Practical Design of Network Coded Multicast over Satellite

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***Abstract*—In this paper, we present the practical design of systematic random network coding (SNC) for multicast over satellite. In particular, the satellite coverage is over a large geographical area that consists of several users. These users may face different channel conditions and undergo different packet loss rates. In this work, we show two regions of transmission. First, the general multicast region where all users subscribed to the multicast channel can recover all the data packets and second, we identify the multi-unicast region where only users with good channel conditions can recover all the data packets but those with bad channel accept transmission losses. Further, we present the architectural and encapsulation feasibility of SNC at satellite-dependent and satellite-independent layers. We derive theoretically and by simulation the benefits of SNC against state-of-the-art end-to-end coding at these layers. Our results show that: (i) SNC at link layer can achieve up to 26.90% and 24.26% higher maximum achievable rates for multicast and multi-unicast, respectively and (ii) SNC at application layer requires up to 6.66% and 30.02% smaller available network bandwidth for multicast and multi-unicast respectively.**

***Index Terms*—Network coding; Multicast; Satellite communication; Achievable rates.**

I. Introduction

This paper extends our earlier work [1]. In this paper, we consider more detail analysis of system model and introduces the results and analysis for multicast and multi-unicast using network coding at both satellite-dependent and satellite-independent layers.

Network coding (NC) extends traditional network operations from routing and store-and-forward to more powerful operations that allow for coding information at intermediate nodes. NC was first introduced in [2], where it is shown that NC can achieve the min-cut capacity of the error-free networks by allowing coding at intermediate nodes. In subsequent work [3], it is also shown that the linear encoding and decoding operations at intermediate nodes are sufficient to achieve the min-cut max-flow performance. These results lead to the investigation of efficient linear encoding and decoding strategies for network coding. As a result, it is shown that random linear network coding (RNC) [4], where information packets transmitted in the network are random linear combinations of the original data packets, is asymptotically capacity achieving.

The main philosophy of using RNC as a capacity-achieving network coding scheme is that it allows the practical application of network coding in a distributed manner. For example, there is no need of a centralized architecture to take care of coding coefficients used at all the nodes. The coding coefficients are generated at random and they are send within the packet headers [5]. Although RNC has several theoretical benefits, it has three main limitations for the practical applications of network coding: high complexity, high delay, and high overhead. The high decoding complexity is due to the use of Gaussian elimination (GE) algorithm. In order to solve a system of linear equations, GE needs to solve a densely filled decoding matrix. This results into the high decoding complexity. The high delay is due to the time the receiver waits for the arrival of the complete block in order to start the decoding process and the high overhead is due to the coding coefficients, which are attached as a side information with the coded packets [5] in practical network coding schemes. Therefore, RNC is not the most efficient form of network coding and because of all the aforementioned constraints, it can adversely effect the applications like real-time video streaming, where there can be impairments in video quality due to high complexity and delay.

Recently, systematic network coding (SNC) [6]-[14] has been investigated as a powerful practical network coding solution. If systematic coding is used, the sinks can receive both uncoded and coded packets. There are three main benefits of using systematic coding. Firstly, the sink has to decode only the packets, which have not arrived in their original form. Hence, some rows of the decoding matrix are singleton and contain only one non-zero element. In this case, the decoding is done over a sparse decoding matrix that contains several zero elements that reduces the decoding complexity significantly. Secondly, the sink does not have to wait for complete block to start recovering packets. The packets that are received in their original form, are recovered instantly, which decreases the overall per-packet delay. Finally, the systematic packets do not have overhead of coding coefficients, as these are not the encoded packets. This reduces the overall overhead significantly. Therefore, SNC can overcome all the limitations imposed by RNC. In this work, we explore the benefits of using SNC for transmission over satellite.

A. Network coding over satellite

In the literature, there are specifically two directions of work on the use of network coding for transmission over satellite. The first direction focuses on throughput improvement. In [15][16], network coding is shown to enhance throughput by load balancing and allocating coded packets across different beams in multi-beam satellite systems. Further, works in [17][18] take advantage of orthogonal transmission available using multi-link reception in multi-beam satellite systems. In [17][18], network coding is shown to provide benefits in terms of higher throughputs subject to the location of user terminals. In [19][20], network coding is shown to improve end-user quality-of-experience for video streaming application over relay satellites.

