An Adaptive Video Streaming For The Best Possible Delivering Experience To Multiple Users In Wireless Networks

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ABSTRACT

Efficient support of video streaming in a wireless network established by varied clients permits the users in the network area to define adaptively the quality level of the desired video segments. When the user asks the server to send a packet, the server should begin streaming in a low resolution to avoid latency, packet loss... etc. The paper will propose an effective scheme to evaluate and adjust the performance needed under realistic assumptions of network parameters for each user. It has two parts; the first part applies adaptive video streaming to multiple clients through the Rayleigh fading channel and the issue faced on the Quality of Service (QoS) of video being displayed and the overall system was built to measure the performance for each user and concluded that the video was clear but never match with the High Definition (HD) input provided. The second part is the simulation; Adjusting the performance needed for each user by applying the error correction for the adaptive streaming using Reed Solomon algorithms. The results are enhanced to achieve the required QoS in addition to the video streaming adaptation. These results conclude the better results for parameters of video streaming system such as latency, bit rate through Rayleigh channel, delay factor of adaptive streaming, and efficiency.

1. Introduction

Recent mainstream video streaming key to utilize services of a Content Delivery Network (CDN) providers must discover novel multimedia streaming solutions to provide streaming services to meet the ever-increasing demands of the public. The users request access to high-quality video on any device and any location without having to wait and they could be more discouraging. To reach this level of service, CDN providers need solutions that scale easily because it is difficult to know when and where content will become "obsolete". They need solutions that intelligently adapt to dynamic network environments to offer their services in any format. They also need multimedia streaming solutions able to ignore delay, congestions, and loss of internet packets, real problems that will not however deny users to go elsewhere when the page load slowly or video is interrupted (Muhammad, et. al., 2021). Suppliers wishing to provide the best services in video streaming possible, as different applications - for example, telephone, email, and streaming video - can use the same network, you need to observe how network resources will be shared to match the requirements of each service. Using QoS, different video streaming can use

channels without consuming each other's bandwidth (BW). The term 'Quality of Service' refers to various technologies such as the Rayleigh which can identify the type of data in a packet and thus divide the packets into categories of traffic that can be prioritized for streaming. The main advantages of a network that operates with Rayleigh are the ability to prioritize traffic and allow the distribution of high priority streams before lower priority streams, as well as greater reliability in a network, controlling the amount of BW, an application can then use and therefore monitor the BW contention between applications. Video traffic often considered a high priority and requiring low latency, is an ideal case in which Reed Solomon can be used to ensure rapid responses to move demands. The necessary condition for using Rayleigh in a video network is that all channels meet QoS. Some of the important QoS parameters that interest us usually are BW, Latency, and Jitter (Mohammed, 2021).

Many services need a minimum BW to proceed, as in the case of voice-over IP and video streaming. With more BW available, the stream will be faster with a clear voice. The time observed by a packet to reach the destination need greatly helps to stream with high latency which also harms a little video. As too high, it also causes delays to open streams. Often, a latency problem is observed as a "lack of BW". The first channel had only twice the BW of the second, but they looked to have ten times more, simply because their latency is much lower (RTT from 30ms). Jitter causes more noticeable streaming problems, even though it troubles any connection since the channel assumes slight variable latency and considers the packet lost if it delays arriving. In the case of video streaming, the application has the option to implement a "jitter buffer", which is the maximum length of the time between the entrances of a packet at the transport system's interface (Mohammed, et. al., 2021), (Hyunmin, et. al., 2004). It makes the video play consecutively, even when there is jitter, but has the serious disadvantage of increasing perceived latency. The Adaptive Media Delivery (AMD) solution allows CDN operators to provide instant access, high quality, and seamless streaming video services around the world and on any device (Mary, et. al. 2015). Optimized through Adaptive Bitrate Streaming (ABR), the solution can deliver streaming video services on different networks, fixed or mobile, and at different rates. Availability, scalability, reliability, and unparalleled presence (Ali, et. al., 2021) are the characteristics of the AMD solution for the distribution of streaming video in various formats such as Microsoft Smooth Streaming, Adobe HDS, Apple HLS, and MPEG DASH (Kevin, 2021). The AMD solution combines all the necessary features for video services in high-quality streaming to decrease the impact of network congestion, latency, and packet loss with its edge servers deployed in over-network providers stations (Truong, et al., 2012). With our multiple points of evaluation, we can bring the content of your users for high-quality streaming. The quality of experience are events measured make history in terms of capacity and scale. The AMD solution is the result of this experience and allows us to have a direct start on user requirements. Being able to provide capacity and distribution scale to meet demand has permitted us to attract the best results, this large setup distinguishes that if a video becomes "out-of-date", it must be able to deliver reliable streaming services to broadcast it. (Chih-Lin, et. al., 2021)
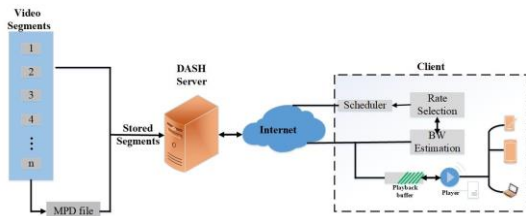
Adaptive streaming model

Fig.1. Block Diagram of Client-Side Rate Adaptation Streaming.

As seen in figure 1 (Muhammad et. al., 2021), the client-side diagram for adaptation streaming sends the request (HTTP GET) to the HTTP server to download the video chunks with the rate selection module. The server downloaded the Media Presentation Description (MPD) file to attain all the knowledge about the quality levels before the first video segment. The HTTP client receives the video and the downloaded segments are stored in a playback buffer and then feed to the player decoder. The playback buffer is connected with each independent video segment.

