = PSNR without impairments
X1 = PSNR with Periodic Drop 1 out of 500
X2 = PSNR with Delay Gaussian 1ms to 10ms
X3 = PSNR with Jitter 1ms to 10ms

ANOVA

	df	*SS*	*MS*	*F*	*Significance F*
Regression	3	66.984	22.328	2.021	0.230
Residual	5	55.245	11.049		
Total	8	122.228			

	Coefficients	*Standard Error*	*T Stat*	*P-value*	*Lower 95%*	*Upper 95%*
Intercept	23.101	10.959	2.108	0.089	-5.070	51.271
X1	-0.723	1.621	-0.446	0.674	-4.890	3.444
X2	0.239	0.695	0.343	0.745	-1.548	2.025
X3	0.654	0.787	0.831	0.444	-1.369	2.677

CODEC 10Mbps/Network 100Mbps Without Impairments

The PSNR values contained in Table 8 provide the basis for a regression analysis of non-impaired sequences, and the subsequent summary in Table 8a provides statistical insight into how increases in bit rate and bandwidth might affect output PSNR levels.

The regression statistics presented in Table 8a are significantly more remarkable than that of Tables 6a and 7a. The Multiple R value of .99 indicates an almost perfect, positive correlation between the dependent variable and the derived combination of the predictor or independent variables, and the equally impressive values for the R Square and Adjusted R Square validate the high proportion of variance in Y that can be associated with a variance in X1, X2, and X3.

Table 8 – PSNR Values for All Configurations Without Impairments

No Impairment 10Mbps/100Mbps	CODEC Back-to-Back	CODEC Back-to-Back with Switch	No Impairment 3.5Mbps/10Mbps
Y	X1	X2	X3
27.18	22.77	22.90	25.83
28.85	25.83	25.85	28.03
24.43	23.67	17.09	23.84
29.34	26.16	26.00	28.39
23.21	22.71	22.77	22.79
20.98	21.47	21.38	20.60
24.70	22.70	22.63	23.90
34.21	34.99	34.69	34.05
26.48	24.69	24.82	25.37

Y = PSNR without impairments (10/100)
X1 = PSNR without impairments (CODECs Back-to-Back)
X2 = PSNR without impairments (CODECs Back-to-Back via Switch)
X3 = PSNR without impairments (3.5/10)

The ANOVA F-test values confirm that one or more of independent variables are equal to zero, and the associated P-test value of 0.0 is statistically significant.

The *t* statistic values in Table 8a are significant in that they show a low ratio of each regression coefficient to its estimated standard deviation. In addition, for two of three two-tailed tests the upper and lower limits of the 95% confidence interval span zero and are therefore statistically significant.

It can be concluded with 95% confidence that the PSNR output of the non-impaired sequences within the 10Mbps/100Mbps environment do indeed add information meaningful to the regression equation, and are significantly related to the PSNR outputs of the non-impaired sequences within the other configuration environments.

Table 8a – All Configurations Without Impairments: Regression Analysis Summary

SUMMARY OUTPUT

Regression Statistics		
		Y = PSNR without impairments (10/100)
Multiple R	0.999	X1 = PSNR without impairments (CODECs Back-to-Back)
R Square	0.998	X2 = PSNR without impairments (CODECs Back-to-Back via Switch)
Adjusted R Square	0.997	X3 = PSNR without impairments (3.5/10)
Standard Error	0.226	
Observations	9	

ANOVA

	df	*SS*	*MS*	*F*	*Significance F*	
Regression	3	121.973	40.658	796.065	0.000	
Residual	5	0.255	0.051			
Total	8	122.228				
	Coefficients	*Standard Error*	*T Stat*	*P-value*	*Lower 95%*	*Upper 95%*
Intercept	0.978	0.544	1.798	0.132	-0.420	2.376
X1	-0.239	0.062	-3.830	0.012	-0.400	-0.079
X2	0.004	0.038	0.105	0.920	-0.093	0.101
X3	1.218	0.060	20.136	0.000	1.063	1.374

PQR Metric Regression Analysis

The values that comprise the PQR regression matrices are presented in Tables 9 through 11. As was the case with the PSNR metric, the derived values representing the variables Y and X1 through X3 are the result of the transmission of the nine video sequences over the dual subnet configuration "C" under varying CODEC bit rates and available network bandwidth.

CODEC 3.5Mbps/Network 10Mbps With Impairments

The summary output for the PQR results within the 3.5Mbps/10Mbps environment provided in Table 9a is similar to that of the respective PSNR output, in that a high degree of correlation appears to exist between the PQR outputs of non-impaired sequences (dependent variable) and the PQR levels resulting from the transmission of impaired sequences.

In this case, the difference between the values derived for R Square and Adjusted R is nominal enough to accept R Square as a valid expression of the high proportion of the variance in Y associated with the variance in X1, X2, and X3.

The ANOVA "F" gives the overall F-test of H0: X1, X2, and X3 = 0, versus the alternative that at least one of the independent variables does not equal zero. The associated P-value is equal to .05 and at the 95% confidence level H0 is not rejected.

Table 9 – PQR Values for CODEC Bit Rate 3.5Mbps/Network Bandwidth 10Mbps

No Impairment	Periodic Drop (1 out of 500)	Delay (Gaussian Dist. 1 – 10ms)	Jitter (1 - 10ms)
Y	X1	X2	X3
10.18	10.59	33.81	31.26
8.46	20.82	33.49	34.76
10.27	12.23	34.65	34.74
7.84	17.34	30.97	30.69
12.49	20.12	34.92	33.60
13.79	20.50	38.50	38.64
12.56	17.90	32.52	31.13
6.44	9.86	23.47	23.43
9.77	16.11	32.01	32.10

Y = PQR without impairments

X1 = PQR with Periodic Drop 1 out of 500
X2 = PQR with Delay Gaussian 1ms to 10ms
X3 = PQR with Jitter 1ms to 10ms

All values within the 95% confidence interval span zero, and none of their respective P-values are greater than .05.

Table 9a – 3.5Mbps CODEC/10Mbps Network: Regression Analysis Summary

SUMMARY OUTPUT

Regression Statistics		
Multiple R	0.871	Y = PQR without impairments
R Square	0.758	X1 = PQR with Periodic Drop 1 out of 500
Adjusted R Square	0.614	X2 = PQR with Delay Gaussian 1ms to 10ms
Standard Error	1.501	X3 = PQR with Jitter 1ms to 10ms
Observations	9	

ANOVA

	df	*SS*	*MS*	*F*	*Significance F*
Regression	3	35.364	11.788	5.233	0.053
Residual	5	11.262	2.252		
Total	8	46.627			

	Coefficients	*Standard Error*	*T Stat*	*P-value*	*Lower 95%*	*Upper 95%*
Intercept	-4.683	4.334	-1.081	0.329	-15.825	6.458
X1	0.150	0.169	0.887	0.415	-0.284	0.584
X2	1.167	0.499	2.338	0.067	-0.116	2.450
X3	-0.797	0.530	-1.504	0.193	-2.159	0.565

This indicates with 95% confidence that the predictor variables do not add meaningful information to the regression equation and verifies that the independent variables are not significantly related to the dependent variable. In terms of PQR values, the results associated with the impaired sequences are not good predictors of the non-impaired PQR values.

CODEC 10Mbps/Network 100Mbps With Impairments

The PQR regression matrix for the 10Mbps/100Mbps with impairments is presented in Table 10, and as with the previous PQR summary output, the derived regression summary output in Table 10a also has similarities to its respective PSNR counterpart.

