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Video Streaming: Challenges to Innovation

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Abstract

With the tremendous use of internet throughout the whole world, multimedia services provided are widely increased. And it will again increase in future. Transmission of Multimedia over the internet require the quality in all respect .But till wireless channels are more unreliable as compared to wired channels & bandwidth of these wireless channels changes with time. Due to different types of receivers, it is also difficult to obtain efficiency for video multicast. This paper discusses the sender-to-receiver Quality of Service provisioning for scalable video streaming traffic delivery over existing method. The Prototype architecture is proposed & validated which explores the joint use of packet prioritization and scalable video coding (SVC) together with the appropriate mapping of UMTS traffic classes to the Different traffic classes. We develop an analytical model to determine tradeoff's between network parameters using CSMA MAC protocol. On top of the analytical model, we use an optimization approach for determining the optimal number of APs in a cluster and the optimal separation distances between APs. Several approaches have been proposed in order to address the end-to-end QoS both from the network perspective, like UMTS QoS traffic classes, and from the application perspective, like scalable video coding and packetized prioritization mechanisms. This paper is to find the end-to-end QoS problem of scalable video streaming traffic delivery. We develop an frequency planning to improve the capacity of network and by eight unique video sequences and two scalable encoders, demonstrates the quality gains of scalable video coding(SVC).

Keywords:- SVC, QoS, BER, SAMM, JSCC.

1. INTRODUCTION

Real-time video transmission is largely achieved through applications that impose an end-to-end delay constraint on the video stream. Those real-time applications include conversational applications such as videoconferencing, distance learning, and video phony. Such applications usually have strict end-to end delay constraint, less than 200 milliseconds. Real-time applications may also include streaming applications. Those applications allow the start of video playback before the whole video stream has been transmitted with an initial setup time usually of a few seconds. All those applications require real-time playback. That is, once the playback starts, it must be continuous without interruption. Real-time video applications have gained increased popularity since the introduction of the first commercial products for Internet video streaming in 1995. As wireless networks are quickly becoming an important component of the modern communications infrastructure, Internet protocol (IP)-based architecture for the fourth-generation (4G) wireless systems[1] grows to be the provider of the next generation wireless services such as voice, high-speed data, Internet access, and multimedia streaming on all IP networks. The high bit rate support in the WIMAX standard makes it possible to transmit video in WIMAX [2]. All the above-mentioned cutting-edge developments, however, confront the high technical hurdles associated with high bit rate, quality of service (QoS), and real-time requirements of video applications. Quality of Service (QoS) means the capability of a network to provide better service to selected network traffic over various technologies. For technologies like Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these underlying technologies. The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics. Also important is to make sure that providing priority for one or more flows does not make other flows fail. QoS technologies provide the elemental building blocks that will be used for future business applications in campus, WAN, and service provider networks. This article outlines the features and benefits of the QoS provided by the Cisco IOS QoS[].

1.1 Challenges for Real-Time Video Transmission

Generally speaking, the main challenge to the real-time video communications is how to reliably transmit video packets over error-prone networks, where meeting the transmission deadline is complicated by the variability in throughput, delay, and packet loss in the network. In particular, a key problem of video transmission over the existing Internet and wireless networks is the incompatibility between the nature of the network conditions and the QoS requirements (such as those pertaining to bandwidth, delay, and packet loss) of multimedia applications. With a best-effort approach, the current IP network was originally designed for data transmission, having no guarantee of QoS for multimedia applications. Similarly, the current wireless networks were designed mainly for voice communication, which does not require as large bandwidth as video applications do. Different types of IP applications have different types of quality impairment under the same network conditions, and therefore call for different QoS. For example, applications such as web browsing, data file transfer, and electronic mail, are not sensitive to delay. However, for the deployment of multimedia applications with video stream, which is more sensitive to delay but more tolerant to packet loss, the lack of QoS guarantees in today's Internet and wireless networks introduces huge complications. Specifically, several technological challenges need to be addressed in designing a high-quality video transmission system. First, to achieve acceptable delivery quality, transmission of a real-time video stream typically has a minimum bandwidth requirement. However, the current Internet does not provide bandwidth reservation to meet the bandwidth requirement. At the same time, compared to wired links, wireless channels are much noisier due to fading, multi-path, and shadowing effects, which result in a much higher bit error rate (BER) and consequently an even lower throughput. Second, in the Internet, a packet can be lost due to congestion caused by buffer overflow and excessive delay. In wireless networks, a packet with unrecoverable bit errors is usually discarded at the link-layer according to the current standards. This difficulty is not as severe for traditional IP applications such as data transfer and email, where reliable transmission can always be achieved through retransmission, as it is for real-time video applications, where retransmission-based techniques may not be available due to the tight delay constraints. Third, the network resources are limited and may vary with time and space. Network resources include transmission bandwidth, buffers in the routers and switches, buffers at the sender or the receiver end, computation capability for encoding, decoding and transcoding, transmission cost in networks with pricing charge enabled, transmission power in wireless communications, delay, etc. Some constraints on resource, such as buffer size, computation speed, and display precision at the user end, but other constraints in that they aim to make the system stable or to treat other users in the communication system fairly. One example of a soft constraint at hand is the TCP-friendly protocol used to perform congestion control for media delivery applications, where the source bit rate is constrained so that all kinds of traffics can fairly share the network resources.

