ISSN (Online) : 2278-1021
ISSN (Print) : 2319-5940

International Journal of Advanced Research in Computer and Communication Engineering
Vol. 3, Issue 7, July 2014

A Review of Network Coding and Multimedia Traffic over Wireless Networks within 2013

Mahmoud Riad Mahmoud[1], Imane Aly Saroit[2], Walid Mohamed Abdelwahab[3]

Professor, Mathematical Statistics department, Institute of statistical studies and research, Cairo University, Egypt [1]

Professor, Information Technology department, Faculty of Computer and Information, Cairo University, Egypt [2]

PG Scholar, Computer Science and Information, Institute of statistical studies and research, Cairo University, Egypt [3]

Abstract: Multimedia transmission over wireless networks has grown in recent years, so there is more attention from research community. Providing a high quality multimedia transmission over wireless is a challenge, as it is surrounded with strict timing constraint and high bandwidth demand. In addition, wireless communication is associated with challenges like limited bandwidth, interference and mobility, which make it more difficult and challenging. Network Coding is technique, which allows and encourages mixing of data at intermediate network nodes. A receiver sees these data packets and deduces from them the messages that were originally intended for the data sink. Many studies have been performed on video streaming over wireless networking. Most of these studies were intended to increase the performance and efficiency of wireless networks in order to satisfy the highly increased data traffic requirement. We noticed that during the past two years, many scholars tried to use network coding in order to provide a better environment for video traffic over wireless networks. So, this paper gives a literature review of the most related work with providing better performance of multimedia streaming over wireless networks. This paper is limited only for 2013's researches.

Keywords: network coding research, wireless networks, multimedia streaming, video transmission.

I. INTRODUCTION

Requirements for high quality media streaming is increasing every day. So, many researches were conducted and even under conduction to gain enhancements in multimedia streaming. Video users are expected to grow rapidly in next year's. It is mentioned that such blossom will be more four hundred and fifty million by 2014 [1]. Specifically the area of using network coding in multimedia streaming is a new research area [2].

Generally multimedia streaming is a challenging due to many factors such as loss sensitivity, delay and high bit rate; and it became more challenging when multimedia is transmitted over wireless networks. Scholars proposed many approaches to overcome these obstacles like new network protocols and coding techniques (channel, source and layered).

1.1 Network coding

In the year 2000, network coding was introduced as a new paradigm for handling information of networks by Ahlswede et al [3]. All communication networks today make a basic assumption that information is separate. Thus, information is transmitted over networks and shares the network medium like cars share the highway. Such information can be data packets or signals. A data stream is completely independent but it may share network resources. In networks field, most network functions especially data storage, routing and error control are based on such assumption.

Network coding can be seen as an alternative to traditional routing which has recently emerged. In network coding, the network nodes have the ability to combine the received packets, and then it forwards such packets to the neighboring nodes. That can result in reducing number of transmissions hence improving the throughput [5]. In wired and wireless networks, network coding can be illustrated by the following well known examples.

The first one (Figure 1-a) illustrates a multicast scenario. There is are two sources S1 and S2 which want to transmit data packets a and b to two receivers X and Y. R is an intermediate node which creates new packet $a \oplus b$ then forwards it to X and Y, rather than relaying a and b. the outgoing link from R, will be used once to transmit a and b; and X will recover b by $b = a\oplus (a \oplus b)$ and Y will recover a, by same concept. In traditional routing, two individual transmissions should have made in order to transmit a and b.

There is a saved transmission slot which can be used to communicate a new data packet; so network coding increasing throughput of such network. Therefore, network coding reduces the delay, as there is no need to keep waiting for the two transmissions from node R. moreover network coding reduces the energy consumption as it saved a transmission slot.

The second example (Figure 1-b) illustrates that there are two wireless nodes X and Y which are in the range of S (base station). X and Y want to exchange data packets a and b, and they can't communicate directly.

As shown in the diagram X forward a to S and then Y forwards b to S. with using network coding S creates $a \oplus b$ and broadcasts to both nods X and Y, as illustrated in the wired example.

