*International Journal of Engineering & Technology*, 7 (3.34) (2018) 226-230

International Journal of Engineering & Technology

*Website: www.sciencepubco.com/index.php/IJET*

***Research paper***

# Efficient Streaming of Video Frames over Heterogeneous Wireless Network

[1]K.Lavanya, [2] Dr.R.Kanthavel, [3] Dr.R.Dhaya

**Abstract**

Transferring high quality video stream through a heterogeneous wireless network has many challenges due to the varying data rate and round trip time(RTT) involved in the network and the more stringent quality of service(QoS) requirements of the multimedia application such as on time delivery , minimum delay etc. User datagram protocol (UDP), a connectionless, unreliable transport protocol has been widely used for transferring video frames.UDP cannot guarantee reliable information delivery and may lead to packet loss. The packet loss, increases with time varying bandwidth availability in the heterogeneous wireless networks. Transmission control protocol(TCP) can be an alternate transport layer protocol to provide reliable delivery of video information. But, using TCP for wireless networks has limitations due to misinterpretation of packet loss, frequent link failure, asymmetric link behaviour etc.,. which minimizes the throughput and in turn degrades the quality of the video frames transmitted. This paper analyses an efficient scheme to use Heterogeneous Environment Retransmission algorithm with SCTP (HERTS) to transfer video data in the heterogeneous environment. By using multi-homing and multi-streaming feature of Stream control transmission protocol (SCTP), the packet delivery rate, through put and delay requirements can be optimized. The transport layer model suggested in this paper aims at reducing the occupancy of the retransmitted packets in the link, by using a separate end to end path allotted for retransmission.

***Keywords**: SCTP, multi-homed network, Heterogeneous wireless network.*

## 1. Introduction

Wireless environment has become heterogeneous due to the deployment of devices operated using various wireless radio access options available such as 3G cellular mobile networks, WIMAX(IEEE 802.16),WiFi(IEEE 802.11) etc,.The features of these access technologies differ from each other in many ways.WiFi provides limited coverage area and mobility support for the users.Cellular mobile network supports user mobility to a greater extent,but the bandwidth is inadequate to support video streaming applications which have high QoS requirement.Wimax can provide extended coverage but the available capacity is insufficient for video traffic.

| NAME | CHANNEL BANDWIDTH IN MHz | DATARATE IN Mbps |
|---|---|---|
| LTE | 1.4 - 20 | 50 |
| WIFI | 20 | 54 |
| WIMAX | 1.25 - 20 | 40 |

**Fig. 1:** Comparision of various Access networks

The differences in data rate support, available bandwidth etc, makes the wireless network heterogeneous in nature. Some of the challenges of heterogeneous networks are (i) to predetermine the theoretical capacity of the network (ii) Hand off (iii)Interoperability of the different wireless radio access technologies (iv)Mobility of the user device (v)Qos and (vi)Interference between the radio signals..
The mobile video streaming has started dominating the internet due to the excessive growth of mobile devices and multimedia applications. The problems faced by these applications in wireless networks include varying channel quality or bandwidth, high bit error rates and varying Quality of Service (QoS). This paper focuses on the transport layer options to achieve better quality and performance of the multimedia applications, especially video services.
UDP, a traditional connectionless protocol being used for video streaming in most of the cases, does not establish an end to end connection between the end nodes.
In UDP, the source need not wait for an acknowledgement or feedback from the receiver and therefore without the knowledge of the data delivery, the source continuously transmits the data, thus the service provided is unreliable. There is no congestion avoidance mechanisms involved in UDP, which may lead to packet loss.The packet loss in video streaming application will create gaps in the delivery of video streams,which in turn affects the quality of the received video. In addition, Since UDP provides only the basic transport layer functions,it can be used only along with any application layer protocol toprovide video streaming function support.
Transmission Control Protocol (TCP) ,the next transport layer used in TCP are too much conservative, such that even a single packet loss makes the TCP sliding
window protocol to invoke the congestion control mechanism and thereby reduces the throughput to a greater extent.In a wireless network, packet losses occur mostly due to the channel failures like shadowing, fading etc.These channel failures are also misinterpreted as loss due to congestion and the send window size is reduced considerably.As a consequence of variation in the throughput the quality of the transmitted video information may be degraded.In addition ,
The recent researches to overcome these drawbacks, focus on using the stream control transmission protocol (SCTP) as the transport layer solution. SCTP manages connections across mobile terminals more efficiently. The delivery of video frames in wireless networks can be improved by exploiting the multi