We have two case studies that show methods for watching the video streaming in two clients

First, PC1 is viewing two video streams from cameras 1 and 2, each sending images at 2.5 Mbit/s. Unexpectedly, PC2 starts a file transfer from PC3. In this situation, the file transfer will attempt to use the full 10 Mbit/s capacity between routers 1 and 2, while video streams will attempt to keep their total of 5 Mbit/s. The BW assigned to the observation system can no longer be guaranteed and the video frame rate will be expected to fall. At worst, FTP traffic will consume all offered BW.
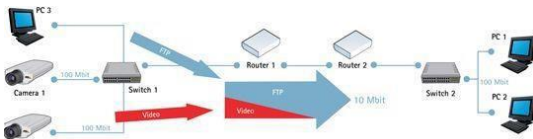

Fig. 2. Streaming case study 1

Second, Router 1 has been configured to offer up to 5 Mbit/s of the 10 Mbit/s presented for video transmission. FTP can use 2 Mbit/s, and HTTP and all other traffic can use a maximum of 3 Mbit/s. With this division, video broadcasts will always have the essential BW available. File transfers are considered less important and acquire less BW, but there will be BW available for web browsing and other traffics. Only these maximum values are valid when network congestion occurs. If there is idle BW, it can be used by any type of traffic.
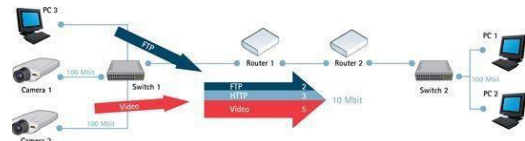

Fig. 3. Streaming case study 2

2. Rayleigh Fading Channels

In probability theory and statistics, the Rayleigh distribution is a continuous distribution function and presented as a two-dimensional vector that has its two components, orthogonal, independent, and follows a normal distribution. Its absolute value will then follow a Rayleigh distribution (Fernandez &Ferrero, 2016). In the case of complex numbers with independent real and imaginary components, this distribution can happen and the following is a normal distribution.

The probability density function is (Fernandez &Ferrero, 2016):

$$f(x|\sigma) = \frac{x \exp\left(\frac{-x^2}{2\sigma^2}\right)}{\sigma^2} \qquad (1)$$

Their hope is

$$E[X] = \sigma\sqrt{\frac{\pi}{2}} \qquad (2)$$

Its variance:

$$V[X] = \frac{4-\pi}{2}\,\sigma^2 \qquad (3)$$

The maximum likelihood estimated the parameter for sigma is given using:

$$\sigma \approx \sqrt{\frac{1}{2N}\sum_{i=0}^{N} x_i^2} \qquad (4)$$

The signaling channel is a logical and physically essential element within a signaling system. The term channel also indicates each independent way capable of carrying a signal with its features. If a given system divides the communications in frequency-division multiplexing (FDM) and Time-division multiplexing (TDM), each part of band B and of time (slot) will be a channel. When the channels are obtained from the physical resources division called physical channels. Their names rather than logical channels consisting of independent information flows are an example for GSM: BCCH, SCH, TCH, RACHMS, etc. (Jahangir et. al., 2000). The physical channels are then to be understood as the transport rails, while the logical channels as a subdivision of the transmission potential of offers by the same physical channel. The logical channels mapped on the physical channel then end up with rules that depend on the particular telecommunication system implemented (Federica & Alessandro, 2021). The example of the physical channel shown here is also an example of multiplexing of a system; in fact, deriving a physical channel means at least deciding on the multiplexing, but in addition, a physical channel can also characterize by assigning a specific modulation and/or coding and/or property data to that particular resource.

2.1 Channel as a physical system

From the point of view on system, or logical-functional-behavioral, the theory of the channel of communication systems is a system that receives in input (inputs) a signal x(t) and produces as output in output a signal y(t). In

the linear case, it is therefore described by an impulse response h(t), a transfer function H(s), and a frequency response H(f). In the case of the signal channel transfer function and the frequency response, it is typically time-variant in a random way under the mutability of the physical propagation conditions. The frequency band (f) extension for the frequency response defines the bandwidth (B) of the channel, while the shape of the frequency response illustrates the possible distortion of the input signal x(t) in that band. To these parameters is finally add the channel attenuation and the signal propagation delay which regulates the phase shift in the reception of the carrier signal, parameters included in the transfer function.

2.2 Ideal channels and real channels

An ideal transmission channel should have a sufficiently broadband and uniform, both in amplitude and in phase, to contain the spectrum of the information signal without distorting it and should be able to transfer at any distance without introducing degradations in the electrical power source. Real channels typically are present degradation factors such accurately attenuation band distortion and noise (Ashfaq, et. al., 2021). In particular, when the channel introduces distortion in the band is said to be dispersive or one or more frequency components of the transmitted signal and propagating in it suffer effects of altered attenuation (Amplitude distortion) and/or different phase shifts (Phase distortion or Spectral broadening).

2.3 Noise in the channel

To the signal x (t) transmitted in the channel, it is associated then in real cases a noise n(t) of the random type which makes the behavior of the same type of random channel; the noise is often merged in amplitude to the valuable signal. A noise always present, both in the channel in the transmitter-receiver devices, is the thermal noise due to molecular thermal agitation of the physical constituents of the channel (Saeed, 2000). The presence of noise will result in the introduction of distortion on the transmitted signal that, together with the in-band distortion proper to the channel, can lead to the change of the waveform. Then, the transmitted signal with distortion of information contained in it is in an analog transmission in a digital transmission through a decoding error in reception due to inter-symbol interference (Salah, 2021). The downstream receiver, using decision, must therefore go back to the original signal transmitted by estimating the error and eliminating Forward Error Correction (FEC) or request retransmission (ARQ).

2.4 Interference

Another type of interference on the channel that creates signal distortion is the interference due to other unwanted information signals in the band of the useful signal (Salah, 2021).