ISSN (Online) : 2278-1021
ISSN (Print) : 2319-5940
International Journal of Advanced Research in Computer and Communication Engineering
Vol. 3, Issue 7, July 2014

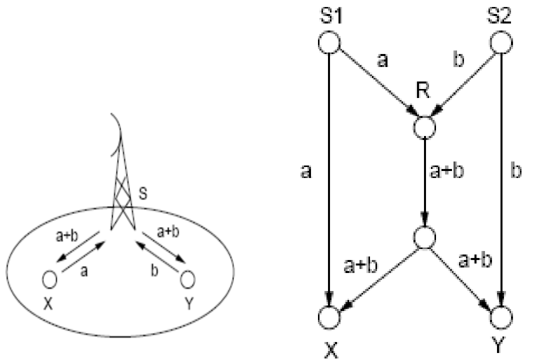

Figure: 1a Network Coding Figure: 1b Wireless Network Coding

From the above example it can be concluded that NC allows nods to encode the incoming information before sending it. Such coding is based on some scheme, that make nodes code and decode data packets. Linear is the simplest coding scheme, as it regards a block of data as whole in a vector over a specific bas field. All nodes are allowed to apply linear transformation to such vector before passing it on. [3] Pointed out that linear network coding is proved to be an optimum scheme for encode Information. The linear equations in network coding are solved by performing Gaussian elimination, which is an algorithm for solving systems of linear equations in linear algebra.

With random network coding scheme, the global encoding vector (coefficients) is generated randomly from a finite field F2s, in a fully independent and decentralized manner [4].

II. 2013'S RESEARCHES

2.1 Expanding Window Network Coding

Greco, et al [8] proposed a network coding framework for video delivery over wireless networks. Such framework is able to guarantee a good trade-off between resiliency to losses and timely delivery. This work designed by combining Expanding Window Network Coding (EWNC), Multiple Description Coding (MDC), and a novel Rate-Distortion Optimised (RDO) scheduling algorithm.

Expanding Window Network Coding (EWNC) is a network coding technique that been proposed recently. The main idea of EWNC, is to use Gaussian elimination at the receiver side in order to provide instant decodability of packets. Multiple Description Coding (MDC) is a well-established joint source-channel coding paradigm based on splitting a media content into N sub-streams, referred to as descriptions. Any description can be independently decoded for representing the content, but the quality improves with the number of descriptions. Video MDC has been proven to be a valuable tool to cope with packet losses in wireless networks. MDC was mainly designed for reducing the probability of generating non innovative packets. Rate-Distortion Optimised (RDO) is the main and novel idea of the framework. RDO is scheduling algorithm maximize the expected video quality.

They reported a comparison with the reference techniques under a few different simulation conditions; basically it includes one scenario with the proposed network coding framework, and one without network coding. The quality of service measurement was based on (PSNR) Peak signal-to-noise ratio which is an engineering term for the ratio between the maximum possible power of a signal and the power of corrupting noise that affects the fidelity of its representation.

They observed that, introduction of the scheduling, jointly with the possibility of mixing packets across descriptions, significantly improves the performance of video streaming perceived by the user. Such framework provides well-combined solution, but on the other hand it lacks for a practical implementation.

During our research we found Nemoianu et al. [9] have proposed almost same framework idea, for streaming multi-view videos over unreliable wireless channels. In fact, Nemoianu et al. concur with Greco et al. [8] and stress the importance of combining Expanding Window Network Coding (EWNC), Multiple Description Coding (MDC), and Rate-Distortion Optimised (RDO) scheduling algorithm, in order to enhance streaming of multimedia over wireless networks.

2.2 Vehicular Content Distribution with Network Coding

Seung [10] discusses different strategies for enhancing multimedia performance; one strategy was mainly to effectively deliver multimedia data over the Internet with mobility by making use of network coding. He divides every video stream into multiple files then he used network coding to generate random linear combination of data packets within a file. Once enough data packets are received for such file, the file is decoded then it is passed to the video player on the laptop or smartphone to play in the proper order, by using the mobile media player plugin for Windows operating system. He implements network coding in a general basis, as the implementation was for all data transfer. Such implementation meant to ensure that only innovative packets are exchanged between Access points (APs) and vehicles. Specifically, he developed a novel Vehicular Content Distribution (VCD) system which helps to enable high-bandwidth content distribution over vehicular networks. VCD consists of some vehicles, access points with and without Internet access, web servers or any content server on the Internet and a controller. Some of these access points may form a mesh network.

He considers two performance metrics, total download size and total amount of time the video can play. His simulation assumes a content server is published on the Internet and it has all the content, while all vehicles and APs are initialized without any content. He studied the implication of the proposed strategy for smartphones and desktop. At the end of his research, he concludes that network coding can enhance data traffic for both desktop and smartphone, but he pointed out that NC incurs much lower cost on the desktop than on the smartphone.

ISSN (Online) : 2278-1021
ISSN (Print) : 2319-5940
International Journal of Advanced Research in Computer and Communication Engineering
Vol. 3, Issue 7, July 2014

2.3 Random Linear Network Coding with Secure Practical Network Coding

Moreover, Scaria and Suresh [11] proposed a novel system for enhancing wireless network coding in terms of performance with security improvements. Their proposed system has combined RLNC (Random Linear Network Coding) and SPOC (Secure Practical Network Coding) in wireless transmission for improving the confidentiality and integrity of message.

