

EVALUATION BROADCASTING VIDEO SIGNALS BY WIRELESS NETWORK PARAMETERS BETWEEN CLIENT/SERVER

Media Sabah Azzat¹ and Ergun Erçelebi²

¹ Department of computer science, College of Science, University of Zakho, Kurdistan region – Iraq.

² Dept. of Electrical and electronics Engineering, Faculty of Engineering, University of Gaziantep, Turkey.

(Accepted for publication: August 16, 2015)

Abstract:

This paper focuses on designing and implementing a system for recently emerging video transmission by wireless networks parameters. Such parameters are considering packet loss rate, bandwidth and delay, as well as the quality of server (QoS), which eventually contributed to the user's the quality of experience (QoE). The signals of the captured video by cameras send to the server-host then to be displayed and broadcasted to the client-host. This system overcomes the several problems as: delay, packet loss rate and bandwidth by resources and destinations through controlling the permissions and broadcasting, as well as to the monitoring the configuration of the system during broadcasting. The advantage of this system is to improve application software to be used in a friendly use with as maximum as possible of flexibility and full optional controlling to get complete features of the video broadcasting systems. The system controls the problems of delay, loss rate and bandwidth between source and destination, through controlling the permissions, broadcasting and monitoring when configuration of the system during broadcasting. The synchronization between two sides of the wireless network was achieved when incoming signals from the cameras to the server-side and broadcasting these signals to the client-side. Eventually this system enables the administrator to monitor the dataflow from server-side to client-side and apply compressing with many compression techniques on the incoming and broadcasted signals.

Keywords: Video Broadcasting, Client/Server, Quality of Service, Compression Techniques, Quality of Experience and wireless network.

I. INTRODUCTION

The increasing used for video transmission applications has generated a great deal of interest in demand of end users for higher quality videos. Video files is an example for contain a mixture of everything such as: text, graphics, audio, images, metadata and, animation data. It is finding today used in many applications such as surveillance, video phones, video conferencing, video streaming, entertainment, distance learning, tele-medical applications, sports, security, education programs and video on demand (Kalaivani R. & Manicha C., 2013). The use of QoE requirements for users with wireless environments has been earning interest. The main cause to be success of the new multimedia applications is QoE or the perceived quality of streamed videos. The user's perception of service quality is the prime criterion for quality evaluation of multimedia applications (Asiya K. et al., 2010).

II. RELATED WORK

In order the proposed system depended in this paper to the related works, which are:

In 2002, Servetto et al. (2002), proposed a new architecture for a video multicast system, they presented the design of the different components of this system, and showed results obtained in a real implementation. Their feeds consisted of video encoded. Decoding was performed on Linux PCs, and the quality of their reconstructed signals degraded gracefully with the speed of the CPU on which the receiver runs.

In 2009, Hinds (2009), designed and implemented a video conference system over the hybrid type peer-to-peer networks. The proposed system has hybrid type Peer to Peer networks based client/server, where client/server was used for interchange of account management, the real time video conference was used to client list and situation information and Peer to Peer. The traffic of server side needs to reduce it, and reduce the load of a network because the multimedia data was decentralized by hybrid peer-to-peer networks

to client side. Also, through high speed networks the proposed system was implemented and checked by the multi-party video conference system using application software and communication protocol.

Khan et al. (2009), presented the content-based video quality prediction for MPEG4 video streaming over wireless networks. The video is transmitted over mobile/wireless networks, there are many parameters that affect video quality but their conjoint effect is not well identified and understood. The content of video has an impact on quality of video under same network conditions. The work help in the development of a reference-free video prediction pattern and QoS control methods for video over wireless/mobile networks.

In 2011, Hidayat & Wilson (2011), provide a basic concept Impact of ad hoc network parameters and conditions on video conferencing quality. The ad hoc wireless Network has limited bandwidth and suffers from errors caused by highly mobile nodes as well as the absence of a dedicated routing device. The video conferencing applications need to observe the effect of various ad hoc wireless network parameters and conditions on the quality of video. They have to demonstrate that bandwidth, congestion and video rate significantly affect the video quality of video conferencing sessions on ad hoc wireless networks.

