Multiple Descriptions Coding Technique for Improving Bandwidth in Multimedia Traffic using NS-2

Sapna Dandia[1] Nitin Naghwal[2]
[1,2]Department of Electronics and Communication Engineering
[1,2]IKGPTU (Universal Group of Institute), Lalru, Chandigarh (Mohali), India

Abstract— MDC splits one source of information in multiple entities, where each of the entities is decodable in a stand-alone fashion. The more of these entities are received the more information of the original source can be restored. MDC essentially offers a better solution for multimedia traffic class in multicast networks. However the research demonstration the implementation with a stricter receiver configuration where the receiver under consideration is getting one out of four streams of multicast and PSNR is evaluated extensively. The paper presents novel method based on traffic weight to improve performance of multimedia traffic with the implantation of Technique. These results are presented in tabular form for easier understanding and interpretation. The implementation of MDC proves helpful in bandwidth provisioning.

Keywords: MDC, PSNR, MP OLSR, NS2

I. INTRODUCTION

Wireless communication is a rapidly increasing business in communication profile. Its growth has exponentially increased in the past years and still broadening. From the latest technology to our daily needs, from our imagination to the reality, it has gained all the attention. Indeed, many organizations heavily depend for their information transfer on wireless networks for the efficient and constructive output. This explosive growth of wireless networks have now conjugated with smart phones, tablets, palmtops, laptops and desktops to give it a profound growth both as stand-alone system or as a part of the larger system [9].The first wireless communication was developed in Pre-industrial age when information was transferred in the form of smoke signals or flashing mirrors. Then came telegraphs(by Samuel Morse in 1838), later telephones and thus radio communications was born which enabled data transfer over long distances with better quality, less power and very less time consumption .Early wireless communication transmitted analog signals but now signals are composed of high frequency radio waves in the form of digital signals. A digital radio signal transmits a continuous stream of bits or group of bits called as "packet" and the type of radio is called packet radio. The first wireless network named ALOHANET, developed in University of Hawaii in 1971 was implemented with 7 computers over four islands to communicate bi-directional over a central hub in Oahu .But the invention of Ethernet in 1970's, took away the attention over radio transmissions. It allowed data transfer @10mbp and in 1985, FCC (Federal Communications Commission) commercialized its manufacturing for scientific and research. But the Ethernet could not interfere with ISM (medical frequency band) which led to poor performance, low security, lack of standardization and high costs. But the next generation of Ethernet i.e. IEEE 802.11 provides better performance but low data rates. In 1999 advancement over Ethernet was introduced as IEEE 802.11a called as LAN (Local Area Network) mainly to improve data rates .Till date the most prosperous application of wireless communication is cellular communication [9]. The first voice communication started in 1915, between New York and San Francisco and in 1946 it was introduced in 25 cities of US. But due to its capacity limitations AT&T Bell Laboratories developed the cellular concept. The concept explains that the power of signal decreases with increase and distance and that the same frequency can be reused at some distance with very low interference [8]. In 1990, digital cellular system came into existence known as second generation cellular system. Here the signal transmission was digital based i.e. in the form of 0's and 1's, with better speed, efficiency and low costs. The second generation was mainly developed for voice services but later it supported data services as well such as internet, short message sending(SMS), emails, multimedia message service(MMS), etc. Further advancements are being adopted with better voice and data services, better efficiency and low costs. The current wireless system has constantly overcome the drawbacks of the old ones with new emerging trends. Starting from wired communication to wireless communication systems are constantly evolving new ideas. And cellular communication has evolves satellite communications. The first artificial satellite was introduced in 1957 by Soviet Union with Sergei Kerulen as chief designer. Since then thousands of satellites have been introduced in the space by over 50 countries. And the latest wireless short-distance area networks or low cost-low power radios named Zigbee, Bluetooth and UWB to support short distance communication. Its characteristics designed to be same as that of IEEE standard 802.15 named WPAN (Wireless Personal Area Networks) [10]