They pointed out that, RLNC can be used to reduce the overhead of the entire data packet flow in terms of end-to-end delay, hence it has a great effect in increasing the performance of video streaming. . Each symbol or data packet is sent with the global encoding data vector. When the received matrix has full rank, the receivers decode the original data packets using Gaussian elimination. Their proposed secure practical network coding gives a strong way to utilize the integral security of random NC. SPOC reduces the number of cryptographic operations for the secret communication. This is achieved by performing locking and unlocking operations. Locking or protecting is performed only to the source coefficients that are required to do decoding for the linearly encoded data packets. Unlocking is performed to the coefficients, which don't compromise hidden data. The unlocking helps the proposed SPOC to allow intermediate nodes to perform network coding operations.

Their packet format consists of two basic parts, first is the global encoding vector that is kept in the header. Second is the payload part, which is divided into vectors based on the used field size. They treated a video stream as a partitioned stream of group of pictures in a constant period.

On the other hand, the provided model lack of any time of implementation even simulation with a network simulator. It just provided a schema of the main operations of such model.

2.4 Practical Online Scheduling Algorithm with Network Coding

Oh [12] proposes a practical online scheduling algorithm for mobile video streaming to multiple users with network coding capabilities. She build here idea from the fact that watching videos over mobile devices such as smartphones and tablets has been attracting interest from users, and demand for mobile video streaming is increasing. However, existing wireless technologies (e.g., WiFi, WiMax, or LTE) cannot support this impending demand.

She used realistic scenarios in order to study mobile video streaming over unreliable networks with network coding possibilities. She built here model upon recent work of network coding that demonstrates the capabilities of network coding in increasing the throughput of networks. Moreover, she investigated video quality peak signal-to-noise ratio (PSNR) over wireless networks, and she studied how mixing data packets from different data flows into a single data packet, can increase the information per transmission.

Namely, the main idea is that when the transmitted data flows are mobile video streams, network codes should be selected in a way that maximize the network throughput along with the video quality (PSNR). Finally, this work proposed a practical online scheduling scheme to sub-optimize not only the network throughput but also video quality with the reliability gain of network coding over unreliable wireless links. In order to achieve such a suboptimal solution, the theoretical analysis uses the partially observable Markov decision process (POMDP).

In this model, each AP will construct and broadcast the best network code GNCp,k,i (GoP-aware Network Coding), which consists of packet p of I-frame XOR-ed together with some other packets and kth network code set at APi. Implementation and analysis results show that this scheme closely approximates the optimal solution for both mobile video quality and throughput. However, there was no practical implementation or even simulation.

2.5 Random Network Coding with Multi Generation Mixing

Modi [13] employed random network coding (RNC) with Multi Generation Mixing (MGM) for MPEG-4 video traffic Wireless Ad hoc Networks. Modi pointed out that Network Coding is applied at different layers for instance Physical Layer, MAC layer, Network & MAC Cross layer, Transport layer and Application layer; and he said that all can enhance the performance of wireless Networks.

He designed his solution by considering that Generation indicates how long and for how many packets a node should wait before encoding a packet. Information is in the form of packets encoded using randomly generated coefficients over finite field. So, receiver needs to store packets and coefficients until the last packet successfully received by the receiver which introduces delay. So, sparse connectivity and high losses are considered to be the main degrading factors on the performance of network coding.

He stated that, the loss of one frame may result in loss of many frames. In case of video transmission over wireless networks, video is a sequence of Group of Pictures (GoP). GoP specifies the order in which intra- and inter-frames are arranged and in his model it is composed of I (Intra-coded), B (Bidirectional) and P (Predictive) frames. I frame is referenced by P and B frames. If I frames of multiple GoPs are encoded in one encoded packet then the loss of such encoded packet results in loss of many frames. This can be overcome through the concept of Multi Generation Mixing. For video transmission over wireless networks, if packets of I frames of same generation are encoded with the encoded packets of P frames then with B frames then loss of one encoded packet will not results in loss of video because receiving any (enough) packets can recover I frames. Such Multi Generation Mixed packet provides protection to packets of I frames and improves performance even though there is high losses of packet.

His simulation results showed that, the loss of one or more I packets are recovered through successful decoding of

ISSN (Online) : 2278-1021
ISSN (Print) : 2319-5940
International Journal of Advanced Research in Computer and Communication Engineering
Vol. 3, Issue 7, July 2014

either IP packets or IBP packets. As the results show that, the packet delivery fraction and the delay are much better in case of using network coding with MGM. However, in this work, no real video traffic is used for transmission, it considers only static network nodes, and there is no practical implementation of random network coding.