Halkawt R. (2013), the designed and implemented of a proposed system for broadcasting video-signals from multi-sources to multi-destinations using fundamentals of clients/server host and clustering. The system achieved the improved application software to be used friendly with flexibility and optional controlling to get complete features of the video broadcasting systems. The administrator to observe the dataflow from server-side to clients-side and using compressing with many compression techniques on the incoming from server-host for broadcasted signals. Also, the server-side is responsible on supervision full operation transmission video broadcasting and using network local area networks (LAN).

In 2014, Bruzgiene et al. (2014), provided impact of video streaming on quality of experience over 802.11n dual band networks with experimental evaluation. Can be access even video broadcasting services, however, video

streaming over wireless networks is especially difficult due to the factors, influencing the wireless communication process: the signal force, the interference from other wireless links. It is necessary to analyze the parameters of video streaming over 802.11 wireless networks, evaluating the influence of different frequency bands on the characteristics of video quality of experience: mean opinion score, peak signal to noise ratio and others.

Ibrahim M. (2014) presented a proposed system modified video broadcasting. The thesis was built depending on the provided multilevel client/server principles with peer-to-peer technology for video broadcasting system. The system included multi-sources connected with IP-cameras via main-server to destinations/sources CSs which work as middle-level, to clients as final destination-level.

In conclusion, the proposed system of the current paper provides all the facilities stated in the previous works that can be considered as an aggregate system of all the previous works. Hence, this system has source and destination, receives signals from server and broadcast this signals to client and transmission video conference by using parameters for wireless networks. Also, many other flexible options are provided with proposed system in order to be more efficient and powerful.

III. VIDEO BROADCASTING

To systematic as a sequence or images of frames that are sent to the subscriber and displayed at a constant frame rate is video streaming the video content. The video ingredient is coupled with a multi-channel audio that is also structured as a series of audio frames which is included in the video content. While there are different transmission and buffering rate from the network and the client station video player streaming real-time video. The video content might be characterized by many parameters including pixel color depth, video format, coding scheme and frame inter arrival rate (Gill et al., 2011). Dissemination video of real-time has loss rate, delay, and bandwidth (Waqas D., 2010):

- **Rate:** Packet loss rate can probably make the presentation annoying to human eyes, or, in some cases, make the presentation impossible. Many

different type of loss might be happened. To achieve adequate visual quality, the packet loss ratio is required to be kept below a threshold. Although, real time video has a loss requirement, the current internet does not give any loss assurance. In particular, the packet loss ratio could be very high during network congestion, leading to critical degradation of video quality (Waqas D., 2010).

• **Delay:** To understand delay, the server sends data to client, and goes into a state where it listens to Acknowledgment (ACK) receipt of data from client. After this operation the current time on server computer is going to be considering Time1. Client receives server data and sends an ACK to sever. Server receives ACK and gets current time, which is Time 2. Time delay is assumed to be: $(\text{Time } 2 - \text{Time } 1)/2$ (Halkawt R., 2013). This is usually not subject to stringent delay restraints; real-time video needs a bounded end-to-end delay. That is, every video packet must arrive at the destination in time to be decoded and displayed. Because real-time video must be played out in a timely fashion, if the video packet does not arrive on time, then it is useless and can be considered lost, because its time slot for being played has been passed. Although, real-time video requires timely delivery, the current internet does not offer such a delay guarantee. In particular, the congestion in the internet could provoke an excessive delay, which exceeds the delay requirement of real-time video (Waqas D., 2010).

• **Bandwidth:** To realize quality of presentation, sending of real-time video contents demand a minimum bandwidth. The congestion occurs when the sender disseminates multimedia contents faster than the available bandwidth, which cause a drop in video quality and ultimately induced a packet loss. On the other hand, the available bandwidth then receives produces sub-optimal video quality if sender transmits slower than that. Moreover, traditional routers typically do not actively participate in congestion control, a congestion collapse can cause from immoderate traffic, which can further degrade the throughput of real-time video (Waqas D., 2010).

