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OPTIMAL ADAPTATION OF INTERNET VIDEO STREAMING

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Abstract

The main challenge of this work is how to get optimal video streaming over the Internet. This paper describes the main objective of the adaptive video streaming over Internet based High Efficiency Video Encoding (HEVC)/H.265 codec, network mechanism observation, network performance measurements and how the network effecting on video streaming by network monitoring. Where; the main job is how to incorporate video coding with the internet protocols to reach the goal of video editing quality based on internet environments. H.265 codec with optimal configurations is tested for three raw videos with different categories of content to compress them with low bitrate and acceptable quality. The videos streaming is tested over the two IP networks, Unicast and Multicast. To obtain an appropriate map from network performance evaluation and how it effects on the video quality, Real-Time Streaming Protocol (RTSP) streaming protocol, Real-Time Control Protocol (RTCP) controlling protocol and Real-Time Protocol (RTP) transmitting protocol are configured. The network measurements, packet loss, delay jitter and latency, show that different video categories can be streamed probably by incorporating the encoder with the network to work as silence network.

Keywords: HEVC/H.265, QoS, RTP Protocols, UHD video, Unicast and Multicast.

1. Introduction

Digital distribution of video streaming content has grown extremely in the last few years. The diversity of services and devices with growing popularity of higher quality Ultra High Definition (UHD) video to be transmitted for several kinds of terminal devices like mobile, tablet, personal PCs and High Definition Television (HDTV), is the main challenge of multimedia providers [1].

Today the available bandwidth is the main challenges where unlimited number of users shared with. In this work, to make compromisation between increasing number of users with limited bandwidth, adaptation of Internet video streaming using HEVC is analysed to get the better of the traffic bandwidth load balancing. The HEVC standard becomes the candidate to perform different goals, not just the compression efficiency but also it is designed to be used in several services, products and applications, also it is an integrated system that make the data transporting easily and flexibility of removing redundant data, as well as have the ability of executing by parallel processing architectures [2].

In the next subsections a layered H.265 is implemented, where each layer is of specific spatial resolution of parameters that are directly affects on the bitrate and quality, also the Robustness, Repeatability, Accuracy, Generality, Efficiency and Quality are the main properties of each feature as a result it will affect the resolution and quality of the video [3]. The analysis and implementation include the procedure of selecting the H.265 parameters to find the optimal bitrate and quality for each level. As a layered encoder, the selection of the proper level is based on the channel buffer statues with a proposed controller. At the same time, the reconstructed video almost in the proper resolution is based on the user terminal device application.

This technique of video encoder has made video access to end user with high quality, easily and taking into account the status of the channel. Today network technologies have led the video streaming services are available at any situation of network condition and at any terminals. However, video has required qualified network of services to avoid the problems of packet loss, delay and jitter that led to degradation in video quality [4]. Recently the nature of the internet real-time variable bit-rate led the designer start work with variable bit-rate encoded video.

That what will be detailed in the later paper sections with the network performance observation to evaluate its effect on video streaming. Generally, the calculation of objective mode was represented by measuring the Peak Signal-to-Noise Ratio (PSNR) and Bit Rate (BR) [5].

2. Layered HEVC/H.265’s

The proposed system is mainly consisting of the controller incorporated with encoder described in Fig. 1. Such system will serve the bandwidth reservation for video streaming with UHD resolution especially for widely usage Internet, also unexpected number of users at the channel cause a variation of bandwidth availability, all that reasons generates a congestion in the network which make the status of channel is bad. To eliminate this problem there is a controller using to avoid congestion when the network is loaded.

In this paper UHD (4K) is the goal. H.265 encoder with its features and parameters that effect on BR and PSNR are used to find the optimal value of them for different formats. The raw video, 4K “3840*2160”, is subsampled into the other formats, 1080i/p “1920*1080”, 720p “1280*720”, 4CIF “704*576”, CIF “352*288”, QCIF “176*144”. As parallel process all formats are encoded with H.265 make the control terminal selects the proper format based on channel buffer status. As design steps, each of these formats is worked of optimal BR and PSNR. When the channel is busy with number of users at a time, the video sequence should be transmitted with one of these formats be less than UHD. Adapting the sending bitrate over internet [6] needs to modify the syntax of network layer to include the transmitted format [7].

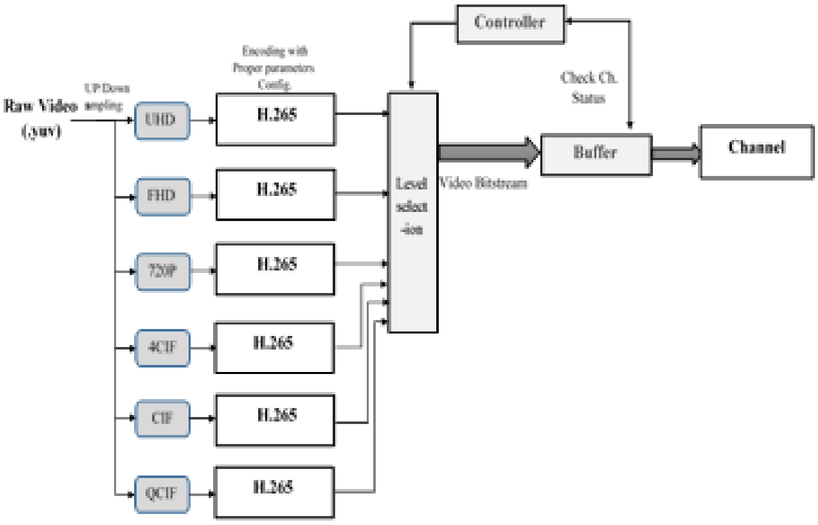

Fig. 1. The controller was used to select proper format [7].