homing and multi streaming features of SCTP. These features make the multihomed mobile terminal with multiple network interfaces, to use them concurrently to enhance the performance of video transfer. The real time needs and drawbacks of the transport layer protocol have set certain objectives, to be achieved which includes i)reliable multi-media data transfer ii) in order packet delivery iii)reduced end to end delay and packet loss iv) reduce receiver buffer block and v) assured quality.
The video service is characterized by high transmission rate, large bandwidth requirement and tight delay constraint. But the link resources available in the wireless environment are limited and varying with time. The bandwidth requirement can be satisfied by aggregating the available bandwidth of different networks in the heterogeneous environment in to a large logical channel with the multistreaming feature of SCTP.
This paper analyses the features of SCTP, a transport layer protocol for video streaming over heterogeneous wireless network in order to achieve the above set objectives. The remaining sections of the paper has been organised as follows: section II discuss about the related works by various researchers, section III deals with the overview of the general architecture for video streaming application, section IV lists out the important features of SCTP supporting high definition video streaming, section V concludes the paper with the experimental set up and future work.

# 2. Related Works

The relevant research works on video transmission using UDP, TCP and SCTP have been explained here.[1] analyzes the mechanisms with TCP for improving the delay friendliness of CBR applications and suggested that the better delay performance can be acheived using packet splitting.[2] proposes a new complete user datagram protocol (CUDP), which utilizes channel error information measured to support error recovery at the packet level. [3] Developed a model for TCP streaming systems and mainly focused on buffer occupancy, which depends on the TCP arriving rate and the playout rate. The proposed framework allows to estimate the frequency of buffer overflow events .[4] explores the different factors that degrade users' quality of experience(QoE) in video streaming service that use TCP as transmission protocol .[5] investigates the problem of mobile video delivery using MPTCP in heterogeneous wireless networks with multihomed terminals and suggested that in order to achieve the optimal quality of real-time video streaming, the path asymmetry in different access networks and the disadvantages of the data retransmission mechanism in MPTCP have to be considered.The authors have presented a progressive flow rate assignment algorithm for improving good-put and achieve high PSNR value.But this work do not concentrate on the balanced load distribution to different paths.
[6] have formulated a ‘Concurrent Multipath Transfer - Content Aware’ (CMT-CA) solution featured by scheduling based on the estimated video parameters and proposed a Markovian decision process(MDP) based algorithm to improve PSNR and reduced end-end delay . But ,this method have not dealt with the improvement in Good put . In [7], Utility Maximization theory for flow rate allocation method has been dealt with to minimize the end to end delay with better video quality and proposed a quality-aware adaptive concurrent multipath transfer solution to provide maximum throughput . [8]S. Han et al, (2011) developed a packetization -aware fountain code for high quality video streaming. .[9] proposed an analytical approach to focus on video streaming traffic and to evaluate the packet level for improving performance of a multipath transmission scheme, which sends video traffic bursts over multiple available channels in a probabilistic manner.
[11] Analyses the performance of multimedia distribution when making use of two multi-homing SCTP-based approaches: Single Path Transfer and Concurrent Multi-path Transfer, in which a single or all paths within an association are used simultaneously for data transmission.[12] have proposed an effective end-to-end virtual path construction system, which exploits path diversity over heterogeneous wireless networks. In [13] a comparative study of the trade-off between performance and computational complexity is made and a convex programming-based algorithm has been proposed to fetch good performance efficiency suitable for real time applications.
The features of UDP & TCP discussed so far clearly shows that they might not be a best option for streaming video applications, which have tight constraint over delay, packet loss and quality. The succeeding section discuss the features of SCTP for video streaming applications in detail.