Table 10 – PQR Values for CODEC Bit Rate 10Mbps/Network Bandwidth 100Mbps

No Impairment	Periodic Drop (1 out of 500)	Delay (Gaussian Dist. 1 - 10ms)	Jitter (1 - 10ms)
Y	X1	X2	X3
8.25	24.34	11.32	25.50
7.47	25.74	21.62	19.36
9.58	25.19	19.22	15.86
5.52	20.47	19.10	20.85
11.19	25.60	25.52	27.61
13.01	25.79	20.16	27.34
10.42	21.00	28.34	25.85
6.23	16.93	7.44	6.96
8.84	18.78	11.08	17.59

Y = PQR without impairments

X1 = PQR with Periodic Drop 1 out of 500

X2 = PQR with Delay Gaussian 1ms to 10ms

X3 = PQR with Jitter 1ms to 10ms

The regression value Multiple R indicates only a modest correlation between the dependent variable and the derived combination of independent variables, and the unremarkable R Square and Multiple R statistics provides no evidence that variances in Y can be attributed to any variances in X1, X2, or X3.The ANOVA statistics in Table 10a indicate that at least one of the predictor variables does not equal zero, and is validated with 95% confidence as evidenced by the F-test's associated P-value of .35.

In addition, all values within the 95% confidence interval span zero and none of the *t* statistic P-values is less than .05, so it can be inferred that the independent variables are not significantly related to the dependent variable and are not accurate predictors of same at the 95% confidence level.

Table 10a – 10Mbps CODEC/100Mbps Network: Regression Analysis Summary

SUMMARY OUTPUT						
Regression Statistics		Y = PQR without impairments				
Multiple R	0.674	X1 = PQR with Periodic Drop 1 out of 500				
R Square	0.454	X2 = PQR with Delay Gaussian 1ms to 10ms				
Adjusted R Square	0.127	X3 = PQR with Jitter 1ms to 10ms				
Standard Error	2.238					
Observations	9					
ANOVA						
	df	*SS*	*MS*	*F*	*Significance F*	
Regression	3	20.863	6.954	1.388	0.348	
Residual	5	25.051	5.010			
Total	8	45.915				
	Coefficients	*Standard Error*	*t Stat*	*P-value*	*Lower 95%*	*Upper 95%*
Intercept	1.397	5.443	0.257	0.808	-12.596	15.389
X1	0.167	0.304	0.549	0.607	-0.615	0.949
X2	0.034	0.152	0.220	0.835	-0.358	0.425
X3	0.152	0.171	0.889	0.415	-0.287	0.591

CODEC 10Mbps/Network 100Mbps Without Impairments

Table 11 provides the values upon which a regression analysis of the output PQR levels of non-impaired sequences is performed, and as was the case in Table 8a, the subsequent summary output report in Table 11a provides statistical insight into how increases in bit rate and bandwidth might affect output PQR levels.

The regression statistics found in Table 11a are significant with respect to the correlation coefficient and coefficient of determination. Mirroring the results found in Table 8a for PSNR, the .99 value of Multiple R indicates an almost perfect positive correlation between the dependent and independent variables.

Table 11 – PQR for All Configurations Without Impairments

No Impairment 10Mbps/100Mbps	CODEC Back-to-Back	CODEC Back-to-Back with Switch	No Impairment 3.5Mbps/10Mbps
Y	X1	X2	X3
8.25	14.89	14.68	10.18
7.47	11.55	11.56	8.46
9.58	11.47	18.97	10.27
5.52	12.30	12.32	7.84
11.19	14.82	14.74	12.49
13.01	14.58	14.57	13.79
10.42	16.31	16.35	12.56
6.23	6.44	6.47	6.44
8.84	11.26	11.28	9.77

Y = PQR without impairments (10/100)
X1 = PQR without impairments (CODECs Back-to-Back)
X2 = PQR without impairments (CODECs Back-to-Back via Switch)
X3 = PQR without impairments (3.5/10)

There is a nominal difference between R Square and Adjusted R Square indicating a high proportion of the variance in Y as being associated with variances in X1, X2, and X3. The ANOVA statistics reveal the highly significant P-value of 0.0.

Interpretation of the regression coefficients indicates that no values within the 95% confidence interval span zero, and therefore H0: X1, X2, and X3 = 0 is rejected. Coefficients X1 and X3 have associated P-values that are less than .05, which implies that they are significantly related to the dependent variable at the 95% confidence level.

Table 11a – All Configurations Without Impairments: Regression Analysis Summary

SUMMARY OUTPUT

Regression Statistics		Y = PQR without impairments (10/100)
Multiple R	0.995	X1 = PQR without impairments (CODECs Back-to-Back)
R Square	0.990	X2 = PQR without impairments (CODECs Back-to-Back via Switch)
Adjusted R Square	0.984	X3 = PQR without impairments (3.5/10)
Standard Error	0.305	
Observations	9	

ANOVA

	df	*SS*	*MS*	*F*	*Significance F*
Regression	3	45.450	15.150	162.872	0.000
Residual	5	0.465	0.093		
Total	8	45.915			

	Coefficients	*Standard Error*	*T Stat*	*P-value*	*Lower 95%*	*Upper 95%*
Intercept	0.189	0.496	0.380	0.720	-1.087	1.465
X1	-0.406	0.069	-5.864	0.002	-0.584	-0.228
X2	0.012	0.045	0.259	0.806	-0.104	0.127
X3	1.345	0.081	16.571	0.000	1.137	1.554

It should be noted that not all of the video quality tests performed throughout the course of this study were included in the final series of regression analyses discussed within this chapter. One hundred and thirty five additional tests were also conducted on the nine video sequences within the CODEC 3.5Mbps/Network 10Mbps configuration to introduce supplementary impairments into the video stream. Impairment distributions incorporating increased proportional periodic packet drops, conditional packet bursting parameters, fixed and uniform delay levels, and additional jitter values were applied and their outcomes documented, to create a more comprehensive record of baseline PQR and PSNR measurements.

Time and equipment limitations did not allow for the same set of additional tests to be performed using the configuration featuring the CODEC bit rate setting of 10Mbps with 100Mbps of available network bandwidth. The resulting comprehensive list of PQR and PSNR values for all tests performed within the 3.5Mbps/10Mbps configuration is found in Appendix A.

Furthermore, a set of graphical representations of the test results that were statistically analyzed and discussed in this appears in Appendix B. These graphs may provide a helpful visual context that can be applied to the testing metric results, further adding clarity to the subsequent analytical interpretations discussed herein.

Chapter Synopsis

The test results and statistical derivations reviewed within this chapter were evaluated within the context of the research hypotheses (one foundational and three impairment-specific) developed in the first chapter of this dissertation. Identification of specific multiple regression statistics, as well as Analysis of Variance (ANOVA) observations, and interpretations of regression coefficient values were also presented here for each testing metric (PSNR and PQR). In addition, six regression matrices developed for this study were discussed within the framework of the acquired PSNR and PQR values from each of the video quality tests.

The testing environment created for this study was described in detail to facilitate a clearer understanding of the specific experimental design utilized. The video quality testing methodology was also described in this chapter, as was the use of varying levels of bandwidth availability and CODEC bit rate settings, and the packet-specific impairments introduced into the transmitted video stream.