2.5 Nonlinear effects

Depending on the type of channel created in a non-linear effects variable function of the power of the transmitted signal, which has the most typical effect that inter-modulation, which in turn, is a form of distortion by non-linearity of the signal. Rayleigh and Rician fading channels are convenient models of real-world phenomena in wireless communication (Sánchez. et. al., 2020).
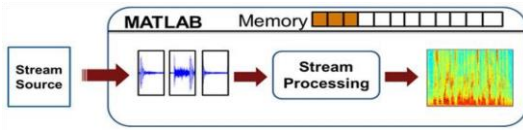
3. Simulation Code, Results, and Discussion

Fig. 4. Video streaming processing in Matlab

During the process of adaptive streaming with a source that offers streaming signals to the clients, the Rayleigh encoding and decoding system will process the data and also compression and performance factors applied. A proper memory allocation will be done to ensure that congestion will not degrade of quality and packet drop.

Fig. 5. Streaming server and Rayleigh encoder

Fig. 6. Client streaming reception through Rayleigh

As seen figure 5 includes the origin of the streaming over the Rayleigh fading channel, the received streams in the client panel are shown in figure 6. Background algorithms are added to track the plotting parameters. Now collective performance parameters like delay, throughput, collective QoS, and link efficiency are plotted to a terminal using a graphical representation.

Adaptive entry point

Based on the main reasons for the high BW request, the most cited is the increase of the data related to the video streaming traffic. In particular, to an increase in numbers and bits, it is the high-definition video about 50% of consumption for the downstream BW, with the aggregation of the two clients easily exceeding 30%. This figure still does not equal that of total availability perhaps because clients do not still present in reception on this level.

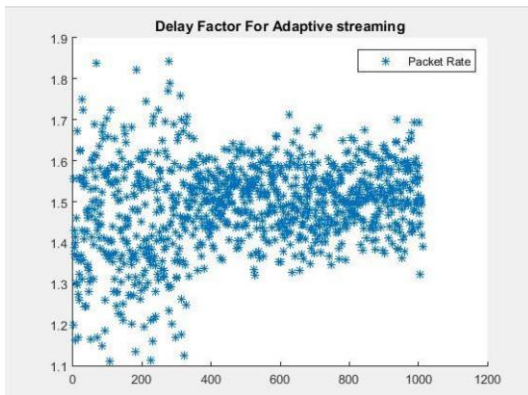## Delay factor for adaptive streaming

Fig. 7. Delay factor for adaptive streaming

As seen in figure 7, the load of around 6Mbps on the shared link, here the packet rate corresponds to 1000pps per 1.5msec, still, the high rate corresponds to 2ms in peak time achieved. This clearly shows the significant improvement after the initial load to queue the packets. Hog Rayleigh segmentation during the encoding and decoding helps to achieve the goal of quality enhancement. It was observed also that the use of two or more cascaded error correction codes could provide additional performance gains. While this increases the transmission rate requirements and frequency band occupancy in the channel, the input redundant symbols make the most protected sign, in the sense that they prevent the system noise substantially affecting the transmission quality. The digits of the numerator indicate the number of original input symbols in the FEC encoder (uplink transmission), while the denominator shows the total number of symbols corrected in anticipation that leave the encoder. The greater the BW occupied by a channel, the higher the load recorded.

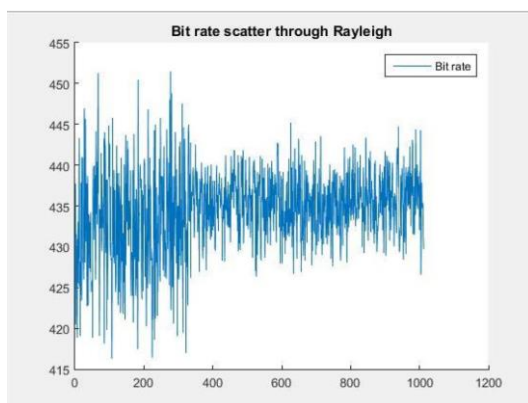## Bit rate through Rayleigh fading

Fig. 8. Bit rate through Rayleigh

A bit rate scatters type in which the video blocks move describing an elliptical path in the vertical plane containing the direction of propagation of the wave signal. At the upper end of the elliptical path, the particles move in opposition to the direction of propagation direction and based on the travel route in the direction of propagation. Since waves are dispersive Rayleigh, and different lengths propagate with different speeds, they are useful for

evaluating the variation of delay (variance) with depth. Rayleigh waves make up most of the stream recorded as a surface wave. It is produced when a video signal passes through dispersion much smaller than the length of the frames of the video signal beam particles. To disperse, what happens is that all the video signal does not reach the receiver. As an example, the Rayleigh scattering is the main reason that the signal improves throughout. But if the particle size is larger than the length, the video signal is not separated and all lengths are dispersed. Therefore, the Rayleigh scattering depends on particle size and length of the video signal, which leads to the Rayleigh coefficient. According to the output, it was found that the degree of dispersion is inversely proportional to the length of the video signal which causes a longer length dispersion to find smaller, and vice versa.
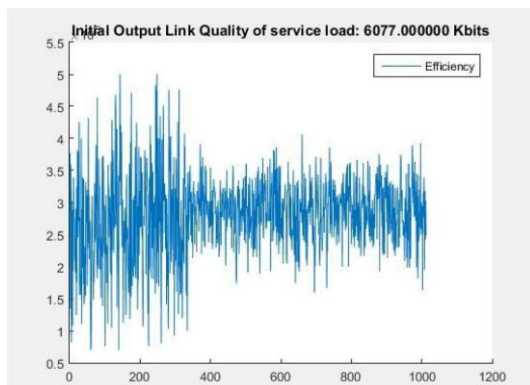
Efficiency measure through packet loss