# 3. Stream Control Transmission Protocol (Sctp)

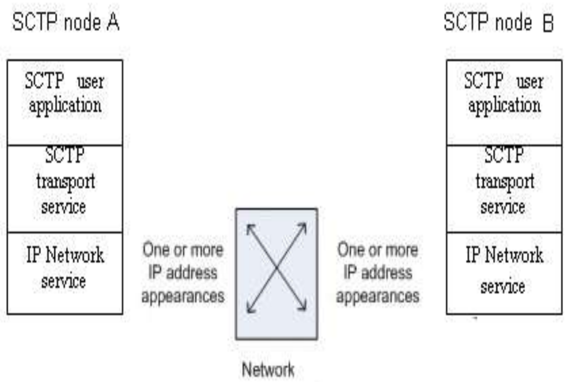


**Fig. 2:** SCTP Association between two end nodes

Stream control transmission protocol (SCTP) [14] is an efficient transport layer protocol that can be used for transferring multimedia data.SCTP is a connection oriented protocol that uses checksum, a sequence number and a selective retransmission mechanism to provide reliable data transfer operating over a connectionless packet network like IP. Some of the features of SCTP includes i)error free data transfer with acknowledgement ii) message fragmentation based on the network’s Maximum transfer Unit(MTU) size iii) In order delivery of user messages within multiple flows iv) multiplexing of differnt user messages into a single SCTP chunk and iv) link-level adaptability to faults . It utilizes the multi-homing and multi streaming features for transferring user data.The ability of the SCTP association to support more than one paths between the two end points enables a single user message to be transferred as multiple streams through different independent paths.

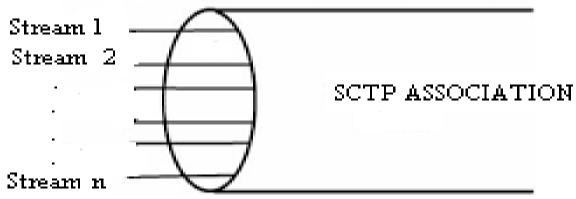


**Fig. 3:** SCTP Association with multiple streams

The SCTP Association can establish a connectivity between the server and client of video streaming application. This single connectivity between the end nodes is capable of supporting multiple streams of information through paths formed, by using their multiple IP addresses. A multi-homed node has multiple IP addresses and one of these addresses is identified as primary address and the connectivity to this address forms the primary path to reach the corresponding node. The end nodes exchange their list of addresses during the connection initiation process. This feature of SCTP makes its association more fault tolerant during network failures. One of the IP addresses is considered to be the primary destination and if it becomes unreachable, the SCTP sender moves to an alternate IP address as destination to complete the transfer.A single SCTP association supports multiple network connections so that it provides an uninterrupted continuous data delivery in any mobile network.SCTP can be used as an efficient transport layer protocol to reduce the gap between Quality of service (QoS) requirements for video information transfer and efficient exploitation of the available resources. Here, the retransmission of lost packets need not follow the same path as that of the actually transmitted packets. The resources are utilized effectively by selective retransmission of the missed data. In SCTP,the connectivity of a session is continuously monitored by using the heart beat message.SCTP [15] adopts the congestion control and flow control techniques from TCP.If message from a stream is lost, the messages from the other streams need not be delayed in the receiver buffer until the retransmitted unit of the lost message reaches the destination. Thus, avoids the problem of receive buffer blocking as in traditional TCP.