1.2 Existing System

In our existing system, proposed the Joint source-channel coding, Similar to error resilient source coding. In general, JSCC is accomplished by designing the quantize and entropy coder for given channel. For image and video signals, JSCC focuses on the optimal bit allocation between source coding and channel coding ,given channel loss characteristics. JSCC is studied for wavelet image transmission over the Internet, where the source and channel coding bits are allocated to minimize the expected end-to-end distortion. In JSCC for Internet scalable video is studied, where error resilient source coding and FEC are jointly considered[3]. The similar problem of scalable video transmission is also over bit-based channels. We take ARQ into account for non-scalable video. Real-time video applications, such as videoconferencing, video phony and on-demand video streaming, have gained increased popularity. However, a key problem of video transmission over the existing Internet and wireless networks is the incompatibility between the nature of the network conditions and the QoS (Qualify of Service) requirements (such as those in bandwidth, delay, and packet loss) of real-time video applications. Cross-layer design is a natural approach to deal with the incompatibility problem. This approach aims to efficiently perform cross-layer resource allocation (such as bandwidth, transmission energy, and buffers) by increasing the communication efficiency of multiple network layers.

Demerits:

1)Video quality reduced.

2)Video streaming time is increased.

1.3 Literature Survey

Optimal rate allocation for scalable video multicast over Wi-MAX, by Hsin-Yu Chi, Chia-Wen Lin and Yung-Chang Chen[4] has been proposed a new wireless broadband standard, which is capable of delivering very high data rate and covering wide area. The above said paper propose a Genetic Algorithm (GA) to reduce computational complexity when finding optimal modulation. The proposed method achieves lowest average distortion and the reduced computational complexity. According to the size of video layer, SS distribution, and available symbols; our method adaptively changes the modulations of each video layer in each group of picture (GOP) time. To ensure each receiver could receive all data of BL, we choose the minimum one of the modulation groups whose SS number is not zero in a multicast session. As a result, the PER of BL for receiver should be made zero. After allocating the available symbols to the data of BL, the remaining symbols are allocated to ELs according to the order of layer index. Furthermore, we assume the given number of available symbols must be larger than the symbols required for sending the BL data. In our GA-based algorithm, we use the Roulette wheel selection algorithm to perform fitness-proportional selection according to the fitness value of the chromosomes. The requirement is that the fittest individuals have a greater chance of survival than weaker ones. Besides, we use the one-cut-point crossover method to obtain offspring with a crossover rate. Based on the mutation rate, some portion of genes will be changed. To ensure that the high fitness chromosome is not lost from generation to generation, the best chromosome is copied into the new generation.

1.4 A Cross-Layer Design for Scalable Mobile Video

This paper presents SoftCast, a clean-slate design for mobile video[5]. This paper aims to improve the robustness and scalability of mobile video. The paper presents SoftCast, a cross-layer design of mobile video that both compresses the video and protect it from errors and losses. SoftCast adopts an integrated design for video and PHY layer coding, making the whole network stack act as a linear transform. The lack of scalability and robustness in today’s mobile video stems from the existing design of the network stack. Specifically, mobile video is impacted by two layers in the stack: the application video codec, which compresses the video, and the physical layer, which protects the video from channel errors and losses. This paper aims to improve the robustness and scalability of mobile video. SoftCast starts with a video that is represented as a sequence of numbers, with each number representing a pixel luminance. It then performs a sequence of transformations to obtain the final signal samples that are transmitted on the channel. The crucial property of SoftCast is that each transformation is linear[5]. This ensures that the signal samples transmitted on the channel are linearly related to the original pixel values. Therefore, increasing channel noise progressively perturbs the transmitted bits in proportion to their significance to the video application; high-quality channels perturb only the least significant bits while low-quality channels still preserve the most significant bits. Thus, each receiver decodes the received signal into a video whose quality is proportional to the quality of its specific instantaneous channel.

