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ASRNC over Wireless Networks

Walid Mohamed Abdelwahab[1]

PhD Candidate, Computer Science[1]

Abstract: In our contemporary world, 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 challenging as it surrounds with strict timing constraint and high bandwidth demand. The Wireless communication is associated with challenges like limited bandwidth, interference and mobility which make it more difficult and challenging. In this paper we propose a new adaptive scheduled random network coding (ASRNC) scheme for video streaming over wireless IEEE 802.11 protocol. Such model is built on the fact that network coding can increase the quality of networks in terms of throughput enhancement and delay reduction. The model is practically implemented and integrated with network simulator (NS-2). BonnMotion is used as a movement generator so it helps to create different movements' scenarios. Movable and Static scenarios of the underlining wireless network were integrated with NS2 in different movement' speeds. Dynamic of movement scenarios were include the random waypoint and random street. The performance evaluation factors of the quality of multimedia streaming includes calculation of latency, jitter and packet delivery rate. The proposed ASRNC has increased multimedia streaming quality with both static and mobile movements, as it was compared to simplified multicast forwarding (SMF) as an optimized traditional broadcast scheme.

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

I. INTRODUCTION

Video users are expected to grow rapidly in next years. According to [1] the volume of video content transmission is expected to be much more increased in 2020. Specifically the area of using network coding in multimedia streaming is a new research area [2].

As we can see from the below figure, Video transmission traffic will be 4.2 times than non/video traffic in 2025 and 6 times in 2030 [1]

Estimation of mobile traffic by different service types globally

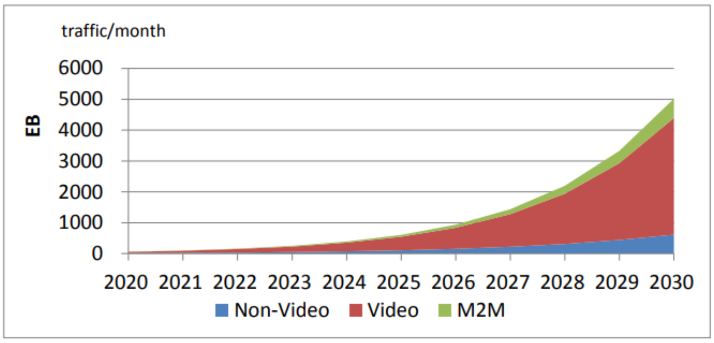

Figure 1 Video demand increase 2020-2030

Generally, multimedia streaming is a challenging due to many factors such as delay and high bit rate, loss sensitivity; 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).

Channel coding has proposing techniques to increase the reliability of the multimedia transmission as it requires high bandwidth, for instance using FEC (Forward Error Correction) techniques [18]. Channel coding has proposed error-resilient coding techniques in order to provide high quality of video streaming in loosely environments. [5, 6]. Dealing with heterogeneity and time varying nature of the Internet was overcome by some other techniques like layered video coding techniques. It tries to adopt the bit rate to the available bandwidth [7, 5]. The following section briefly illustrates the main two broadcasting protocols of this paper network coding and simplified multicast forwarding.

1.1 Network coding

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 [3]. In wired and wireless networks, network coding can be illustrated by the following well known examples.

The first one (Figure 2-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 ⊕ 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⊕ (a ⊕ 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 2-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 ⊕ b and broadcasts to both nods X and Y, as illustrated in the wired example.

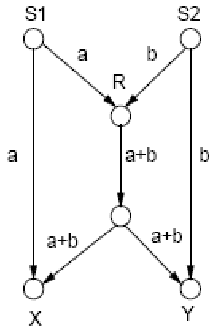

Figure: 1a Network Coding

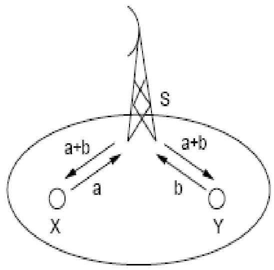

Figure: 1b Wireless Network Coding

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].

1.1 Related Work

Recently several researches have been published to show network coding benefits in many areas including video streaming, wireless networks, sensor networks and content distribution. Some of these applications are briefly described below.

Using network coding techniques in wireless networks was outlined by researchers like [9]. They have provided a model for broadband wireless access (BWA) networks which increases the throughput of the overall network. That was by developing an analytical model in order to evaluate the mean decoding delay and the mean good put of the proposed network coding solution. The Frame-by-frame Adaptive Systematic Network Coding (FASNC) which they have designed doesn't have an implementation from programming perspective. The results showed that it lowers the decoding delay and reduction of buffer requirements which is very important especially for delay-sensitive applications such as multimedia applications.

[10] Proposed a network coding scheme to maximize the bandwidth efficiency of broadcast sessions over WLAN and WiMAX. That was based on markov decision process MDP, they showed the improvement of such approach over traditional techniques. But this was only for static stations which is not likely to be practical.

