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Proposed Network Coding Solution for Multimedia Streaming over Wireless Networks

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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 novel dynamic random network coding (DRNC) scheme for video streaming over wireless mesh network which follows IEEE 802.11 protocol. Such scheme is built on the fact that network coding can increase the throughput of the network. The scheme is practically developed and integrated with NS-2 simulator. We 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 way point 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. The performance evaluation of the quality includes calculating latency, jitter and packet delivery rate. 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.

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

I. Introduction

Video users are expected to grow rapidly in next years. 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).

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 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

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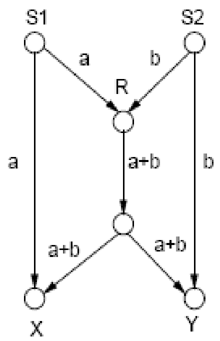
individual transmissions should have made in order to transmit a and b. There is a saved transmission slot which can be used to communicate a new data packet; so network coding increasing throughput of such network. Therefore, network coding reduces the delay, as there is no need to keep waiting for 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.



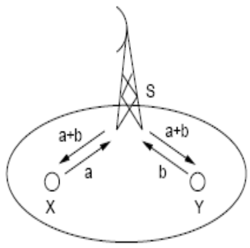


Figure: 1a Network Coding   Figure: 1b Wireless Network Coding

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].

1.2 Simplified Multicast Forwarding

SMF is an optimized multicast routing protocol which provides a plan for forwarding data packets from source to destination. SMF consists of two main components, first is Duplicate packet Detection (DPD) and Rely Set Selection (RSS) [8], [23].

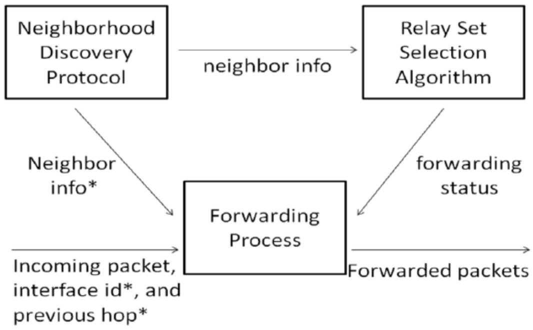


Figure 2 SMF Components

1) Multicast Duplicate Packet Detection (DPD): DPD is mainly used in the forwarding process in order to make sure that the received packets are not previously received. So it determine whether to drop a packet or not. There are two main duplicate packet detection mechanisms a- header content identification-based DPD (I-DPD) which uses a combination of packet headers and flow state in order to estimate temporal uniqueness of a packet. b- hash-based DPD (H-DPD) which employs hashing of the selected header fields with payload for the same effect.

2) Relay Set Selection (RSS): Relay Set Select produces a reduced relay set which is used in relying information across network. There are several rely set algorithm which are supported by SMF, for instance Source-based Multi-Point Relay (S-MPR) and Essential Connected Dominating Set (ECD).

DPD and RSS use Neighbourhood Discovery protocol for collecting and discovering information about nodes in a network.

II. 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

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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 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.

III. PROPOSED DRNC SCHEME ANALYSIS

As illustrated before network coding is allowing nodes to combine incoming information (data packets) before sending it. Practically data blocks are treated as a vector within a predefined base field. From theoretical perspective, NC has two main parts. 1) Global encoding matrix (Gt), such matrix can be either fixed or randomly selected. 2) Original data or information (X). The result Y can be generated by multiplying Gt by X. In Random network coding, Global encoding matrix plays a crucial role in resolving the original data packets at receiver side. When the size of such field is large, the probability of solving matrix will be high accordingly. But on the other hand, the overhead (field size) should be low as possible in order to achieve higher throughput. Many scholars such as [19] 28 and 216 are appropriate field sizes. A finite field (GF(pn)) can be defined as a field with a finite field order. Such order is always a power of a prime or prime. [20] Pointed out that intermediate nodes randomly and independently choose the coefficients of the received combination from a Galois Field (GF). The receiver should receive an amount of combinations of packets that equals to the number of source processes in order to decode the encoded packets. According to [21] the main operations of RNC are encoding and decoding (figure 3).