IV. CLIENT/SERVER

Although, client/server is still developing as a form of computing, there are some essential concepts. The client/server is presenting feature a single logical picture to the user, because it contains cooperative processing capabilities that physically split the processing performed by the client from that performed by the server. They must have processing capabilities and make requests to servers therefore client work-stations need to be intelligent. A set of special tasks for any other device requesting their services doing that the servers perform. The work-stations and servers connect together by networks. To form the client/server architecture ties the components together by software applications (Guynes & Windsor, 2011). A client is any process that request exact services from server process. The client is known as the interface application, usually the end user react with the client process. The server is any process that request exact services to the client and also supports multiple and synchronize with the client request. The server is known as interface application, usually that server process provides the base services for the client process (Subhash, 2009). Fig1 shows the client/server single.



Fig.1: The Client/Server Single (Bruzgiene et al., 2014).

V. VIDEO COMPRESSION

The compression is a reversible transformation of data to a format that demand fewer bits, that the data can be stored or transmitted more efficiently. The data size in compressed shape (C) close to the original size (O) the ratio of compression is $(R=C/O)$. The opposite of this operation, decompression, achieve an origin replica from the original data then the compression is lossless. Lossy compression, used on image data, the

original image does not allow reproduction of an exact replica from it, but has a higher compression ratio. Then lossy compression allows only an approach of the original to be created. For image compression, the fidelity of the approximation decreases as the compression ratio increases. The success of data compression depends on the data itself and some data types are inherently more compressible than others. Usually most compression algorithms exploit this property and some elements within the data are more common than others, known as repetition. The more successful the compression of the data is possible to be, greater the redundancy within the data. Fortunately, digital video contains a great deal of redundancy and thus is very suitable for compression. The compressed data are often known as an encoder or coder, while a device that decompresses data is known as a decoder. A device is known as a codec that acts as both a coder and decoder (Biswas, 2008).

VI. Proposed System Components

The system is organized with two hardware-sides (server and client). The server-side consists of one host with N-cameras; the client-side has one host. The server-host is connected with three cameras, with an acceptable resolution degree. The first camera is a webcam of a laptop live video; other cameras are joined with the server by USB cables. The server-host function is controlling for all operations in the system, in order to start IP numbering with the server-host for connections to the client-host. Ordinarily, the client-host must have facilities and robustness of the treating with video signals. However, it is important that the server-host has facilities higher than the client-host, and then connected by a wireless network. Fig. 2 illustrates the organization of the proposed system:

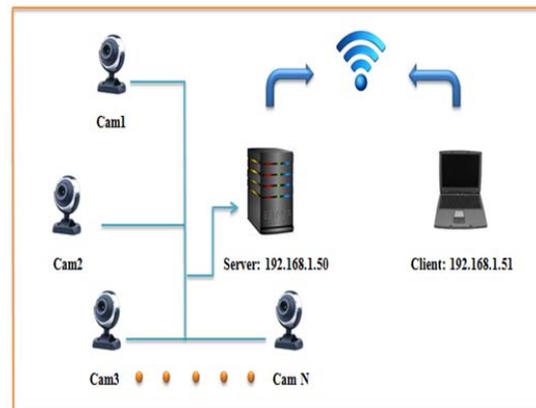


Fig.2: Organization Structure of the Proposed System.