[11] Stressed on the traditional paradigm of a single source that can serve a monolithic network and the content delivery to a single device is more obsolete. They pointed out to importance of taking into consideration Quality of Experience (QoE) while designing the delivery mechanisms especially for multimedia communication. They proposed a solution using network coding to overcome slow delivery over wireless and the buffering length limitations. They

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used two servers one is costly and other is free in order to develop a control policies, such policies switch between the two main servers (costly and free). So they attempt to attain the quality of experience metrics at the lower cost.
[12] evaluated the use of network coding along with a compression video slandered (H.264/MPEG-4 AVC). Their simulation result which was based on NS-2 simulator shows that network coding improves the overall quality by reducing packet loss and improving the throughput. Their quality matrix is based on used peak-signal-to-noise-ratio (PSNR).
[13] Proposed a network coding solution (NicroCast) for video streaming over smartphones. They designed and evaluate MicroCast system which is based on using two network interfaces per participated phone. A cellular interface used to connect the video server and Wifi interface to connect all other participated phones. It was implemented using seven android phones and it shows significant benefits in terms of video transmission performance.
[14] Proposed a mechanism which combines network coding and multi-path routing in order to provide a more reliability in forwarding data over wireless networks. As, the number of the control messages (required messages for routing discovering) has been reduced; and even the transmission time has been reduced. Such mechanism was designed for low-cost, reliability without retransmission, low-power and multifunctional sensors over wireless networks.
Some scholars pointed that network coding can be used effectively to overcome the lost packages during video transmission. For instance [10] used NC for retransmitting the lost packages, and that make them able to send more information per transmission.
In 2017, a new scheme has been proposed by [26], they tried to enhance the quality of video streaming by integrating multiple generation scheduling and network coding. Its results shows that improvement is achieved but it did not take into consideration mobility of nodes and it was not have a practical implementation of network coding.
In addition to that research, [27] in that thesis, he studied a scalable live multimedia broadcast over a certain wire-less network. He assumed the nodes receive multimedia packets from a static base station. He designed two IDNC algorithms, each one has it is own priory. Such prioritized algorithms are proved in increasing the number of video layers at the devices compared to the existing network coding schemes. Moreover, [28] concurs with [27], as it is built on the fact that encoded video has unequally important packets. Therefore, it plays a great role in the overall video quality.

II. PROPOSED ASRNC SCHEME ANALYSIS

The emergence of Network Coding has brought about a metamorphosis in thinking of network communication systems. As, it allows network nodes to forward, process and code the incoming independent information flows.

2.1 Dynamic Random Network Coding

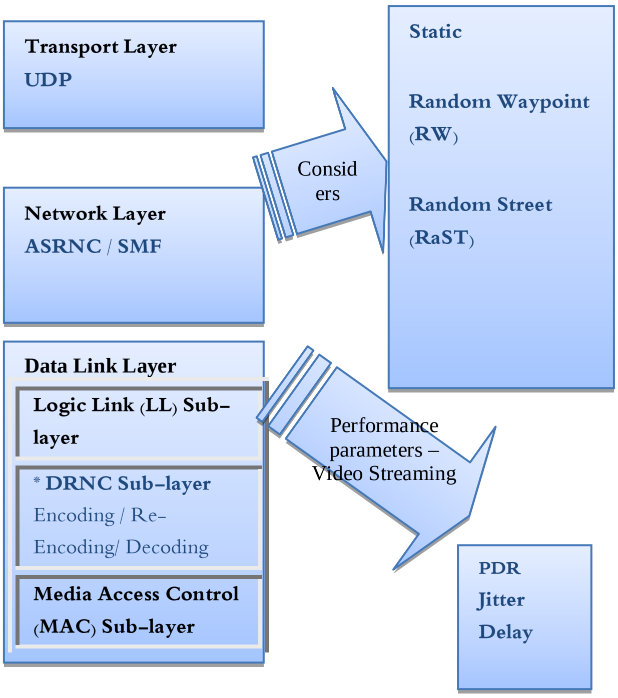

Figure 2 : Dynamic Random Network Coding Solution

As expressed before network coding allows nodes to combine incoming information (data packets) before sending it. We propose a Dynamic Random Network Coding (ASRNC) for broadcasting video stream data packets over wireless networks. The proposed solution is carefully designed based on the random network coding concepts. Practically in our ASRNC, data blocks are treated as a vector within a predefined base field. As illustrated in figure 2, our approach uses the UDP in the transport layer and either ASRNC of SMF in the network layer. Encoding and Decoding in case of using ASRNC is done in the data link layer. It considers static, random waypoint and random street movements. Its main performance factors are PDR, Jitter and Latency.

The proposed ASRNC is not just a traditional network coding, but it is random and dynamic. Therefore, ASRNC has a specific packet format. The following subsections introduce our ASRNC specifications and components in more details.

1) 2.1.1 Random Network Coding of ASRNC

Simply from practical perspective, Network Coding has two main parts. a) Global encoding matrix (Gt), that matrix can be selected as fixed or random, however it is randomly selected in ASRNC. b) Original data or information (X). Then multiplying Gt by X generate the result.