1.5 Layered Resource Allocation for Video Broadcasts over Wireless Networks

This paper aims to combine adaptive modulation and coding with layered video coding to improve the quality of video services to users experiencing different radio conditions[6]. This paper proposes an optimal radio resource allocation algorithm which maximizes a general performance metric for a video session in polynomial time. In this paper, optimal allocation algorithm runs in polynomial time, making it suitable for practical video services, and maximizes a general utility function. Also an alternative formulation of utility can be adapted to meet different circumstances. In this paper, a new layered resource allocation is proposed which fully utilizes the AMC and layered coding facilities in broadcast standards. With Adapting a generalized performance metric to accommodate various quality measures, we will describe an algorithm which maximizes the quality of each video session, while guaranteeing a maximum video quality to all users; and this algorithm runs in polynomial time under any given resource budget. We show that a system-wide optimal resource allocation can be obtained with a simple two-step decomposition of allocation that deals with each video session and then with the system as a whole. We also present a sub-optimal system-wide allocation algorithm of reduced computational complexity, which is useful when the distribution of users radio conditions changes frequently. This work extends existing broadcast and multicast services by the introduction of the layered resource allocation, which enables fine-grained control over coverage and video quality when many users are experiencing heterogeneous radio conditions. Our optimal allocation algorithm runs in polynomial time, making it suitable for practical video services, and maximizes a general utility function, so that alternative formulation of utility can be adapted to meet different circumstances.

1.6 Source-Adaptive Multilayered Multicast Algorithms for Real-Time Video Distribution

Multilayered encoding, however, is not sufficient to provide high video quality and high network utilization. Adaptive techniques capable of adjusting the rates of video layers are required to maximize video quality and network utilization. We propose a class of algorithms known as source-adaptive multilayered multicast (SAMM) algorithms. SAMM algorithms exhibit good performance in terms of good put, video quality and scalability while requiring only a minor amount of buffer allocation[7]. This paper also introduces a network architecture defining the source, receiver and network functions necessary to support SAMM algorithms. This architecture includes feedback mergers, which consolidate the contents of feedback packets as they return to the source. Feedback mergers are used to prevent feedback implosion, a scalability problem that occurs as an increasing number of receivers return congestion feedback to a single source. This architecture also requires that routers implement some form of priority packet discarding, to ensure that, in case of congestion, packets from less important video layers are discarded before packets from more important layers. In order to prevent SAMM video flows from negatively impacting the performance of other adaptive flows in the network, routers must also isolate flows based on service class. If this extraneous traffic shares first-in-first-out (FIFO) queues with competing traffic that is adaptive (e.g., TCP flows), then the adaptive flows may experience an unfair degree of discarding or delay within the network. Receiver-driven algorithms do not share this deficiency with sender-driven algorithms, because they send each layer of video in a different flow and allow for the pruning of flows that have no downstream receivers. One way to correct this deficiency of the sender-driven algorithms is to isolate video traffic from other traffic.

1.7 An adaptive modulation method for multicast communications of hierarchical data in wireless networks

This paper propose a new adaptive modulation method for multicast communications. The method separates data into hierarchical layers according to its importance, and transmits the layers of data in the same frame adaptively[8]. The number of slots for each physical layer mode is optimized, and a simple closed form solution is obtained. The relative size of each layer within a frame and the data rates the layers are sent are chosen to maximize the total sum of data rates for a given outage probability. The first criterion is the sum of data rates that are successfully detected at the receiver. The perceived quality of hierarchical multimedia is proportional to the number of successfully received layers by each user. Thus, the sum of data rates can be considered as the sum of perceived quality of the multicast service. The second criterion is the outage probability, which is the probability that a receiver cannot receive the base layer at a required BER. A low outage probability ensures a high availability, which is of particular interest to the service providers. It separates a set of data into several hierarchical layers according to their importance, and the transmitter sends the layers of data at different data rates, or modes, in each data frame. The base layer that contains the most important information is sent at the lowest data rate while the higher layers are sent at higher rates. The relative size of each layer within a frame and the data rates the layers are sent; are chosen to maximize the total sum of data rates for a given outage probability. We performed simulations for 802.11a wireless LAN, which show that the performance of the proposed method.