VII. Designing of The Proposed System

The proposed system is designed to get a successful broadcasting video-conference transmission by wireless network parameters to be achieved in an efficient and robust way. This is useful to provide service of the sending/receiving video signals. The general structure of the proposed system consists of seven stages: initializing server host (ISH), detecting connected cameras (DCC), initializing connected client-host (ICCH), detecting connected client-host (DCCH), start broadcasting (SB), controlling-option (CO), and system interface map (SIM). Fig. 3 shows the main stages of the general structure for the proposed system:

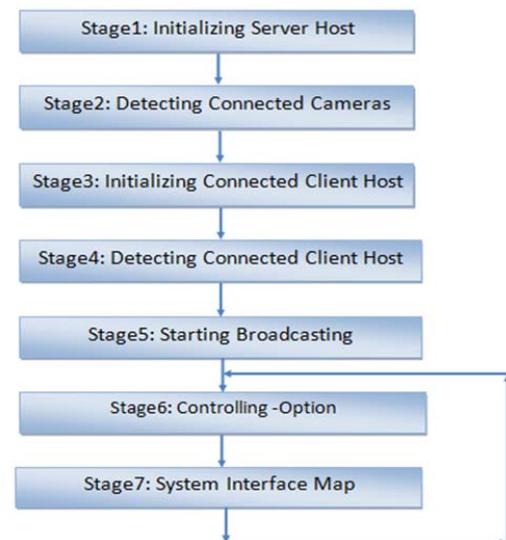


Fig.3: Main Stages of the Proposed System.

Initializing server host is first stage of the proposed system, which is responsible for function the server-host and it is lead each of cameras and client side. When the ISH-stage is complete the second stage is started detect the connected cameras host. The ICCH-stage is responsible for initialize to connect with client-side, then to ready for receiving video signals that incoming from server-side. The client-host job is only to display the received signal therefor the software tools is used in the client-side similar to the server-host, but without compression techniques because there is no need for compressing techniques. After client initialization stage is complete, the server starts to detect the connected client-host. The SB-stage will be started of the viewer to the client, when the DCCH-stage is correct completion. This stage is described the controlling option from the server-side related with the broadcasting signals sent to the client-side in the viewer-window. Server software automatically detects information about the wireless network of the system. The administrator can be monitoring the CO-stage by selecting CO-button which it will appear on the displayed viewer-window for detected camera. The finally stage of the general structure is SIM-stage provides the capability of the interfacing between server and client side. Start server broadcasts video signals of any detected camera for detected client, the interface map will path by connecting between them with a line on the interfacing map as an indicator to starting of broadcasting signal.

VIII. Implementation of The Proposed System

The main focus on server-client communication provide socket listen on a signal port for client. Initial connect is by the combination of two sides through IP-port pair then ability to broadcasting video signals form server-side to the client-side. The compression techniques used to reduce the size of video signals in the hard disk or broadcasting video signals to exploit the network bandwidth. The server processor has ability to know the size of data when operation sent and received. As well it will need a connection map to responsible for communication and sending/receiving data of the video signals. The transmission control protocol

and the internet protocol (TCP/IP) are defining as protocol set because it contains many various protocols and therefore many various ways for computers to connect between them. A TCP/IP socket protocol considers endpoint communication of the specific connection. It is defines by IP protocol and a port which is a particular TCP connection or the listening state. A TCP connection is actually identified by the tuple: (source_address, destination_address, source_port, destination_port). The wireless communication synchronous to a provide service endpoint. This is because connect is specific from both of its local and remote endpoints, allowing traffic to be routed to a service. The TCP server depends on the window's sockets, to easier work. The server agrees client communications and can send and receive data. The server spawns a listen for client and it can accept as connections. This GUI-form consist of viewer window with options facilities for system controlling. When the initialization operation is completed, the server will display a GUI-form to support the administrator to control the incoming/outgoing video signals with high facilities, firm technique and useful manner. Fig4 Shows GUI-form of server-host:



Fig.4: GUI-form of Server-Host.

