The Influence Of Network Coding On The Performance Of Wireless Networks: A Survey

Article · June 2014

2 authors:

Nandhini Vineeth
BMS College of Engineering

H S Guruprasad
BMS College of Engineering

ISSN:2320-0790

An International Journal of Advanced Computer Technology

The Influence Of Network Coding On The Performance Of Wireless Networks: A Survey

Nandhini Vineeth[1], H.S.Guruprasad[2]

Department Of Computer Science And Engineering, BMSCE, Bangalore, India

Abstract: Network coding has been a challenging concept in networks compared to the most prevalent store and forward technique. This has been quite successful in both the wired and wireless arena. The enhancements achieved in the performance of various wireless networks with the usage of network coding are identified and a survey has been done. The paper also covers the survey on the usage of network coding on video streaming. The various forms of network coding like linear, random, dynamic, symbol level, generation based and intersession are discussed here.

Keywords: Wireless Networks, Network Coding, Linear Network Coding, Random Linear NC, Generation Based NC,DynamicNC,Inter-SessionNC

I. INTRODUCTION

This section gives an introduction to the areas where the works on network coding have been seen across.

A. Wireless Networks

Wireless networks are formed when devices connect with the other devices through electromagnetic energy in the air and start their communication using radio waves. These networks can be classified into infrastructure based networks and adhoc networks. Infrastructure based networks depend on an access point for all their communication whereas adhoc networks are self-organizing networks. Vehicular Adhoc NETworks (VANETs) [1] are a special type of Mobile adhoc NETworks (MANETs). These are the networks formed among vehicles for their communication and some of the works discussed here are on this.

B. Video

Digital video is obtained from a sequence of still images. The successive images received at a rate higher than 10 to 15 per second can give the feel of smooth motion. Huge volumes of data need to be handled for digital video used for transmission purposes. Hence compression algorithms are used before the video data is transmitted. MPEG-4 [2] is an optimal compression standard which supports different codecs. This has the advantages of giving better compression rates of natural video by adapting compression quality say main objects given more bits for compression than the background images. H.264 achieves much better video compression but complexity is more [3].

C. Video Streaming

Video transmission can be done in two ways. One is through interactive video in which the two way communication is possible and the other is streaming video where video contents are transmitted from one source to one or multiple receivers. The streaming server encodes the video sequences to be transmitted into small packets which are played in the receivers [4, 5].

D. Network coding (NC)

Network coding is the concept of mixing up packets whose contents are not dependent on one another by the relay nodes that are originated by different sources. After mixing up the packets they are transmitted to the destination in the network. These packets can either be the ones from the other nodes that are to be forwarded or may be its own. The network coding is applied interestingly in various diverse areas like information coding, networking, adhoc networks, distributed storage and many more. The basic concept of network coding can be understood with a simple example. Nodes that are out of transmission range of each other when wish to communicate, take the help of intermediate nodes. Such a scenario is shown in Fig I with a normal transmission between 2 nodes A and B. When A wants to transmit data to B, the transmissions encountered are A→C, C→B (2 transmissions) and similarly when B wants to transmit to A, transmissions are from B→C, C→A totaling to four. In network coding scenario shown in fig II, A and B send their data to C where C acts as a relay node and mixes the packets using say XOR and broadcasts the XORed data which results only in 3 transmissions. The receiver nodes retrieve the original packets which they are awaiting for by XORing the coded packets with their own. The number of transmissions here is thus reduced and hence improves the performance in terms of throughput, reduced congestion, delay etc.

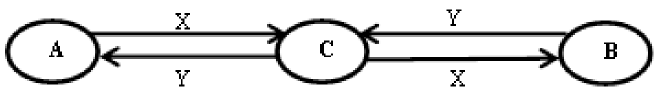

Fig. I Transmission of messages without network coding

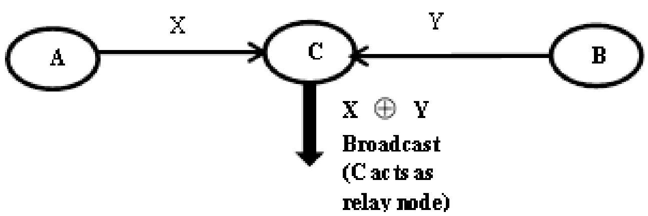

Fig. II. Transmission of messages using network coding

In another scenario as in Fig III, where the source nodes wants to multicast (more than one destination nodes) multiple packets, the relay node needs to know about the packets to be mixed up so that the destinations will be able to decode the required data. The source S wants to transmit packets P1 and P2 to two destinations say D1 and D2. It transmits P1 to I1 and P2 to I2. These send the same packets to their neighbors. The node I3 receives both the packets, apply network coding i.e. XOR P1 and P2 and sends the coded packet to I4. I4 transmits this coded packet to D1 and D2. D1 recovers P2 by XORing the coded packet with P1 that it has already received from I1. D2 also recovers P1 using the same procedure. Now D1 and D2 get the packets P1 and P2 sent from S. The dotted lines show the transmission of coded packets. Such types of network coding are termed as XOR coding.