Fig. 9. packet loss through link efficiency

A Stability in packet flow as per figure 9 indicates the full usage of available link efficiency for a specific period. Although it is known that the 1/2 FEC code signal results in a more robust and protected to interference and noise ratio than with FEC 3/4 since it implies more FEC (for each bit input to the encoder FEC, an extra is included), their use imply a reduction of channels (less Mbps) that the channel can put on an encoder. The results showed an improvement of 3.3Mbps over an original signal (uncorrected) that could be obtained by using error correction techniques. In other words, a stream that normally needs a BW of 1.8Mbps in media for receiving a signal without correction could use a smaller BW (3Mbps smaller), the signal was added to error correction techniques, with the same previous quality. The signal modulated MPEG-4 contains special codes that the receiver uses to check (and correct if necessary) if all transmitted information bits were received correctly. This early correction of technical errors (FEC) creates a robust signal with advantages over other digital signals.
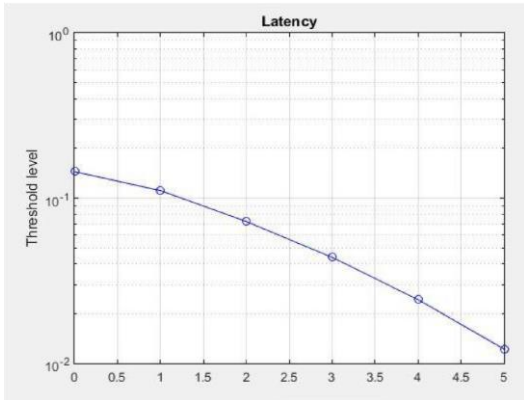
Latency curve through an adaptive channel

Fig. 10 latency curve

As seen in figure 10, A stable decrease in latency in the client response shows the efficiency of the algorithm applied over the channel. The symbol transmission rate which determines the quality of reception of a signal is a channel width function available. The greater the BW occupied by a video encoder channel, the higher the latency to its transmission. The commitment, then, will depend on the final quality (the receiver) signal to be obtained, and the throughput you can make. Although some frames are initially able to handle streaming of traffic, the content was often transmitted at 12Mbps or higher bit-rate, disadvantageous for the BW. The results adjustment for the switching parameters of the flow and streaming/recording, as well as the most important aspect, the control of the bit-rate.

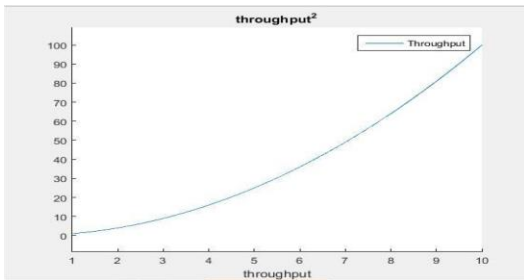## Throughput through an adaptive channel

Fig.11. Throughput curve

The performance on adaptive streaming is composed of capacity levels, delay, and throughput. In a fading channel, it is very important to maintain optimum levels in these components, since different blocks of flows generated by users, devices, or applications can be strongly affected in their activities due to changes in performance levels. It is then understood performance architecture is the set of mechanisms that are used to configure, operate, manage, and list the resources available on the network that support traffic flow information. It is the amount of time it takes to transfer a unit of information through the system from a source to a destination. Users using applications or interactive real-time expect the network delay is minimized. Even so, applications using voice or video are expected to have maximum levels of throughput. The term used when changes occur in the throughput is known as peak ratio and this causes changes in the video. The throughput is a point that must

be given much importance in the technologies involved in the transmission of video, but they are especially crucial for slow links (Iqbal & Olariu, 2021). As we see here, our throughput is increasing on time so that performance is increased.
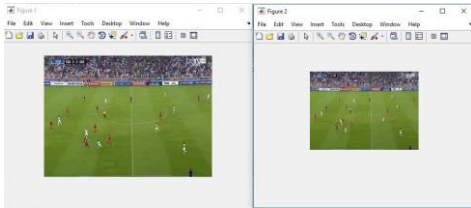
3.1 Video-on-demand phase:

Fig. 12. Video on Client

From these two videos shown in figure 12, we conclude that client 1 has a good channel (wider BW) and got good video quality while client 2 has a bad channel (tinier BW) and got lower quality.
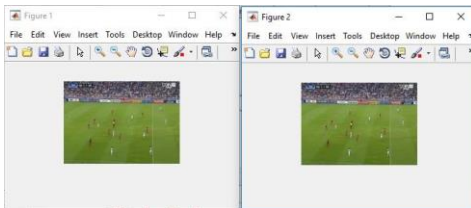

Fig. 13. Video on the client after a period

Client 1's BW was reduced (perhaps the client move away from the access point) and the system detect this change and started transmitting video with lower quality, while client 2 kept going with the previous situation.
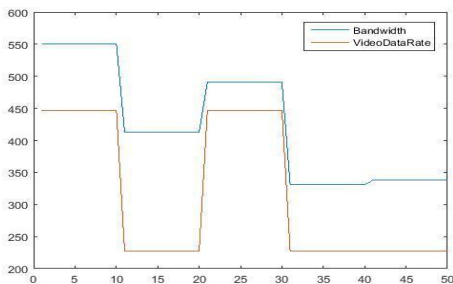

Fig. 14. Video Data Rate & Bandwidth

This curve illustrates the BW & video data rate relationship, the system checks BW every interval of time and can provide suitable video to the client to prevent latency.

Suggest that the server should have more than one copy of the video to stream, the copies have different resolutions from each other so that users' devices can request resolution according to its available BW. The system works in two phases:

Initial phases (setup phase)

First, to establish the streaming the system should follow the following procedures:

1- The client should send a request packet.

2- The server will respond with two test packets to measure the BW.