The client-host can be receiving video signals that incoming from server-host. The client-host job is only to display the received signal. Therefore, the software tool is used in the client-side similar to the server-host, but without compression techniques because there is no need for compressing techniques. The GUI-form for client-host consists of window that has camera light as an indicator with three-color options: Red, Yellow and Green. The client-side can receive

video signals and display on viewer, then send acknowledgment to the server-side and it can compute result the data from create result table. Fig5 shows the display GUI-form for client-host system:

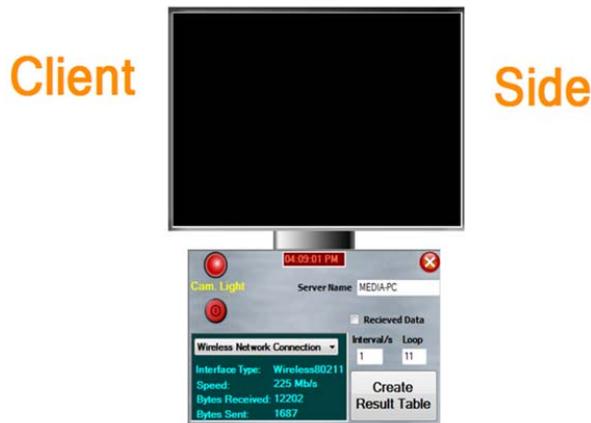


Fig.5: Display GUI for Client-Host System.

IX. Results of The Proposed System

The administrator can calculate and observe the dataflow rate, then choose one camera that sending to client-host. The data of video signals flow from server-host to client-host through (10 second) with different resolutions and with full quality. Fig6 illustrate data flow. The data received of video signals at client-host through (10 Second) with different resolutions and with full quality as shown in fig 7. Table1 describes data sending without compression techniques. While Table2 reporting data receiving with full quality and without compression.

Table1: Data Sending with Full Quality and without Compression.

Time (Second)	Resolution (160*120)	Resolution (224*240)	Resolution (640*360)	Resolution (640*480)	Resolution (1280*720)
1	158	852	1196	1929	3130
2	333	1622	2524	3557	5345
3	490	2389	3126	5026	7296
4	648	3256	4123	5902	9956
5	822	4035	5176	7378	11691
6	979	4813	5868	8378	13162
7	1135	5678	7128	9423	14789
8	1306	6455	7929	11013	16298
9	1465	7235	8775	11989	18363
10	1619	8039	10013	13326	20935

Table2: Data Receiving with Full Quality and without Compression.

Time (Second)	Resolution (160*120)	Resolution (224*240)	Resolution (640*360)	Resolution (640*480)	Resolution (1280*720)
1	157	831	1174	1557	2976
2	315	1602	2498	3478	5530
3	473	2380	3104	4445	7768
4	647	3235	4098	5883	9454
5	804	4015	5148	6976	11392
6	961	4794	5847	7976	12682
7	1132	5638	7004	8647	13972
8	1291	6438	7886	9812	15401
9	1446	7218	8743	10364	16561
10	1600	8009	9986	11762	18702

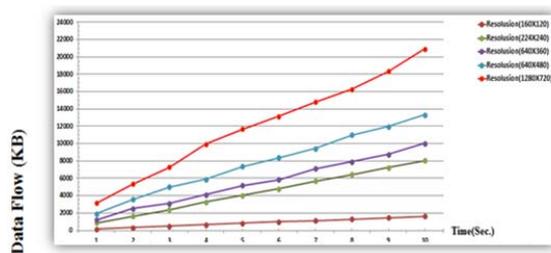


Fig.6: Data Flow of video signals from server-host to client-host and with full quality.

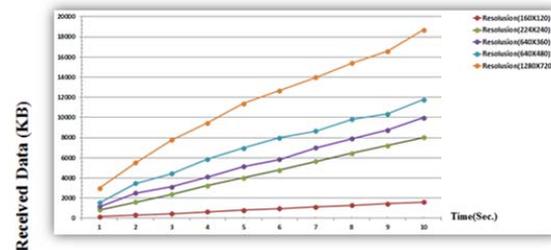


Fig.7: Received Data of video signals at client-host and with full quality.

The result of the dataflow for broadcasting video signals from server-side to client-side using compression techniques and full quality are mentioned in table 3: JPG compression, DivX 6.8.5, Cinepak by radius, MPEG compressor, Microsoft RLE video codec and full quality without compression through (10 seconds), with default resolution (640x480). Fig8 describes all specifics for compression techniques by suitable curves. Table4 compression ratio for dataflow (JPG codec, DivX 6.8.5 codec, Cinepak codec by Radius, MPEG compressor and Microsoft RLE) with default resolution (640x480 pixels) through 10 seconds. Fig9 illustrates all specifics for compression techniques by suitable curves by

compression ratio of the broadcasting video signals for (JPG codec, DivX 6.8.5 codec, Cinepak codec by Radius, MPEG compressor and Microsoft RLE) with default resolution (640x480 pixels) through 10 seconds.