2. METHODOLOGY

2.1 Cross-Layer Resource Allocation

We can address the above challenges by enforcing error control, especially through unequal error protection (UEP) for video packets that are usually of different importance. Error control techniques, in general, include error resilient source coding, forward error correction (FEC), retransmission, power control, network QoS support, and error concealment. To maximize the error control efficiency, limited network resources should be optimally allocated to video packets which typically requires cross-layer design. The traditional layered protocol stack, where various protocol layers can only communicate with each other in a restricted manner, has proved to be inefficient and inflexible in adapting to the constantly changing network conditions. For the best end-to-end performance, multiple protocol layers should be jointly designed and should be able to react to the channel conditions in order to make the end-system network-adaptive. In addition, conventional video communication systems have focused on video compression, namely, rate-distortion optimized source coding, without considering other layers. While these algorithms can produce significant improvements in source-coding performance, they are inadequate for video communications over hostile channels. This is because Shannon's separation theorem, that source coding and channel coding can be separately designed without any loss of optimality, does not apply to general time-varying channels, or to systems with a complexity or delay constraint. Therefore, recent video coding research has been focused on the investigation of joint design of end-system source coding with manipulations in other layers, such as channel coding, power adaptation in wireless networks, and QoS support from the network (e.g., differentiated services networks and integrated services networks). One of the main characteristics of video is that different portions of the bit stream have different importance in their contribution to the quality of the reconstructed video. For example, in an MPEG video bit stream, I (Intra) frames are more important than P (Predictive) and B (Bi-directional predictive) frames. If the bit stream is partitioned into packets, Intra-coded packets are usually more important than Inter-coded packets. If error concealment is used, the packets that are hard to conceal are usually more important than easily concealable ones. In the scalable video bit sstream, the base layer is more important than the enhancement layer. Therefore, UEP is naturally preferable in video transmission. UEP commonly enables a prioritized protection for video packets through different levels of FEC and/or retransmission. UEP can also be realized through prioritized transmission by techniques based on transmitter power adaptation, channel/path diversity. Thus, in different network infrastructures, we consider different applicable error control components to support UEP. In particular, we jointly consider source coding and cross layer resource allocation to perform optimal UEP.

This Paper is about the end system design for real-time video applications. Such end system design consists of three major components: video codec, rate control or congestion control, and error control. This paper focuses on error control. More specifically, our focus is on the interaction between the video encoder and the underlying layers. Thus the decoder-based techniques, such as post-processing, joint source-channel decoding, and receiver-driven channel coding are not discussed in this dissertation. With respect to cross-layer design at the sender side, traditionally, there are two approaches to address the challenges described above for video communications. In the networking community, the approach is to develop protocols and mechanisms to adapt the network to the video applications. One example is to modify the mechanisms implemented in the routers/switches to provide QoS support to guarantee bandwidth, bounded delay, delay jitter, and packet loss for video applications. Another example is to use additional components such as an overlay network. The networking community approach is beyond the scope of this paper. In the video community, on the other hand, the approach is to adapt the end system to the network, which is what we employ in this paper. We focus on the end-system design for video communications based on the current, recently proposed, and emerging network protocols and architectures. We assume that the lower layers provide a set of given adaptation components; from the encoder's point of view, these components can be regarded as network resource allocation. Based on the assumption that our encoder can access and specify those resource allocation knobs, the major contribution of this paper is a proposed general resource-distortion optimization framework, which assigns network resources from multiple layers to each video packet according to its level of importance. Depending on different adaptation components, this framework is embodied in the forms of joint source-channel coding, joint source coding and rate adaptation, joint source-channel coding and power adaptation, joint source coding and packet classification etc. Within the framework, error resilient source coding, channel coding, and error concealment are jointly considered. In particular, we study the following problems using the proposed framework: joint source-channel coding (JSCC) for Internet video transmission, joint source-channel coding and power adaptation (JSCCPA) for wireless video transmission, and joint resource allocation and packet classification (JRAPC) for network video transmission. Besides single layer video source coding, we also consider scalable video source coding for further error resilience. In addressing these problems, we propose efficient algorithms for obtaining the optimal solutions. The simulation results, as expected, demonstrate the benefits of joint design of source coding and cross layer resource allocation. The proposed framework proves to be general and flexible. It allows for the comparison of different error control techniques (such as pure FEC, pure retransmission, and hybrid FEC and selective retransmission), and different packetization schemes. For example, in the JSCC work, FEC and application layer retransmission can each achieve an optimal result depending on the packet loss rates and roundtrip-time. But when the two are jointly employed in the proposed hybrid technique, improved results are obtained due to the increased exibility[9]. The framework is also applicable to the emerging network architectures, in which source coding can be jointly designed with packet classification. The proposed framework provides an optimization benchmark against which the performances of other sub7-optimal systems can be evaluated. It also provides a useful tool for assessing the effectiveness of different error control components in practical system design. For example, in the JRAPA study, the simulation results suggest that channel coding and power adaptation each has its effective working region. Thus, in a practical wireless video streaming system, under certain conditions, adjusting only channel coding or power control, but not both, might be adequate to achieve near-optimal results.