Patel and Narmawala [14] concur with [12] and stress the importance of using network coding and Multi Generation Mixing (MGM) in improving the network performance. In Network Coding with MGM goal is to enhance decodable rates in situation where losses prevent efficient propagation of sender packets. MGM allows the cooperative decoding among the different generations of a mixing set which enhances decodablity.

They have simulated the proposed protocol in NS2 simulator. The simulated network contained 20 to 40 wireless nodes which move randomly. The average speed of a node was 30 m/s. The communication range of a node was 100 m. after conducting such simulation with and without the proposed solution, they found protocol using network coding with multi generation mixing outperforms conventional scheme and scheme using network coding in terms of delivery delay, delivery ratio and throughput.

2.6 COPE Network Coding for Voice Application and Multimedia

Pertovt et al [15] proposed a network coding based solution for dealing with delivery of voice application and multimedia sessions over the packet-switched broadband Internet protocol (IP) networks in real time. Their research investigates the benefits of using wireless network coding for VoIP application in WMNs. They present network coding for WMNs used to decrease the network delay, which plays a great role in video and voice streaming improvement. They pointed out that, transmitting voice over a wireless mesh networks (WMNs), the analog voice signal has to be digitalized, encoded, and packetized. They utilize the two facts, first is Voice over Internet protocol (VoIP) is used for transmitting voice signals over a packet-switched Internet protocol (IP) networks in real-time. Second, is network-coding exploits the broadcast nature and enhances network throughput of the wireless medium.
The proposed solution is based on the practical network coding procedure (COPE); that encodes two or more packets in a single transmission based on the nodes knowledge on what information (which packets) neighboring nodes have. Such procedure was tested in a real WMN deployment, which is of a particular importance. Their simulation considers a typical network topology for WMNs with 10 wireless nodes and 3 neighbors per each node; all such nodes were static. They have just compared the simulation with COPE (NC) and without NC.

Their simulation results show that network coding can enhance and improve the VoIP performance in WMNs especially when the investigated network is highly loaded or congested. Network coding decreases the network delay while the influence on jitter is small. The network coding benefits is supposed to be dependent on the used VoIP codec. Such codec converts an audio signal into compressed digital form for transmission and then back into an uncompressed audio signal for replay, but it require higher traffic loads per second.

Their performance matrix includes the calculation of End-to-End (ETE) delay and packet delay variation or jitter, which have a major impact on the VoIP. They concluded that the combination of Network coding procedure with various VoIP codecs is used to observe the impact on network delay and jitter of the VoIP application. Moreover, Network coding can decrease network delay and jitter and network coding benefits are codec dependent.

2.7 Novel Distributed Rate Allocation Algorithm with Network Coding

Bourtsoulatze et al [16] proposed a novel distributed rate allocation algorithm for delivery of multimedia data in mesh networks. The algorithm is based on inter-session network coding. The network users decide locally on the optimal coding decisions and rates for each combination of packets that they request from their parents. Each user requests the content of one of the available sources. The decisions are based on the minimization of the average decoding delay of the node and its children nodes and require only a minimal communication overhead.

They validated the performance of their distributed rate allocation algorithm in different video streaming scenarios using the NS-3 network simulator. The results of their simulation showed that the proposed algorithm is capable of eliminating the bottlenecks and reducing the decoding delay of users with limited bandwidth resources. In the context of video transmission, it enables the timely delivery of video data to the network users, hence leads to better average video quality.

2.8 MATIN Framework with Network Coding

Barekatain et al [17] introduced MATIN framework, MATIN is mainly based random network coding and its purpose is to provide efficient P2P video streaming. The MATIN generates the required coefficients matrix without using n2 entries, as it uses just n Galois Field. By this, they make sure that there is no linear dependency among the associated vectors. Then, it sends only one instead of n coefficient entries and it is encapsulated in the header of each encoded data blocks.

They evaluated the performance of the MATIN with P2P live video streaming, by performing a simulation using OMNET++ simulator. They attempt to make the simulation more reliable by performing the simulation under different network conditions; however, they didn’t consider nodes movements. Their performance metrics includes the following factors. First, Video Distortion that indicates the specific capacity of the not playback frames divided by total capacities of all existing video frames of the stream. Second, end-to-end delay that indicates the time elapsed between the transmission a video frame from

ISSN (Online) : 2278-1021
ISSN (Print) : 2319-5940

International Journal of Advanced Research in Computer and Communication Engineering
Vol. 3, Issue 7, July 2014

the source and playing it in a peer. Third, initial start-up delay, such delay indicates the required time for a peer to finish initialize the initial buffer time after joining the network.

To sum up, this work shows how the proposed MATIN decreases the transmission overhead and hence increases the performance of the P2P video streaming.