Table3: Dataflow for Compression Techniques with Full Quality.

Time (Second)	Full Quality (Original)	JPG Compression	DivX6.8.5 Code (4LogicalCPUs)	Cinepak Codec By Radius	MPEG Compressor	Microsoft RLE
1	1929	104	113	83	418	181
2	3557	196	224	185	757	362
3	5026	289	335	257	1095	642
4	5902	383	423	313	1429	906
5	7378	492	514	368	1774	1199
6	8378	591	616	436	2122	1390
7	9423	692	734	489	2459	1581
8	11013	803	847	550	2796	1695
9	11989	901	971	609	3136	1862
10	13326	1002	1068	665	3470	1967

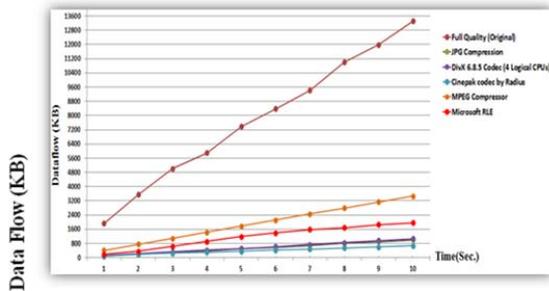


Fig. 8: Dataflow of broadcasting video signal of Original, JPG, DivX 6.8.5, Cinepak by Radius, MPEG Compressor and Microsoft RLE video codec.

Table 4: Compression Ratio for Dataflow.

Time Sec.	Full Quality Original	JPG Comp.	CR% (JPG)	DivX6.8.5 Code 4Logical CPUs	CR% DivX 6.8.5 Codec	Cinepak codec by Radius	CR% Cinepak Codec by Radius	MPEG Comp.	CR% MPEG Comp.	Microsoft RLE	CR% Microsoft RLE
1	1929	104	94.61	113	94.14	83	95.70	418	78.33	181	90.62
2	3557	196	94.49	224	93.70	185	94.80	757	78.72	362	89.82
3	5026	289	94.25	335	93.33	257	94.89	1095	78.21	642	87.23
4	5902	383	93.51	423	92.83	313	94.70	1429	75.79	906	84.65
5	7378	492	93.33	514	93.03	368	95.01	1774	75.96	1199	83.75
6	8378	591	92.95	616	92.65	436	94.80	2122	74.67	1390	83.41
7	9423	692	92.66	734	92.21	489	94.81	2459	73.90	1581	83.22
8	11013	803	92.71	847	92.31	550	95.01	2796	74.61	1695	84.61
9	11989	901	92.48	971	91.90	609	94.92	3136	73.84	1862	84.47
10	13326	1002	92.48	1068	91.99	665	95.01	3470	73.96	1967	85.24

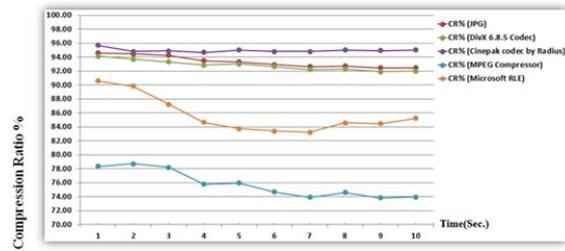


Fig.9: Compression Ratio of the Dataflow (JPG codec, DivX 6.8.5 codec, Cinepak codec by Radius, MPEG compressor and Microsoft RLE)