II. RELATED WORK

A Review of Multiple Descriptions Coding Technique for Error Resilient Video, Multiple Description Coding (MDC) is one of the promising solutions for live video delivery over loss networks. The paper present a review of MDC techniques based on their application domain and it explain their functionality, with the objective of giving enough insight to designers to decide which MDC scheme is best suited for their specific application based on requirements such as standard compatibility, redundancy tenability, complexity, and extendibility to n-description coding. The focus is mainly on video sources but image based algorithms applicable to video are considered as well. It also covers the well-known and important problem of drift and solutions to avoid it. Video transmission over noisy channels has been a challenging problem for more than two decades. Transmission of raw video is not feasible due to the very large bandwidth required and so video compression is inevitable. On the other hand, compressed video is very sensitive to data loss which happens in best-effort networks such as the

Internet. To counter the effect of data loss for video transmission over noisy networks, there are three categories of approaches: a reliable Automatic Repeat Request (ARQ) based transport layer protocol such as TCP, Forward Error Correction (FEC), and Error Resilient Coding (ERC). ARQ and FEC are channel level protections, while ERC can be used as either source level protection, such as Multiple Description Coding (MDC), or as both source and channel level protection known as Joint Source Channel Coding (JSCC) such as Layered Coding (LC) [6] [13]. The Multiple descriptions coding (MDC) is an effective means to combat busty packet losses in the Internet and wireless networks.MDC is especially promising for video applications where retransmission is unacceptable or infeasible. When combined with multiple path transport (MPT), MDC enables traffic dispersion and hence reduces network congestion. The author describes principles in designing MD video coders employing temporal prediction and presents several predictor structures that differ in their tradeoffs between mismatch-induced distortion and coding efficiency. The author also discusses example video communication systems integrating MDC and MPT [14] [15]. Network communication has many separations of functions and levels of abstraction. This is both the cause and product of assigning various design and implementation tasks to different groups of people. In networking, there is the canonical seven-layer open systems interconnection (OSI) reference model. The layers range from the physical layer, characterized by voltage levels and physical connectors, to the application layer, which interacts with the user's software application. All these layers are involved in the example of accessing the web site from an unfettered laptop. Beyond the OSI layering there is a further separation that most people take for granted. This separation is between generating data to be transmitted (creating content) and the delivery of content. The artistic aspect of content generation writing, drawing, photographing, and composing is not an engineering function. However, the engineer has great flexibility in creating representations of audio, images, and video to deliver an artistic vision. This article addresses the generation of content and how it is affected by unreliable content delivery [12] [16]. The Internet is growing quickly as a network of heterogeneous communication networks. The number of users is rapidly expanding and bandwidth-hungry services, such as video streaming, are becoming more and more popular by the day. However, heterogeneity and congestion cause three main problems: unpredictable throughput, losses and delays. The first technique used simple odd/even separation. More recent techniques use prediction, perceptual models and repetition with optimized bit allocations. The inclusion of correlating transforms in a perceptual audio coder is described in some works. Images the recent surge in MD coding was sparked by a pair of image coding papers. It has not received much attention; a two-dimensional analogue of odd/even separation was presented. The correlating transform method was introduced in the context of image coding. Better results obtained with more general correlating transforms appear in [17]. Other transform-based techniques are given in [7]. Progressive image codes are common and very effective. Thus, several research teams have attempted to apply UEP to progressively compressed image sort integrate UEP in touch coders. Examples include. The application of UEP is combined with additional channel coding for wireless channels. MD scalar quantizes are introduced in a wavelet image coder and quantized frames are applied to images in [4] and [17].The Multi Path Optimized Link State Routing (MP-OLSR) can be regarded as a hybrid multipath routing protocol. It sends out HELLO messages and TC messages periodically to be aware of the network topology, just like OLSR. The difference is that MP-OLSR does not always keep a routing table to all the possible destinations. It only calculates the routes when there are data packets need to be sent out. The core functioning of MP-OLSR has two main parts: topology sensing and route computation. The topology sensing makes the nodes get to the topology information of the network, which includes link sensing, neighbor detection and topology discovery. This part gets benefit from MPRs as well as OLSR. By sending the routing control messages proactively, the node could be aware of the topology of the network: its neighbors, 2-hop neighbors and other links. The routing computation uses the Multipath Dijkstra Algorithm to populate the multiple paths based on the information get from the topology sensing. The source route (the hops from the source to the destination) will be saved in the header of the data packets. The medium hops just read the packet head and forward the packet to the next hop. The topology sensing and route computation make it possible to find multiple paths from source to destination. In the specification of the algorithm, the paths will be available and loop-free. However, in practice, the situation will be much more complicated due to the change of the topology and the instability of the wireless medium. So route recovery and loop detection are also proposed as auxiliary functionalities to improve the performance of the protocol. The route recovery can effectively reduce the packet loss, and the loop detection can be used to avoid potential loops in the network. As presented in the state of the art, most of the multipath routing protocols proposed are based on single path version of an existing routing protocol: AODV and AOMDV, DSR and SMR, ASMA. However, the backward compatibility of those protocols with its single path version is not considered. In fact, given the heterogeneous nature of the wireless networks, it is important to make the new protocols backward-compatible. This can make the deployment of the new protocol much easier. So this part also discusses the compatibility between MP-OLSR and standardized OLSR by using IP source routing. The specification of MP-OLSR has been discussed in the 4th OLSR