Opportunistic listening or overhearing is another technique used in many environments. The nodes store a copy of packets they hear for a predetermined duration in spite of the fact that those packets not destined to them. For example, if a node N is waiting to receive a packet P3. It overhears a coded packet say P1 P2. N stores this coded packet. If it is able to get a coded packet P1 P2 P3 within a specified duration, it can XOR both the coded packets to retrieve P3. If not used in the predetermined duration, these are discarded by using Gaussian elimination.

There are also two classifications on network coding. One is the local coding and the other the global. The local coding is said to be done in networks where every intermediate nodes decode the received coded packets and new coded packets are generated. The global coding is said to be done in networks where intermediate nodes code the coded packets directly without decoding them. The final destination does the complete decoding of the packets [6, 7, 8].

II. FORMS OF NETWORK CODING

This section gives an introduction to the various forms of networks coding like linear, random, dynamic, symbol-level, generation based and intersession.

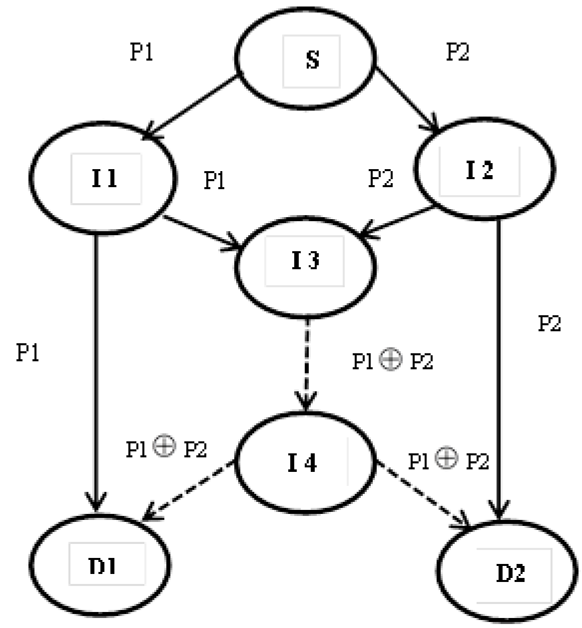

Fig. III. Transmission of multicast messages using network coding

A. *Linear Network Coding (LNC)*

The message to be transmitted is divided into blocks which in turn are a collection of frames and each block is represented in a matrix format which is combined linearly with an encoding vector chosen from the Galois field by the source to form the encoded packet. This is transmitted through the network. The encoding vector seen in the header of the received packets is used for decoding the coded packet. Gaussian elimination is used by the receiver and the actual packet is retrieved [9, 10].

B. *Random linear network coding (RNC)*

When the nodes transmit random linear combination of the packets it is termed as random LNC. The encoding vector is chosen randomly from the Galois field and the procedure is the same as LNC. This is proved to be successful many a times. Rank is another term used here to denote the actual number of frames that are combined to yield a coded packet. A packet is termed as an innovative packet if it is helpful in decoding the received packet and becomes a row in the decoding matrix of the receiver. This is said to increase the rank of the matrix. If it is non-innovative, it is reduced to zeros in the decoding matrix by the gauss elimination method used by the receiver.

The coded packet is identified by a blockid which denotes the minimum frame number of all the frames. The block size refers to the number of frames seen in the block. [11, 12, 13].

C. *Dynamic RNC*

The term dynamic is used as when RNC is used on dynamic scenarios. The RNC has been tested by the usage of dynamic random street and random way point scenarios. The quality of video is more here compared to static and mobile scenarios [14].

D. *Symbol Level Network Coding*

Network coding is done on the physical layer symbols to achieve a better level of granularity. A symbol denotes a small sequence of bits (not the entire packet). The downloading rate, reception reliability and error tolerance are better here as the coding is done in symbol level [15].

E. *Generation Based Network Coding*

The source node sending the file divides the file into blocks said to be generations. While combining, packets belonging to the same generation are to be combined. The coefficients are selected uniformly and randomly from the Galois field. [16]

F. *InterSession Network Coding*

The coding technique in which the packets combined are from different flows (different sources) is termed as inter session network coding and when done from the same flow (same source) it is termed as intra session network coding [17].