3- The client will discard the first packet and turn it on a counter.

4- After receiving the second packet client turn off the counter and calculate the BW using this formula:

$BW = \text{packet size} / \text{time calcualted}$

5- The client will send a packet that contains BW.

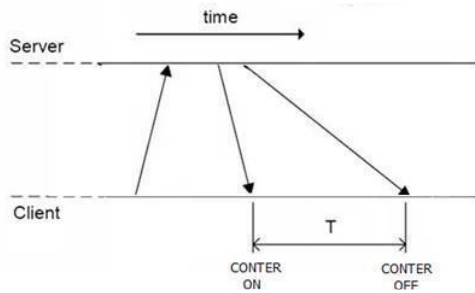6- The server will start streaming video with a suitable resolution.

Fig.*15*. Measurement of the BW

Following this execution, the streaming will start and we have added a dynamic graphical console to identify the input signal, output signal, and jitter, as this is a continuous process, As seen in a snapshot in figure 16.
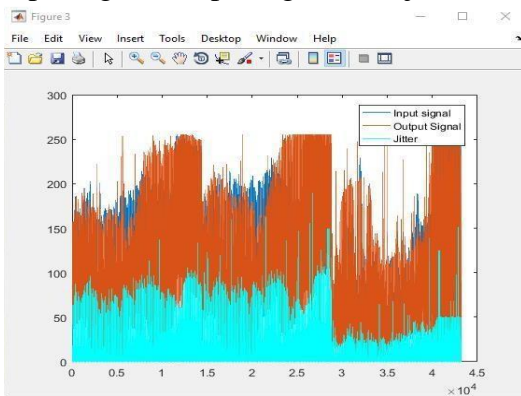

Fig.16. Dynamic streaming analysis

The plot in Figure 16 shows an average of 100usec jitter variation was found with 4 secs of streaming. This is comparatively high as the Reed Solomon algorithm is heavy to execute using arrays. Still, we have tried our level best to keep the ratio under control. As with a high-quality mp4 file, this output is pretty acceptable. At now same time, the original video will be displayed in one panel, and adaptive clients are displayed in another panel as we can see in Figures 17 and 18.
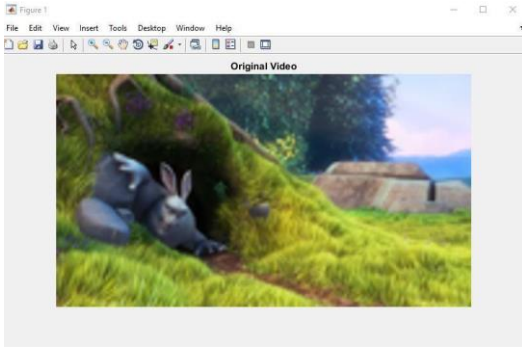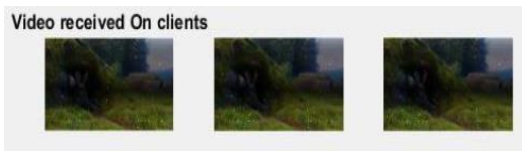
Fig. 17. Original video processed

Fig. 18. The video was received on clients after error correction.

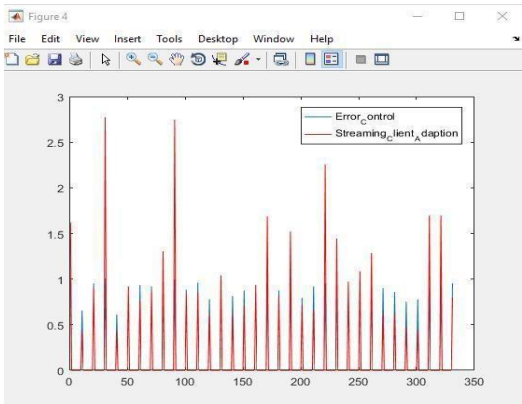## Error control rate and Adaptive rate

Fig. 19. Error correction and adaptive ratio

The error control with Reed Solomon is calculated with a top ratio of zero to one (0 to 1). If the error rate is near 1, it is excellent, if less than 1 seems to be a disrupting event or packet processing error. As seen in figure 19, the blue charted lines show the error control which is almost near to 1 throughout 350 sec of streaming. This is a fantastic result, even this will be more successful when comparing the received video quality on clients. The red notation for adaptive ratio in clients, which is also up to 2.5 out of 3 channels shows the full usage of the channels and efficiency of the Rayleigh fading channel implementation.
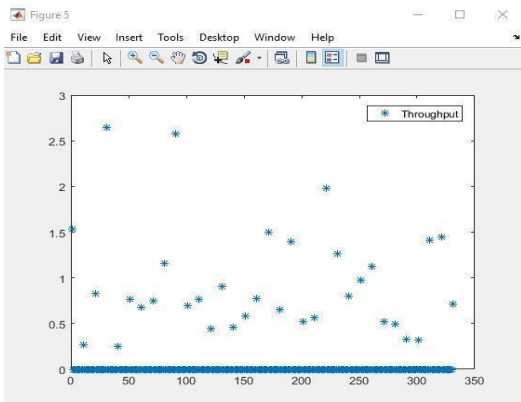
Throughput

Fig. 20. Dynamic throughput

As the streaming is processed through complex and heavy error correction algorithms, figure 20 shows that the throughput cannot be measured in actual vales. With a 350 sec streaming period, we have a good throughput with 250 packets per sec in some streaming stages. This output is also incredible even not that much essential during an error correction experiment.
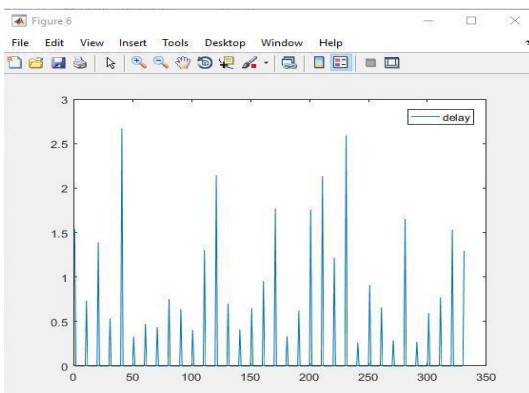
End-to-End Delay