In random network coding, such global encoding matrix plays a great role in resolving original data packets at receiver side. Normally, senders select the coefficients linearly in a random way. Such coefficients taken from a finite field of appropriate size (28). Because when the size of the field is large, the probability of solving such matrix becomes high. On the other hand, the field size (the overhead) has to be low as possible so we can achieve higher throughput. The network node sends the encoding vector within the same data packet; such nodes store the received data packets within their buffers. The receiver should receive an amount of combinations of data packets equals to the number of source processes in order to be able to decode the encoded data packets. the main operations of RNC are encoding and decoding.

With the proposed ASRNC, the code is initialized with the original data X, then create encoding global matrix Gt in a random way, encoded data Y is generated by multiplying Gt by X. Then broadcast take place for both Gt and Y. When receiving a novel (new) packet the protocol adoptively decide either to create novel linear random combination or not. Moreover, ASRNC adapts the best strategy for dealing with video data packets in rapid and dynamic manner. 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.

2.1.2 Adoptive nature of Generations with ASRNC

The adaptive nature of the proposed ASRNC is related to packets generations. In practical random network coding, the generation size g is the number of symbols over which encoding is performed, and it defines the maximal number of symbols that can be combined into a coded symbol. Data is decoded on a per generation level, thus at least g symbols must be received before decoding is possible.

In our ASRNC, 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.

In our ASRNC we understand that such trade-off needs to be set between the size of data generation and the packet delay of the data generation. To sum up, the size of data generation is dynamically adjusted according to the network communication state and the transmission delay of data generation, in real-time.

In addition, we make use of Markov Decision Process, at receiver before decoding, as those packets need to schedule. Such scheduling policy helps a lot at receivers in storage and complexity requirement.

2.1.3 Packet Format of ASRNC

In order to practically implement our protocol in real world, we define our own packet format for ASRNC. From practical point of view, it is very challenging to find space in the IP header due to the very limited space availability. So we have defined our own packet header format where the required information will be stored. As illustrated in figure 3, the ASRNC's packet has four main parts.

First part is the Generation ID which is a 16 bit number used to represent each generation uniquely at the given period of time. The Generation ID is randomly generated by the source node and it should be unique in the network. In order to uniquely identify the original packet in the encoded vector, we used IP address and Sequence Number (each source

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maintains its own sequence number) pair is used. Then we have as many encoding or re-encoding coefficients (encoding vector). At the end, we have the payload part which contains the actual coded packet.

Generation ID	IP address and Sequence Number pair (32 bit IP address & 16 bit sequence number)	Encoding/Re-encoding Coefficients (8 bits per coefficient)	payload

Figure 3 Packet Format for ASRNC

Generally, the packet format is very important to get an effective communication between nodes in the network. There are several known predefined transport protocols that could do the job for us, protocols like TCP, UDP, RTP and RTCP. Different protocols are good for different things, for example TCP is not good for streaming but UDP is.

2.1.4 Transport layer of ASRNC

The priority of providing real-time transmission is higher than the reliability of the transmission for video streaming. UDP is more suitable for delivering video data. In UDP, each packet is sent individually. The sender side does not provide flow control and no guarantee of the successful delivery of a packet so UDP offers minimal transport layer functionalities. It is not concerned about the delivery order of the packets. In brief, short delivery time is the only target for UDP.

In our ASRNC we have used UDP in the transport layer because UDP is simpler than TCP and it gives no overhead for retransmission and congestion control. And basically it is suitable for real-time applications like video streaming, since for a video stream, both real-time transmission and quality are important. It is necessary to provide QoS at another layer while using a UDP transmission. Although UDP does not provide a successful guarantee of delivery, this negative side can be offset by the MAC layer re-transmission mechanism and the ASRNC code. In our ASRNC we employ UDP as a transport layer protocol, so the video streaming application is communicating almost directly with the network layer.

2.1.5 Mobility Support of ASRNC

The proposed ASRNC considers static and dynamic wireless networks. So, it has two different scenarios, one with static scenarios where nodes are stable and not moving around base station(s) and the other with mobility scenarios where all nodes, except base station(s), are moving around. This thesis considers video steaming as one of the main types of multimedia traffic. The multimedia traffic is continually generated by the base station(s) and sent out to several nodes. Furthermore, all nodes can act as intermediate nodes which forward received packets to the nodes that are out of the base station(s) range. In such mobility scenarios we make use of random waypoint mode (RW) and random street mode (RaST).

III.CONCLUSION

The proposed Dynamic Random Network Coding uses UDP in transport layer and performs its random coding and encoding data link layer. ASRNC is based on the concept of random network coding with a dynamic nature and make use of Markov Decision process for scheduling the storage of the incoming packets at receiver side. The proposed ASRNC considers static, random waypoint and random street movements. ASRNC has been implemented and integrated with NS2, and the results showed its enhancement of multimedia streaming according to PDR, latency and Jitter.

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BIOGRAPHY

Walid Mohamed Abdelwahab holds a Master degree in Business Administration, Bedfordshire University, UK. He received a Master degree and PG Diploma in computer science, Cairo University, Egypt. 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.