2.9 Hybrid Video Dissemination Protocol with Network Coding

Naeimipoor [18] proposed hybrid video dissemination protocol (HVDP) which is said to be outperforms other protocols in term of delivery ratio and complies with other quality-of-service requirements for video broadcasting over vehicular environments.

HVDP model combines The Network Coding based Data Dissemination (NCDD) as an application layer technique with two other protocols. First is the Reactive, Density Aware and Timely Dissemination (REACT-DIS) protocol which has been implemented over the network layer. It was used as a receiving-based routing approach that avoids high packet delay and transmission overhead. Second is Media Access Control (MAC) congestion control mechanism over Wireless Access in Vehicular Environment (WAVE) as a reliable approach on top of the link layer. It was used for decreasing packet loss in high congested spots.

His hybrid video dissemination protocol takes advantage of the combined protocols and techniques, in an optimal manner to satisfy the QoS requirement of video streaming in a broadcasting environment. HVDP is a reactive and congestion aware protocol that is mainly based on network coding technique. This scheme attempts to optimize resource utilization and make sure to deliver high quality video to all participant vehicles in the network, while keeping the networking cost as low as possible. This research study attempts to deal with the video streaming challenges in highway scenarios. So, he used the most common way for implementing a highway scenario which is Freeway.

He simulated routing protocols and agent sitting in the mobile nodes with NS-2. EvalVid tool set has been deployed and integrated, in order to evaluate sent and received video files. The EvalVid framework provides some tools to evaluate the quality of video by determining packet and frame loss, in addition to other metrics such as delay, jitter, PSNR were considered.

Results show that network coding at both source and intermediate nodes is an effective solution for reliable and low overhead packet delivery to vehicles with limited distance from the source in low data rate network. Based on his findings, the network coding alone is not adequate to achieve quality of service requirements for dissemination of video in high-density networks with high data rates. However, he stated that it is an efficient solution to combine with other stack layer techniques to provide satisfactory QoS.

2.10 Real-Time Opportunistic Coding with Network Coding

Mowafi and Awad [19] introduced a novel architecture which is based on XOR opportunistic network coding. Such architecture copes with the main performance characteristics that required for real time communication over wireless networks. They named it Real-Time Opportunistic Coding (RTOC). RTOC provides an efficient architecture for applying various techniques in order to satisfy several multimedia application requirements like delay variation, bandwidth, and loss.

RTOC adopts the wireless network coding COPE's basic principles. COPE as a network coding solution has opportunistic listening and opportunistic coding; and it was mainly designed for non-real-time communication. On the other hand, RTOC is designed specifically for real-time communication traffic. Actually, it gives higher priority to the delivery time of data packet, and the inter-packet delay. That was done by attempting to sharply minimize the overhead that is associated with both encoding and decoding processes.

In the opportunistic listening, the wireless network node has an opportunity to overhear the sent packets to other nodes, above and beyond it is allowed to store the overheard data packets for a limited time period. With opportunistic coding, the node looks for opportunities to encode as many packets as possible, ensuring that the recipients can decode their packets.

They developed RTOC Packet Coding Algorithm (RPCA) in order to reduce the overhead along with the data encoding and decoding operations with COPE. RPCA uses a virtual FIFO queue for each neighbour node, where each queue contains the packets destined to that neighbour node. As long as more data packets are encoded, the total number of transmissions becomes less. Hence, that plays an important role in decreasing the number of data packet collisions and transmission overhead. If the data packets to be encoded are different in terms of size, the smaller data packets are padded with zeros. This way, the is no necessary searching for the packets with same size, and there is no packet reordering. Such procedures minimize the data packet encoding time and give a good opportunity to make it suitable for real time traffic.

RTOC was implemented above and below the MAC layer independently of the MAC protocol. RTOC's performance was evaluated using OMNET++; and its evaluation matrix considers end-to-end delay, packet delivery ratio, throughput and jitter. The results illustrate that RTOC has a decent potential to improving the performance of real-time multimedia transmission.

2.11 Joint Inter- layer and Intra-layer Network Coding

Ostovari et al [20] proposed a Multi-layer videos architecture to enhance the performance of providing Video on-demand (VoD) streaming using Network Coding. They give a special consideration to the problem of utilizing helper nodes to minimize the load on the central video servers. The proposed architecture have two versions, one for wired VoD, and extended one for wireless live streaming (LS) applications . They just formulated the problem as a linear programming (LP) optimization using joint inter- and intra-layer network coding (NC). They showed how their method can be extended to suite the case of wireless live streaming. Such a research is lack of any practical implementation or using a simulator.