III. LINEAR NC & RANDOM LINEAR NC

A. *Linear Network Coding*

Park et. al. [18] in their work use random network coding to aim at low cost and low latency. Results show 100% delivery ratio and overhead reduction of 50%. The case study taken here is on multicasting from some cameras to security men on move. If it takes X seconds for one round of report of data, any data to be transmitted should be done within a maximum span of X seconds otherwise the data becomes obsolete. When a very aggressive scheme is used for lost data, it may add to the overhead. The work aims at keeping the packet loss in control while at the same time latency is also kept in check. Codecast works on loss recovery locally and path diversity to achieve its objective. One of the reasons for the increase in end-to-end delay drawback of codecast is the padding done with packets to make all the packets of the same size.

Three significant bits are maintained in the header of each packet**:** vldd, dist and nust. The bit **vldd** indicates if the source of the packet is a multicast receiver or has received some packets from its downstream neighbours with vldd set. When a node receives a predetermined number of packets with vldd not set, it stops forwarding for a preset duration. Thus, the unnecessary nodes are removed from the forwarding graph for a preset duration. The second bit **dist** refers to the maximum number of hops seen in the packets encoded received from the source node plus one for this node and the third bit **nust** denotes number of upstream nodes.

Sub graph selection is the term used to denote the optimal set of forwarding nodes and the frequency of injecting packets. The total number of packets transmitted by the nodes in the system to the number of packets received by the receivers is considered as a significant metric in calculation of the overhead and is shown to be 40% better than conventional multicast. As On Demand Multicast Routing Protocol (ODMRP) [19] is known to be an efficient protocol in lossy channels, performance comparison is done with the same and shown to be better.

A tree structure is built in the conventional multicasting which results in link breakage and hence more packet losses. In codecast, a forwarding structure i.e. A sub graph is used which helps in better delivery ratio.

Hulya et. al. [20] have aimed on the enhancement of the work RaDiO (Rate Distortion Optimization) [21] by giving a distributed solution for delay optimization in intermediate nodes. A mechanism ‘video aware packet delaying’ is designed which intentionally delays the transmission waiting for network coding opportunity in some cases. There could be some exceptions as a situation when two nodes say x and y have packets to be transmitted to the other. Consider x has 2 packets p1 and p2 and y has one p3. The node x transmits the packet p1 when the relay node say z without waiting for an opportunity to network code transmits p1. Now x has p2 and y has p3 which are transmitted to z which now can be network coded and broadcasted to x and y. This could be the best solution here instead of waiting for another packet to be network coded with p1.

The distortion value of every packet can be determined by the video source based on the communication with the intermediate nodes. The flow priority and the packet distortion value influence the overall importance of a packet. The routes are predetermined and the same path is used by all packets of the video. Lagrange’s multipliers help in finding the optimal transmission and delay policies. The complexity and overhead of the system are increased as the multipliers are exchanged among the nodes. When compared with noNC and NC-Radio, the performance achieved is better. More NC opportunities are seen when some selected scenes are delayed and the rate allocation is optimized.

Xiao et. al. [22] in their work deal with Diversity network codes (DNC) where many users send data to a single base station over a block fading channel. DNC are taken over finite fields. The resulting diversity order is shown as 2M-1 where M is the number of users. ie. To achieve a diversity order of 3, any two out of four network code words can be used to reconstruct the user messages. When the linear network coding is used, the Base station is able to retrieve back the data with minimum set of different coded packets. The performance is enhanced in medium to high Signal to Noise Ratio especially when the number of users is more.

B. Random Linear Network Coding

Xingjun et. al. [23] introduce their algorithm in as R^2NC which works with both redundant and random NC.

In the source, all rearranged packets are encoded by RNC and parity packets are generated by redundant NC against packet loss. In the intermediate nodes packets from same link are coded with redundant NC and the combination of packets from different links is done with random NC. The nodes need not know about their neighbours. Only when the global coefficient matrix maintained in the receiver has full rank, decoding is possible. The reliability and efficiency is improved by encoding packets in both application layer and network layer.

Unequal error protection can be obtained with different classification ranks for the group of pictures and per-frame bit rate. Classified packets are assembled into different blocks of packets. When the numbers of classification stages are more, blocks of packets are reduced and this affects the capability of R^2NC with coping with burst packet losses. Rank also plays a significant role. Insufficient ranks may result because of invalid coding vectors.

At the intermediate nodes, the level of redundancy is carefully decided as this influences the decodability. The procedure to assign valid coding vectors to generate global coding Matrix is also decided carefully.

Combination of unequal erasure protection is done by the bit stream rearrangement algorithm and R^2NC influences the priority layers. To enable partial decoding of a block and reduce the impact of the global coding matrix (GCM)'s rank deficiency, the GCM with ladder-shaped partition (LTGCM) is maintained throughout R^2NC process. The procedures to adjust the amount of redundancy after considering the packet loss and link capacity is shown and the order in which we perform the two types of coding at the source node and intermediate nodes is specified.