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 rabid and dynamic adaption of the best strategy for dealing with data packets.

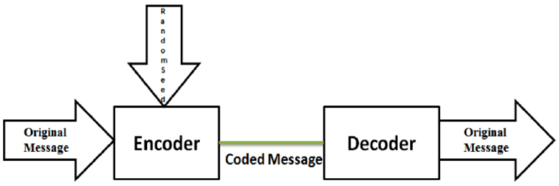

Figure 3, network coding main operations

A. Simulation setup

The setup design and implementation was developed using Network Simulator 2 (NS-2). The simulation is done to evaluate the proposed network coding solution for multimedia streaming traffic in ad hoc wireless mesh networks. It compares the proposed DRNC, with SMF which is one of the most common and optimized broadcast protocols as it was briefed earlier.

Performance criteria evaluation includes calculating latency, jitter and PDR by considering static and mobile scenarios for ad hoc wireless mesh networks.

A number of simulations carried out in this paper fall into two main categories: one is 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 paper 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 act as intermediate nodes which forward received packets to the nodes that are out of the base station(s) range.

[25] BonnMotion 2.0 tool was used to generate mobility scenarios which can run on any platform as it is written by Java programming language. Our scenarios are exported as for NS-2 simulator such movements' information is integrated with TCL scripts of NS-2.

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In such scenarios we make use of random waypoint mode (RW) and random street mode (RaST)
In RW mode, waypoints are the starting and ending positions of a node movement. Such positions are uniformly distributed in a specified domain. The mobility is characterised by a node moving from one waypoint to another in a straight line with constant velocity.

Our Base Stations are randomly surrounded by 100 wireless nodes that move individually around the specified simulation area.

RaST model provides more accurate simulation results; as it is using a vehicular mobility model on real cities, based on the operation of real vehicular traffic. [15]. According to [16, 24] there are two main parameters:

MeshNodeDistance, this is an optional parameter which adds a grid of static nodes to the scenario which represent mesh nodes. That means, it species the distance between two neighbouring nodes (in meters). This was set to 1 (disables adding these static nodes to the scenario).

MaxORSRequestIterations, this parameter route requests to Open Route Service. RaST mode may result in an error for certain pairs of source and destination. Such errors have a high probability to occur especially for positions that are chosen randomly as done within Random Street. Therefore, this parameter limits the maximum number of retries for a route request, where a new random position is generated for each retry.
The default value were used which is 30. (Simply it represents the max iterations until abort). But we have used the following in conjunction with the above. Distance metric for route requests which can be Fastest or Shortest, it is set to Fastest. Speed <minSpeed[m/s]> <maxSpeed[m/s]> it was set as Speed=1 30. MaxPause <maxPause[s]> it was set to MaxPause=30 seconds
The following metrics were used in order to configure mobility scenarios with NS2.
Number of nodes is 100, Minimum speed is 1 m/s, Maximum speed is 30 m/s, Dimensions 1300 * 1300, Model (Static, RW and RaST), and Traffic Type is UDP.
After that we created NS-2 movement files for (Static, Random Waypoint and Random Street mobility models). In order to be more accurate we make 10 scenarios (simulation runs) for each speed (which include speed 1, 5, 10, 15, 25 and 30).