HEVC/H.265’s parameters for video streaming

To test the parameters of H.265 a series of experiments of three types of test sequences (HoneyBee, Jockey, ReadySetGo) with different details and motions using six resolutions (layers) of these sequences, as seen in Fig.1, each sequence is encoded with 100 frames and frame rate 120 fps, there test sequences are encoded to find the optimal configuration of encoder’s parameters at the source device with each resolution. The H.265 parameters: (Quantization Parameter (QP), Constant Rate Factor (CRF), Group of Picture (GoP) and Reference picture (REF)) are tested to find the proper value of these parameters that can work with the three categories of video details that keep the quality of video acceptable at the values (32-40) dB, also the impact of bitrate. To evaluate the performance of encoding system, the general quantitative measure of video quality, peak signal-to-noise ratio (PSNR) [8] is used:

$$PSNR = 10 \log_{10} \frac{MAX^2}{MSE} \quad (1)$$

where the MAX represents the maximum value of pixel, while the MSE is the Mean Square Error [8]:

$$MSE = \frac{1}{MN} \sum_{i=0}^{M-1} \sum_{j=0}^{N-1} [P_{anchor}(i,j) - P_{test}(i,j)]^2 \quad (2)$$

3. Video Streaming Stander

At the current time, the real-time multimedia include video content are spreading over the Internet with high quality for most applications and devices, because the users increasing the use of these applications and devices that lead to developing the network technologies for multimedia distribution. The most important technique, like video streaming, that using Internet Protocol (IP) multicast to distribute the video from server to clients over the Internet. There are different mechanisms of video streaming such as the technology of video coding/transmission at server side then decoding and display at client side to achieve the adaptation of video with varying of bandwidth availability, this kind of network can classify as a heterogeneous and dynamic network [9].

The structure of video streaming as is illustrated in Fig. 2, there are several processes that initiated with media (video and audio) compression and storage space at the streaming server. While the streaming server receives the streaming request from the streaming protocol it selects the appropriate compressed results according to the controlling protocol (application layer Quality of Services (QoS) control) to adapts the media bitstream based on the network situation, then the bitstream is encapsulate with side information by transport protocols to send the encoded data over the Internet. At the decoder side the packets are go to transport protocol then to application layer to be decoded, at the decoder there is a with media synchronization to make a concurrence between audio and video received packets [10].

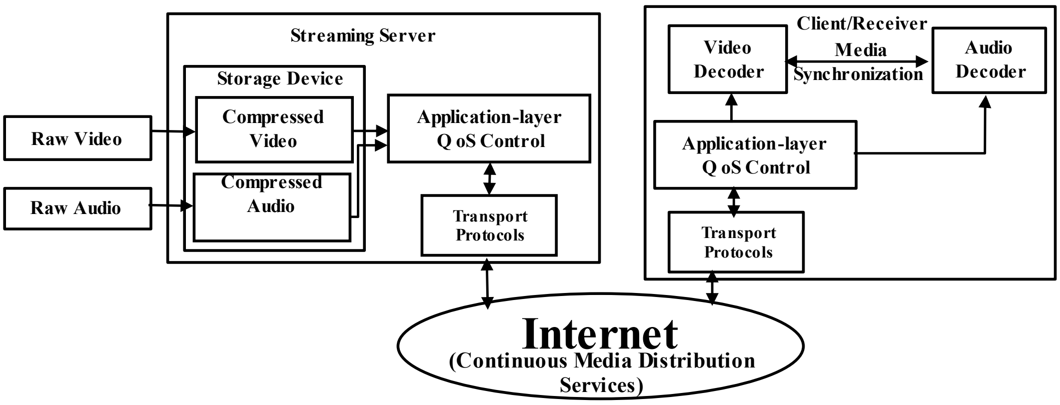

Fig. 2. Video streaming structure [10].

3.1. Unicast and multicast video streaming

The video content needing for hybrid delivered with high quality by using a unicast and multicast linear methods [11]. In this paper there are two scenarios: Unicast with single client receiver and Multicast with multiple client receivers, the performance of streaming with single user is better than multiple users of real-time video and the quality of video was decreasing over the heterogeneous network [12]. When the video sends to multiple clients that preferably use the multicast technology to save the channel bandwidth especially with Internet TV, live activates that needing for multicasting of video streaming. The developing of IP unicast is the IP multicast that produced to delivering the IP packet into the large number of clients as shown in Fig. 3 [13].

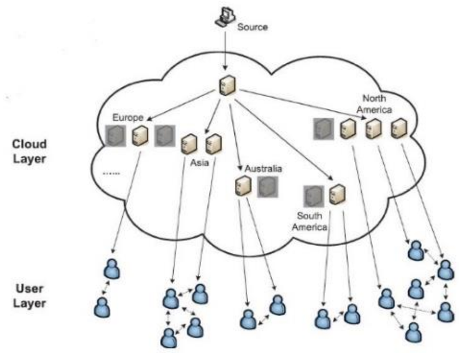

Fig. 3. IP multicasting to the number of clients [13].

Recently studies about the heterogeneous network with multicasting give a high flexibility of joint and leave in group the clients and produced a system with programmable gateway to video processing in strategic positions for saving the requirements of network bandwidth [14].

3.2. Video streaming protocols

Some standard protocols will be applied for the communication between the server and clients of video streaming over the Internet and based on their functions, there are three categories of video streaming protocols: (Network-layer, Transport and Session control) protocols, see Fig. 4 [10]. After data compressed then pass to RTP layer for packetization and add sequence number for information synchronization, the UDP/TCP layer (represent the transport protocol) and the IP layer (represent the network-layer protocol) are the two sequential layers to produce the IP packets to go over the Internet. At the receiver the IP packet pass through invers processing, at the control plane stage there are RTCP and RTSP packets (represent the session control protocol), they are multiplexing in UDP/TCP layer, then go to IP layer to transport over the Internet [10].