2.11 Scalable Video Coding with Network Coding

Al-Hami et al [21] proposed a promising approach for video streaming over wireless, by using video coding (i.e scalable video coding) with network coding. The proposed solution is intended for one-hop wireless LAN network. PDR and end-to-end delay was calculated in order to evaluate the proposed approach against a pure network coding approach. They explained how their approach is beneficial to video streaming; however they have provided very little information of how they have simulated such approach.

2.12 Cognitive radio (CR) Network Coexisting Framework – Network Coding

Hou et al [22] proposed a design framework for achieving efficient multimedia multicast services in cognitive radio (CR) networks. The proposed framework was not only intended to improve the achieved video quality but also it protects the rights of subscribed users. They consider a cognitive radio (CR) network coexisting with a primary network with L primary users. Each primary user owns a licensed channel and has the exclusive priority to access its owned channel. The CR network consists of a secondary base station (SBS) and N secondary user.

Such framework utilizes network coding and superposition coding to reduce the scheduling complexity and account for heterogeneous channel conditions. With network coding, all encoded packets are equally important, thus the SBS does not need to differentiate packets when performing scheduling. With superposition coding, the nodes with good channel conditions can successfully receive more packets such that it can have a larger probability to reconstruct video of higher quality.

Numerical examples show the proposed framework can improve the average received data rate by up to 15%. When achieving the same video quality, the proposed framework can save 30% transmission time comparing with the scenario using direct transmission. On the other hand, the proposed lacks of implementation or simulator evaluation as it is was only evaluated by numerical calculation which is based on a set of assumptions.

2.13 Investigation of the Influence for Network Coding Techniques of Multimedia and Audio Streaming

Saeed et al [23] investigated the influence of network coding techniques of multimedia and audio streaming over wireless networks. They didn't provide any new techniques. They just examines several existing network coding protocols, which meant for ad-hoc wireless mesh networks and they compares their performance factors on multimedia streaming applications. Such comparison was between a network coding technique and optimized broadcast protocols, such as BCast, Partial Dominant Pruning (PDP) and implified Multicast Forwarding (SMF). They performed such simulation using NS-2 simulator, and they considered two main scenarios static and mobility. The mobility scenario was based on random waypoint movements. The simulation considers PDR, latency and jitter as the performance factors.

Their simulation results showed that RLNC based protocol is the best protocol in case of audio and video streaming application as such protocol delivers data packets with lowest latency, highest PDR and low jitter. As a result, RLNC improves and enhance the overall performance of both audio and multimedia streaming applications. On the other hand, using XOR network coding technique is proved to negatively affect the performance of audio streaming application in terms of jitter and latency. In this work, they considered jitter and latency as the main performance metrics for evaluating the quality of service of multimedia streaming applications. So, they have concluded that the overall network performance significantly increases with using random linear network coding scheme.

2.14 Dynamic Random Network Coding

Saroit et al [24] proposed DRNC solution which is supposed to be carefully designed based on the random network coding concepts. Practically in DRNC, data blocks are treated as a vector within a predefined base field. Such approach uses the UDP in the transport layer and either DRNC of SMF in the network layer. Encoding and Decoding in case of using DRNC is done in the data link layer. It is not just a traditional network coding, but it is random and dynamic.

It is random as it uses the random network coding. So It has two main parts. 1) Global encoding matrix (Gt), such matrix can be either fixed or randomly selected, however in DRNC Global encoding matrix is randomly selected. 2) Original data or information (X). The result Y can be generated by multiplying Gt by X. Such global encoding matrix plays a crucial role in resolving the original data packets at receiver side. Senders mainly select the linear coefficients in a random way, from a finite field of appropriate size (28).

It is dynamic as the size of the generation is adjusted dynamically in the real-time. As, if the size of data generation is set to a large value, the average delay of packets in various data generations will increase, which badly affects the quality of service of video streaming data.

On the other hand, if the size of data generation is set to a small value, the average delay of packets in various data generations will decrease. Nevertheless, the smaller size of data generation is badly affects the expected network throughput improvement and data transmission reliability. So in DRNC the size of data generation is dynamically

ISSN (Online) : 2278-1021
ISSN (Print) : 2319-5940
International Journal of Advanced Research in Computer and Communication Engineering
Vol. 3, Issue 7, July 2014

adjusted according to the network communication state and the transmission delay of data generation, in real-time.