The objective of Rezende et. al.'s [24] work is to deliver packets in time for playback. REACT-DIS is the protocol that is able to give less end-to-end latency with an acceptable number of transmissions. NC is used to implement redundancy control. This is a receiver based solution where the intermediate nodes do not change the data but broadcast it to a predetermined percentage of nodes. The dynamic topology, delivery ratio and end-to-end delay are considered.

This is said to be receiver based as forwarding relay nodes are decided by the receiver and not by the sender. The nodes that receive the packets are supposed to broadcast it within a time interval. If it is observed before broadcasting that the packets are already in channel it can decide by itself if a broadcast is required. The decision is based on how many duplicate packets it has heard in the said interval. With a probability inversely proportional to the number of duplicates, it broadcasts. Hence it is observed that in high density areas the broadcasting is lesser and low density areas it is more. To reduce the impact of waiting nodes, when a node once wins as a forwarding node, it continues to be for a predetermined amount of time. Within that if it leaves the range, another node is elected.

Wang et. al. [25] in their work propose a cross layer solution with Network Layer-RNC and application layer-Video Interleaving. One video coding unit is divided into priority levels. Packets from different Video Coding Units (VCU) but with same priority form one generation. Packet-level interleaving scheme called RNC facilitated video interleaving is proposed here. Optimal interleaving degree and the optimal redundancy of each generation are studied. This technique becomes better than RNC for the reasons that the loss is distributed among various VCUs and also that original content can be retrieved easily by temporal/spatial error concealment. The interleaving length is the number of VCUs in a bit stream to be interleaved.

Abedini et. al. [26] aim at designing a minimum cost algorithm to transmit the data from the server where it is generated to a group of wireless devices. The objective here is to stream live content to the entire group of devices where the same frames are played out. Two interfaces seen in the smart phones and tablets which are B2D (Base station to Device) and D2D (Device to Device) are used here. As B2D transmission is expensive both in terms of cost and energy this works aims on doing minimum transmission of B2D channels and use D2D channels to the maximum as the transmission here could be inexpensive. i.e., a couple of devices receive it from the base station and then broadcast them to the other devices. The stream is sent in blocks which are further divided into chunks which are coded using RNC and transmitted. When the receiver receives enough blocks, it decodes and plays out data. The procedure discussed here is that a block created in say frame time f by the server should be played at the received by f+2 else it becomes useless and is dropped off. The device remains idle during this slot.

The selection of the device transmitting in the channel and the duration of transmission are given more significance in this work which could result in a better output. The paper includes simulation and presents the design to implement using android smart phones. In one time slot one B2D transmission and one D2D transmission are allowed as there are two interfaces. To achieve better D2D transmission, concepts from Queuing theory and Foster

Lyapunov stability criterion are used to find a sufficient field size for coding.

Saeed et. al.[27] have compared various NC protocols for adhoc networks with respect to video streaming when broadcast protocols like Bcast etc. are used. This Bcast protocol deals with broadcast of discovery messages transmitted among the nodes with the information of its own and all its neighbours. When a receiving node (say Y) finds that the sending node (say X) does not know one of Y's neighbours then it prepares a discovery message of its own to broadcast. If it receives another discovery message from any of its neighbours it drops it to create another according to the new requirement. Recent packets are buffered by the nodes. If a packet n is received without n-1, it requests its neighbour nodes to retransmit. If the node overhears other nodes transmitting it, it refrains from transmission.

SMF (Simplified Multicast Forwarding) is a multicasting or broadcast protocol. The performance of relays improved as a subset of 2 hop neighbours identified as relaying nodes. These nodes identify a subset of their one hop neighbour as MPR (multipoint relays). Duplicate packets are detected to avoid retransmission of repetitive packets. Partial Dominant Pruning- is same as SMF except that the source selects a subset of nodes for transmission using Partial Dominate Pruning (PDP) algorithm. As no information is carried about MPR, header size is small here. RNC is found to show a better performance than Bcast, SMF, partial dominant pruning.

Greco et. al. [28] have explored a combination of Expanding Window NC [29], Multiple Description Coding and a Rate-distortion optimized (RDO) scheduling algorithm to make a better transmission of video data. The EWNC proves to give a better quality in streaming compared to RNC as the number of packets included in a generation (coding window) is more here. This is done as instant decoding is possible in the receiver side at the cost of arranging the packets in the order taking the RD properties into consideration. Message is divided into sub streams called descriptions. The quality of video increases as the number of descriptions received is more. The global encoding vector which is chosen randomly from the Galois field is used as the random seed. This is mixed up with the original message and the coded packet is transmitted. The parameters such as jitter, latency, packet delivery ratio etc. have been worked on here to enhance the qualities.