CHAPTER 5

CONCLUSIONS AND RECOMMENDATIONS

Over 460 specific video quality tests were conducted across the numerous hardware and network configurations as part of this study, and various levels of subjective quality degradation were observed when packet-specific impairments were applied to the nine video sequences. The levels of observed degradation varied according to the type of impairment applied, the amount of available network bandwidth being utilized, and the particular CODEC configuration under test. The quality degradation taxonomy included but was not limited to, the visible presence of artifacts, jittery and "frozen" frames, and delayed video output. However, the intended deliverable of this research was not the completion of any subjective or qualitative analysis. Rather, as stated in the first chapter, the goal was the measurement of video quality based on the *objective* outcomes provided by the statistically validated interpretation of PSNR and PQR values. It is from those metrics that the conclusions of this study were based.

Conclusion #1

The *foundational* hypothesis (H_0F) used within this study was developed for the purpose of making conclusions about the potential effects that increases in available network bandwidth and CODEC bit rates may have had on the objective quality of video sequences transmitted by way of various hardware and network configurations, without the presence of three packet-specific impairments:

H_0F: *Increases in CODEC bit rates and network bandwidth will have no significant effect on the PSNR and PQR levels of transmitted video sequences that are devoid of packet loss, packet delay, or jitter.*

The comprehensive regression analysis summary output values in Tables 8a and 11a indicated high levels of variable correlation and statistical significance, validating that changes in the independent variables were sufficient predictors of change in the dependent variable. Restated in terms of PSNR and PQR, it can be concluded that based upon the output results, 99 percent of the total variation in the PSNR and PQR output values of the impairment-free sequences that were transmitted over IP (utilizing the highest CODEC bit rate and available network bandwidth), can be explained by the three independent variables, i.e. the combined outputs of the measurement metrics for lower levels of CODEC bit rate and available network bandwidth.

The derived values reflected within the aforementioned regression statistics for both PSNR and PQR also specified a very high level of proportional variance, and any potential upward bias was effectively disputed by the fact that the Adjusted R Square statistics for each metric nearly approached "1". The ANOVA statistics in Tables 8a and 11a are equally significant with the F-test P-values for both metrics calculated as "0".

Regression coefficients X1 and X3 (CODECs back-to-back and CODEC 3.5Mbps/network 10Mbps) for both metrics yielded *t* test P-values that were calculated at less than .05, which indicated statistical significance at the 95% confidence level. However, the X2 coefficients (CODECs back-to-back via network switch) for PSNR and PQR indicated statistical insignificance (0.920 and 0.806 respectively), and would have been excluded if a subsequent regression analysis of X1 and X3 were performed. Therefore, based on these results the null hypothesis H_0F is rejected.

Conclusion #2

Hypotheses H_01, H_02, and H_03 addressed the potential impact of three types of packet impairments on the PSNR and PQR measurements for the CODEC/network bandwidth configurations identified in the third chapter, and are stated as follows:

> H_01: *Dropping 1 out every 500 packets in a periodic distribution will have no significant effect on video stream PSNR and PQR levels.*
>
> H_02: *Delaying packets 1 – 10ms in a Gaussian distribution will have no significant effect on video stream PSNR and PQR levels.*
>
> H_03: *Random jitter of 1 to 10ms will have no significant effect on video stream PSNR and PQR levels.*

Based on the derived regression statistics and output summary values for the PSNR and PQR metrics found in Tables 6a, 7a, 9a, and 10a, the three impairment-specific null hypotheses are not rejected.

The associated regression statistics for the documented PSNR and PQR values indicated deceptively high R Square values, which did not necessarily imply strong goodness-of-fit measurements for either metric. As previously stated, this was due to the tendency of R Square to over estimate the strength of the association when there was more than one independent variable under analysis. Further validation of weak goodness-of-fit was evidenced by the lower Adjusted R Square values of 0.536 and 0.277 for PSNR and 0.614 and 0.127 for PQR.

The regression coefficients X1, X2, and X3 in Tables 6a, 7a, 9a, and 10a each returned standard errors, *t*-statistics, and P-values indicating statistical insignificance at the 95% confidence level. Moreover, the overall test of significance of the regression parameters found within the computed ANOVA values indicated that the F-test results were insignificant and the associated P-values for both metrics were not less than 0.05.[1] It was concluded that the PQR and PSNR values for both CODEC/network bandwidth configurations were jointly statistically insignificant at the 95% significance level.

Recommendations for Future Research

Objective video quality data has been gathered within the course of this study by utilizing the Peak Signal-to-Noise Ratio (PSNR) and Picture Quality Rating (PQR) measurement metrics. This effort was complimentary to the previous research initiatives that used either the PSNR or the PQR measurements as primary determinants of video quality. The concurrent use of both metrics within a single study as was demonstrated here, was notable however, and provided a contextual dichotomy that may add value to those future research initiatives predicated upon the measurement of objective video quality.

The results of this study may also provide ensuing researchers with supplementary insight into the extent of video stream degradation brought about by IP-specific impairments, as well as validating specific assumptions regarding the mitigating effects of increased network bandwidth and/or modified CODEC bit rates. Data of this

[1] Although the P-value of 0.053 returned in Table 9a was within .003 of being considered statistically significant, the apparent "insignificance" of the other derived regression statistics negated the validity of that value.

nature could also be applied to more expansive and sophisticated transmission environments such as Wide Area Networks (WANs), Metropolitan Area Networks (MANs), and fixed wireless networks.

By completing this study within the confines of a controlled laboratory environment, and utilizing one-way streaming video samples and artificially-introduced packet impairments, this project established a viable baseline of video quality values that can be further applied to research initiatives incorporating real-time, interactive (two-way) video transmissions, such as those found in telemedicine and medical informatics applications.

It is also recommended that future studies incorporate and apply the testing metrics (PQR and PSNR) and methodologies used in this project, and add the components of encryption and decryption as possible quality-degrading factors. The increasing incidences of global turmoil and international conflict, and our own homeland security concerns mandate the incorporation of cryptography, encryption, and other communications security components in future research initiatives whose goal is to improve our country's security posture.

Implications For Technology Management

Video, voice, and other multimedia applications that were once only considered within the framework of future strategic initiatives are now common components of many organization's information and data transfer mechanisms, and technology managers must somehow find efficient and effective ways to incorporate them into the existing technology infrastructure.

In real-world situations, subjective evaluations borne of human perception are often in direct conflict with the objective data acquired by way of sophisticated video test equipment. However, this study has shown that with or without the existence of selected packet-specific impairments, increases in bandwidth and CODEC bit rates do in fact improve the objective quality of video over IP.

Subjective evaluations and interpretations are commonly used as methods of gauging end-user satisfaction as it pertains to video streamed over packet switching networks. But the objective results presented here will aide managers in a more practical way, by providing the statistical data necessary to justify increased capital expenditures and additional resource acquisitions that may be required to upgrade existing transmission media or replace outdated network hardware.

Software vendors do not necessarily consider the issues or impacts associated with achieving and maintaining high video quality within packet switching network environments, as they continue to develop multimedia applications that test the limits of transmission speed, system scalability, and network capacity. Therefore, in order to maintain the organization's viability and competitiveness, managers need access to objective information that will assist them in making decisions about their communications infrastructure and justify those decisions to senior executives and governance bodies.

The decision flowchart presented in Figure 7 is a tool designed especially for managers and other decision-makers who may lack specific technical knowledge or networking experience, but are nonetheless interested in understanding and applying the results of this study to their specific IP network environments and video transmission requirements. The flowchart tool takes a high-level approach to improving video quality and provides a number of specific remediation options that should be considered in the presence of packet specific network congestion.