A different direction looks at the application of network coding to counteract packet losses and guarantee higher reliability. It has been shown that network coding together with congestion control algorithm can provide many-fold improvements than existing transport layer protocols [21]. In addition, it has been shown in [22], that unequal-protection aware overlapping network coding together with congestion control algorithms can provide improvements in quality-of-experience (QoE) of video streaming. Further, network coding implementation at link layer has shown to provide several advantages in terms of reliability, complexity, delay, etc. for unicasting over satellite networks with several intermediate nodes [11]-[13]. In [23], it is also shown that it can be used to cope with packet losses, thus counteracting prediction failures of the handover procedures in smart gateway diversity satellite systems.

B. Our contributions

Existing results on the use of SNC has dealt mainly with unicast with a source and a sink connected via several intermediate nodes. In this paper, we extend existing results and investigate the use of SNC for multicast and multi-unicast over satellite. We identify two regions of transmission, one for multicast and another for multi-unicast with one in-network (re)encoding node to increase the achievable rates and the reliability of satellite networks. The one in-network re-encoding node can be a satellite gateway or other node at satellite ground station.

Further, we also present the application of network coding in two different sets of layers of a satellite network protocol architecture. These are satellite-dependent layers (link and physical layers) and satellite-independent layers (application, transport and IP layers). In particular, the satellite-dependent layers are of interest for the system operators while the satellite-independent layers are of interest for application developers who have access to the data flowing in these layers but not to the system operational kernel. Specifically, we present first the implementation of network coding at the link layer of the satellite systems and second, the implementation of network coding at the application layer for better internet communication over satellite systems. The overall architectural design is also presented showing the practical feasibility of SNC over different layers.

The rest of the paper is organized as follows. In Section II, we formalize the system model. In Section III, we describe the proposed SNC scheme for multicast and multi-unicast transmission. In Section IV, we present the theoretical analysis and derivation of theoretical expressions for the reliability and the achievable rate. Section V and Section VI present the implementation of SNC at the link layer and application layer respectively. Section VII presents the simulation results and Section VIII concludes this paper.

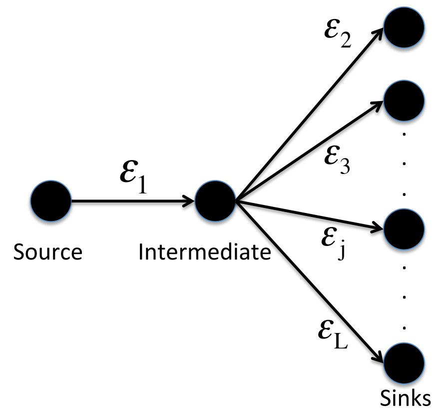


Figure. 1: System model.

II. SYSTEM MODEL

We consider a system topology (Figure 1), where a source is connected to all the sinks via an intermediate node. This system topology is a relevant case in satellite systems where there is one intermediate node, which could be a gateway (or others) and there can be several sink nodes which are the users distributed in a large geographical area undergoing different packet loss rates. Our theoretical derivations and simulation results on the reliability and the achievable rates show the benefits of network coding with respect to state-of-the-art end-to-end forward erasure codes (FEC) codes like Reed-Solomon (RS) codes for both multicast and multi-unicast.

Consider that a source node has K data packets to send to $L-1$ sink nodes. Each packet is a column vector of length M over a finite field $\mathbb{F}_q$. The set of data packets in matrix notation is $\mathbf{S} = \begin{bmatrix} \mathbf{s}_1 & \mathbf{s}_2 & . & . & . & \mathbf{s}_K \end{bmatrix}$, where $\mathbf{s}_t$ is the t^{th} data packet. The source is connected to all the sinks via an intermediate node as shown in Figure 1. All the links are modeled as memoryless erasure channels. There are L links in the network. The erasure probability from the source to the intermediate node is denoted by ϵ_1 and the erasure probability from the intermediate node to the sink node j is denoted by $\epsilon_j, j = 2, .., L$.

We assume there is no feedback from the sinks (or from the intermediate node) due to the inherent large latency of satellite systems. We also assume that packet transmissions occur at discrete time slots such that each node can transmit one packet per time slot. We will also assume that the coding schemes run for a total of N time slots (N is larger than or equal to K) and every node (except the sinks) transmits a packet in each time slot $t = 1, 2, ..., N$.