Workshop 5th OLSR workshop (2009, Vienna, Austria) and recently 78th IETF offered a lot of valuable propositions, from both theoretical and practical points of view. The proposed ideas have been examined in the thesis. In Section the core functionalities, including topology sensing, route computation and packet forwarding are first introduced. Then we present route recovery and loop detection which can be used to improve the packet delivery ratio and delay of the network [18]

A. Methodology

A traffic weight based methodology has been used in our research work as shown in Fig. 1.1

In this proposed approach to MPEG analysis, the first step is to integrate Evalvid with NS-2. Next step involves the selection of the simulation environment/Scenario, as two scenarios have been proposed in this research work. After that the next step is to define the network parameters and evaluate the network performance on the basis of selected network performance parameters. Lastly, the comparative analysis is performed on the basis of the different values of the network parameter The methodology of this project lies around checking the fraction of decodable frames (Q) and the peak signal to noise ratio (PSNR) of the wireless network transmitting multimedia data (MPEG file) by varying the network parameter like bandwidth, fragment size, and channel error rate (CER) which can be easily varied within the networking. [18] And consists of functionality for rate adaptive video, including support for congestion feedback mechanisms such as TCP-Friendly Rate Control (TFRC). The use of the tool-set is illustrated in figure 1.1.

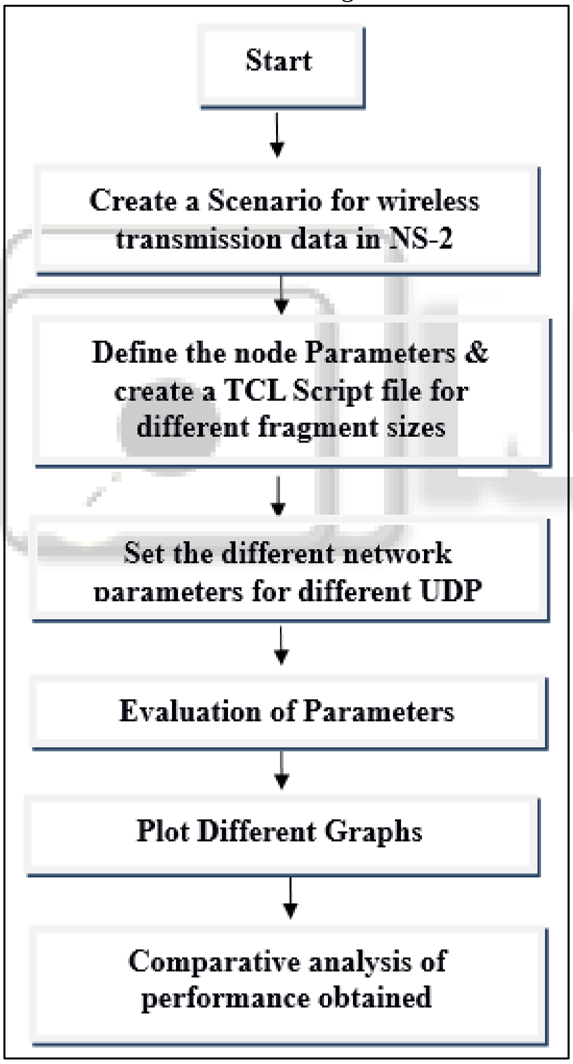


Fig. 1.1: Flow chart of research Methodology

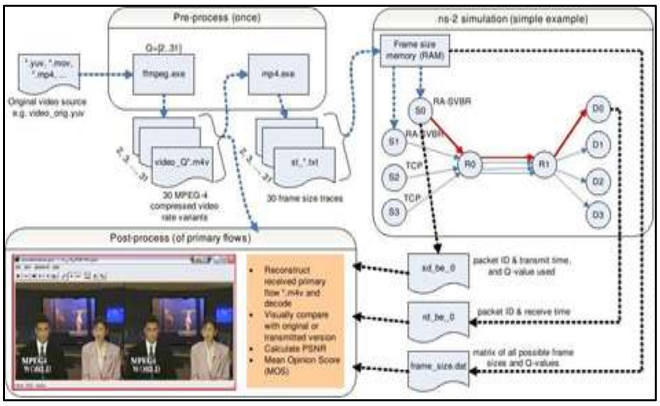


Fig. 1.2: simulation process of the Evalvid-RA framework [2]

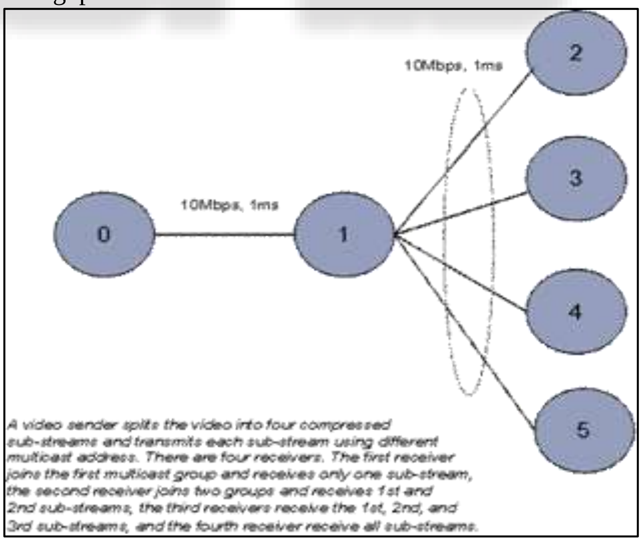
B. Selection of Network and performance parameters

In this research work the analysis of MPEG traffic in mobile ad-hoc networks has been done on the basis of four network parameters. These four network parameters are varied and their effect on the MPEG performance is evaluated. The Three parameters that are used in this work are: Bandwidth, Delay, and Channel Error Rate

III. PROPOSED MODEL

A. UDP and TCP traffic classes along with packet loss rate, throughput and delay Scenario

Under this scenario, a network topology has been proposed in which both the UDP and TCP traffic classes are considered at the source nodes and at the sink node the UDP traffic class is evaluated on the basis of three parameters namely average/peak end-to-end delay, packet loss rate and throughput.



The variations in four network parameters are considered in this scenario. In first case, the bandwidth of the main router to router link (r1-r2) is changed keeping the other network parameters unchanged. In the second case, the delay of the main router to router link (r1-r2) is changed keeping the other network parameters unchanged. In the third case, the error rate of the main link is changed and in the fourth case, the packet size is changed