Variations in CODEC bit rates and network bandwidth such as was demonstrated in this project, could be used in conjunction with the quality thresholds values and the packet impairment data presented previously. The flowchart tool also highlights the importance of recognizing video quality degradation as a by-product of other transmission issues such as high volumes of network traffic, and it emphasizes a continuous process of monitoring video stream outputs and making adjustments as required.

Completing this research endeavor has been both challenging and rewarding, and it is hoped that the information contained within adds practical, pragmatic content to an ever expanding body of knowledge regarding the transmission of video over IP networks. More importantly, it is hoped that the work reflected here is representative of the quality and passion of the many other scientists and researchers who are committed to understanding the relationships between network bandwidth, CODEC bit rates and objective video quality.

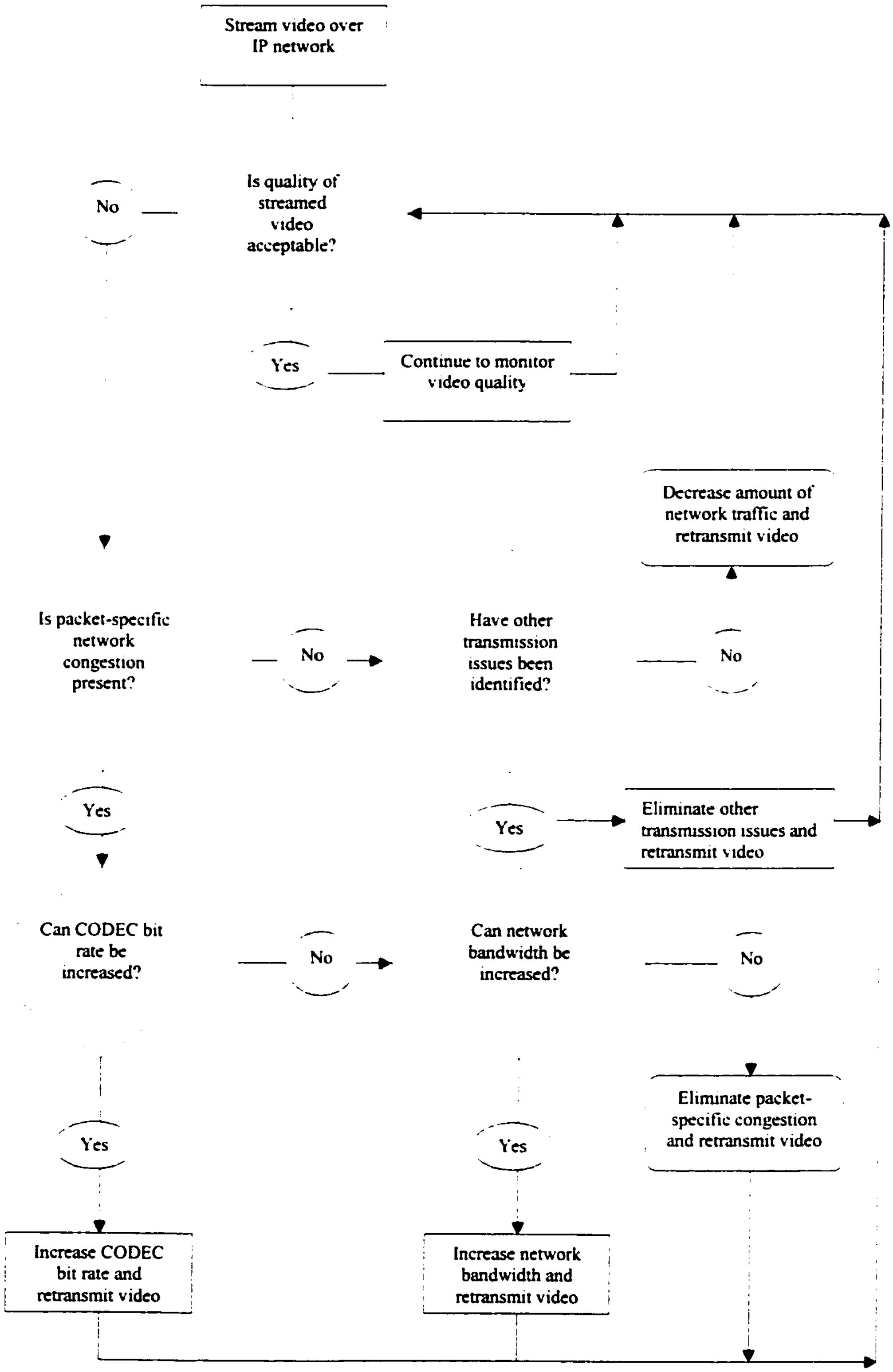

Figure 7 – Video Quality Improvement Decision Flowchart

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APPENDICES

A. Comprehensive Results List (3.5Mbps/10Mbps Configuration)

Sequence	Impairment	PQR	PSNR
Cheer	None	10.18	25.83
Ferris	None	8.46	28.03
Flower	None	10.27	23.84
Football1	None	7.84	28.39
Kiel	None	12.49	22.79
Mobile	None	13.79	20.60
Popple	None	12.56	23.90
Susie	None	6.44	34.05
Tempete	None	9.77	25.37
Cheer	Periodic Drop - 1 out of every 50 packets	30.01	12.39
Ferris	Periodic Drop - 1 out of every 50 packets	30.69	11.70
Flower	Periodic Drop - 1 out of every 50 packets	31.63	10.45
Football1	Periodic Drop - 1 out of every 50 packets	27.20	13.10
Kiel	Periodic Drop - 1 out of every 50 packets	30.07	11.58
Mobile	Periodic Drop - 1 out of every 50 packets	38.56	10.80
Popple	Periodic Drop - 1 out of every 50 packets	28.76	15.24
Susie	Periodic Drop - 1 out of every 50 packets	13.15	18.37
Tempete	Periodic Drop - 1 out of every 50 packets	25.64	15.47
Cheer	Periodic Drop - 1 out of every 100 packets	27.76	12.00
Ferris	Periodic Drop - 1 out of every 100 packets	28.56	12.18
Flower	Periodic Drop - 1 out of every 100 packets	23.77	10.92
Football1	Periodic Drop - 1 out of every 100 packets	28.72	12.43
Kiel	Periodic Drop - 1 out of every 100 packets	28.14	11.65
Mobile	Periodic Drop - 1 out of every 100 packets	28.51	11.34
Popple	Periodic Drop - 1 out of every 100 packets	26.79	16.06
Susie	Periodic Drop - 1 out of every 100 packets	13.45	24.56
Tempete	Periodic Drop - 1 out of every 100 packets	15.57	16.45
Cheer	Periodic Drop - 1 out of every 500 packets	10.59	20.70
Ferris	Periodic Drop - 1 out of every 500 packets	20.82	12.91
Flower	Periodic Drop - 1 out of every 500 packets	12.23	13.11
Football1	Periodic Drop - 1 out of every 500 packets	17.34	13.36
Kiel	Periodic Drop - 1 out of every 500 packets	20.12	12.60