IV. DYNAMIC NC & SYMBOL LEVEL NC

A. *Dynamic RNC*

Imane et. al. [13] aim at a better video quality using dynamic RNC. They claim to be the first to use random street walk mobility model via wireless networks. Quality of video streaming is improved with static and mobile movements compared to the simplified multicast forwarding (SMF). The two components of SMF are duplicate packet detection and relay set selection. LNC considers the data as a vector. This constructs linear equations out of data and uses Gaussian elimination to solve.

The global encoding matrix with random coefficient is combined with the original data and the coded combination is sent to the receiver where it is decoded. The base station nodes generate the video stream and transmit. When receiving the packets, decision is taken dynamically whether to create a new linear random combination. Two scenarios are discussed –mobile and immobile nodes. The parameters worked on are latency, jitter and packet delivery rate. Latency and jitter are considerably decreased. No wait time is encountered in the receivers as no additional buffering is done. Packets delivered are more compared to SMF.

B. *Symbol Level Network Coding*

Yang et. al. [30] have dealt with showing that symbol level NC (SLNC) gives a better performance than PLNC in VANETs. The authors have also worked on VANETs in [31] in which the advantages of SLNC for Live Multimedia Services (LMS) have been taken into consideration. This also makes use of SLNC with coordinated push mechanism (main core) to increase the performance of network. This is shown to give better results than PLNC (Packet Level NC). Minimizing bandwidth cost, dynamically changing network density etc. are taken care. Scenario considered here is a bidirectional vehicle set-AP in the end of the road broadcast multimedia data. One service and one control channel are used here. The communication of OBU and the AP is time slotted which are equal. Communication with GPS is also taken into consideration. Each vehicle synchronizes its clock with GPS. The procedure for symbol coding is that the original streaming content is divided into K (generation size) pieces, each of them with M symbols. NC is done within each generation. Receivers maintain a playback buffer where the receiver coded symbols are kept, a decoding matrix consisting of coding vectors of all the symbols received currently. The rank of each matrix is a symbol rank. Each coded symbol is checked for its usefulness based on its timely arrival, correctness and if it is innovative (increases the symbol rank) and then stored. When sufficient useful symbols are received, decoding is done to get the original symbols by performing Gaussian elimination on the corresponding matrix. Older generations are eliminated and the generations to be played next within alpha seconds are termed as priority generations.

Kim et. al. [32] have proposed a new solution which takes advantage of network coding at the symbol level rather than the packet level to improve throughput in spite of presence of errors. The multichannel wireless networks like WiMax uses the hybrid ARQ. The HARQ is a combination of high rate Forward Error Correction (FEC) and ARQ error control. Opportunistic overhearing cannot be applied here. The Soft-decision Values (SVs) which are the values that help in deciding a received signal to be a zero or one bit are used here. There are two uses for the same. The first is the usage in decoding for the construction of the set of coded blocks. The second is for the counting of dirty blocks. After the packets are received, the confidence of the blocks is checked using SVs. If the decoding fails, the confidence level of blocks is checked. An appropriate threshold value is selected. If the confidence level is below a specific threshold, the blocks are marked as dirty blocks. As the retransmissions are done with smaller number of bits with shorter transmission delays, the results show a shorter packet delivery time compared to the SOFT [33] and HARQ.

Sangki et. al [34] have presented SYNC (Symbol Level Network Coding) where a new concept of piggybacking a new packet with a retransmitted packet is done. Their implementation is on a software defined radio [SDR] platform. The results show a better performance in terms of throughput compared to normal retransmissions and SOFT.

V. GENERATION BASED NC & INTER-SESSION NC

A. Generation Based Network Coding (GBNC)

A file in source is divided into blocks called generations in generation based NC (GBNC). Generally, generation sizes are fixed. Normally large size of generation increases delay but maximizes NC and is preferred in lightly loaded networks. The smaller generation size is recommended in heavily loaded networks as it decreases delay with compromised throughput. Youghourta et. al. [35] in their work DYGES recommend a dynamic size for generations. The size of the generation is dynamically decided based on the size of the network, current congestion and losses in the network. The parameter values of current generation transmission decide the size of the next generation. The main objective is to maintain a steady delay around a given threshold termed as thresh, while maximizing the throughput. The generation based NC goes with a restriction that the packets combined are always from the same generation. Randomly and uniformly selected coefficients from a Galois field are used here. Gaussian elimination is used to recover back the original packets in the receiver. As the generation size is significant in various environments, DYGES changes the size dynamically.
Youghourta et. al. extend their work to RDYGES [36] for recovery of lost ACK. Opportunistic listening is used by the node sending the ACK to check if its neighbour has forwarded it. If not, it is retransmitted. When the destination is sure that its ACK has been received by the source, it sends a Self-Ack which is used to inform others that ACK is not lost and it stores the same in its own table.