## 4. Overview of a Digital Video Transmission System Using Sctp

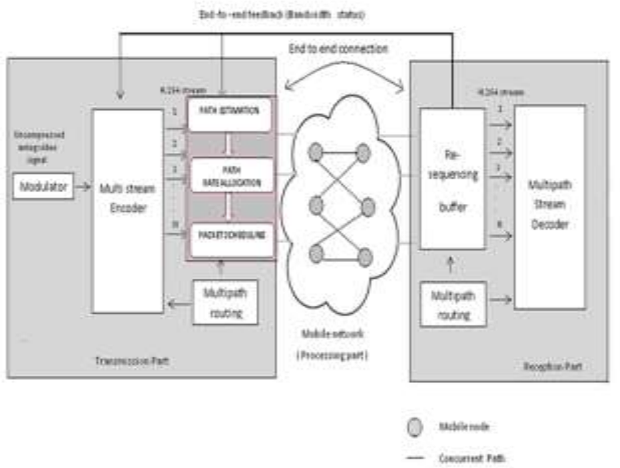


**Fig.4:** Overview of Digital Video transmission system

Digital Video frames transmission system has three major parts, namely transmission section, mobile network and receiver section. The transmitter section is responsible for forwarding the video streams to the receiver through the heterogeneous network.It consists of a multi stream encoder and a traffic scheduler. The digitized raw video signal to be transmitted is fed in to the multi-stream encoder. The received signal is encoded into N- multiple streams of H.264 standard to provide a high resolution video stream in to the traffic scheduler.The traffic scheduler in turn comprises of three modules namely: path estimator, path rate allocation and packet scheduler. The path estimator determines the path behaviour in terms of delay, bandwidth, packet loss rate etc,. The path rate allocator based on the estimated path status, assigns rate limit for each of the paths.Finally,the packet scheduler buffers the packets in the order in which they are to be served in each path.Multipath routing protocols deployed at both transmitter and receiver maintains state information about the multiple end to end paths and finds multiple routes between source-destination pair. The transmission of the information through the mobile network is carried out through the transport protocol, which is found to be more effective for video streaming. At the receiver end, the streams arrived are delivered to the re-sequencing buffer where it is arranged in sequence order. The video signal extracted from the buffer, is then decoded and displayed .

During the flow, the lost or damaged packet, if any should be retransmitted to avoid holes in the received information, which may affect the quality of the displayed video. The lost packet is identified by the sender with the non-reception of the corresponding acknowledgement before the expiry of the retransmission timer. These retransmitted packets play a vital role in introducing jitter in the transmission. If more packets are retransmitted then, worst the receive buffer blocking problem.

## 5. Proposed System

The proposed **H**eterogeneous **E**nvironment **R**etransmission **A**lgorithm with **S**CTP (HERTS) algorithm, uses multi-homing and multi-streaming features of SCTP to retransmit the packets through a separate path. If the primary path (P1) allotted for the transmission is not active, the sender waits for the path to recover within a prefixed path recovery time(PRT1). If the timer corresponding to PRT1 saturates, the transmission is accomplished using secondary path. During the failure of secondary path(SP1),the sender waits till PRT2 for the path to recover. Here the recovery time of both the primary and secondary path is estimated dynamically based on the exponential averaging of the successive PRT values. The gap in the transmitted information can be reduced by using packet interleaving method and thus providing guaranteed quality for the information at the reception.

The challenging task is to find the shortest path in the movable environment .This problem can be considered as the following cases as discussed below.

Case I) If the packet is a retransmitted one.

Case II) If the packet is transmitted for the first time

(i) Failure of primary path leads to choose the secondary path

(ii) If the secondary path fails

The encoded video signal is to be sent through a mobile network by adapting one of the above mentioned cases. The ultimate aim of using SCTP with multi-homed concurrent scheme is to ensure the better quality of the transferred video at any cost and the end to end delay is low.