Sequence	Impairment	PQR	PSNR
Mobile	Periodic Drop - 1 out of every 500 packets	20.50	13.39
Popple	Periodic Drop - 1 out of every 500 packets	17.90	17.34
Susie	Periodic Drop - 1 out of every 500 packets	9.86	26.51
Tempete	Periodic Drop - 1 out of every 500 packets	16.11	15.94
Cheer	Periodic Drop - 1 out of every 1000 packets	22.14	12.44
Ferris	Periodic Drop - 1 out of every 1000 packets	8.48	28.09
Flower	Periodic Drop - 1 out of every 1000 packets	10.78	16.72
Football1	Periodic Drop - 1 out of every 1000 packets	24.27	13.17
Kiel	Periodic Drop - 1 out of every 1000 packets	13.36	15.60
Mobile	Periodic Drop - 1 out of every 1000 packets	23.68	12.45
Popple	Periodic Drop - 1 out of every 1000 packets	13.08	19.36
Susie	Periodic Drop - 1 out of every 1000 packets	7.48	28.59
Tempete	Periodic Drop - 1 out of every 1000 packets	14.19	21.87
Cheer	Burst Drop - 1% (1 to 1)	25.58	12.42
Ferris	Burst Drop - 1% (1 to 1)	29.83	11.71
Flower	Burst Drop - 1% (1 to 1)	31.89	10.61
Football1	Burst Drop - 1% (1 to 1)	22.34	12.99
Kiel	Burst Drop - 1% (1 to 1)	27.45	11.86
Mobile	Burst Drop - 1% (1 to 1)	35.32	11.03
Popple	Burst Drop - 1% (1 to 1)	25.44	15.96
Susie	Burst Drop - 1% (1 to 1)	12.64	25.28
Tempete	Burst Drop - 1% (1 to 1)	22.36	15.76
Cheer	Burst Drop - 1% (1 to 5)	28.63	11.86
Ferris	Burst Drop - 1% (1 to 5)	30.33	11.73
Flower	Burst Drop - 1% (1 to 5)	30.05	10.60
Football1	Burst Drop - 1% (1 to 5)	25.00	13.24
Kiel	Burst Drop - 1% (1 to 5)	32.15	11.16
Mobile	Burst Drop - 1% (1 to 5)	36.69	11.12
Popple	Burst Drop - 1% (1 to 5)	25.60	16.33
Susie	Burst Drop - 1% (1 to 5)	7.62	25.04
Tempete	Burst Drop - 1% (1 to 5)	19.10	15.48
Cheer	Delay - Fixed 10ms	10.17	25.79
Ferris	Delay - Fixed 10ms	8.47	28.08
Flower	Delay - Fixed 10ms	10.30	23.93
Football1	Delay - Fixed 10ms	7.82	28.37
Kiel	Delay - Fixed 10ms	12.51	22.76
Mobile	Delay - Fixed 10ms	13.81	20.56
Popple	Delay - Fixed 10ms	12.57	23.87
Susie	Delay - Fixed 10ms	6.44	34.16
Tempete	Delay - Fixed 10ms	9.78	25.33

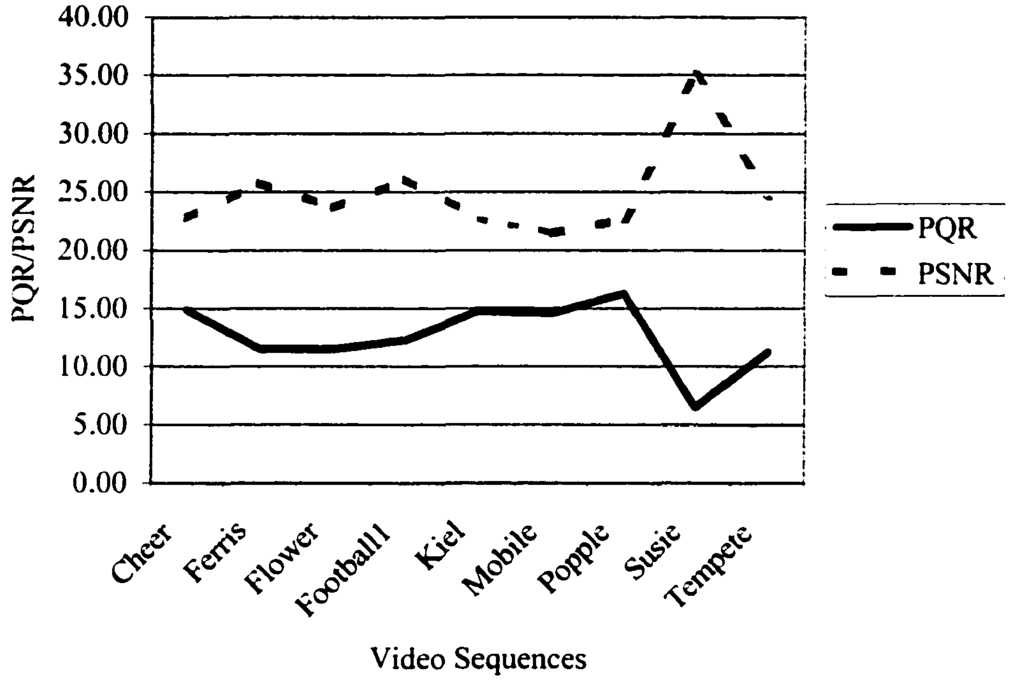
Sequence	Impairment	PQR	PSNR
Cheer	Delay - Fixed 500ms	10.24	25.85
Ferris	Delay - Fixed 500ms	8.47	28.12
Flower	Delay - Fixed 500ms	10.29	23.80
Football1	Delay - Fixed 500ms	7.83	28.41
Kiel	Delay - Fixed 500ms	12.47	22.70
Mobile	Delay - Fixed 500ms	13.81	20.55
Popple	Delay - Fixed 500ms	12.59	23.94
Susie	Delay - Fixed 500ms	6.44	34.09
Tempete	Delay - Fixed 500ms	9.78	25.41
Cheer	Delay - Fixed 9,999 ms	29.57	11.69
Ferris	Delay - Fixed 9,999 ms	29.37	12.30
Flower	Delay - Fixed 9,999 ms	29.86	12.10
Football1	Delay - Fixed 9,999 ms	29.82	12.63
Kiel	Delay - Fixed 9,999 ms	34.65	7.15
Mobile	Delay - Fixed 9,999 ms	38.48	10.95
Popple	Delay - Fixed 9,999 ms	32.55	14.39
Susie	Delay - Fixed 9,999 ms	23.49	17.91
Tempete	Delay - Fixed 9,999 ms	25.83	13.01
Cheer	Delay - Gaussian 1ms to 5ms	24.25	12.43
Ferris	Delay - Gaussian 1ms to 5ms	25.91	13.41
Flower	Delay - Gaussian 1ms to 5ms	14.71	13.43
Football1	Delay - Gaussian 1ms to 5ms	25.77	12.65
Kiel	Delay - Gaussian 1ms to 5ms	26.97	11.80
Mobile	Delay - Gaussian 1ms to 5ms	19.20	12.03
Popple	Delay - Gaussian 1ms to 5ms	26.76	16.42
Susie	Delay - Gaussian 1ms to 5ms	8.32	28.48
Tempete	Delay - Gaussian 1ms to 5ms	17.60	17.27
Cheer	Delay - Gaussian 1ms to 10ms	33.81	11.95
Ferris	Delay - Gaussian 1ms to 10ms	33.49	11.40
Flower	Delay - Gaussian 1ms to 10ms	34.65	9.70
Football1	Delay - Gaussian 1ms to 10ms	30.97	12.80
Kiel	Delay - Gaussian 1ms to 10ms	34.92	11.23
Mobile	Delay - Gaussian 1ms to 10ms	38.50	10.67
Popple	Delay - Gaussian 1ms to 10ms	32.52	14.48
Susie	Delay - Gaussian 1ms to 10ms	23.47	17.93
Tempete	Delay - Gaussian 1ms to 10ms	32.01	13.07