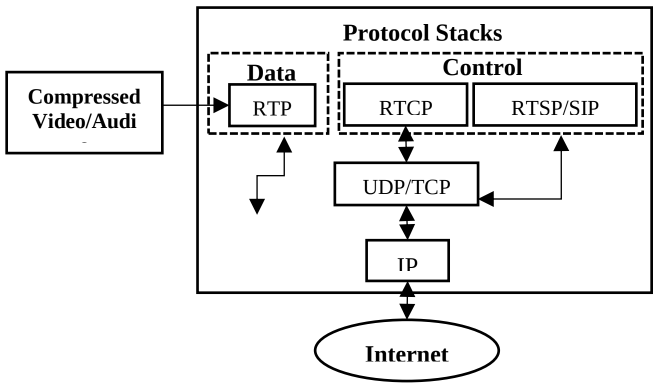

Fig. 4. Multimedia streaming protocol stacks [10].

The family of transport protocols are (TCP, UDP, RTP and RTCP), the basic function was applied by TCP and UDP, but the RTP and RTCP applied over them. In this paper the Real-Time protocol (RTP) is used for video multicasting transmission from server to clients based on UDP (User Datagram Protocol) [14], it has two concepts: application layer framing and integrated layer processing. While it’s designed to support function to end-to-end Internet transport for applications work with real-time [10], also the RTP was support several functions such as: Source identification, Payload type identification, Sequence numbering and Time-stamping, while the RTCP protocol was designed for controlling and incorporation with RTP, also have several functions: basically QoS feedback, Participant identification, Control Packets scaling, Inter-media synchronization and Minimal session control information [10].

4. Network performance evaluation

The heterogeneity of networks and devices make the transmitting of data packets over the Internet so difficult because the variability of data throughput, delay and loss. While the network parameters are affected on the video streaming transmission from server to client, so it necessary to evaluate the performance of network with some important parameters and Quality of Services (QoS) metrics: (bandwidth, Delay, Jitter and Packet loss) that affected on the video quality, this metrics can be measured by real-time monitoring the network [15].

5. Methodology and Experimental Results

Video encoding model

Based to video encoding model, the operations flow is implemented in Fig. 5 shows the proposed model flowchart stages, HEVC/H.265 video source and controller.

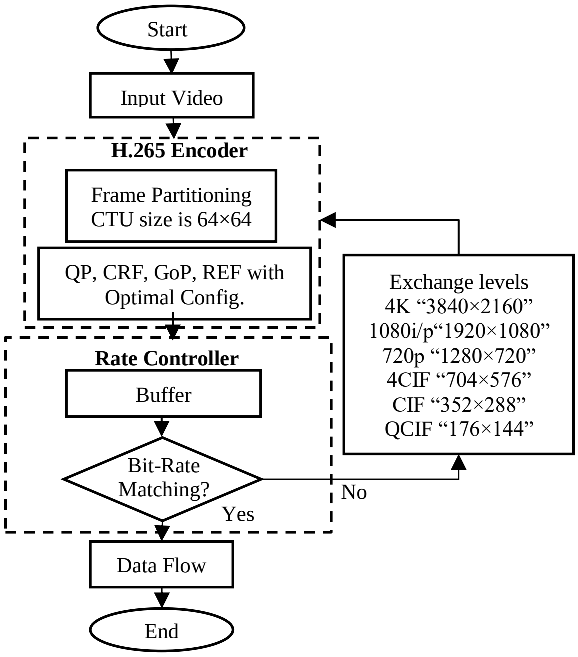

Fig. 5. Flowchart of the proposed encoding system.

The proposed system mainly consists of the H.265 encoder with incorporated controller, see Fig. 5. Such system will serve the bandwidth reservation for video streaming with UHD resolution.

Table 1 gives sample of experimental tests to find the range of use QP for each resolution at different motion details. For example, at 4k layer the value of QP (46 – 48) leading to accepted range of PSNR (36-33) of encoded video to be transmit over the channel. While at the smallest resolution (QCIF) achieve PSNR more than 36dB, it is reached to 40 dB, because the BR at this layer less than its value at 4K layer. The controller works to check the network condition then return instruction to the encoder to select the layer at appropriate value of QP based on motion details and layer type.

Table 1. QP range for each resolution.

Size	HoneyBee			Jockey			ReadySetGO			Rang	
	QP	PSN-R (dB)	BR	QP	PSN-R (dB)	BR	QP	PSN-R (dB)	BR	QP	PSN-R (dB)
4K							40	36.8	7380.6	46	36.9
	46	36.5	792.4	46	36.9	1863.7	44	35.1			
	48	35.7		48	35.7		48	33.2		48	33.2
1080i/p	42	36.9	453.7	42	36.2	1291.5	36	36.9	5042.3	36	36.9
	44	35.9		44	35.2		40	34.8			
	48	33.9		48	33.2		44	32.8			
							48	30.8		48	30.8
720p	36	38.5	446.7	36	37.9	1592.0	32	37.6	4951.5	36	38.5
	40	36.5		40	36.0		34	36.6			
	44	34.4		44	33.9		36	35.5			
	48	32.3		48	31.8		40	33.4		48	31.8
4CIF	32	39.2	403.8	32	38.6	1591.1	28	38.6	5121.2	32	39.2
	36	37.0		36	36.6		32	36.3			
	37	36.4		40	34.5		36	34.1			
	40	34.9					40	32.1		40	32.1
CIF							24	39.8	3417.0	24	39.8
	28	39.1	257.7	28	38.7	1179.4	28	37.1			
	32	36.7		32	36.4		29	36.5			
	36	34.3		36	34.2		32	34.7		36	34.2
QCIF				20	42.4	1309.8				20	42.4
	24	40.2	152.9	24	39.72		20	41.8	1952.3		
	28	37.6		28	37.0		24	39.0			
	29	36.9		29	36.4		28	36.2		29	36.4

6. The Proposed System of Video Streaming Over IP Network

As a second part of the proposed system is how to get overall video streaming over Internet by integrating the encoder operation with the network protocols to examine the network performance and its effects on the video streaming quality for different video test sequences. One basic issue is the network path condition from server to client and impacts of video quality when it suffers from packet loss, latency, bandwidth congestion and delay jitter, this work focus on how transmit the video with low-Bitrate and with taking into account the network status.