B. Performance parameters

The management of routing protocols is with the following significant Quality of ervices (QoS) metrics for routine measures:
All results of such parameters are calculated by the average of 10 with the different movement scenarios. Our performance parameters compromise:

1 Packet Delivery Ratio (PDR)

PDR is describing the data loss rate that can be seen by transport layer. According to [17] PDR is the ratio of the total number of packets received (created by application layer sources) and the number of the received packets at the sink (destination).

$$PDR = \frac{\sum_{i=1}^{N} ReceivedPackets_i}{NumberOfSourcePackets \times NumberOfReceivers}$$

*N is the number of network nodes. *i is the number of received packets.
Generally higher PDR results in higher video quality.
2- Latency
Latency (average end-to-end delay) is the average amount of time elapsed between the source sending a packet and all receivers decoding the packet.

$$Jitter = \frac{1}{J-1}\sum_{i=1}^{J}(X_i - X)^2$$

*J is the number of received packets. *Xi is the received packet latency. *X is the average latency.
The highest acceptable latency for the video streaming applications was set as 200 ms, based on [22]
3- Jitter
Jitter is the variance of the latency between data packets from the same data flow. It can be illustrated as

$$Latency = \frac{\sum_{i=1}^{N} TimeDecoded_i}{N} - TimeSent$$

*N is the number of receiving nodes for such broadcasted packet.
Jitter plays a great role in real time applications such as video streaming, so it is vital to have smaller jitter in order to provide a high quality of data flow.

C. Static simulation results

The simulation results for SMF and DRNC are illustrated by the following diagrams. In diagram 1.1, PDR with DRNC is higher than the one with normal SMF (with DRNC PDR is greater than 0.8 in all ten investigated scenarios). Multimedia streaming requires high PDR as it have large data packets and requires high data rate. The main reason for such high PDR is that DRNC combine (Code) more than one packet together, therefore it increase the throughput and hence the PDR.

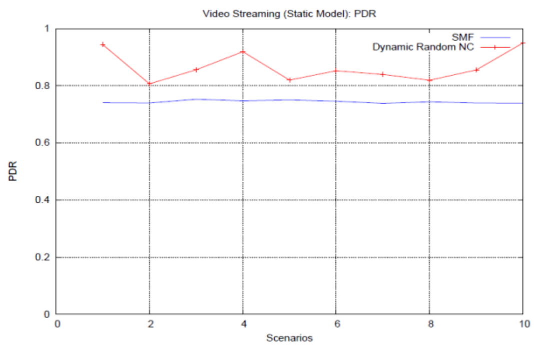

Diagram 1.1 PDR for DRNC and SMF – static model

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For the following diagrams 1.2 and 1.3, it is easy to notice that both are much less with DRLNC than with normal SMF. However extra coding in receiver or sender side badly affects the delay. But the significant increase of the throughput and the dynamic nature of the implemented code with the code efficiency played an important role in such results.

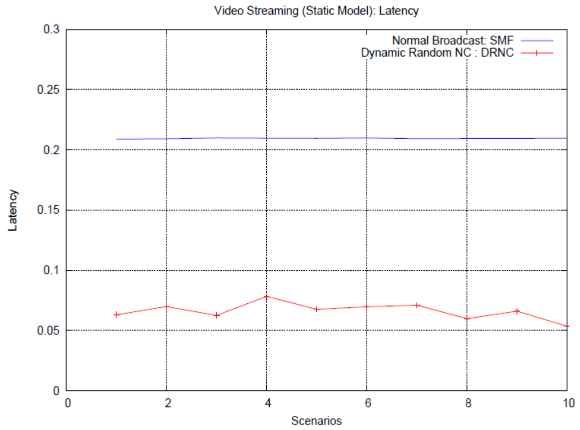

Diagram 1.2 Latency for DRNC and SMF – static model

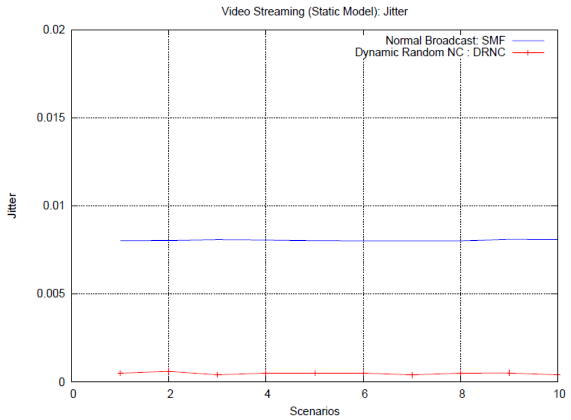

Diagram 1.3 Jitter for DRNC and SMF – static model

D. Mobility simulation results

From the following diagram, PDR with DRNC is still above 0.8 in random waypoint and random street movement scenarios. This is mainly because random network coding is dependent on the randomly selected coefficients. These coefficients are added to the packets' headers during encoding process to be used in at the sinks in decoding the encoded packets. That means that the performance of the protocol won't be highly affected by the mobility of the notes.