III. SYSTEMATIC NETWORK CODING FOR MULTICAST AND MULTI-UNICAST

A. Encoding at the source node

The SNC encoder sends K data packets in the first K time slots (systematic phase) followed by $N-K$ random linear combinations of data packets in the next $N-K$ time slots (non-systematic phase). Let,

$$\mathbf{X} = \mathbf{SG}$$

represent K systematic packets and $N - K$ coded packets transmitted by the SNC encoder during N consecutive time slots. The generator matrix

$$\mathbf{G} = \left[\begin{array}{cc} \mathbf{I}_K & \mathbf{C} \end{array} \right]$$

consists of the identity matrix $\mathbf{I}_K$ of dimension K and $\mathbf{C} \in \mathbb{F}_q^{K \times N-K}$ with elements chosen randomly from a finite field $\mathbb{F}_q$. The code rate is given by $\rho = \frac{K}{N}$.

B. Re-encoding at the intermediate node

The SNC re-encoder performs re-encoding operations at every time slot and sends N packets to the sink nodes. Let,

$$\mathbf{X}_I = \mathbf{X}\mathbf{D}_1\mathbf{T}$$

represent N packets transmitted by the SNC re-encoder during N consecutive time slots, where $\mathbf{D}_1 \in \mathbb{F}_q^{N \times N}$ represents erasures from the source node to the intermediate node and $\mathbf{T} \in \mathbb{F}_q^{N \times N}$ represents the re-encoding operations at the intermediate node.

The erasure matrix $\mathbf{D}_1$ is an $N \times N$ diagonal matrix with every diagonal component zero with probability ϵ_1 and one with probability $1 - \epsilon_1$.

The re-encoding matrix $\mathbf{T}$ is modeled as an upper triangular matrix. The non-zero elements of $\mathbf{T}$ are selected as follows. During the systematic phase, if a packet $\mathbf{s}_t$ is lost, i.e., $\mathbf{D}_1(t,t) = 0$ then the non-zero elements of the t^{th} column of matrix $\mathbf{T}$ are randomly selected from $\mathbb{F}_q$. This represents that if the systematic packet is lost from the source node to the intermediate node, then the intermediate node transmits a random linear combination of the packets stored in its buffer. If a packet $\mathbf{s}_t$ is not lost, i.e., $\mathbf{D}_1(t,t) = 1$ then the t^{th} column of matrix $\mathbf{T}$ is the same as the t^{th} column of identity matrix $\mathbf{I}_N$. This represents that the intermediate node forwards this systematic packet to the sinks. During the non-systematic phase, the intermediate node sends a random linear combination of the packets stored in its buffer and all the non-zero elements of last $N - K$ columns of $\mathbf{T}$ are chosen randomly from the finite field $\mathbb{F}_q$.

C. Decoding at the sink nodes

Let $\mathbf{Y}_j = \mathbf{X}_I\mathbf{D}_j, j = 2, 3, ..., L$ represents N packets received by the sink node j where $\mathbf{D}_j$ represents erasures from the intermediate node to the sink node j. $\mathbf{D}_j$ is $N \times N$ diagonal matrix of the same type as $\mathbf{D}_1$ but with erasure probability ϵ_j. If the sink node j does not receive any packet in time slot t then the t^{th} column of $\mathbf{Y}_j$ is a zero column.

The overall SNC coding strategy can be expressed using a linear operation channel (LOC) model, where the output at the sink node j is $\mathbf{Y}_j = \mathbf{SGH}_j$ where $\mathbf{H}_j = \mathbf{D}_1\mathbf{T}\mathbf{D}_j$ represents the transfer matrix from the source to the sink j. We assume that the coding vectors are attached in the packet headers so that the matrix $\mathbf{GH}_j$ is known at the sink j. However, the overhead, due to the attached coding vectors, is kept low due to the use of systematic coding (coding vectors are not attached with the systematic packets). The decoding is progressive using the Gaussian Jordan algorithm as in [11]. All the K data packets are recovered when K innovative packets are received at the sink j, i.e., $rank(\mathbf{GH}_j) = K$.

IV. THEORETICAL ANALYSIS

In this section, we present the theoretical expressions of the average reliability and the average achievable rate of the considered topology. First, let us note that the capacity of the topology is the min-cut of the network, which is given by $\min_j(1 - \epsilon_j)$.