6.1. The selection mechanism of suitable videos at the server

The Real Time Transport Protocol (RTP) and the Real Time Control Protocol (RTCP) are developed to distribute the multimedia over internet. So this work suggests using the RTP protocol that provides the functionality of transforming the real-time data over network but without flow control, while using the RTCP to give feedback about the network path status to allow this system to changing the video according to network condition. Both of these protocols support the availability of many-to-many communication, but for more scalability of applications a one-to-many architecture of distribution as video multicasting. Such network architecture tries to achieve better scalability and performance with suitable feedback of Quality of Service (QoS).

Figure 6 describes the scenario of distributed video with Real-Time streaming over IP network, where the RTP is based on underlying protocols and running over UDP.

Source server for video streaming in Sub-Net 1 and client at Sub-Net2 side using RTSP protocol for streaming and transmitting by RTP protocol after selects an appropriate level of video by using RTCP protocol according to network status. While the player requests the streaming content at the server, the server accept that by RTSP and send the video streaming to the client using RTP. At the server there is the network emulator tool that is using to monitor the network path from server to client and emulate the network parameters conditions.

With RTCP protocol that which gives the reports as feedback about QoS with reflected method that which broadcast the Sender Report (SR) to all participants in the one-to-many way, while the receiver generates the Receiver Report (RR) and send it by one-to-one way does not have ability to send with one-to-many. The sender (server) receives the RR packet from each receiver (client) in unicast connection (one-to-one way), so the sender listening to these reports and collecting them and releases to each member by using multicast group. The RR report has the QoS information that going to the source server to learn it about the channel status that helps it to select an appropriate level of the encoded video.

6.2. Methodology of network streaming system and tools

The network architecture shown in the Fig. 6 is applying in the Graphical Network Simulation (GNS3). The scenario is proposed in this work that based on the network topologies and using the protocols that now existing and implemented in several cases. This work using model c7200 of router.

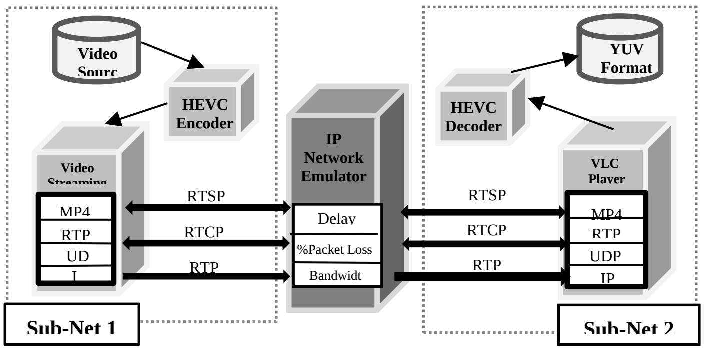

Fig. 6. The network configuration of video streaming.

The Qemu (Virtual Machine) template is used to install the server with Windows Server 2012 R2 by uploading its image file. The used memory is of 4096 MB with increasing one CPU of operating system type 64-bit. The used server has one adapter for network (can be increased) with Ethernet (type e1000) interface connection. Also installing the Client by the same minor. However, the CISCO router with c7200 is installed by Dynamips IOS router template with some properties.

6.3. The structure model of video streaming

The structure model of the proposed system includes two scenarios: Unicast and Multicast that, where their elements and how to be configured will be described in the following sub sections.

Network experiment-environment of unicast and multicast video streaming

This section represents the main components of the experimental set-up. The video is streamed through two scenarios; Unicast and Multicast Video Streaming. At the Unicast, as shown in Fig. 7, the video is streaming from source (server) to destination (client) as one-to-one way, through the channel (Router), while at the Multicast video streaming, as shown in Fig. 8, the video is streaming from source (server) to destination (four clients) as one-to-many way, through the channel.

The network test-environment as shown is a LAN`s structure that consists of streaming server, client and network emulator, see Figs. 7 and 8. While all hosts are connected by Ethernet and Fast Ethernet, where the bandwidth standard of Ethernet is 10 Mbps and the bandwidth standard of Fast Ethernet is 100 Mbps, the type of connections are based on the host interface and what the GNS3 supports in the used version of this work. The server uses the VLC-player software to stream an appropriate video sequence, which is prepared at six levels of resolutions. The GNS3 is network emulation tool that can be emulated for different conditions such as network`s delay jitter, loss and available bandwidth.

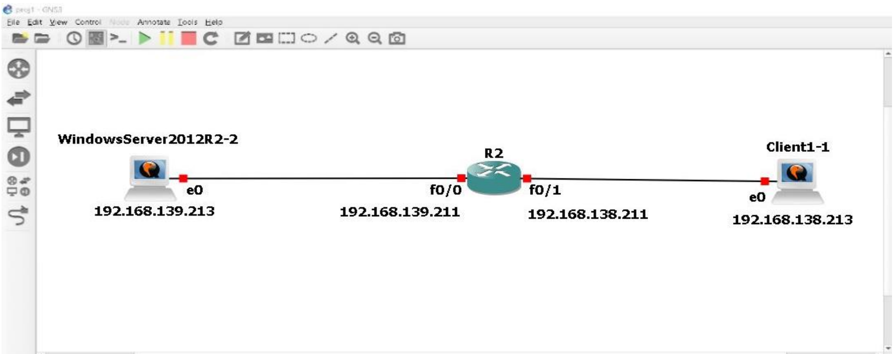

Fig. 7. Unicast network for video streaming over the internet.