B. UDP traffic class along with average PSNR and Fraction of decodable frames scenario

Under this scenario, a network topology has been proposed in which only the UDP traffic class is considered at the source node and at the sink node the performance is evaluated on the basis of two parameters namely: average PSNR and fraction of decodable frames. The variations in two network parameters are considered in this scenario. Under this scenario, the Queue algorithm in the proposed network topology is changed four times keeping the fragment size unchanged. This process is repeated three times at the three different values of fragment size for comparative analysis. evaluate is nothing but the combination of NS-2 and evalvid. The simulation consists of two nodes node -1 and node -2 which are exchanging a multimedia file (i.e. MPEG File). Node -1 is source and node-2 is a sink. The traffic is routed using AODV Protocol. [21]The total size of the MPEG file is sent approximately 50Mb. The simulation setup is designed to analyze the performance metric-"fraction of decodable frames (Q)" and "PSNR". The data transfer takes place from node-1(source) to node-2(sink) in packets. The number of packets depends upon the frame size. If the frame size is small the number of packets is large i.e. fragmentation is more, if the frame size is large the number of packets will be small i.e. the fragmentation is less. The analysis reveals that upon increasing the number of frame size for constant channel error rate the fraction of decodable frames (Q) will be increases. We have tested the MPEG-4 Traffic on the following parameter:

Bandwidth	Fragment Size	Channel Rate Error
0.5 Mb	256 bytes	0.1
1.0 Mb	512 bytes	0.3
2.0 Mb	1024 bytes	0.6

Table 1.1: Different parameter for testing MPEG-4

The transfer of MPEG files transfer from node-1 and node-2 take place using UDP (user data gram protocol) as there no establishment of connection between the nodes and no acknowledgement is issued after the transfer of data and real time transport protocol (RTP) is used as traffic source. The sink agent attached at node-2 receives the data from node-1. The above given parameter altered as mentioned and the performance parameters like fraction of decodable frames (Q) ,(PSNR) peak signal to noise ratio and peak end to end delay is measured. The further section describes the effects of above mentioned parameters on the network.

IV. Results and Analysis

The simulation was run and following observations were made

A. Results for Packet loss rate

1) Total Packet loss:

Packet loss is the failure of one or more transmitted packets to arrive at their destination. In digital communication, Packet loss is distinguished as one of the three main error types, the other two being bit error and spurious packets caused due to noise. A problem of packet loss is much more complicated in Mobile Ad hoc Networks because wireless links are subject to transmission errors and the network topology changes dynamically.

Steps for calculation of packet loss rate

$$PacketLoss\ \ Rate = \left(\frac{no.ofpacketst\ \ ransmitted\ \ - no.ofpacketsr\ \ eceived}{no.ofpacketst\ \ ransmitted} \right) * 100$$

2) Results of calculation of PSNR for varying Fragment Size

The simulation was run with varying fragment size (256kb, 512kb and 1024kb). The Min peak signal to noise ratio (PSNR= 32.683753) observed against fragment size of 256kb while max peak signal to noise ratio(PSNR= 35.985632) was recorded for fragment size 1024kb.

Fragment Size	1024kb	512kb	256kb
PSNR	35.985632	34.317582	32.683753

Table 5.1: Showing PSNR value for different fragment size

3) Results of calculation of PSNR for varying Fragment Size

The max peak signal to noise ratio (PSNR = 40.626631) was observed for queue type REM and min peak signal to noise ratio(PSNR = 32.945678) was observed for Drop tail queue type .

AQM	Drop tail	SFQ	REM
PSNR	32.945678	36.963241	40.626631

Table 5.2 showing PSNR values

4) Results of calculation of PSNR for varying Bandwidth

The max peak signal to noise ratio (PSNR = 33.215414) was reused against bandwidth 10mb while min peak signal to noise ratio(PSNR = 28.346172) was observed while bandwidth was reduced to 2mb.

Bandwidth	10mb	5mb	2mb
PSNR	33.215414	30.561355	28.346172

Table 5.3 showing bandwidth with PSNR

B. Evaluation of frame loss in video transmission varying frame size.

The values for loss against frames I,P,B & OA were observed to be 37.5,34.29,0and 34.3 in case of frame size of 1024.While value for loss against frames I,P,B, OA were observed to be 25 , 44.88 , 0 , 44.8 in case of frame size of 512 and the value for loss against frame I, P,B, OA were observed to be 50 , 55.52 , 0 , 55.5 in case of frame size of 256.The overall frame loss was observed to be max with value 55.52 . when frame size was set to be 256.while the overall frame loss was observed to be min while the values 34.3 . when the frame size was set to be 1024 .The frame type 'B' was observed to suffer 0% loss in each case while the frame size was varied as 1024 , 512 , and 256 while the frame type 'P' was observed to suffer max losses as 55.52(256) , 44.88(512) and 34.29(1024) as shown in fig 5.2 & in table 5.4The total no of frames generated were 2000.