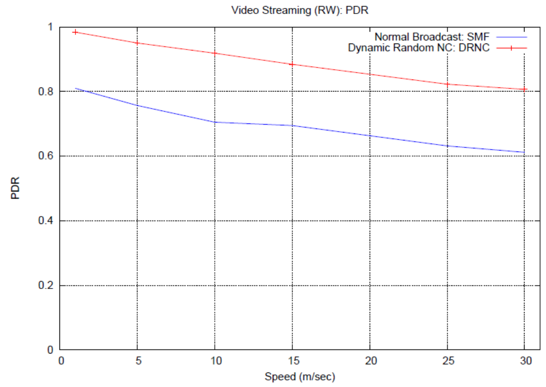

Diagram 2.1 PDR for DRNC and SMF – RW model

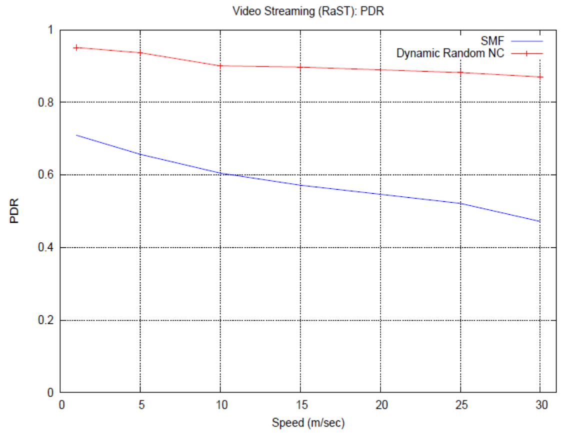

Diagram 2.2 PDR for DRNC and SMF – RaST model

Even for jitter and latency both remain lowest with DRNC. However with normal SMF it becomes much more especially when the mobility speed increases.

First results of Latency of DRNC remain lower than SMF protocol with either random waypoint or random street movements. That is because senders deliver the mixed (encoded) packets to sink nodes in a slightly shorter delivery time. Transmissions of data packets are spread out equally among all nodes. Also, the dynamic nature of our DRNC plays a great role in decreasing latency. The highest data rate nature of video streaming (almost 2000 kbps) has an effect with decreasing latency with DRNC as it is increasing the throughput (because of the reduction of transmissions) however the coding itself is badly affecting latency.

Actually mixing different data packets plays an important role in delivering it to receivers in a significant shorter time. On the other hand encoding (mixing) and decoding take time, but such time is not big enough to make DRNC result in more latency.

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With DRNC there is no additional buffering at the receiver side, so there is no wait time for more information to decode the received encoded packets. As, all required information including coefficients are stored in the encoded packets' headers.

Jitter values are higher especially when the movements speed in above 5 m/sec, this is because the higher probably of having conjunction. Moreover, extra buffering is badly affect jitter with SMF protocol.

DRNC has lowest jitter values for both movements Random waypoint and random street. That is mainly because more buffer in case of SMF. All earlier mentioned factors for latency with DRNC are also have a significant impact on jitter; for instance the dynamic use of coefficients, no additional buffering and the significant increase in throughput.

As with DRNC the number of messages which are successfully delivered per second is significantly higher than SMF.