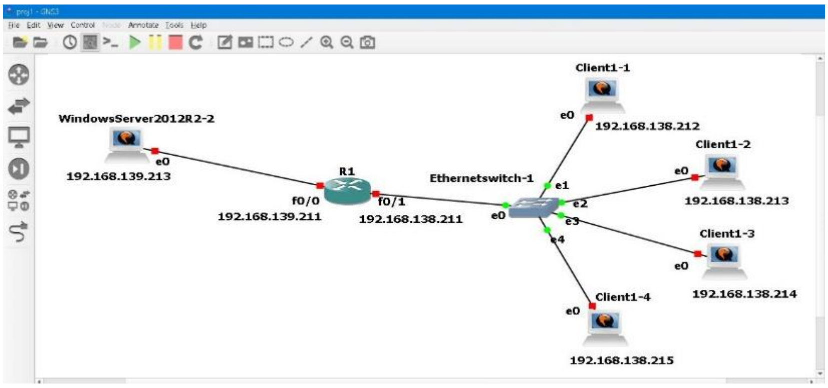

Fig. 8. Multicast network for video streaming over the internet.

6.4. Scenario of video streaming over IP network

The scenario of video streaming over the IP network in this work is consisting of the CISCO routing to perform the video multicasting by using VLC player to each client. The most important knowledge of multicasting is Protocol Independent Multicast (PIM). The real-time video streaming is achieved by using the VLC media server to multicasting the video from server to the group, the router that has been configured with IP addresses is of two interfaces (Fastethernet 0/0 and Fastethernet 0/1). The Fastethernet 0/0 is configured with IP address 192.168.139.211 and the subnet mask is of 255.255.255.0, also the Fastethernet 0/1 with IP address of 192.168.138.211 and the subnet mask is of 255.255.255.0. However, each interface of the network devices should be configured for unicast and multicast video streaming. In the topology of multicasting, the router is configured with Open Shortest Path First (OSPF) protocol to achieve the reachability by activate the backbone area (area 0 or area 0.0.0.0) to connect with all other area.

Therefore, in router configuration of multicasting IGMP group is performed on the (f0/1) interface of the router, it provides an open group, the range of the IP address of multicasting group was (224.0.0.0-239.255.255.255), the IP addresses from (224.0.0.0 to 224.0.0.255) are used for information multicast routing, while the other addresses are used for application programs. In this work IP address (224.4.4.4) will be used with command "ip igmp join-group 224.4.4.4" on interface f0/1 for multicast traffic on this network. After establishing of all these steps of video streaming over IP network by VLC player, practical experiments are executed to evaluate the reliability of the real-time video multicasting from server to 4-clients, for example see Fig. 9 that multicasting HoneyBee video to 4-clients.

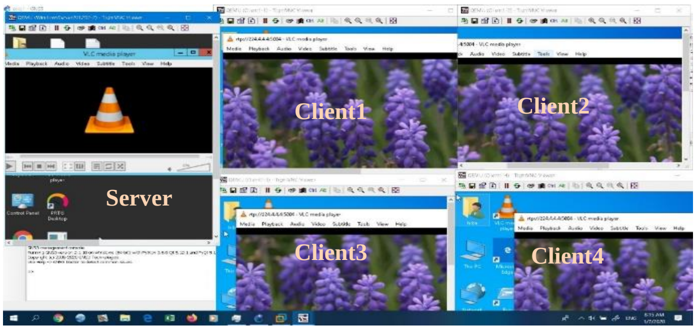

Fig. 9. Real-time HoneyBee video streaming from server to 4-clients.

6.5. Network performance methodology of evaluation

The methodology of network performance evaluation has two categories: (advanced investigation methodology and simple investigation methodology). Advanced reports the metrics like available bandwidth or narrow capacity of network path, while the simple investigation reports the metrics of packets like loss, delay and jitter. These network performance parameters have significant effect on the quality of video streaming, Consequently, this work focuses on how increasing the available bandwidth by decreasing the transmission video's bit-rate by H.265 encoding and select an appropriate level by RTCP protocol according to QoS reports, also measuring the values of network parameters. The Paessler Router Traffic Grapher (PRTG) network monitor applies these measurements, it is monitoring the information technology infrastructure and notifying administrator of system about the problem.

6.6. Video streaming over IP network evaluation

Different experiments are achieved to transmit the test video sequences that are used in this work. The encoded videos at different level of details are applied for both unicasting and multicasting network applications. The PRTG sensors are added to each device, where the used sensors are Simple Network Management Protocol (SNMP) traffic, ping jitter and ping sensor.

6.6.1. The experimental results of advanced and simple investigation methodology of unicasting network

The network uses the RTCP protocol to check the network conditions by reflected method. In this experiment, multi resolutions of video test sequence is streamed to achieve scalability of bitstream. The scalability is performed by finding the optimal configuration for codec's parameters of each resolution. At this work of unicasting video streaming, the 077 Ethernet sensor is used to report the results of data that transmit over this path and show how the available bandwidth changes with the different video resolutions, at the same time the ping jitter sensor and the ping sensor are used to monitoring the amount of the delay jitter and the latency, respectively. For example, Figs. 10-15 show the results for two videos, where the unit of time is in a minute for each figure.

Figures 10 and 11 describe the (077) Ethernet traffic sensor's results of advanced investigation methodology/bandwidth availability.

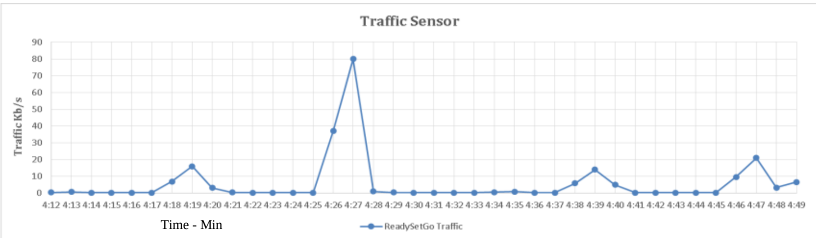

Fig. 10. The traffic of ReadySetGo video streaming over the internet.