Fig. 21. End to End Delay

As the streaming is processed through a complex and heavy error correction algorithm, the throughput cannot be measured in actual vales. With a 350 sec streaming period, we have a stable delay during the streaming. This delay factor is an adaptive value, so the average of 1 to 2 ms delay is achieved with error correction. This value is not much better, but with, a significant quality enhancement, the value is good.
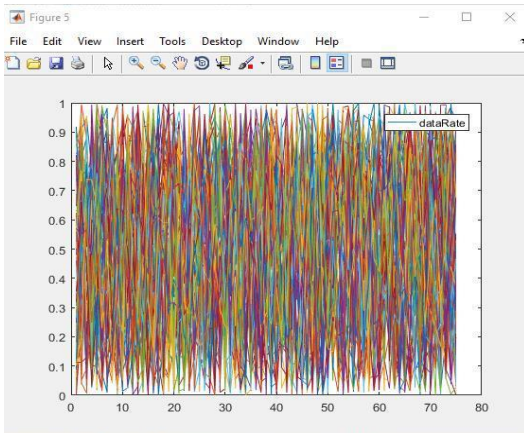
Data rate

Fig. 22. End-to-End Data rate

As the streaming processed data rate is measured with each channel inside the Rayleigh fading channel, the complex graph is generated. Finally, a matrix of intensity that streaming created on each adaptive client. Much populated channel intensity is visible as our output shows the reliable communication that happened through the links. Much better output can be created if we can assign on the video quality, as our paper is keener on QoS, we are satisfied with this output for data rate.

3.2 Streaming phase:

After the server receives the packet with available BW, it will choose a suitable resolution corresponding to the BW to start streaming, but we need to check BW continuously either to increase the video's resolution or reduce it. While the client receives video packets, the client will turn on a counter and after a specific amount of time it will turn it off and see how many bytes will be received, and the available BW was detected and sent to the server (Wafa et al., 2021).
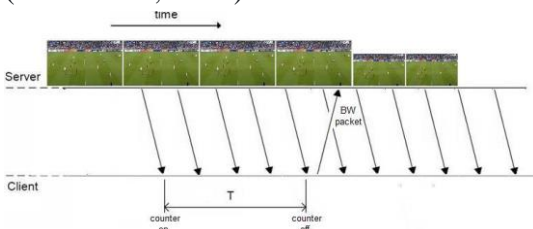

Fig. 23. Live transmission after determining the BW

YouTube live streaming: A case study

Google has already developed and put to the game a live streaming tool, Live YouTube, where you can record from your webcam PC or even your cell phone, quality YouTube videos, and your friends can follow in real-time -Live- what you're filming. It could be an event, a trip, a cool place. You play your videos directly on YouTube. The tool is YouTube Live - http://www.youtube.com/live/, and available to all users, so far they are only for exclusive channels.

This service is intended to transmit audio and video live on the internet (web TV). The plans use media systems technology which is a widely used streaming system whose most widely used broadcast encoder is Adobe Flash Media Live Encoder. The streaming with MS has a flash/html5 player, links to listen on smartphones/tablets, a player for Facebook and the manageable site can be used in different plans.

HD quality and adaptive player: This video plan can transmit up to 3 speeds and screen sizes by the live encoder. The player detects the best quality according to the speed of the viewer and it is also possible to select the quality manually by the player in flash. It is also possible to generate a separate player for each quality. In addition to the flash player in the RTMP protocol, the system generates an adaptive HLS link that opens on systems with IOS, Androids (4.1 onwards), and the html5 player. In html5 player the adaptive features are limited. You can even use RTSP links that are compatible with older versions of Android. Broadcast in high quality for better connections and lower quality to reach audiences who are in more limited connections. The speed limit of the plans goes from 750 to 2000 kbps, depending on the plan you choose.

Standard Quality: This option is recommended for users who are on lower-upload (less than 1000 kbps) connections, where it is not possible to stream HD speed with adaptive. In this way the client transmits at a speed only, not being possible for the viewer to change the quality by the player. Plans are 200 or 400 kbps.

What Quality to choose, the pattern or the HD with adaptive?

Base your decision on the following criteria:

Image quality: The higher the speed, the better the image quality, but the internet needs to be the transmitter and the attendant. In this respect plans with HD quality and adaptive bitrate are superior.

Adaptive bitrate: Plans with this feature allow the client to stream live on more than one speed, enabling high quality for those who have a better connection and a lower quality for those with slower connections. But you need to upload enough as well. In the 1000 kbps HD plan, for example, you can stream video at 736 kbps and 200 kbps with audio at 64 kbps, but you need to upload at 1300 to 1500 kbps since the on-premises upload consumption is the sum of all the encoder.

Upload ability: To make a live stream, your internet connection needs to have a higher upload than the speed of the plan. For example, to broadcast in HD quality of 750 kbps you need to upload from 900 to 1000 kbps. Do some speed testing on your internet to get a better sense of uploading ability, because although many internet providers advertise an upload speed x, in practice it may be different. If you transmit above the upload capacity of your connection, you may experience cuts, loss of connection, and image problems.

Internet Viewer: If your upload allows, it is more appropriate to use the plans with Adaptive HD, because in them you can transmit in high quality for high-speed connections and lower quality for slower connections.

Cost: Plans with the standard quality (up to 400 kbps) cost less because less server link is consumed. It may be more suitable for those who are starting out and still have a low budget to pay for the service.