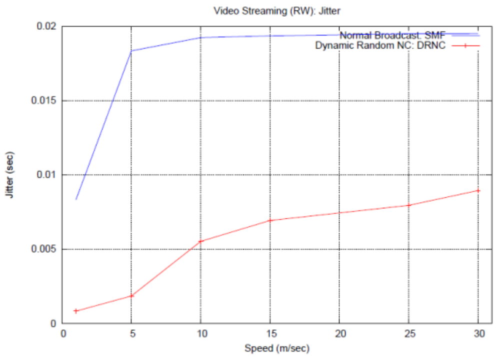

Diagram 2.3 Jitter for DRNC and SMF – RW model

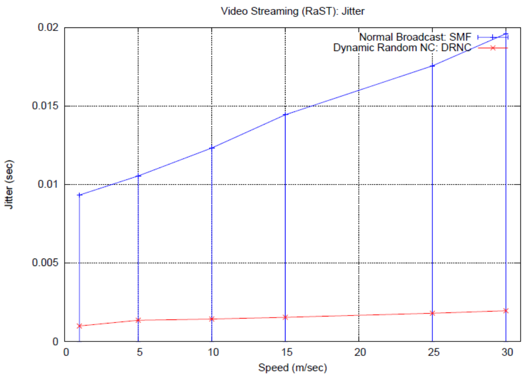

Diagram 2.4 Jitter for DRNC and SMF – RaST model

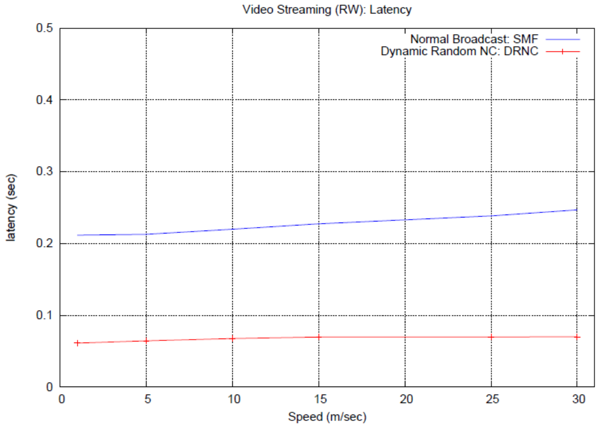

Diagram 2.5 Latency for DRNC and SMF – RW model

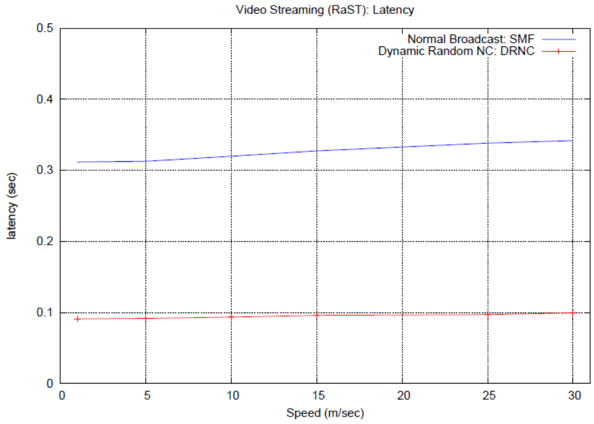

Diagram 2.6 Latency for DRNC and SMF – RaST model

IV. CONCLUSION

In this paper, we proposed a new solution for video streaming over wireless networks. Despite of the challenging nature of wireless and the strict video steaming requirements, our solution shows a significant quality of service. The proposed DRNC which is mainly based on random network coding was evaluated by a practical implementation with taking into consideration static and mobile movements of network nodes. All movements' simulation was based on several velocities in a relatively large network (100 nodes).
The results show that DRNC has lowest values of jitter and latency, and highest values of PDR over optimized broadcasting protocol (SMF). Hence DRNC can be acceptable for multimedia broadcasting over wireless networks.

Future work can be evaluating DRNC with developing the required code modification to be more suitable for WiMAX technologies and cellular systems. As, we believe that our novel scheme can be beneficial with smartphones.

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BIOGRAPHY

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.

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