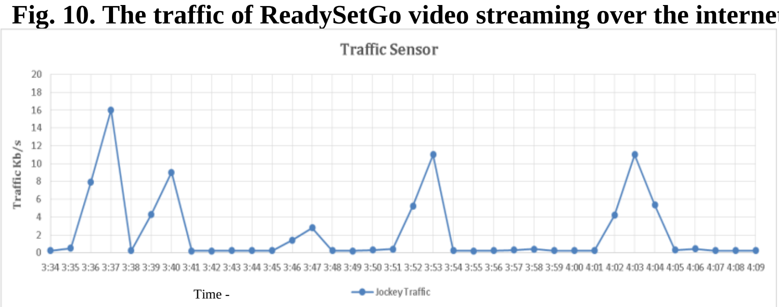

Fig. 11. The traffic of Jockey video streaming over the internet.

While Figs. 12 and 13 describe the ping Jitter sensor's results of simple investigation methodology/delay jitter.

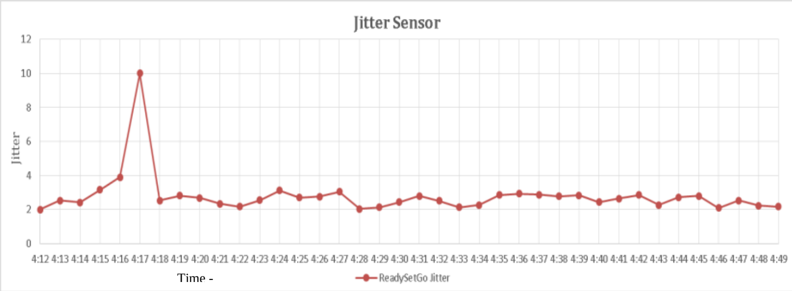

Fig. 12. The delay jitter of ReadySetGo video streaming over the internet.

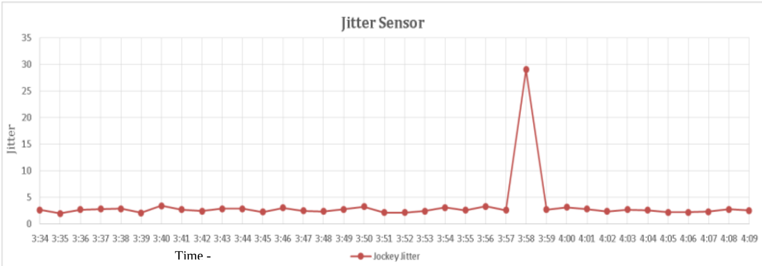

Fig. 13. The delay jitter of Jockey video streaming over the internet.

Figures 14 and 15 describe the ping sensor's results of simple investigation methodology/delay latency.

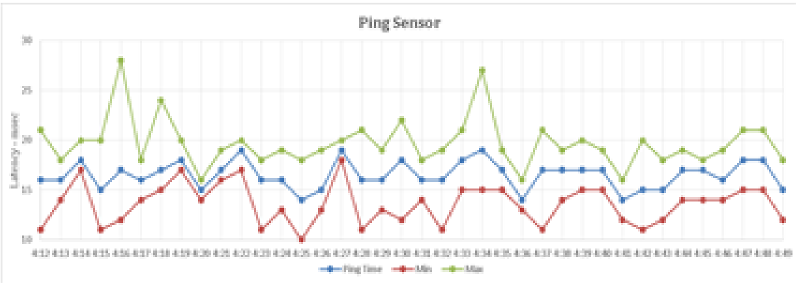

Fig. 14. The delay latency of ReadySetGo video streaming over the internet.

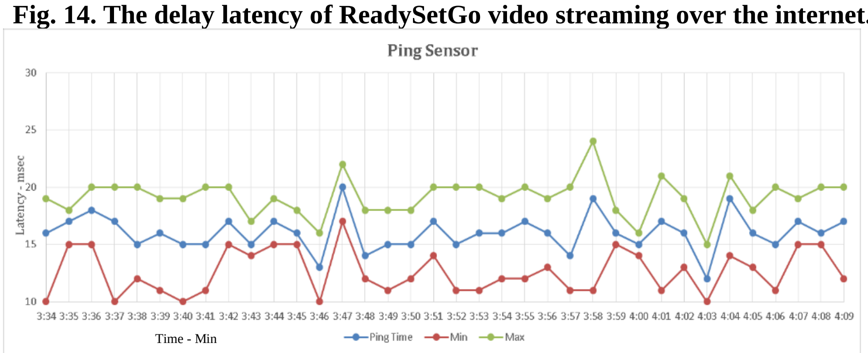

Fig. 15. The delay latency of Jockey video streaming over the internet.

6.6.2. The experimental results of advanced and simple investigation methodology of multicasting network

These experiments describe the variation for results obtained between different categories of video. This variation of results is the main motivation to produce the QoS map of the network with all types of videos such as that results breakdown in this sub section. The three categories of motion details for their test sequences have different behaviour, where the high motion details are suffered from packet loss, latency and delay jitter more than the other two kinds of video, so the poor network has more negative impact on this kind of category.

At the experiments of advanced investigation methodology about the network bandwidth availability, according to the packet size that transmitting over the multicast network path the SNMP sensor (007 Ethernet traffic) reports the IPv4 packet. However, when the available bandwidth is share among number of clients and the number of users is increasing over the network that led to network congestion, so the controller selection of appropriate level of video less than UHD to be transmitted over the network is the solution. Figure 16 shows the results of network traffic when the ReadySetGO test sequence is sharing among the four clients at the same time. The lower level has Bitrate little than higher level (UHD) supports the idea of increasing the bandwidth availability for other users to receive the real-time video streaming.