Sequence	Impairment	PQR	PSNR
Cheer	Delay - Uniform 1ms to 10ms (1ms)	10.21	25.79
Ferris	Delay - Uniform 1ms to 10ms (1ms)	12.09	16.36
Flower	Delay - Uniform 1ms to 10ms (1ms)	21.14	13.08
Football1	Delay - Uniform 1ms to 10ms (1ms)	7.83	28.39
Kiel	Delay - Uniform 1ms to 10ms (1ms)	12.49	22.79
Mobile	Delay - Uniform 1ms to 10ms (1ms)	25.98	14.20
Popple	Delay - Uniform 1ms to 10ms (1ms)	16.14	16.90
Susie	Delay - Uniform 1ms to 10ms (1ms)	6.44	34.13
Tempete	Delay - Uniform 1ms to 10ms (1ms)	9.77	25.44
Cheer	Jitter – 1ms (fixed)	10.20	25.85
Ferris	Jitter – 1ms (fixed)	11.02	18.59
Flower	Jitter – 1ms (fixed)	10.28	23.82
Football1	Jitter – 1ms (fixed)	26.87	6.04
Kiel	Jitter – 1ms (fixed)	31.90	5.83
Mobile	Jitter – 1ms (fixed)	34.29	9.48
Popple	Jitter – 1ms (fixed)	29.68	14.25
Susie	Jitter – 1ms (fixed)	19.90	13.12
Tempete	Jitter – 1ms (fixed)	27.12	13.11
Cheer	Jitter – 3ms (fixed)	10.18	25.87
Ferris	Jitter – 3ms (fixed)	8.47	27.94
Flower	Jitter – 3ms (fixed)	10.29	23.77
Football1	Jitter – 3ms (fixed)	7.82	28.34
Kiel	Jitter – 3ms (fixed)	12.46	22.73
Mobile	Jitter – 3ms (fixed)	13.80	20.56
Popple	Jitter – 3ms (fixed)	12.58	23.90
Susie	Jitter – 3ms (fixed)	6.48	34.23
Tempete	Jitter – 3ms (fixed)	9.78	25.46
Cheer	Jitter – 5ms (fixed)	10.20	25.81
Ferris	Jitter – 5ms (fixed)	23.50	16.42
Flower	Jitter – 5ms (fixed)	10.30	23.77
Football1	Jitter – 5ms (fixed)	7.83	28.36
Kiel	Jitter – 5ms (fixed)	12.47	22.73
Mobile	Jitter – 5ms (fixed)	13.82	20.55
Popple	Jitter – 5ms (fixed)	12.57	24.05
Susie	Jitter – 5ms (fixed)	6.47	34.16
Tempete	Jitter – 5ms (fixed)	9.77	25.43

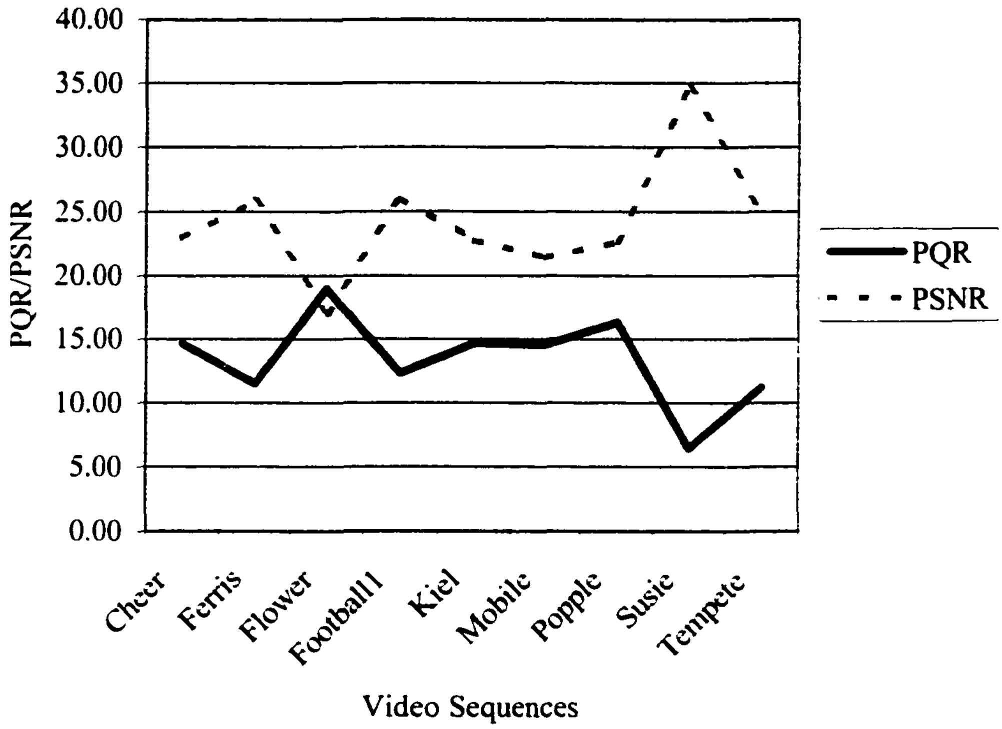
Sequence	Impairment	PQR	PSNR
Cheer	Jitter - 10ms (fixed)	10.20	25.85
Ferris	Jitter - 10ms (fixed)	8.46	27.99
Flower	Jitter - 10ms (fixed)	10.29	23.79
Football1	Jitter - 10ms (fixed)	7.81	28.36
Kiel	Jitter - 10ms (fixed)	12.49	22.78
Mobile	Jitter - 10ms (fixed)	13.83	20.60
Popple	Jitter - 10ms (fixed)	12.56	24.02
Susie	Jitter - 10ms (fixed)	6.47	34.27
Tempete	Jitter - 10ms (fixed)	9.77	25.35
Cheer	Jitter – 0ms to 1ms	34.46	6.06
Ferris	Jitter – 0ms to 1ms	30.23	10.14
Flower	Jitter – 0ms to 1ms	29.05	9.83
Football1	Jitter – 0ms to 1ms	29.88	5.96
Kiel	Jitter – 0ms to 1ms	31.00	6.06
Mobile	Jitter – 0ms to 1ms	35.50	4.97
Popple	Jitter – 0ms to 1ms	28.21	13.36
Susie	Jitter – 0ms to 1ms	21.65	7.73
Tempete	Jitter – 0ms to 1ms	19.58	9.65
Cheer	Jitter – 1ms to 10ms	31.26	11.71
Ferris	Jitter – 1ms to 10ms	34.76	11.70
Flower	Jitter – 1ms to 10ms	34.74	3.25
Football1	Jitter – 1ms to 10ms	30.69	12.79
Kiel	Jitter – 1ms to 10ms	33.60	11.24
Mobile	Jitter – 1ms to 10ms	38.64	5.50
Popple	Jitter – 1ms to 10ms	31.13	8.39
Susie	Jitter – 1ms to 10ms	23.43	17.92
Tempete	Jitter – 1ms to 10ms	32.10	12.94