2.2 Proposed System

In proposed methodology to challenges in QoS provisioning for wireless video transport, we use an adaptive framework, which specifically addresses scalable video transport over wireless networks. This proposed method to presents a novel method for QoS provisioning via the use of the average reward reinforcement learning, which can maximize the network revenue subject to several predetermined QoS constraints. The proposed method to investigate the end-to-end QoS provisioning for scalable video streaming technique. Congestion management, queue management, link efficiency, and shaping/policing tools provide QoS within a single network element. Service levels refer to the actual end-to-end QoS capabilities, meaning the capability of a network to deliver service needed by specific network traffic from end to end or edge to edge. The services differ in their level of QoS strictness, which describes how tightly the service can be bound by specific bandwidth, delay, jitter, and loss characteristics.

Merits:

a)

he video quality to be increased and improve the frequency planning to rectify the collision issue for the video processing in the WMN.

b)

o improve the management of QoS.

Flow Diagram:

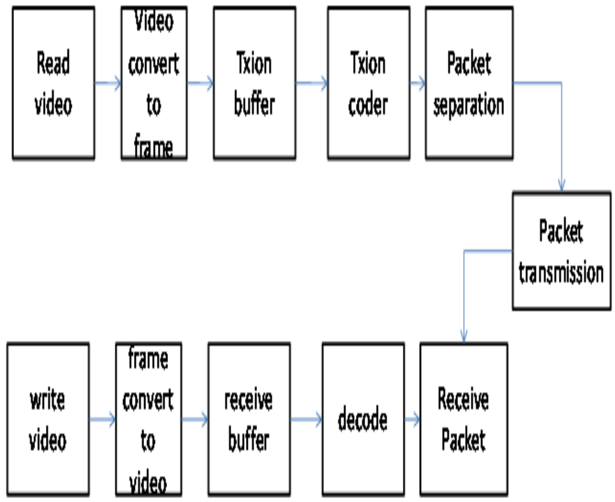

Figure 1. Flow Diagram

As shown in figure above, Read video block load the video files. Video files are converted into frames at video to frame conversion block. Conversion frame stored in buffer because to avoid data loss & frames convert into packet in transmission coder block. Packets are separated for data transmission. We Transmit the data with the help of WIMAX protocol. Then receiver receives the packets & Rearrange it. Received packets are again stored in buffer at receiver side. The packets are then converted into frames & the frames are converted into video. Finally write the video in receiver system. We used Matlab video processing to develop an application. Matlab provides its own functionality for video & image processing[10].

2.2.1 Proposed Modules:

Input

Video to frame conversion

Adaptive modulation/demodulation

Transmitter/receiver

Frame to video conversion

QOS constraints

Input

In this module, analysis of the input video format and file size is done. Some applications create collections of images related by time, such as frames in a movie, or by (spatial location, such as magnetic resonance imaging (MRI) slices). These collections of images are referred to by a variety of names, such as image sequences, image stacks, or videos. The toolbox represents image sequences as four-dimensional arrays, where each separate image is called a frame, all frames are the same size, and the frames are concatenated along the fourth dimension. mmread function will be used for loading and showing the input video seTquence.

Video to frame conversion

The loaded video sequence should be converted into frames (i.e. still images) using mmreader function. To read the videTo information, we are using the read function which reads the total information about the given video file. It convert multiple number of frames.

Adaptive modulation/demodulation

Group the incoming data bits into codeword's, one for each symbol that will be transmitted. Map the codeword's to attributes, for example amplitudes of the I and Q signals (the equivalent low pass signal), or frequency or phase values. Adapt pulse shaping or some other filtering to limit the bandwidth and form the spectrum of the equivalent low pass signal, typically using digital signal processing. Perform digital to analog conversion (DAC) of the I and Q signals (since today all of the above is normally achieved using digital signal processing, DSP).
Generate a high frequency sine carrier waveform, and perhaps also a cosine quadrature component. Carry out the modulation, for example by multiplying the sine and cosine waveform with the I and Q signals, resulting in the equivalent low pass signal being frequency shifted to the modulated pass band signal or RF signal.