Let us now define η as the residual erasure rate of any link that could be achieved after the overall coding and decoding operations. The reliability of the link is given by $(1-\eta)$. Based on the definitions of the residual erasure rate and using the definition of achievable rate from [25], we define the average achievable rate of the considered topology as,

$$R_{av} = \rho\,(1 - \eta_{av}) \tag{1}$$

with,

$$\eta_{av} = \frac{1}{L-1}\sum_{j=2}^{L}[1 - (1 - \eta_1)(1 - \eta_j)] \tag{2}$$

as the average reliability of the considered topology where η_1 is the residual erasure rate from the source node to the intermediate node and η_j is the residual erasure rate from the intermediate node to the sink node j. Here,

$$\eta_l = \epsilon_l(\phi_{l1} + \phi_{l2}), l = 1, 2, ..., L. \tag{3}$$

where ϵ_l is the probability by which a systematic packet is lost at link l.

Let us now derive the residual erasure rate η_l. Note that the residual erasure rate is zero if the decoding is sucessful and all the data packets are recovered. Further, a decoding failure will result into a finite value of residual erasure rate. The decoding failure can be due to two reasons. First, when the number of total packets received is less than K. The probability of this event is represented as,

$$\phi_{l1} = Pr(A < K - 1) \tag{4}$$

where A is the total number of packets received. Second, when the number of total packets received is more than K but they are not linearly independent to have successful decoding. This event corresponds to a not full rank decoding matrix and its probability is represented as,

$$\phi_{l2} = Pr(A \geq K - 1)Pr(rank(\mathbf{GH}_j) < K) \tag{5}$$

Assuming Bernoulli distributed erasures and using the probability mass function $\alpha(f, v, p) = \left(\begin{array}{c} v \\ f \end{array}\right)(1-p)^f(p)^{v-f}$, we have,

$$\phi_{l1} = \sum_{f=0}^{K-1}\alpha(f, N-1, \epsilon_l), \tag{6}$$

$$\phi_{l2} = \sum_{f_1=0}^{K-1} \alpha(f_1, K, \epsilon_l) \sum_{f_2=K-f_1}^{N-K} \alpha(f_2, N-K, \epsilon_l)\left(1-\pi\left(f_2, K-f_1\right)\right). \quad (7)$$

where f_1, f_2 are the number of systematic and coded packets received, respectively, and $\pi\left(f_2, K-f_1\right)$ is the probability of f_2 coded packets having $K-f_1$ degrees of freedom. When there are f_1 degrees of freedom from f_1 systematic packets and $K-f_1$ degrees of freedom from f_2 coded packets then $rank(\mathbf{GH}) = K$. Using (equation 7, [26]), we have the exact expression for $\pi\left(f_2, K-f_1\right) = \prod_{f_3=0}^{K-f_1-1} 1-q^{f_3-f_2}$. From the above expressions, we can obtain the average residual erasure rate and average achievable rate using (2).

In the next sections, we will present the practical application of network coding at different layers of the satellite systems. In particular, we consider the application at link and application layers of the satellite systems.

V. Implementation in link layer of satellite systems

A. State-of-the-art link layer protocols in satellite systems

The current state-of-the-art link layer protocols in the satellite systems provide efficient encapsulation of network layer (IP) protocol data units (PDUs) over the physical layer frames. For example, generic stream encapsulation (GSE) protocol [27] in digital video broadcasting by satellite - second generation (DVB-S2) based systems is used as a link layer protocol to encapsulate network layer IP packets.

The existing link layer forward erasure correction (LL-FEC) frameworks in the satellite systems are mainly based on RS or Raptor codes [28]. However, the main limitation of the existing frameworks is that they operate only in end-to-end fashion and do not utilize the coding opportunities at the intermediate node. In this section, we will present an architectural and encapsulation framework to enable link layer systematic network coding (LL-SNC) at the source and at the intermediate node of the satellite systems.

B. LL-SNC architecture and encapsulation

In Figure 2, we present the complete information flow with LL-SNC architecture and LL-SNC encapsulation, where IP packets are transmitted from the source and recovered at the sinks. This figure represents the case when there is only one sink in the network. When there are several sinks, the same LL PDUs are transmitted from the intermediate node to all the sink nodes.

At the source, the network layer IP packets are encapsulated into an LL-SNC frame. The LL-SNC frame consists of an application data table (ADT) to store IP PDUs, a network coding data table (NCDT) to store network coded packets and a coefficient data table (CDT) to store coding coefficients. The data from the LL-SNC frame is then encapsulated into LL PDUs. The LL PDUs are then encapsulated into the physical (PHY) frames.