B. Graphical Representation of PQR/PSNR Values

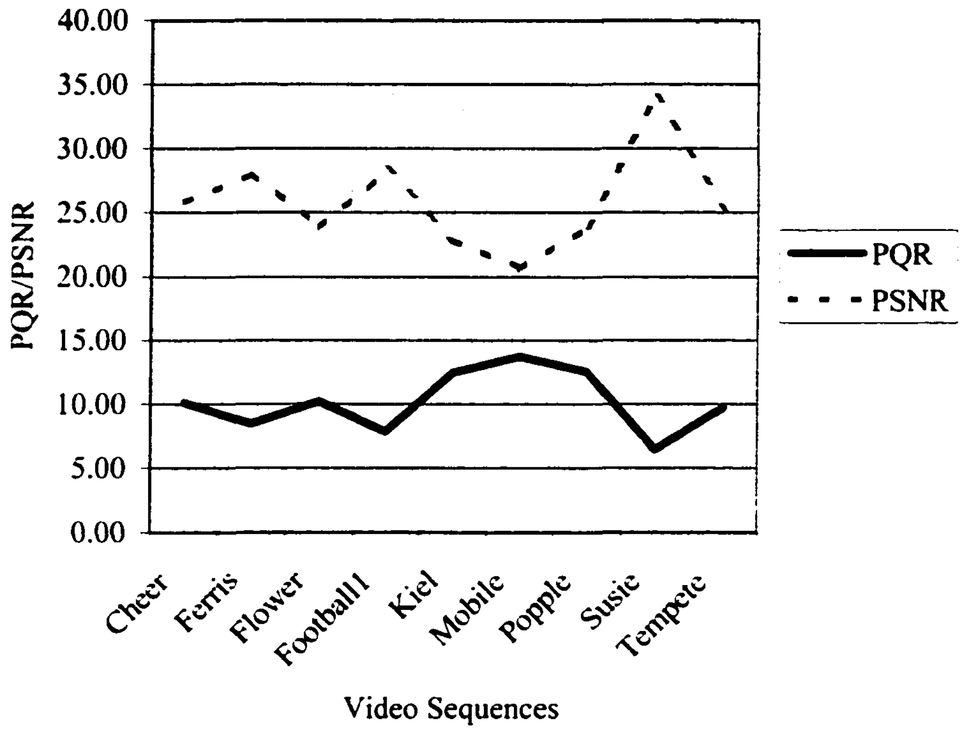
CODEC 1.5Mbps Back-to-Back Without Impairments



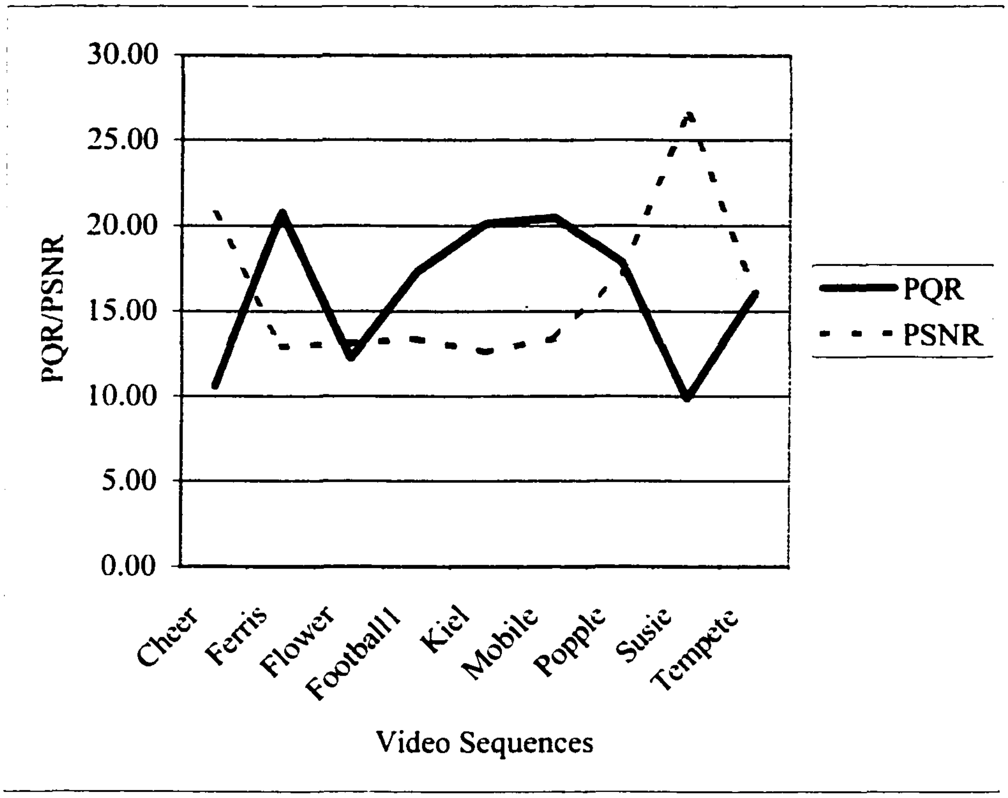
CODEC 1.5Mbps Back-to-Back Via Switch Without Impairments



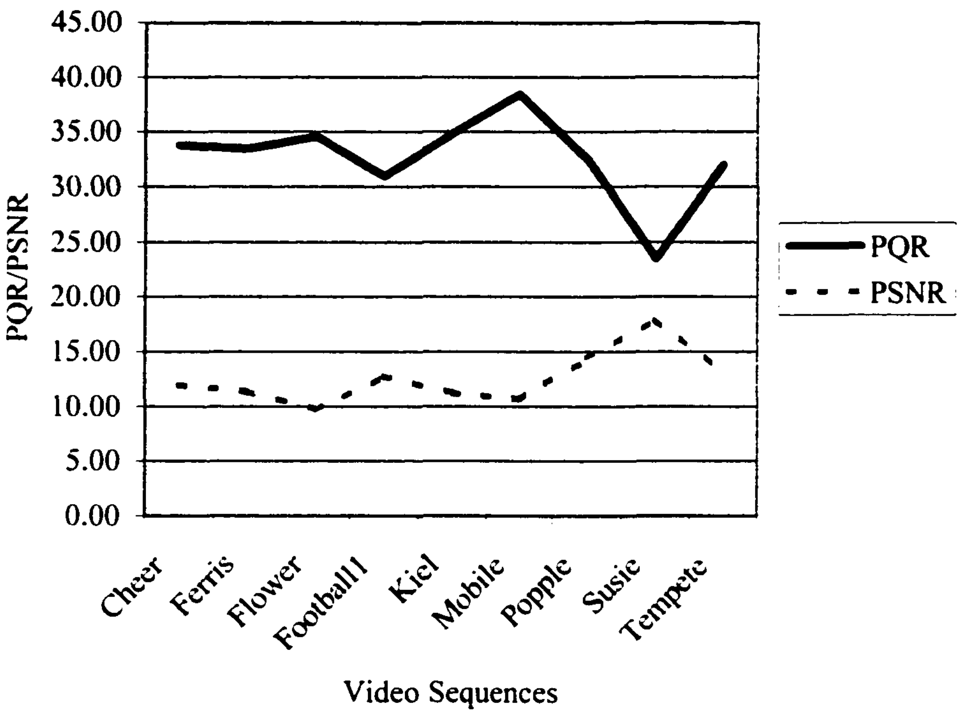
CODEC 3.5Mbps/Network 10Mbps Without Impairments