Transmitter/receiver

This module contain the data transmitter and receiver blocks in wimax.
It required receiver antenna and transmitter antenna.
It transmit the data in analog waveform.
It required the IP address for communication process.

Frames to video conversion

The loaded image sequence/frames should be converted into video (i.e. still images) using video reader function.

To read the image information, we are using the read function which reads the total information about the given image file. It converts the multiple number of frames into video.

QOS constraints

Real-time video transmission typically has requirements on quality of service (QoS). However, wireless channels are unreliable and the channel bandwidth varies with time, which may cause severe degradation to video quality. In addition, for video multicast, the heterogeneity of receivers makes it difficult to achieve efficiency and exibility.

2.2.2 System Requirement

Hardware Requirement(Minimum):

- Pentium IV – 2.7 GHz
- 1GB DDR RAM
- 250Gb Hard Disk

Software Requirement:

- Operating System : Windows XP or win7
- Tool : Matlab Version : r2013a

3. RESULT

From the above proposed method, we will get a best quality video in following way:

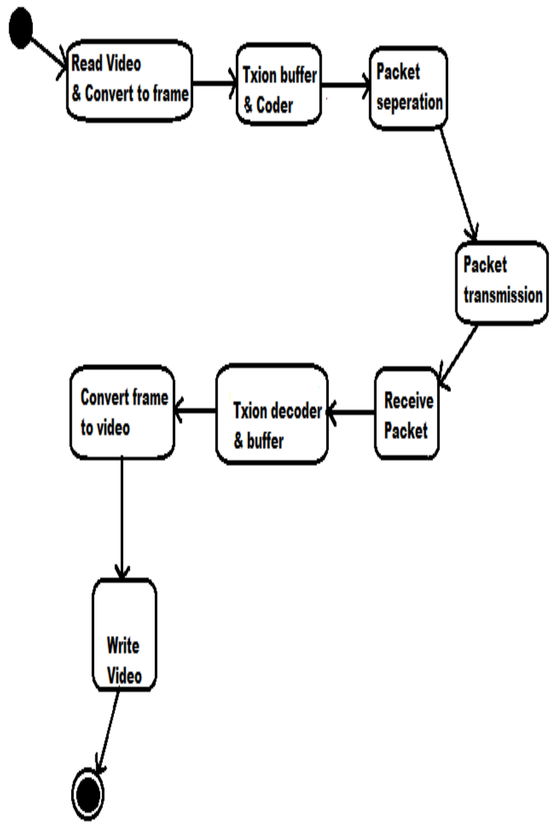

Figure 2. Activity Diagram

Input video is converted to frames & then stored in txion buffer in coded format. Coded frame packets are separated & transmitted to client. Client decodes received packets & store in txion buffer. The frames from packet are converted to video & write to client.

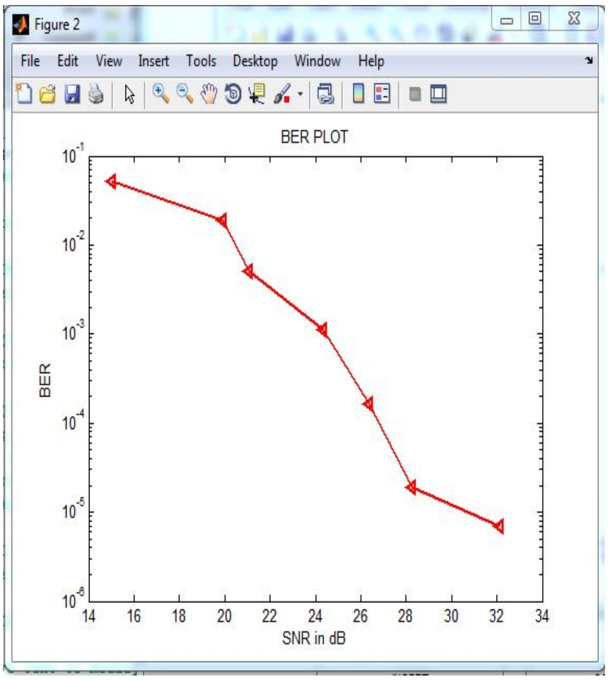

Figure 3. BER Plot of sample

BER plot shows BER vs SNR ratio of final video reached to destination in their respects.