While in the simple investigation methodology that measures the network performance parameters, in this work the PRTG sensors are used to monitoring the delay jitter and latency, they are increasing with multicasting video streaming among clients as that shown in Figs. 16-23. These parameters are effect on the video quality that transmit over the Internet especially with high spatial resolution and motion details. The videos with high and medium motion details (ReadySetGo and Jockey test sequences) are observed at sensors and described, where the traffic events are recorded and displayed by the figures below and the unit of time is in a minute for each figure.

Figures 16 and 17 described the (077) ethernet traffic sensor's results of advanced investigation methodology/bandwidth availability.

While the Figs. 18 and 19 describe the ping Jitter sensor's results of simple investigation methodology/delay jitter.

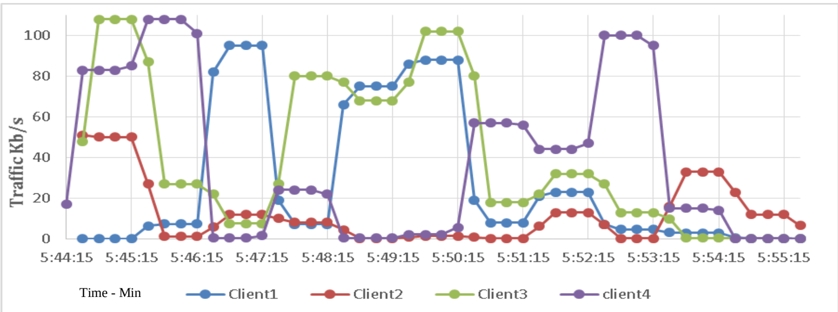

Fig. 16. The traffics of client (1, 2, 3 and 4) for ReadySetGo video streaming.

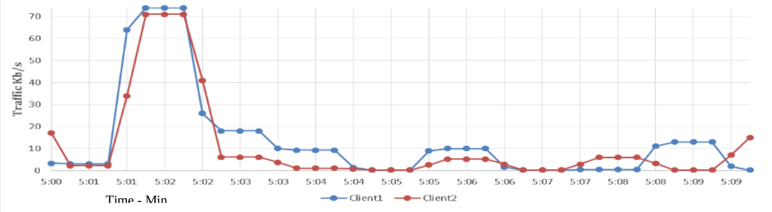

Fig. 17. The traffics of client (1and 2) for Jockey video streaming.

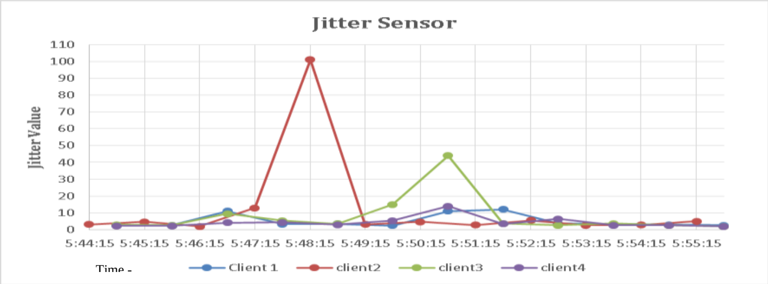

Fig. 18. The delay jitter of client (1, 2, 3 and 4) for ReadySetGo video streaming.

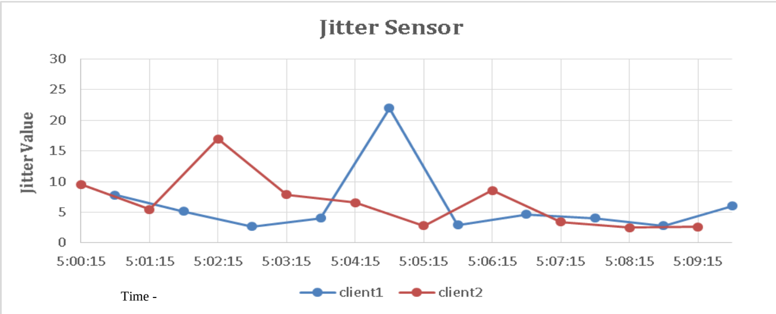

Fig. 19. The delay jitter of client (1and 2) for Jockey video streaming.

Figures 20-23 Ping sensor: the results of simple investigation methodology/delay latency.

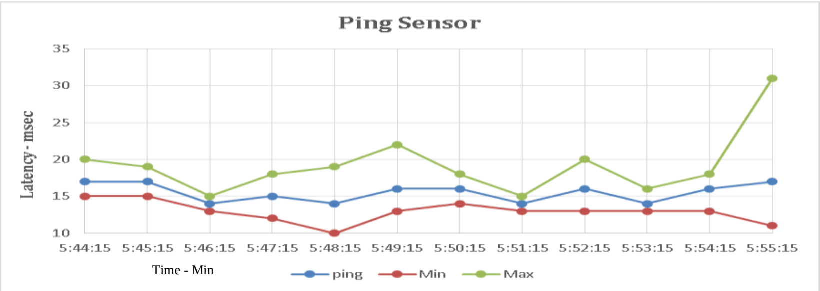

Fig. 20. The delay latency of client1-ReadySetGo video streaming.

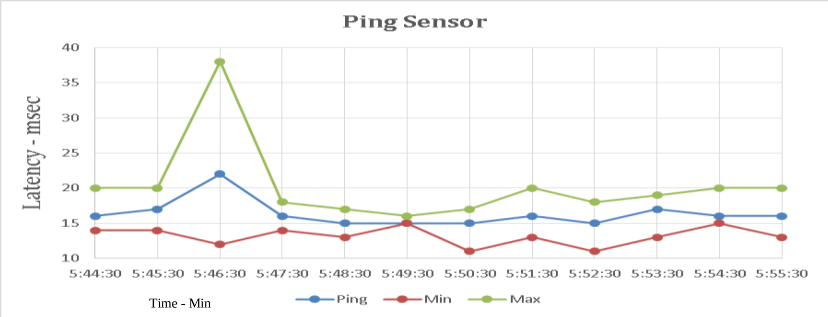

Fig. 21. The delay latency of client2-ReadySetGo video streaming.