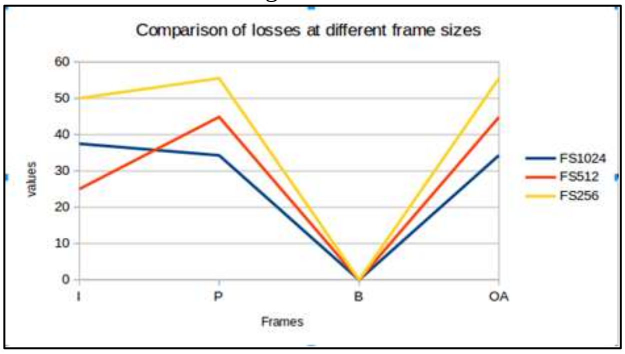


Fig 5.2: Showing values for different frame types against frame sizes

Frame Type	FS1024	FS512	FS256
I	37.5	25	50
P	24.29	44.88	55.52
B	0	0	0

Table 5.4: showing values for different frame sizes

MDC implementation essentially required multiple receivers that are provided by the different streams as mentioned above in the simulation scenario description. The value of PSNR is calculated at node 2 which is receiver 1 out of 4 receivers and the least privileged receiver. Getting a finite value of PSNR on these nodes represents successful implementation MDC. In order to optimize the performance further changes were made in the network parameters and follow the results which are as stated. While calculating the PSNR we are varying one parameter and keeping the other parameters constant so that the effect of the varying network parameter is analyzed.

V. Conclusion

The variation in fragment size reveals that keeping the moderate value of fragment size gives highest PSNR. As the fragment size decreases PSNR also decreases. The PSNR is also not the highest at the highest fragment size as no of packets forming a frame is less and the loss of even one frame can cause the PSNR to decrease. The default queue management technique Drop-tail gives the worst PSNR, as we varied the AQM, SFQ and REM the PSNR increased giving the highest value at REM which is around 40. The variation in bandwidth gives obvious results as the bandwidth decreases, PSNR also decreases. The research provides general guidelines on how to improve performance of multimedia traffic with the implantation of MDC. More changes at network parameters can be done in order to enhance the performance or PSNR. The effort results can be used as a guiding light to perform implementation of MDC on more complicated multimedia traffic classes like streaming video and high definition videos.

References

[1] Dán G, Fodor V, Karlsson G Are Multiple Descriptions Better Than One? In: Boutaba R, Almeroth K, Puigjaner R, Shen S, Black J (eds) Networking 2005. Networking Technologies Services, and Protocols Performance of Computer and Communication Networks Mobile and Wireless Communications Systems,vol 3462. Lecture Notes in Computer Science. Springer Berlin Heidelberg, pp 684-696. doi:10.1007/11422778_55 (2005)

[2] Fumagalli M, Lancini R, Stanzione A Video transmission over IP by using polyphasedownsampling multiple description coding. In: Multimedia and Expo, 2001. ICME 2001.IEEE International Conference on, 22-25 Aug 2001. pp 1095-1098. doi:10.1109/icme.2001.1237917 34 Shirani S Content based multiple description image coding. Multimedia, IEEE Transactions on 8 (2):411-419. doi:10.1109/tmm.864349 (2006)

[3] Greco C, Cagnazzo M, Pesquet-Popescu B Low-Latency Video Streaming With Congestion Control in Mobile Ad-Hoc Networks. Multimedia, IEEE Transactions on 14 (4):1337-1350. doi:10.1109/tmm.2012.2195480 (2012)

[4] Huihui B, Yao Z, Ce Z Multiple Description Video Coding using Adaptive Temporal SubSampling. In: Multimedia and Expo, 2007 IEEE International Conference on, 2-5 July 2007 2007. pp 1331-1334. doi:10.1109/icme..4284904 (2007)

[5] Jing W, Jie L H.264 Intra Frame Coding and JPEG 2000-based Predictive Multiple Description Image Coding. In: Communications, Computers and Signal Processing PacRim 2007. IEEE Pacific Rim Conference on, 22-24 Aug. 2007 2007. pp 569-572. doi:10.1109/pacrim.2007.4313300 32, (2007).