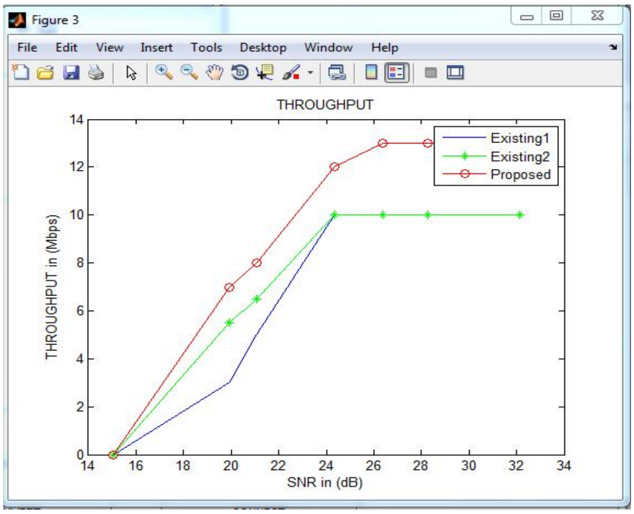

Figure 4. Throughput of sample

Throughput in Mbps with respect to SNR in db. It shows Noise rate present in received video & throughput. Here throughput must be high & noise ratio must be very low. This graph shows us our received video quality.

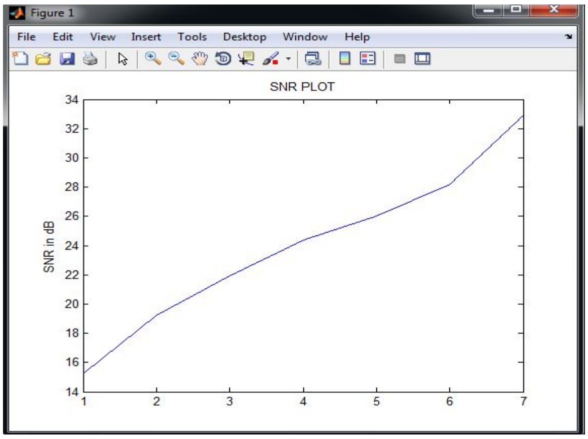

Figure 5. SNR Plot of sample

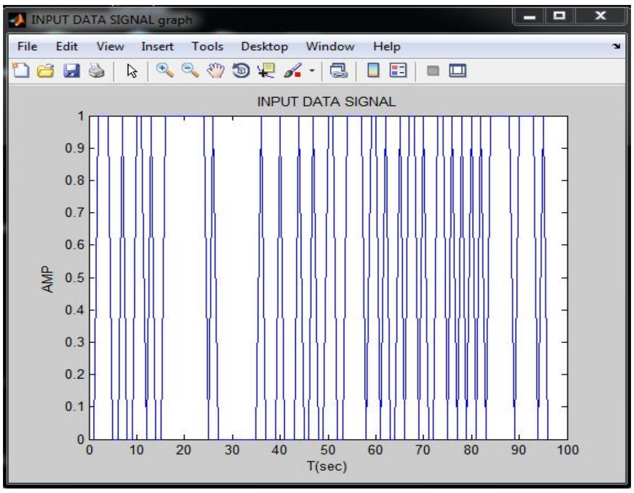

Figure 6. Input Signals of sample

This graph is display at sending side/server side. It shows input data signals with respect to time in seconds.

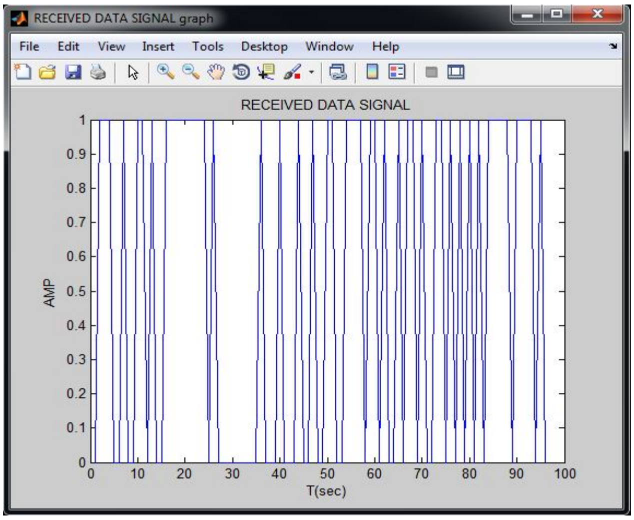

Figure 7. Received Signals of sample

Received signals are signals received at client for video transmission & this graph show signal quality in terms of AMP with respect to time in seconds. If received signals are as same as input/transmitted signals then we can say video is delivered/streamed in best quality. Difference in input signals & received signals is tolerable for real time video streaming. Here we have achieved our best quality of video.