With the proposed DRNC, the code is initialized with the original information X, then create random encoding global matrix Gt, encode data Y by multiplying Gt by X, then send or broadcast both Gt and Y. When receiving a new (innovative) packet the protocol dynamically decide whether to create new linear random combination or not, above and beyond the rapid and dynamic adaption of the best strategy for dealing with video data packets. Simply, once the source generates the video data packet, it is encoded with a random coefficient from GF, the resultant packet is called coded packet. This packet is re-encoded with other coded packets in the generation (if they exist) before transmission. If the existing coded packets are further encoded with random coefficients from the finite field, such resultant packet is called a re-encoded packet. All the packets are stored locally in generations in the form of a decoding matrix. Each row of the matrix contains the coefficients of the coded/re-encoded packet.
DRNC's scheme is practically implemented; it is developed and integrated with NS-2 simulator. Imane et al. have used BonnMotion in order to create different movements' scenarios. Static and dynamic scenarios of the underlining wireless network were investigated in different speeds of the movements. Dynamic scenarios include the random waypoint and random street. So, this work is not just providing a novel scheme; but it is also the first study to evaluate network coding over random street movements via wireless mesh networks, moreover this study is one of the limited studies for providing practical implementation of network coding. The performance evaluation of the quality includes calculating latency, jitter and packet delivery ratio.

It has been proved that DRNC has increased video streaming quality with static and mobile movements, as it was compared to simplified multicast forwarding (SMF) as an optimized traditional broadcast scheme.

REFERENCES

[1] O. Oyman, "Enabling Mobile Video Services over WiMAX and LTE", a tutorial. Proc. of IEEE Vehicular Technology Society of the Institute of Electrical and Electronics Engineers, 2010.
[2] Athina Markopoulou and Hulya Seferoglu, "Network Coding meets Multimedia: Opportunities and Challenges", University of California, Irvine, USA, 2009.
[3] [3] S. R. Li, R. W. Yeung, N. Cai, "Linear Network Coding", Proc. of IEEE Transaction on Information Theory, vol. 49,no. 2, pp. 371-381, 2003.
[4] T. Ho, M. Medard, R. Koetter, D. Karger, M. Effros, J. Shi, and B. Leong. A random linear network coding approach to multicast. IEEE Transactions on Information Theory, 52(10):4413–4430, October 2006. (Cited on pages 28, 42, 48, 92, 132, 144 and 145.)
[5] Christopher Miles, "Network Coding Theory and Matroid Representability", Departmental Honors Thesis, Lafayette College Department of Electrical & Computer Engineering, May 2013.
[6] G. D. L. Reyes, A. Reibman, S. Chang, and J. Chuang, "Error resilient transcoding for video over wireless channels," IEEE Transactions on Multimedia, vol. 18, pp. 1063–1074, June 2000.
[7] W. Li, "Overview of fine granularity scalability in mpeg-4 video standard," IEEE Trans. Circuits and Systems for Video Technology, vol. 11, no. 3, pp. 301–317, 2001.
[8] Claudio Greco, Irina Delia Nemoianu, Marco Cagnazzo, Beatrice Pesquet-Popescu, "A Network Coding Scheduling For Multiple Description Video Streaming Over Wireless Networks", European Signal Processing Conference (EUSIPCO), Bucarest : Romania, hal-00781305, version 1, 19 Mar 2013, <http://hal.archives-ouvertes.fr/docs/00/78/13/05/PDF/main.pdf>.
[9] Irina Delia Nemoianu, Claudio Greco, Marco Cagnazzo, Beatrice Pesquet-Popescu, "Multi-View Video Streaming Over Wireless Networks With Rd-Optimized Scheduling Of Network Coded Packets", Visual Communications and Image Processing (VCIP) Conference, San Diego, CA : United States, hal-00802334, version 1, 19 Mar 2013, <http://hal.archives-ouvertes.fr/docs/00/80/23/34/PDF/main.pdf>.
[10] Yousuk Seung, "Optimizing Mobile Multimedia Content Delivery", PhD. Dissertation, Faculty of the Graduate School of The University of Texas at Austin, 2013.
[11] Chinchu Mary Scaria and S. Suresh, "Safe and Sound Network Coding for Wireless Video Streaming", International Journal of Electronics and Computer Science Engineering (IJECSE), Vol. 1, No. 2, PP. 590-596, <http://www.ijecse.org/wp-content/uploads/2013/01/Volume-1Number-2PP-590-596.pdf>.
[12] Hayoung Oh, "Practical Online Scheduling for Mobile Video Streaming on Smartphones", Smart Computing Review Journal , Vol. 3, No. 1, PP. 33-41, February 2013, <http://www.smartcr.org>.
[13] Jayesh n. Modi, "comparative study and analysis of multimedia traffic over ad hoc network", International Journal of Research in Engineering & Technology (IJRET), Vol. 1, Issue No. 2, July 2013, PP. 25-34, <www.impactjournals.us>.
[14] Ankit Patel and Zunnun Narmawala, "Network Coding with MGM based Anycast Packet Transmission in Vehicular Ad-Hoc Networks", International Journal of Computer Science and Technology IJCST, Vol. 4, Issue No. 2, June 2013, PP. 508-512, <www.ijcst.com>.
[15] Erik Pertovt, Kemal Alič, Aleš Švigelj and Mihael Mohorčič, Performance Evaluation of VoIP Codecs over Network Coding in Wireless Mesh Networks, Proceedings of the 2013 International Conference on Electronics and Communication Systems, PP. 43-49,<http://www.europment.org/library/2013/rhodes/bypaper/ECS/ECS-03.pdf>
[16] Eirina Bourtsoulatze, Nikolaos Thomos and Pascal Frossard, "Distributed Rate Allocation in Inter-Session Network Coding", IEEE Transactions on Communications, arXiv: 1212.5032 v1 [cs.NI], PP. 1-34, <http://arxiv-web3.library.cornell.edu/abs/1212.5032v1>.
[17] Behrang Barekatain, Dariush Khezrimotlagh, Mohd Aizaini Maarof, Hamid Reza Ghaeini, Shaharuddin Salleh, Alfonso Ariza Quintana, Behzad Akbari and Alicia Trivin˜o Cabrera, "MATIN: A Random Network Coding Based Framework for High Quality Peer-to-Peer Live Video Streaming", PLOS ONE Journal, Vol. 8, Issue No. 8, August 2013, PP. 1-17, <www.plosone.org>.
[18] Farahnaz Naeimipoor, "Video Streaming and Multimedia Broadcasting over Vehicular Ad Hoc Networks", M.Sc Thesis in Electronic Business Technology, Faculty of Graduate and Postdoctoral Studies, University of Ottawa: Canada 2013. < http://www.ruor.uottawa.ca/>.
[19] Moad Y. Mowafi and Fahed H. Awad, "Opportunistic Network Coding for Real-Time Transmission over Wireless Networks", Network Protocols and Algorithms Journal , 2013, Vol. 5, issue No. 1, PP. 1-19, <http://macrothink.org/journal>.
[20] Pouya Ostovari, Abdallah Khreishahy, and Jie Wu, "Multi-Layer Video Streaming with Helper Nodes using Network Coding", Proc. of the 10th IEEE International Conference on Mobile Ad-hoc and Sensor Systems (MASS 2013) October, PP. 14-16, <http://www.cis.temple.edu/>.
[21] Mo'taz Al-Hami, Abdallah Khreishah, and Jie Wu, "Video Streaming Over Wireless LAN With Network Coding", Proc. of the 12th IEEE International Symposium on Network Computing and Applications, PP. 21 - 24 August 2013, <http://www.cis.temple.edu/>.
[22] Fen Hou, Zhaofu Chen, Jianwei Huang, Zhu Li and Aggelos K. Katsaggelos, "Multimedia Multicast Service Provisioning in Cognitive Radio Networks", Wireless Communications and Mobile Computing Conference (IWCMC), 2013 9th International, PP. 1175 - 1180, <http://ieeexplore.ieee.org/>.
[23] Basil Saeed, Chung-Horng Lung, Thomas Kunz and Anand Srinivasan, "Multimedia Streaming for Ad Hoc Wireless Mesh Networks Using Network Coding", Int. J. Communications,