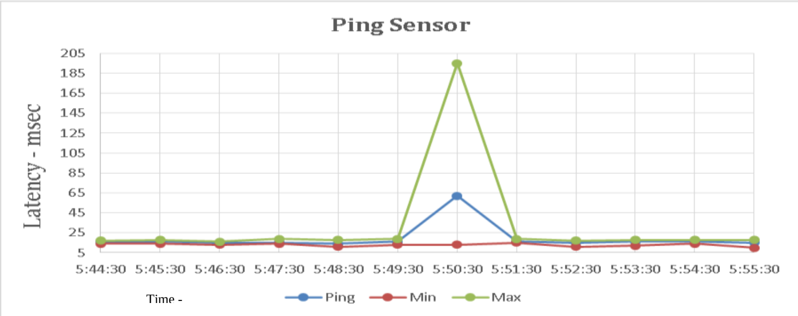

Fig. 22. The delay latency of client3-ReadySetGo video streaming.

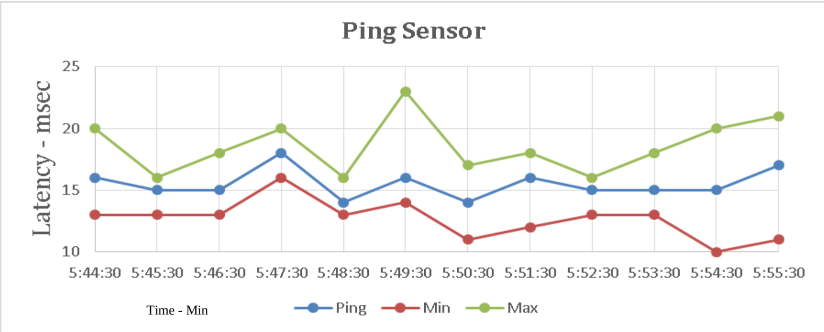

Fig. 23. The delay latency of client4-ReadySetGo video streaming.

7. Network Performance Results Discussion

The results as illustrated for the experiments of unicast and multicast video streaming over the internet give global scope of determining the map of QoS for the network performance. Generally when the users' number is increasing the bandwidth availability is decreasing, which is the important QoS situation especially when the UHD is transmitting.

Throughout the experiment of this work at the unicast network with low details video streaming over the network, the maximum value of the total traffic is reached to 10 Kb/s according to the bandwidth sensor's reports, while the video medium details video's packet is reached to 16 Kb/s, but the high details video's packet is reached to 80 Kb/s. Also the values of the jitter range from 1 to 5 for their videos and the results from the ping sensor represents the values of latency when received the data at the client, the values of the ping time range from 12 msec to 26 msec, while the maximum of latency at the ReadySetGO is 28 msec, the ReadySetGo video suffers from delay values more than other two types of videos.

The results obtained from the multicasting network when streaming three categories of the video content from server to four clients as shown in Figs. 16-23 above, the max value of the packet was achieved to 25 Kb/s for HoneyBee and 74 Kb/s at the Jockey while at the ReadySetGo achieved to 108 Kb/s. The jitter values at four client's range (2-22) for the HoneyBee test sequence, at the jockey the jitter range (2-22), while the range of jitter at ReadySetGo with high motion details (2-101), it suffers from jitter more than other two types of videos also the delay at this video achieved to 195 msec more than other two videos.

Clearly, the significant different between the results of bitrate, jitter delay and delay are due to the different video's content categories that is reflected on encoding scheme, buffer size, video (duration, size, frame rate and bitrate), and finally number of end users at the network.

8. Conclusions

Throughout the implementation of the proposed video streaming over the Internet system and the measurements methodologies, there are several conclusions have been taken based on the practical results.

In this study an adaptation of the UHD video streaming over the Internet when transmitting to the clients is the goal due to limitation in bandwidth when the number of users is increasing on the network. The work takes the problem solution in two topics, encoding and Internet streaming. The streaming of video with accepted range of PSNR of (32-40) is the point strength of the proposed encoding system that is directly reflected to solve the problem of Internet sharp fluctuations.

So this paper dealing with controlling the streaming of and video encoding and network performance evaluation by determining the network parameters. The system performance is optimized when the adaptive compression and adaptive layer selection are applied. The optimal configurations of codec's parameters and intelligently select the video's level of spatial resolutions based on the network bandwidth availability result of benefits to end users video quality.

The proposed algorithm experiment results show the processing time reduction and improving of the received video quality.

Nomenclatures

M	Width of picture
MAX	Maximum value of pixel
MSE	The value of Mean Square Error
N	Height of picture
P_{anchor}	Single pixel of raw picture
PSNR	The quality of video sequence
P_{test}	Single pixel of compressed picture

Abbreviations

4k	4000 pixels (UHD)
BR	Bit Rate
CIF	Common Intermediate Format
dB	decibel
FHD	Full High Definition
GNS3	Graphical Network Simulation
GoP	Group of Picture
HD	High Definition
HDTV	High Definition Television
HEVC	High Efficiency Video Coding
IOS	Internetwork Operating System
IP	Internet Protocol
IPTV	Internet Protocol Television
ISO	International Organization for Standardization
OSPF	Open Shortest Path First
PRTG	Paessler Router Traffic Grapher
PSNR	peak signal-to-noise ratio
QCIF	Quarter Common Intermediate Format
QoS	Quality of Services
QP	quantization parameter
RR	Receiver Report
RTCP	Real-Time Control Protocol
RTP	Real-Time Protocol
RTSP	Real-Time Streaming Protocol
SNMP	Simple Network Management Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UHD	Ultra-High Definition
VLC	Variable Length Coding
YUV	Luminance-Bandwidth-Chrominance

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