Xiaofu Wu et. al. [37] focus on transforming non-zero delay networks to zero-delay networks considering the drawbacks of generation based network coding. The way the topology of the network needs to be changed based on the memory available in the intermediate nodes is shown. The influence of the same on the performance is also shown mathematically.

B. InterSession Network Coding

Hulya et. al. [38] focus on optimally allocating flow rates between users and codes.

Rate control in wireless networks which is considered to be a utility maximization problem with one hop intersession NC is shown to have a distributed solution. When NC is used, the achievable rate gets extended which affects the rate allocation. Sources can solve the rate control problem in a decentralized way. Flow rates decided by the higher layers may affect the NC opportunities. The utility is maximized when any two flows in opposite direction are coded together. Hence this work is on finding appropriate rate allocation for video over wireless.

The authors here propose that some scenes can be introduced additional delay to optimize rate allocation over longer time scales. As video streams are divided into scenes, the rate and priority given to each scene which depends on the video content need to vary. Eg. In a cricket match, hit of a six is to be given more importance than showing the spectators.

The rate control problem is solved by decomposing the same into sub problems at each source. Every video stream is divided into scenes which are featured by its duration, rate and utility. The total rate and utility are increased with an optimization interval T which is decided by every source. The transmission of scenes can be done in parallel instead of sequential. This makes the flows separate sharing channel capacity during T. To maintain the video quality without it being dropped below certain limit, a minimum rate is required to be met by every scene. Thus rate control and NC have been used up to achieve better quality.

1) ***Intra and inter session NC:*** The technique in which the packets of the same flow is combined is termed as intra session NC and that of different streams termed as inter session NC. Here in this work, Seferoglu et.al. I^2NC [39] have tried a combination of both of these to achieve a better performance. Parity packets are generated using intra session packets. The disadvantage of COPE [4] is that each node needs to have information of the packets the neighbour nodes are holding. This becomes troublesome in lossy networks. Here in this work, the authors say that only the loss probabilities of the overheard packets are required to proceed with their technique. Two schemes termed as I^2NC-state which requires knowing the state of the neighbours and I^2NC-stateless, where only the loss rate of links of their neighbours are to be known. An adaptation layer is considered at the interface between the TCP and the coding being done. The throughput has been checked for all possible combination of TCP and the two schemes have been checked with various topologies like X, Cross, Wheel etc., The performance is much better compared to no NC and COPE as it is loss resilient and does not depends on the knowledge of the neighbours state.

Tracey et. al. [40] present dynamic algorithms for wireless networks which are centralized power control algorithms and also algorithms for distributed session scheduling, routing and network coding at every node which are distributed. Wired networks are also considered where all the above algorithms are completely distributed. Back pressure routing. The backpressure algorithm is given more significance here which operates in slotted time. This is seen in environments the packets from multiple data streams arrive and must be delivered to appropriate destinations. In backpressure routing during every time slot, the direction to route the data is decided based on the difference in the queue sizes of the source and destination of links. Independent sources and correlated sources are considered and separate policies are suggested for each of the types like feedback about the amount of data received from each source. This paper works on the combination of this backpressure algorithm and network coding for multicast networks to show reduced interference effects.

VI. CONCLUSION

This survey paper gives an overview of how the basic concept of network coding has been used up by the researchers in different possible ways to enhance the performance of wireless networks. The parameters of throughput, delay, packet delivery ratio, jitter etc. have been worked on.

REFERENCES

[1]. Jarupan, Boangoat, and Eylem Ekici. "A survey of cross-layer design for VANETs." *Journal Ad Hoc Networks* Volume 9, Issue 5, July 2011, pp:966-983, DOI:10.1016/j.adhoc.2010.11.007

[2]. Yan, Bo, Kam W. Ng. "A survey on the techniques for the transport of MPEG-4 video over wireless networks." *IEEE Transactions on Consumer Electronics,* Volume 48, Issue 4, Nov 2002, pp: 863-873, DOI: 10.1109/TCE.2003.1196414

[3]. Stockhammer, Thomas, Miska M. Hannuksela, Thomas Wiegand. "H. 264/AVC in wireless environments." *IEEE Transactions on* Circuits *and Systems for Video Technology,* Volume 13, Issue 7, July 2003, pp: 657-673, DOI: 10.1109/TCSVT.2003.815167.

[4]. Guo, Meng, Mostafa H. Ammar, and Ellen W. Zegura. "V3: A vehicle-to-vehicle live video streaming architecture." *Pervasive and Mobile Computing* Volume 1 Issue 4 September 2005 pp: 404-424, DOI: 10.1016/j.pmcj.2005.08.001

[5]. Mittal, Meenakshi. "A Study of Live Video Streaming over Highway Vehicular Ad hoc Networks." *International Journal of Computer Applications* Volume 1, Issue 21, Jan 2010, pp: 86-90, DOI:10.5120/46-643.