The experimental set up is built with 2 end nodes namely sender S and receiver R

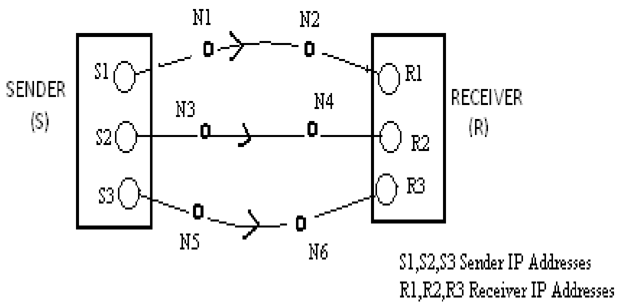


**Fig.5:** multi-homed mobile network scenario

Sender → Server node that transmits video information

Receiver → Client node that video information

Primary Path (P1) S1→N1→N2→R1

Secondary Path 1 (SP1) S2→N3→N4→R2

Secondary Path 2 (SP2) S3→N3→N4→R3

In this setup, Path P1 is considered as the primary path, PS1 is considered as the secondary path and SP2 is the path for retransmitted packets.. Till a network or link failure is met, all the new data packets are transmitted through the primary path(P1).If the packet to be sent is a retransmitted packet,it follows path SP2.When P1 is not reachable, the sender has to wait for PRT1(path recovery time) for the path to recover.If the path P1 is not recovered,the packets are redirected through the secondary path(SP1). For the failure of SP1, the sender has to wait ,till PRT2 exceeds .If SP1 recovers, the transmission of data chunks continue

. Otherwise it checks for the availability of the primary path and if this path is also not available the transmission is terminated.PRT values are initially fixed based on the time taken for reception of the heart-beat acknowledgement chunk.Thereafter,it is estimated based on the exponential average of the PRT values as follows:

PRT = SPRT (K+1) + $\Delta$ ; ($\Delta$=a constant)
SPRT (K+1) = $\alpha$ PRT (K+1) + (1-$\alpha$) SPRT (k); K=0, 1, 2, 3,

$\alpha$ is the smoothing factor($0< \alpha<1$).The value of $\alpha$ closer to one, gives more weight for the recent values of PRT.

In the case of terminated transmission, the higher layer can trigger a retransmission. In general, these retransmitted packets are also queued in the same buffer as the newly transmitted packets. This may lead to variations in throughput, increases the end to end latency and delay variations of the packets. In order to overcome these drawbacks, using SCTP multi-streaming feature, the retransmitted packets can be send through a separate path PS2 for. The retransmitted packets are queued in a separate buffer and these packets will not affect the actual transmission of new packets and thus reduces the through put variations, packet delay and jitter in the transmission of packets.

***HERTS Algorithm***

**Step: 1** Sender (S) starts the video transmission process
**Step:2** Checks whether the packet is a retransmitted one.
Case(I) If it is a retransmitted packet, sender chooses SP2;
S→ SP2; S3→N5→N6→R3
**Step:3** Case(II) If it is not a retransmitted packet, sender chooses P1;
**Step:4** Case (i) If P1 is active, packets are transmitted through P1;
S→ P1; S1→N1→N2→R1
Else, wait till PRT1 for P1 to recover and continue the transmission through P1
**Step: 5** Case (ii) If P1 does not recover till PRT1, sender chooses SP1;
S→ SP1; S2→N3→N4→R2
Else, wait till PRT2 for P2 to recover and continue the transmission through SP1
Else if, SP1 is not recovered, terminate transmission.

## 6. Experimental Setup and Results

The environment for the proposed system is created with two multi-homed hosts with one primary path and two secondary paths. The Wi-Fi interface is used as the primary path(P1),cellular and Bluetooth interfaces the secondary paths SP1 and SP2 respectively. The video encoder standard used is H.264/AVC.The encoding rate of the video stream considered is 30 frames per second. The test sequences used were blue sky,park and river.These sequences have different temporal and spatial characteristics which can be seen in their video quality and encoding rates.

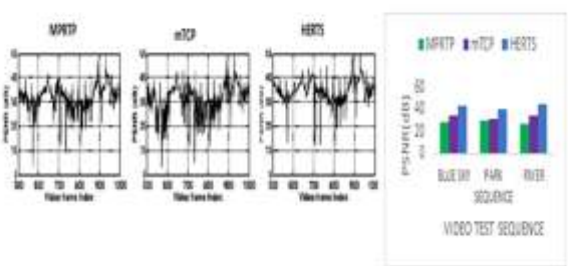

**Fig 6 a:** Comparison of PSNR results

The simulation was carried out for the below mentioned five cases.
**Case i)** If a packet is to be retransmitted **,** Sender (S) initiates the transfer through the path SP2
**Case ii)** For a new packet,Sender (S) initiates the transfer through the primary path P1..
**Case iii) For a new packet,**If P1 is not active or not recovered within PRT1, Sender transmits through the path SP1.
**Case iv)** If SP1 is not active and has not recovered within PRT2, terminate transmission