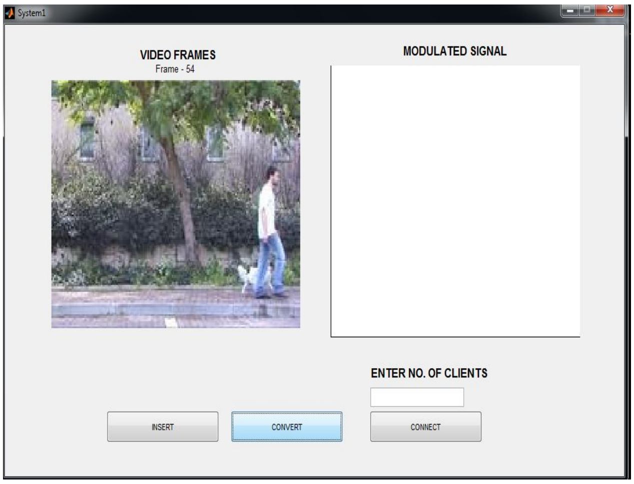

Figure 8. Server Side/Sending Side Window of sample

This is video sending window. Here video is loaded & converted to frames. The number of frames created are shown with the effect of Convert button. Then we have to connect no. of client to whom we want to connect.

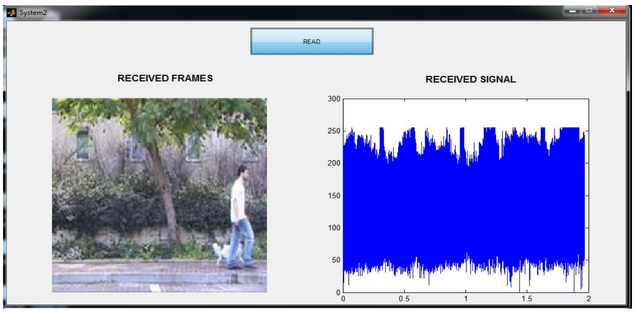

Figure 9. Client Side/User Side of sample

At client side, when we click on 'Read button' ,client receives frames from sender & convert it to video. This window also shows received signals. All the above graphs are changing as input video is changed & as per quality of video. For example, We have taken a .avi video of 4MB size & 3 seconds duration. When we have loaded it & converted then that video is converted in to 54 frames. The number of frames varies with the size of video. After a connection with client & client read the video; frames are transmitted & video display on screen. This transmission of frames & their conversion into video is happening in very fast speed i.e. in fraction of seconds. Also we have investigated different graph plots of network parameters. These network parameters/graph are different for different videos.

4. DISCUSION/ ANALYSIS

This technique is useful in live video streaming applications like video conferencing (i.e. skype application), to watch video on youtube etc. It can be used in live traffic monitoring system ,web cam chatting, CCTV for security. Like this, Now a days many more applications are used where live video streaming is required. In this applications, we can use this technique to improve video quality with performance of application. Future scope **: At Live Video Streaming:** This technique is also very useful where we want to make an artificial intelligent robot. As we know, In a future, we can use an automatic robots for many uses of human being. We can use them for war, we can use them to work in very hygienic conditions like as worker in dangerous mines, we can use them for automatically find a suspect & to kill them(i.e. terrorist) etc. So in these fields of work, we can add artificial intelligence technique in them but to find a target & identify it automatically ; we have to load the image/video from robots front view through the camera situated in robots eyes, scan that image/video automatically, identify fixed target & then shoot it. All this activity should be happen in a fraction of second. In this mission/activity of robot, video recording , video storage, live video streaming to server from where robot is going to operate or where members of that particular mission are locate, is first and very important task. So, this technique will be useful in such respect. It is an one example. This technique has many more real time uses.

5. CONCLUSION

In this paper, We have investigated various factors affecting live video streaming. These are external & internal noise, SNR, BER, signal behavior, frequency, quality etc. We have investigated challenges in real time video transmission & checked for its solutions like cross layer resource allocation, Layered Resource Allocation, Source-Adaptive Multilayered Multicast Algorithms, An adaptive modulation method for multicast communications. For the scalability with the QoS requirement, an optimization approach has been proposed to maximize the ratio of total capacity to total cost .We have developed a model with frequency planning & QoS constraints. We have used modulation /demodulation technique for signal transmission & video to frame ,frame to video conversion technique . The problem of video leakage in conferencing/streaming is removed with video compression. We got a fantastic result which is shown above with graphs ,as an example. We conclude that, this technique is very effective & also have future scope.

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