X. Discussion

In this paper, the proposed system provides facilities stated in the previous works that can be as an aggregate system to have source and destination by wireless network. Also, receiving the incoming video signals from the source and broadcasting them to the destination, using compression techniques, sending added texts, controlling and monitoring the system immediately. This system can send a dataflow rate of (3130 KB/Sec.) and receive a data rate of (2976 KB/Sec.) with normal frame rate, a loss ratio with data sending and data receiving of (4.95 %) and video compression of (305 KB/Sec). There is no such previous work exactly. So, in order to make a comparison between the results depended in this work and those of the previous works, there are several works as follows:

1. In (Servetto et al., 2002), resource and destination used with a real implementation. The capacity of video encoded at about 366 KB/sec.
2. In (Halkawt R., 2013), the dataflow of broadcasting signal of one-camera to one-client. The system can send a dataflow rate of (1997 KB/Sec.)
3. In (Ibrahim M., 2014), included multi-sources connected with IP-cameras to destinations/sources CSs. The system sends a dataflow rate of (1214 KB/Sec.).

XI. Conclusion

The most important of conclusion from an efficient system has shown the ability of connecting of source to destination for broadcasting video-signals is designed and implemented via wireless network depending on

the principles of server/client. The software implementation of this system is structured with full flexibility, robustness and friendly utilized with the capabilities of compressing and text addition to the broadcasted signals. Send (control and data) messages are dependent on enabling the server for receiving signals from the connected cameras and apply the necessary processing on them, as well as, transmitted audio signals with video, then broadcasting them to the client-side. It provides the ability of connecting N of resources (cameras) via destination (client) depending on the capabilities of the dedicated interface channel. The obtained results, it becomes obvious that the rate of dataflow has increased with different resolution, especially using compression techniques.

XII. REFERENCES:

- Asiya K., Lingfen S., Emmanuel J., & Emmanuel I. (2010). QoE-driven Adaptation Scheme for Video Applications over Wireless Networks. *IET Communications Special Issue on Video Communications over Wireless Networks*, 4(11), 1337-1347.
- Biswas, A. K. (2008). *Performance Analysis of Video Compression Algorithms for Transmission over Network*. (PhD), Indian Institute of Technology, Kharagpur.
- Bruzgiene, R., Narbutaite, L., Adomkus, T., & Cibulskis, R. (2014). The Impact of Video Streaming on Quality of Experience over 802.11 n Dual Band networks: Experimental Evaluation. *Elektronika ir Elektrotechnika*, 20(6), 158-161.
- Gill, R., Farah, T., & Trajkovic, L. (2011). Comparison of WiMAX and ADSL performance when streaming audio and video content. *OPNETWORK*.
- Guynes, C. S., & Windsor, J. (2011). Revisiting client/server computing. *Journal of Business & Economics Research (JBER)*, 9(1).
- Halkawt R. (2013). *Design and Implementation of a Proposed Video-Broadcasting System*. (MSc), University of Zakho.
- Hidayat, A., & Wilson, C. (2011). Impact of Ad Hoc Network Parameters and Conditions on Video Conferencing Quality. *International Journal of Video & Image Processing and Network Security IJVIPNS-IJENS Vol: 11 No, 1*, 21-27.
- Hinds, J. A. S. (2009). *Real-time video streaming using peer-to-peer for video distribution*. (MSc), University of Pretoria.
- Ibrahim M. (2014). *Modified Approach for Video Broadcasting System*. (MSc), University of Zakho.
- Kalaivani R., & Manicha C. (2013). Object Detection in Video Frames Using Various Approaches. *International Journal of Advanced Research in Computer and Communication Engineering*, 2(9), 157-160.
- Khan, A., Sun, L., & Ifeachor, E. (2009). Content-based video quality prediction for MPEG4 video streaming over wireless networks. *Journal of Multimedia*, 4(4), 228-239.
- Servetto, S. D., Puri, R., Wagner, J.-P., Scholtes, P., & Vetterli, M. (2002). *Video multicast in (large) local area networks*. Paper presented at the INFOCOM 2002. Twenty-First Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE.
- Subhash, C. Y. (2009). *An Introduction to Client/Server Computing: New Age International Pvt Ltd Publishers*.
- Waqas D. (2010). *Distribution Agnostic Video Server*. (MSc), School of Information and Communication Technology.