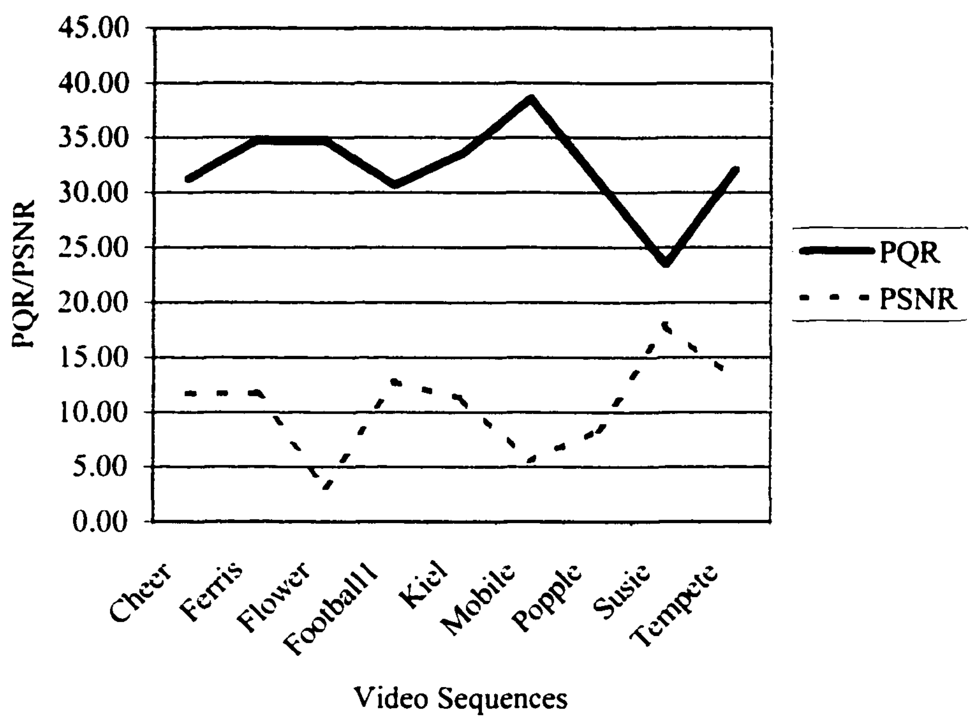
CODEC 3.5Mbps/Network 10Mbps Periodic Drop 1 – 500 Packets



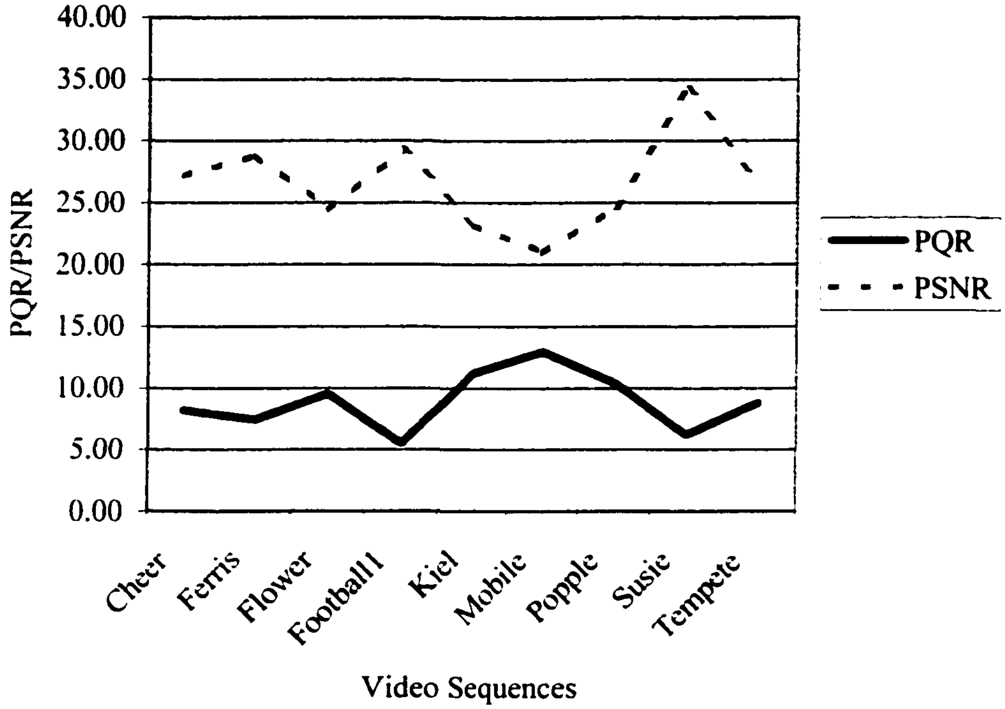
CODEC 3.5Mbps/Network 10Mbps Gaussian Delay 1ms – 10ms



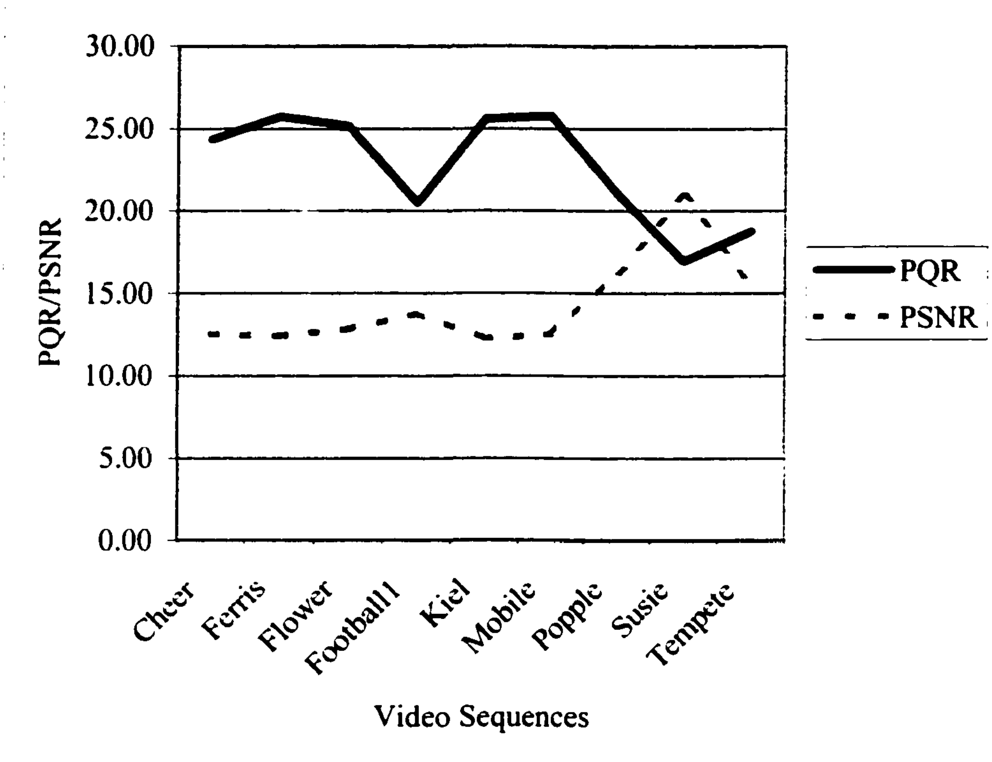
CODEC 3.5Mbps/Network 10Mbps Jitter 1m – 10ms



CODEC 10Mbps/Network 100Mbps Without Impairments



CODEC 10Mbps/Network 100Mbps Periodic Drop 1 – 500 Packets



CODEC 10Mbps/Network 100Mbps Gaussian Delay 1ms – 10ms