Schema of transmission: In the block diagram below we do not include players with the adaptive video, they are more to visualize the quality of the image at different speeds.
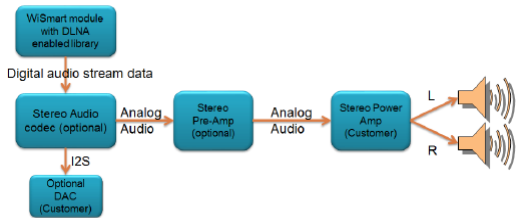

Fig. 24. YouTube Audio DLNA (digital living network alliance)

Live transmission and without Auto VJ visual mixing software

YouTube live without Auto VJ: In this system, the images are generated from an encoder installed on the customer's local computer and transmitted at more than one speed by the encoder (adaptive bitrates), so that the viewer has more than one image quality option according to the Internet available. You can capture images from webcams and capture boards. To use in the programming files recorded in the micro, use some encoders that have this feature, such as Wirecast, Vidblaster, among others. The standard encoder (Adobe Flash Media Encoder) only takes images from live sources.

YouTube Live with Auto VJ: In addition to allowing live streaming, the system allows you to leave a playlist of videos in MP4 format (H.264 CODEC) and FLV being transmitted by the server. This system does not have the adaptive bitrates, nor in the live mode. You can also record a live stream direct to the server for later use in Auto VJ. It is suitable for anyone who does not need to make live broadcasts or still does not broadcast live all the time. In this way, the client can have 24-hour active programming without having to leave an encoder transmitting live. It is also possible for the same client to make several channels with different contents without having to have a computer for each transmission, as in the system only live. The load is slightly higher because the system involves more server resources such as disk space and processing. The client also needs to convert the videos in the format and speed indicated to include them in the system.

3.3. Error Detection and Correcting

The detector codes and error correctors are transmission errors in the lines because of various influences like thermal noise, impulsive noise, and intermodulation noise. Relying on the transmission medium and the encoding type that is used, other anomalies can be present such as rounding and attenuation noise, as well as line crossing and echo throughout transmission. Two different plans have been designed for the treatment of errors (Ashraf & Aya, 2015). The first plan is error detection in which some redundant bits allow the receiver to detect an error that has occurred, but not what type of error or where. The second plan is redundant information that allows the receiver to deduce what character was transmitted and the receiver can precise a number Limited number of errors.

Error correction can be addressed in two ways:

When the error is noticed in a specific data fragment, the receiver asks the sender to retransmit. The receiver detects the error using enough redundant information to apply the corrector method, it automatically applies the mechanisms necessary to correct that error using redundant bits. The need for a greater number of redundant bits sometimes means that multi-bit correction is not feasible and inefficient because of the high number of bits required. Therefore, the error correction codes are usually reduced to the 1, 2, or 3-bit correction.

3.4. Reed-Solomon Properties

A Reed-Solomon code is RS (n, k) with s-bit symbols which means the encoder takes k symbols from the s bits, and to make a code word of n symbols, it adds parity symbols. There are n-k parity symbols of s bits each. A decoder can correct up to t symbols that contain errors in a code word, where $2t = n-k$.

Example: A popular Reed-Solomon code is RS (255,223) with 8-bit symbols. Each codeword contains 255 bytes of the codeword, of which 223 bytes are data and 32 bytes is parity. For this code you have:

$N = 255, k = 223, s = 8$
$2t = 32, t = 16$

when at least one bit of a symbol is erroneous, a symbol error occurs. As in RS (255,223), 16 symbol errors can be corrected. In the worst case, 16-bit errors occurred in a byte, the decoder corrects 16-bit errors. In the best case, 16 complete byte errors occurred that the decoder corrects 16x8 bit errors. Reed Solomon codes are particularly useful for fixing burst errors.
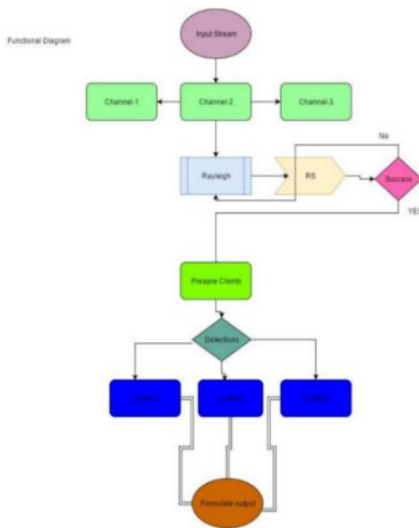
Input is validated and sent through three channels to prepare the Rayleigh formulation. Once the processing is done, in any case of total failure occur, we used to put back the data into array for further processing. This will make the error control better during the Rayleigh fading process.
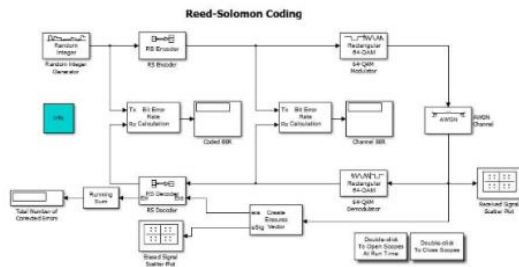
Fig. 25 Reed Solomon coding cycle

Quadrature Amplitude Modulation (QAM) is used in systems that require a high data transfer rate consisting of two carriers that are used in quadrature. This system is used in digital terrestrial TV, cable, and some systems used experimentally by radio amateurs for transmissions in data transfer. In digital transmissions, QPSK (Quadrature Phase Shift Keying) modulation is used for satellite, QAM for cable or terrestrial, and OFDM for terrestrial emission. Some examples for QAM are digital radio links and, high definition digital television. The QAM modulation can be: 4 QAM, 16 QAM, 32 QAM, 64 QAM, 128 QAM, 256 QAM, 512 QAM, 1024 QAM, 2048 QAM, 4096 QAM or more dense.