پوخته:

نه فو كه كو لینه جه ختی لسه ر دیزاینکرن وجیبه جیکرنا نوپترین سیسته می فو گوهاستنا فیدویا بکارئینانا پرا میتزین تورا وایرلیسی. پرامیتزین ژفی جورى ریژا ژدهستدانا پاکجی نیشاندا چارچوئی شه پولاوگیروبونى به رچا فو ووردگریت. وزیده بارى EOS نه وی پالپشتى QOE بوبه کار هیته رى. نه وسیگنالین فیدویانه وین هاتینه گرتن بریکا کامیرادهینه هنارتن بوویستگه هاییشوازی کرنی وپاشی دهینه هنارتن بوبه کار هیته رى. و نه فو سیسته مه کونترول جوره ها ناریشان دکه ت: گیروبون، ریژاژدهست دانا پاکجی. نیشاندا چارچوئی شه پولاوژژیده روسه رچاوبریکا کونترولکرنه په خشی. وهه ردیساچا فیدیرکناسیسته می دده می په خشکرنی دا. باشیافی سیسته می باشترکرناجیبه جیکرنا پروگراما بوبکارئینانوی بشیوه یه کی باش وب زیده ترین ناستی گونجاندى و کونترولکنی ژبودهست فه ئینانا بلندترین ناستی په خشکرنی نه فو سیسته مه. نه فو هه ما هه نگیه دنایه رادووتورین وایرلیس دهاته نه نجامدان ده می سیگنال دهاتن ژکامیرى بولایى سیرفه رى وپه خشى دودره نه نجامی فی چه ندی نه فو سیسته مه شیانی دده ته ریقه به رى بو چا فیدیرکرنادانایانه وین دهینه فه گوهاستن ژسیرفه رى بوبه کار هیته رى وجیبه جیکرنا پروسیسا کومرنا فشاراب جوره ها ته کنیکین فشاری لسه رسیگنالین وهرگرتنى وپه خشکرنی.

تقیم اشارات بث الفیدوا عن طریق معلمات الشبكة اللاسلكية بين العميل/الخادم

الخلاصة:

هذا البحث يركز على تصميم وتنفيذ نظام نقل فيديو الناشئة مؤخراً باستخدام معلمات الشبكة اللاسلكية. هذه المعلمات تتطرق الى معدل فقدان الحزمة، عرض النطاق الترددى والتأخير، فضلاً عن EOS والتي ساهمت فى نهاية المطاف الى QOE للمستخدم. ترسل اشارات الفيديو التي تم التقاطها بواسطة كاميرات الى الخادم المضيف ثم يتم عرضها وبثها الى العميل المستقبل. النظام يتغلب على العديد من المشاكل منها: التأخير، معدل فقدان الحزمة وعرض النطاق الترددى من المصادر والوجهات من خلال السيطرة على التصريحات و البث، وكذلك لمراقبة ترتيب (اعداد) النظام اثناء البث. ميزة هذا النظام هو تحسين تطبيق البرمجيات لاستخدامها بشكل ودى مع اقصى قدر ممكن من المرونة والسيطرة وبخيارات كامل للحصول على ميزات متكاملة لأنظمة بث الفيديو. هذا النظام يتحكم بمشاكل التأخير ومعدل الفقدان وعرض النطاق الترددى بين المصدر والمقصد. من خلال السيطرة على التصريحات المخولة البث والرصد عند تكوين (ترتيب و اعداد) النظام اثناء البث. وقد تحقق التزامن بين الجانبين من الشبكة اللاسلكية عندما ورود الاشارات من الكاميرات الى جانب الخادم وبث هذى الاشارات الى العميل. فى نهاية المطاف هذا النظام يمكن المسؤول من مراقبة البيانات من جانب الخادم الى العميل وتطبيق عملية الضغط مع العديد من تقنيات الضغط على اشارات الواردة والمنبثة.