[6] Javadtalab A, Omidyeganeh M, Shirmohammandi S, Hosseini M A rate control algorithm for x264 high definition video conferencing. In: Multimedia and Expo (ICME), IEEE International Conference on, 11-15 July 2011. pp 1-6. doi:10.1109/icme.6012225 (2011)

[7] Jiann-Jone C, Shih-Chieh L, Ching-Hua C, Chen-Hsiang S, Jyun-Jie J, Chi-Chun LA Multiple Description Video Codec With Adaptive Residual Distributed Coding. Circuits and Systems for Video Technology, IEEE Transactions on 22 (5):754-768. doi:10.1109/tcsvt.2179459 (2012)

[8] Kibria R, Kim J H.264/AVC-based multiple description coding for wireless video transmission. Paper presented at the International Conference on Communications (2008)

[9] Kobayashi M, Nakayama H, Ansari N, Kato N Robust and Efficient Stream Delivery for Application Layer Multicasting in Heterogeneous Networks. Multimedia, IEEE Transactions on 11 (1):166-176. doi:10.1109/tmm.2008.2008933 (2009)

[10] Moo Young K, Kleijn WB Comparative rate-distortion performance of multiple description coding for real-time audiovisual communication over the Internet. Communications, IEEE Transactions on 54 (4):625-636. doi:10.1109/tcomm.2006.873071 (2006)

[11] Padmanabhan VN, Wang HJ, Chou PA, Sripanidkulchai K Distributing streaming media content using cooperative networking. Paper presented at the ACM/IEEE NOSSDAV, Miami, FL, USA. (2002)

[12] Reibman AR, Yao W, Xiaoxin Q, Zhimei J, Chawla K Transmission of multiple description and layered video over an EGPRS wireless network. In: Image Processing 2000. Proceedings 2000 International Conference on, 10-13 Sept 2000. Pp 136-139 vol.132. doi:10.1109/icip.899246 (2000)

[13] Shirani S, Gallant M, Kossentini F Multiple description image coding using pre- and post-processing. In: Information Technology: Coding and Computing, 2001. Proceedings International Conference on, Apr 2001. pp 35-39. doi:10.1109/itcc.2001.918761 (2001)

[14] Tillo T, Olmo G Low complexity pre post processing multiple description coding for video streaming. In: Information and Communication Technologies: From Theory to Applications, 2004. Proceedings 2004 International Conference on, 19-23 April 2004 pp 519-520. doi:10.1109/ictta.2004.1307860 (2004.)

[15] Vijayalaksmi M, Linganagouda Kulkarni “A Layered Optimization Approach for Efficient MPEG Video

Transmission Over Wireless Networks" International Journal of Electronics Communication and Computer Engineering Volume 3, Issue (1) NCRTCST, ISSN 2249 –071X (2012)
[16] Wu JC, Peng KJ, Lu MT, Lin CK, Cheng YH, Huang P, Yao J, Chen HH Hot Streaming: Enabling scalable and quality IPTV services. Paper presented at the Proc. IPTV Workshop Conjunction 15th Int. World Wide Web Conf., (2006)
[17] Yiting L, Gibson JD Routing-Aware Multiple Description Video Coding Over Mobile Ad-Hoc Networks. Multimedia, IEEE Transactions on 1(1):132-142. doi:10.1109/tmm.2010.2089504 12 Min Q Zimmermann R (2010) an Adaptive Strategy for Mobile Ad Hoc Media Streaming. Multimedia, IEEE Transactions on 12 (4):317-329. doi:10.1109/tmm.2010.2046275 (2011)
[18] Zhang M, Liu W, Wang R, Bai H A Novel Multiple description video coding Algorithm. Paper presented at the International Conference on Computational Intelligence and Security, (2008.)