At the intermediate node, the payload of correctly received LL PDUs is stored in the LL-SNC frame. The IP PDUs are stored in the ADT of the intermediate node. The coded packets and the corresponding coefficients are stored in NCDT and CDT of the intermediate node. When the intermediate node receives LL PDU without error, it sends the LL PDU to the sink node and also stores it in the LL-SNC frame. When the intermediate node receives LL PDU with errors, it discards the LL PDU and generates new coded packet and coding coefficients as explained in Section III. These new coded packets and the corresponding coefficients are stored in NCDT and CDT of the intermediate node.

At the sink node, the correctly received LL PDUs are stored in the LL-SNC frame. The IP PDUs are stored in the ADT of the sink node. The coded packets and the coding coefficients are stored in NCDT and CDT, respectively. The progressive decoding is performed and the lost IP PDUs are recovered. These IP PDUs are then passed to the upper layers.

The encapsulation process is similar to the encapsulation of LL-FEC over GSE [28] but with some modifications to accomodate network coding parameters. A source block (ADT) consists of K columns and stores ADUs. Now, ADUs are arranged column wise starting from the upper left corner. If an ADU does not fit in one column, it continues at the top of the following column and so on. If the ADT is not completely filled then the zero-padding bytes are inserted in last column to fill it completely. Each ADU is then encapsulated in a single or multiple RTP packets. The FEC block (NCDT) contains $N-K$ columns with $N-K$ coded packets and the coefficient block (CDT) contains $N-K$ columns with $N-K$ set of coding coefficients. Each coded packet from NCDT and the corresponding coding coefficients from CDT are encapsulated in one RTP packet. The first K bytes of RTP payload contain K coding coefficients followed by the corresponding NCDT column. The value of K is signaled through the RTP header of the RTP packet. Finally, CRC-32 is added with every RTP packet to detect errors in RTP packets at the receiving end. Now, if an ADU is lost, then the corresponding part of the column or the complete column is also lost. The progressive decoding is performed and lost columns (or lost part of columns) in ADT are filled with the recovered data.

VI. Implementation in application layer of satellite systems

A. State-of-the-art application layer protocols in satellite systems

The current state-of-the-art application/transport layer protocols primarily include transmission control protocol (TCP) based schemes, which are most commonly used to guarantee reliabiity. TCP used retransmission mechanism to recover from packet losses. However, TCP and TCP variants have limited performance over satellite due to long round-trip times (RTT) [31]. In addition, for the real-time multimedia delivery, specifically for audio and video transmission, TCP is not