[6]. R. Ahlswede, N. Cai, S. R. Li, and R. W. Yeung. "Network Information Flow" In IEEE Transactions on Information Theory, Volume 46 Issue 4, July 2000, DOI:10.1109/18.850663.

[7]. Katti, Sachin, et al. "XORs in the air: practical wireless network coding." *IEEE Transactions on Networking",* Volume 36, Issue 4, June 2008. DOI:10.1109/TNET.2008.923722

[8]. Hu, Sachin Katti Dina Katabi Wenjun, Hariharan Rahul Muriel Médard. "The Importance of Being Opportunistic: Practical Network Coding for Wireless Environments.", Newsletter ACM SIGCOMM Computer Communication Review, Volume 36, Issue 4, October 2006, DOI: 10.1145/1159913.1159942.

[9]. Li, S-YR, Raymond W. Yeung, and Ning Cai. "Linear network coding." *IEEE Transactions on* Information *Theory",* Volume 49, Issue 2, February 2003 pp: 371-381, DOI: 10.1109/TIT.2002.807285.

[10]. Fragouli, Christina, Jean-Yves Le Boudec, and Jörg Widmer. "Network coding: an instant primer." *ACM SIGCOMM Computer Communication Review,* Volume 36, Issue 1 January 2006 pp: 63-68, DOI: 10.1145/1111322.1111337

[11]. Najeem, M., and C. Siva Ram Murthy. "On enhancing the random linear network coding." *17th IEEE International Conference on Networks, (ICON),* Dec 2011, 10.1109/ICON.2011.6168483

[12]. Ho, Tracey, et al. "The benefits of coding over routing in a randomized setting", *IEEE International Symposium on Information Theory,* July 2003. DOI:10.1109/ISIT.2003.1228459

[13]. Ho, Tracey, et al. "A random linear network coding approach to multicast." *IEEE Transactions on Information Theory,* Volume 52, Issue 10, Oct 2000, pp: 4413-4430. DOI:10.1109/TIT.2006.881746

[14]. Saroit, Imane Aly, 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,* Volume 2, Issue 7,July 2013

[15]. S. Katti, D. Katabi, H. Balakrishnan, and M. Medard, "Symbol-level network coding for wireless mesh networks", *Newsletter ACM SIGCOMM Computer Communication Review* Volume 38 Issue 4, October 2008, Pages 401-412, DOI: 10.1145/1402946.1403004

[16]. Lee, Uichin, et al. "Understanding Processing Overheads of Network Coding Based Content Distribution in VANETs.", International Journal IEEE Transactions on parallel and distributed systems, Volume 24 Issue 11 November 2013 pp:2304-2318, DOI: 10.1109/TPDS.2012.306

[17]. Wang, Chih-Chun, and Ness B. Shroff. "Intersession network coding for two simple multicast sessions." *Proc. Annual Allerton Conference on Communication, Control, and Computing.* September 2007.

[18]. Park, Joon-Sang, et al. "Codecast: a network-coding-based ad hoc multicast protocol." *IEEE Wireless Communications*, Volume 13, Issue 5, October 2006, pp: 76-81, DOI: 10.1109/WC-M.2006.250362

[19]. Yi, Yunjung, et al. "On-demand multicast routing protocol (ODMRP) for ad hoc networks." *IETF MANET Working Group Internet-Draft* (2002).

[20]. Seferoglu, Hulya, and Athina Markopoulou. "Delay-optimized network coding for video streaming over wireless networks." *IEEE International Conference on Communications (ICC)*, May 2010, pp: 1-5, DOI: 10.1109/ICC.2010.5502651

[21]. Sullivan, Gary J., and Thomas Wiegand. "Rate-distortion optimization for video compression", *IEEE Signal Processing Magazine*, Volume 15, Issue 6, November 1998, pp: 74-90, DOI: 1053-5888/9

[22]. Xiao, Ming, and Mikael Skoglund. "Multiple-user cooperative communications based on linear network coding." *IEEE Transactions on Communications*, Volume 58, Issue 12, December 2010, pp: 3345-3351, DOI: 10.1109/TCOMM.2010.092810.100045

[23]. Zhang, Xingjun, et al. "Joint redundant and random network coding for robust video transmission over lossy networks." *Journal Mobile Information Systems* Volume 8, Issue 3 July 2012, pp: 213-230, DOI: 10.3233/MIS-2012-0139.