**Delay performance**: The end-to-end delay is measured to reflect the delay performance. The end-to-end delay of a video frame includes the transmission delay and the waiting time at both sender and receiver side. It is measured from the time a video frame is generated to the time it can be decoded. A low end to end value guarantees a better video quality in real time video applications.

**Peak signal to noise ratio**: PSNR is the standard metric to measure video quality.It is a function of the mean square error between the actual and the reconstructed video frames. High PSNR means good image quality and less error introduced to the image. In case of loss less compression PSNR will be high.

Fig. 6.a & 6.c plots the comparison of simulation results for HERTS with the reference schemes multipath TCP (mTCP) and multipath real time transport protocol (MPRTP).The proposed HERTS algorithm outperforms the existing solutions in video quality and delay. It is found to have PSNR and end to end delay values in the range of 40-45 dB and 101-117 msec respectively whereas the reference schemes mTCP and MPRTP have PSNR and end to end delay values as 32-35 dB &124-140 msec and 27-30 dB & 114- 133 msec respectively.

| SCHEMES | METRICS | | | | | |
|---|---|---|---|---|---|---|
| | BLUE SKY | | PARK SEQUENCE | | RIVER | |
| | PSNR(dB) | End to end delay(msec) | PSNR(dB) | End to end delay(msec) | PSNR(dB) | End to end delay(msec) |
| **MPRTP** | **28** | 114 | **30** | 125 | **27** | 133 |
| **mTCP** | 35 | 124 | 32 | 132 | 34 | 140 |
| **HERTS** | 43 | 104 | 40 | 109 | 45 | 117 |

**Fig 6.b:** Comparison of PSNR and end to end delay for various video sequence

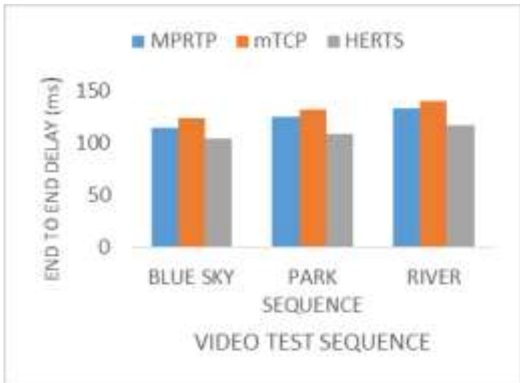


**Fig 6.c:** Comparison of delay performance

Fig. 6.b shows the measured values for peak signal to noise ratio (PSNR) and end to end delay of the reference and the proposed schemes. The increased (PSNR) and the delay performance is an evidence for the quality of the video transmitted through the wireless network with the proposed HERTS algorithm.

# 7. Conclusion

Stream control transmission protocol (SCTP) has been suggested as a better transport layer solution for video streaming in wireless network. The HERTS algorithm proposed minimizes the packet delay incurred due to path failure by using an alternate path with the multi-homing and multi-streaming support of SCTP and improves the quality of the video transmitted which is proved with increased PSNR and reduced end to end delay of the packets. The packet delivery rate, through put and delay requirements of the video transmission are also optimized. Since the PRT estimation is done dynamically by considering the status of the network, it is well suitable for real time video streaming applications

# References


[1] Eli Brosh, Salman Abdul Baset, *Graduate Student Member, IEEE*, Vishal Misra, *Member, IEEE*, Dan Rubenstein, *Member, IEEE*, and Henning Schulzrinne, *Fellow, IEEE* “The Delay-Friendliness of TCP for Real-Time Traffic”IEEE/ACM Transactions On Networking, Vol. 18, No. 5, October 2010.