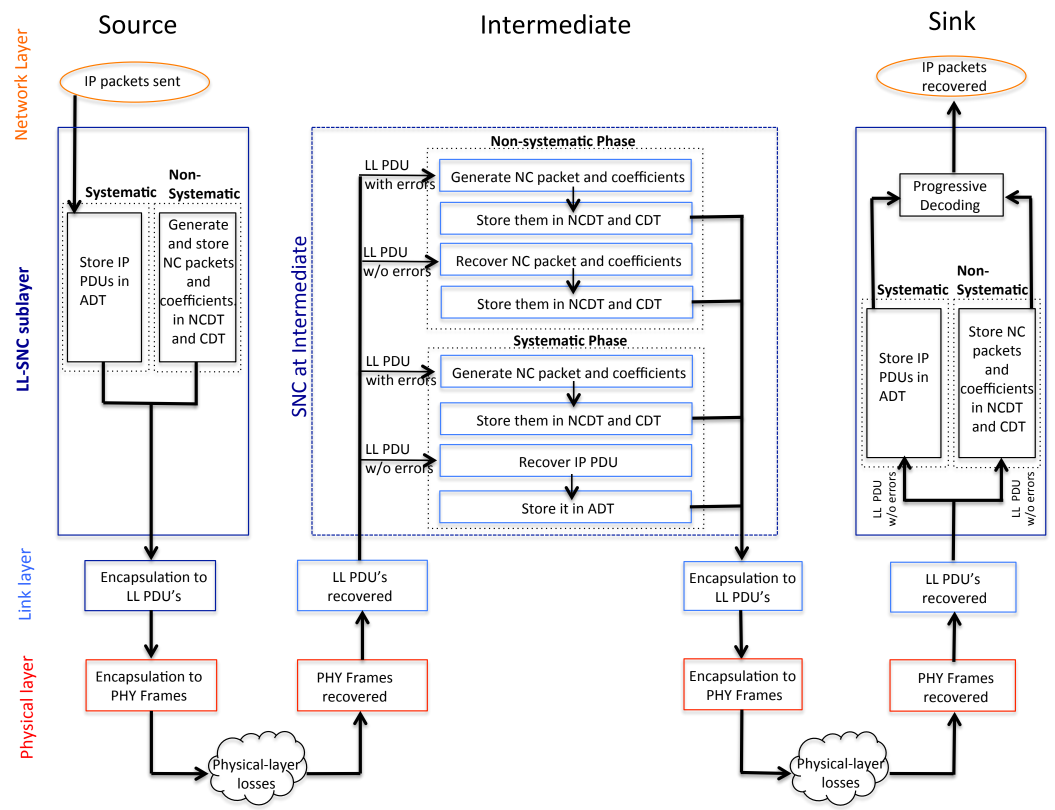


Figure. 2: Flow diagram with LL-SNC architecture and LL-SNC encapsulation in satellite system.

suitable for timely delivery as it requires heavy feedback and acknowledgements [32], [33], [34]. Another disadvantage of TCP is its compatibility with the mutlicast and broadcast communication. There may exist parallel TCP connections with the sender to different receivers, but each TCP connection can require transmission of different lost packets that will result into a resource-intensive solution.

Therefore, most of the real-time protocols for the multimedia delivery are built on the top of user datagram protocol (UDP). As the reliability is not guaranteed with UDP, forward erasure correction (FEC) codes are used to recover from packet losses. Specifically, Reed-Solomon (RS) code, which is a maximum distance separable (MDS) code, is optimal in terms of erasure correction performance and mainly used as a start-of-the-art FEC scheme. However, there are mainly three limitations of using RS codes. Firstly, a construction of the RS code is based on a finite algebraic arithmetic, therefore, the receiver has to wait for all the packets to start the decoding process. Secondly, the constraints on the RS coding parameters, make its incompatible with adaptive solutions and finally, when there are multiple hops where intermediate nodes can perform coding, they have to first decode the original packets and then re-encode to generate the coded packets. This results into an additional delay because of decoding and then re-encoding the original packets at every node. In this section, we will present an architectural and encapsulation framework to enable application layer systematic network coding (AL-SNC) at the source and at the intermediate node of the satellite systems.

B. AL-SNC architecture and encapsulation

The encapsulation process is similar to the encapsulation of RS codes over real-time transport protocol (RTP) [30]. In addition, there are some modifications to accomodate network coding parameters. In Figure 3, we present the modified encapsulation process to be used for the network coding. An application layer source block (AL-ADT) consists of K ADUs in K columns. The number of rows in the source block is $M = E + 2$, where E is the length of the largest ADU. The columns, which do not have the largest ADU, are filled with zeros to be completely filled. Each column can be considered as a data packet. The first two bytes of each column in the source block contain the length of the corresponding ADU. ADUs are then encapsulated into RTP packets. The first two bytes and the zero paddings are not sent over the network. The application layer FEC block (AL-NCDT) contains $N-K$ columns with $N-K$ coded packets and the application layer coefficient block (AL-CDT) contains $N-K$ columns with $N-K$ set of coding coefficients. FEC packets and coefficients are then encapsulated into RTP packets. Each RTP packet contains RTP payload, RTP header and FEC payload ID. This FEC payload ID is used for signaling the coding parameters like source block ID, FEC packet ID, values of K and N, etc. The CRC-32 is added with every RTP packet to detect errors in RTP packets at the receiving end. At the receiver, the values of coding parameters are extracted from the FEC payload ID. Now, if ADUs are lost then the complete columns are lost. So, if FEC decoding succeeds, the receiver recovers