[24]. Rezende, Cristiano, Richard W. Pazzi, and Azzedine Boukerche. "A reactive solution with a redundancy-based error correction mechanism for video dissemination over vehicular ad hoc networks." *Proceedings of the 15th ACM international conference on Modelling, analysis and simulation of wireless and mobile systems*, pp: 343-352, October 2012. DOI:10.1145/2387238.2387296

[25]. Wang, Hui, and C-C. Jay Kuo. "Robust video multicast with joint network coding and video interleaving." *Journal of Visual Communication and Image Representation, Volume* 21, Issue 2, Feb 2010, pp: 77-88, DOI: 10.1016/j.jvcir.2009.06.008

[26]. Abedini, Navid, et al. "Realtime streaming with guaranteed QoS over wireless D2D networks." *Proceedings of the fourteenth ACM international symposium on Mobile ad hoc networking and computing*, pp: 197-206, JUL-AUG 2013, DOI: 10.1145/2491288.2491308

[27]. Saeed, Basil, et al. "Multimedia Streaming for Ad Hoc Wireless Mesh Networks Using Network Coding.", *International Journal of Communications, Network and System Sciences Volume* 6 Issue 5, May 2013, PP. 204-220 DOI: 10.4236/ijcns.2013.65024

[28]. Greco, Claudio, et al. "A network coding scheduling for Multiple Description video streaming over wireless networks." *Signal Processing Conference (EUSIPCO), 2012 Proceedings of the 20th European*. IEEE, Aug 2012, pp: 1915-1919, ISSN: 2219-5491.

[29]. Nybom, Kristian, Stefan Gronroos, and Jerker Bjorkqvist. "Expanding window fountain coded scalable video in broadcasting." *IEEE International Conference on Multimedia and Expo (ICME)*, July 2010, *pp: 516-521*, DOI: 10.1109/ICME.2010.5583861

[30]. Yang, Zhenyu, Ming Li, and Wenjing Lou. "Codeplay: Live multimedia streaming in vanets using symbol-level network coding." ,*The 18th IEEE International Conference on Network Protocols*, pp:223-232, DOI: 10.1109/ICNP.2010.5762771

[31]. Li, Ming, Zhenyu Yang, and Wenjing Lou. "Codeon: Cooperative popular content distribution for vehicular networks using symbol level network coding." *IEEE Journal on Selected Areas in Communications*, Volume 29, Issue 1, August 2012, pp: 223-235, DOI:10.1109/JSAC.2011.110121

[32]. Kim, Ronny Yongho, Jin Jin, and Baochun Li. "Drizzle: cooperative symbol-level network coding in multichannel wireless networks.", *IEEE Transactions on Vehicular Technology*, Volume 59, Issue 3, March 201, pp: 1415-1432, DOI:10.1109/TVT.2010.2040095

[33]. Woo, Grace R., et al. "Beyond the bits: cooperative packet recovery using physical layer information." *Proceedings of the 13th annual ACM international conference on Mobile computing and networking*, September 2007, pp: 147-158, DOI:10.1145/1287853.1287871

[34]. Yun, Sangki, Hyogon Kim, and Kun Tan. "Towards zero retransmission overhead: A symbol level network coding approach to retransmission.", *IEEE Transactions on Mobile Computing*, Volume 10, Issue 8, Aug 2011, pp: 1083-1095, DOI:10.1109/TMC.2010.248.

[35]. Benfattoum, Youghourta, Steven Martin, and Khaldoun Al Agha. "DYGES: A network-aware Generation-Based Network Coding for multicast flows." *IEEE Conference on Vehicular Technology Conference (VTC Fall)*, Sep 2012, pp: 1-5, DOI:10.1109/VTCFall.2012.6399304

[36]. Benfattoum, Youghourta, Steven Martin, and Khaldoun Al Agha. "QoS for real-time reliable multicasting in wireless multi-hop networks using a Generation-Based Network Coding.", *Journal Computer Networks : The international journal of computer and telecommunications networking*, Volume 57, Issue 6, April 2013, pp: 1488-1502, DOI:10.1016/j.comnet.2013.02.005

[37]. Wu, Xiaofu, Chunming Zhao, and Xiaohu You. "Generation-based network coding over networks with delay." *IFIP International Conference on Network and Parallel Computing, 2008, October 2008*, pp:365-368, DOI:10.1109/NPC.2008.57

[38]. Seferoglu, Hulya, and Athina Markopoulou. "Distributed rate control for video streaming over wireless networks with intersession network coding." *17th International Packet Video Workshop, m=May 2009*, pp: 1-10, DOI: 10.1109/PACKET.2009.5152165

[39]. Seferoglu, Hulya, Athina Markopoulou, and K. K. Ramakrishnan. "Intra-and Inter-Session Network Coding in Wireless Networks." ,*Proceedings IEEE, INFOCOM 2011*, April 2011, pp: 1035-1043.

[40]. Ho, Tracey, and Harish Viswanathan. "Dynamic algorithms for multicast with intra-session network coding." *IEEE*

Transactions on Information Theory, Volume 55, Issue 2, Feb 2009, pp: 797-815, DOI: 10.1109/TIT.2008.2009809