[2] Haitao Zheng, Member, IEEE, and Jill Boyce, Member, IEEE, “An Improved UDP Protocol for Video Transmission Over Internet-to-Wireless Networks” IEEE Transactions On Multimedia, Vol. 3, No. 3, September 2001

[3] Jinyao Yan, Member, IEEE, Wolfgang Mühlbauer, and Bernhard Plattner, Member, “IEEE Analytical Framework for Improving the Quality of Streaming Over TCP” IEEE Transactions On Multimedia, Vol. 14, No. 6, December 2012 1579

[4] Demóstenes Z. Rodríguez, Julia Abrahão, Dante C. Begazo, Renata L. Rosa, and Graça Bressan, “Quality Metric to Assess Video Streaming Service over TCP considering Temporal Location of Pauses” , IEEE Transactions on Consumer Electronics, Vol. 58, No. 3, August 2012

[5] Jiyan Wu, Member, IEEE, Chau Yuen, Senior Member, IEEE, Bo Cheng, Member, IEEE, Ming Wang, and Junliang Chen,“Streaming High-Quality Mobile Video with Multipath TCP in Heterogeneous Wireless Networks” IEEE Transactions On Mobile Computing, Vol. 15, No. 9, September 2016

[6] Wu, J.Yuen, C.; wang, M.; Chen, Jun-liang, “Content-Aware Concurrent Multipath transfer for high-definition video streaming over heterogeneous wireless networks”,in Parallel and distributed systems, IEEE transactions on , vol.pp, no.99, pp.1-1,Jan 2015.

[7] Chang Qiao Xu, Tianjiao Liu, Jianfeng Guan, Gongke Zhang, and Gabriel-Miro Muntean“CMT-QA:Quality-Aware adaptive concurrent multipath data transfer in heterogeneous wireless networks “ , IEEE transactions on mobile computing, vol. 12, no. 11, November 2013

[8] S. Han, H. Joo, D. Lee, and H. Song, “An end-to-end virtual path construction system for stable live video streaming over heterogeneous wireless networks,” IEEE J. Sel. Areas Commun., vol.29,no. 5, pp. 1032–1041, May 2011

[9] Wei Song, and Weihua Zhuang , “Performance Analysis of Probabilistic Multipath Transmission of Video Streaming Traffic over Multi-Radio Wireless Devices,” IEEE Transactions on wireless communications, vol. 11, no. 4, April 2012

[10] Scott Pudlewski and Tommasomelodia,”A tutorial on encoding and wireless transmission of compressively sampled videos “, IEEE communications surveys & tutorials, vol. 15, no. 2, second quarter 2013

[11] ChangQiaoXu, Enda Fallon, Yuansongqiao, Lujiezhong, and Gabriel-Miro Muntean, “Performance Evaluation of multimedia content distribution over multi-homed wireless networks,” IEEE transactions on broadcasting, vol. 57, no. 2, June 2011

[12] Sang chun Han, Hyunchuljoo,Dongju Lee, and Hwang jun Song, “An end-to-end virtual path construction system for stable live video streaming over heterogeneous wireless networks,”IEEE journal on selected areas in communications, vol. 29, no. 5, May 2011.

[13] B. Rong, Y. Qian, K. Lu, R. Hu, and M.Kadoch, “Multipath routing over wireless mesh networks for multiple description video transmission,” IEEE J. sel. areas commun., vol. 28, no. 3, pp. 321–331, Mar. 2010.

[14] RFC 4960 Stream Control Transmission Protocol

[15] Thomas Dreibholz, Erwin P.Rathgeb,Irene Rungeler,Robin Seggelmann, Michael Tuxen and Randall R.Stewart,” Stream control Transmission protocol:past,current and future standardization activities”,IEEE communication magazine April 2011.

[16] Chih-Heng ke , Ce-Kuen shieh , Wen-Shyang Hwang and Artur Ziviani,” An Evaluation Framework for More Realistic Simulations of MPEG Video Transmission” Journal Of Information Science And Engineering 24, 425-440 (2008).