Gaussian noise is a statistical noise whose probability density function (PDP) is equal to the normal distribution, which is also known as Gaussian distribution. The probability density function p of a Gaussian random variable z is given by

$$p_G(z) = \frac{1}{\sigma\sqrt{2\pi}} e^{-\frac{(z-\mu)^2}{2\sigma^2}}$$

Where z is the gray tone

The main sources of Gaussian noise in digital images are illumination problems or high temperature during acquisition or transmission problems. In digital image processing, Gaussian noise can be reduced by spatial filter techniques, which soften the noise in the image with the disadvantage of erasing it a bit. Such techniques are median filter and Gaussian filter.

The video rate and figure attributes must be calculated to initiate channel and channel capacity and the frame size should be adjusted for faster processing. Slow processing resulted in packet loss during the Rayleigh and reed Solomon algorithms.

3.5. Live video streaming by using Wowza

Wowza Streaming Engine is strong, customizable, and accessible server software that controls reliable streaming of high-quality video and audio to any device anywhere. Use it to form audio and video requests and facilities that deliver attractive streaming for live events, news, training, and on-demand videos with security and confidence even deployed in the cloud. Any video format can be delivered in multiple formats and accepted by Wowza Streaming Engine with the highest possible quality to any connected device anywhere. It can push live video to CDNs and select unicast or multicast transmission.
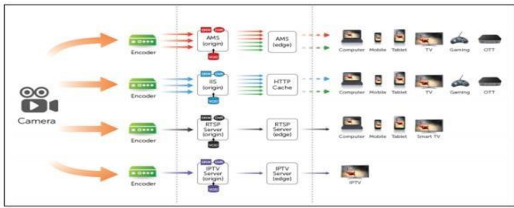
Fig. 26. Conventional, segregated streaming workflows

Wowza Streaming Engine can stream a single H.264 encode (live or on-demand) to all client types and push the requested stream in client-specific encoders and servers. This reduces operational charges related to the hardware needed to keep and simplify the management and deliver the best possible watching experience to customers. You can select from a wide-ranging of conventional live RTSP/RTP, MPEG-TS, and RTMP-based encoders, or accept videos from H.264 IP cameras.

Fig. 27. Wowza Streaming Engine accepts video from a variety of sources and outputs to multiple formats

Wirecast is a live video streaming tool by Tele stream. It permits users to build live or on-demand distributes for the web and provides the source that controls real-time switching between multiple live video cameras, while dynamically mixing in other media (such as QuickTime movies) to create professional broadcasts for live or on-demand distribution on the internet.
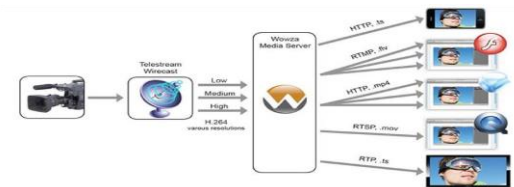

Fig. 28. Single encoding/delivery workflow with Wirecast and Wowza Media allows simultaneous delivery to multiple clients

Wowza Streaming Engine application will represent the developer that accepts the live video from Wirecast then provides many recent techniques for performance enhancement by presenting many resolutions and determining some curves as we will see in the results part.
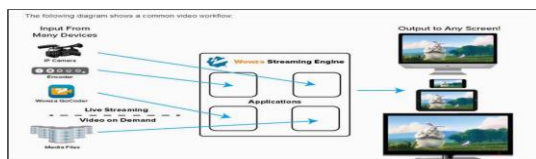

Fig. 29. The performance improvement by using Wowza Streaming Engine

Wowza Streaming Cloud is the live streaming service to viewers of any format on any device with high-quality from a camera or encoder. It represents the provider that announces the live video online and allows many viewers to watch this live video.

Fig. 30. Wowza Streaming Cloud

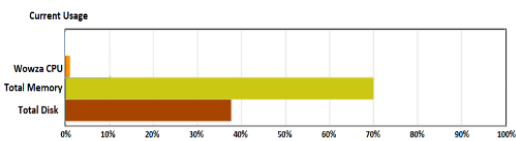Fig. 31. Wowza Streaming Engine -Live video stream

Fig. 32. Wowza Streaming Engine - Server Monitoring

Fig. 33. Wowza Streaming Engine - Network

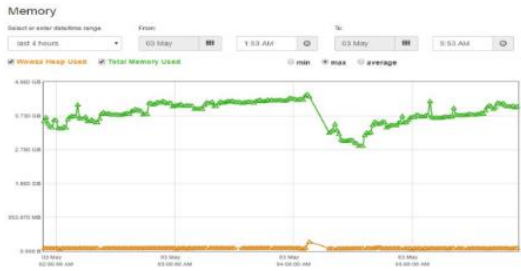Fig. 34. the curve of Wowza – CPU Usage

Fig. 35. Wowza Streaming Engine – Memory Usage

Fig. 36. Wowza Streaming Engine - Total Disk Usage

Conclusion and Future Work

A reliable scheme to evaluate and regulate the performance required under realistic assumptions of network parameters for adaptive video streaming to multiple clients through the Rayleigh fading channel and the overall system was built to evaluate the performance for each user and concluded that the video was clear but never match with the High Definition (HD) input provided. Moreover, Adjust the performance needed for each user by applying the error correction for the adaptive streaming using Reed Solomon algorithms. The results enhanced the performance and concluded the better results to achieve the required QoS for parameters of video streaming system such as latency, bit rate through Rayleigh channel, delay factor of adaptive streaming, and efficiency. In future work, the security feature can be added to the proposed adaptive streaming model to make a secure system for the video transmission

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