Network and System Sciences, May 2013, No. 6, PP. 204-220, <http://www.scirp.org/journal/ijcns>.

[24] Imane Aly Saroit, Mahmoud Riad Mahmoud and Walid Mohamed Abdelwahab, "Proposed Network Coding Solution for Multimedia Streaming over Wireless Networks", International Journal of Advanced Research in Computer and Communication Engineering Vol. 2, Issue 7, July 2013, pp 2582- 2588

BIOGRAPHIES

Mahmoud Riad Mahmoud obtained his Ph. D in Biostatistics from the University of North Carolina at Chapel Hill in 1973. He has been with Cairo University, Egypt since that time. He is professor at Cairo University. His major area of interest is Multivariate Statistical Inference and Characterizations.

Imane Aly Saroit received her Ph.D in 1994, from Faculty of engineering, Communication department, Cairo University. She worked at Cairo university since 1989; she is now a professor at the Information Technology department and also she is a vice dean for Student Affairs at the Faculty of Computers and Information, Cairo University. Her researches interests are focused on computer networks, specially wireless and mobile Networks.

Walid Mohamed Abdelwahab holds a Master degree in Business Administration, Bedfordshire University, UK. He received a PG Diploma in computer science, Cairo University, Egypt in 2004 and a BSc in Computer in 2002. He is certified for Sun java programmer, IBM system administrator, Websphere Portal, SOA designer, SAP Reports and Oracle SQL. He worked for IBM as a senior software engineer, and currently he is software engineering manager at Informatics Oman. Moreover, he is doing IT consultation for several organizations. His research interests include software engineering, information security and leading & managing integrated software systems.