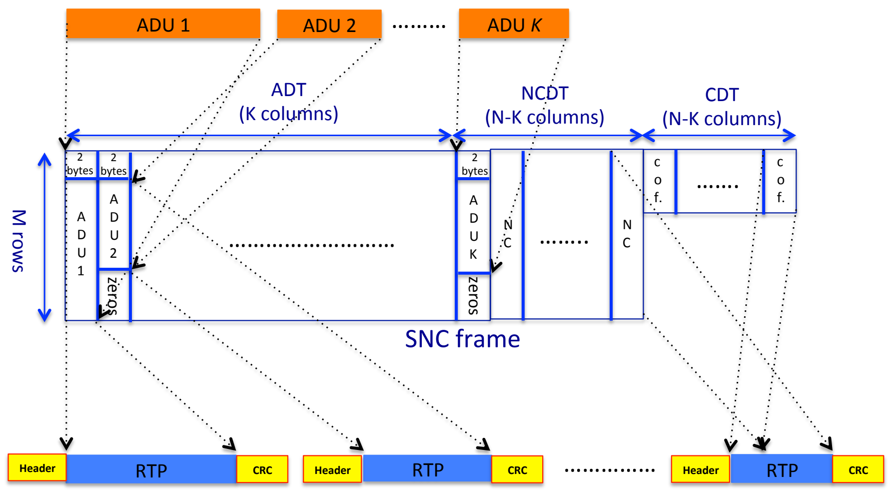


Figure. 3: Encapsulation of application layer data units over RTP with network coding.

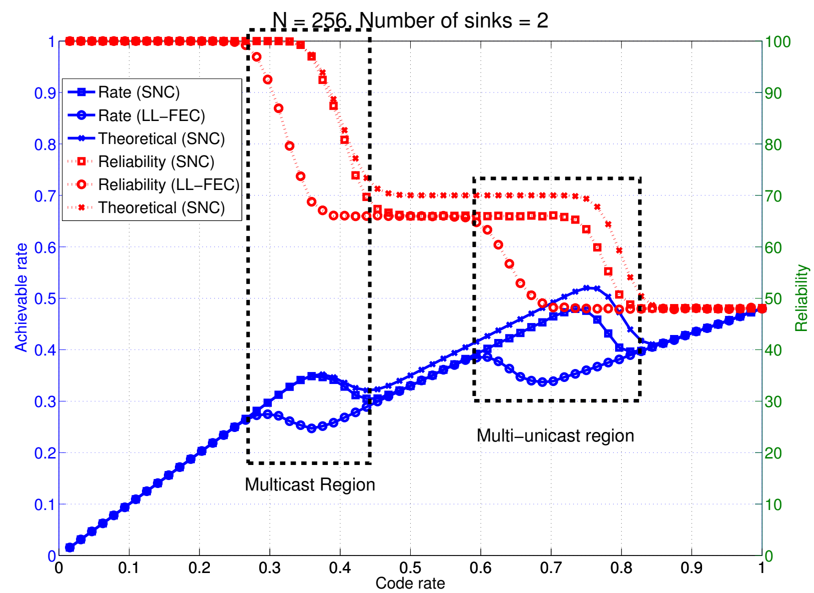


Figure. 4: LL-SNC multicast and multi-unicast region with two sinks.

ADUs by filling the erased columns. The initial two bytes are used to remove zero padding from the data packets to recover the ADUs.

VII. Simulation results

A. Performance metrics

1) Achievable rates and reliability: The theoretical expressions of the average achievable rate and the average reliability are derived in Section IV. In this section, we will present the simulation results on the average achievable rate and the average reliability. In the results, we also compare the simulation results with the theoretical expressions derived in Section IV.

2) Average delay per-packet: If a packet $\mathbf{s}_t$ is transmitted by the source at time t_j and it is recovered at the sink at time t_r then packet $\mathbf{s}_t$ incurs a delay δ_t where, $\delta_t = t_r - t_j$. For the block of K packets, the average delay per-packet is given as, $\triangle = \frac{\sum_{t=1}^{K} \delta_t}{K}$. Note that the delay is evaluated only for the packets that are recovered at the sink.

TABLE I: MAXIMUM ACHIEVABLE RATES FOR MUTI-CAST AND MULTI-UNICAST

N	Application	Sinks	LL-SNC	LL-FEC	Gain
256	Multicast	2	0.3486	0.2747	26.90%
256	Multi-unicast	2	0.4794	0.3858	24.26%
256	Multicast	10	0.3575	0.3043	17.48%
256	Multi-unicast	10	0.6745	0.5404	24.81%
50	Multicast	2	0.3035	0.2553	18.88%
50	Multi-unicast	2	0.3852	0.3579	7.63%
50	Multicast	10	0.3107	0.2751	12.94%
50	Multi-unicast	10	0.5406	0.4990	8.34%

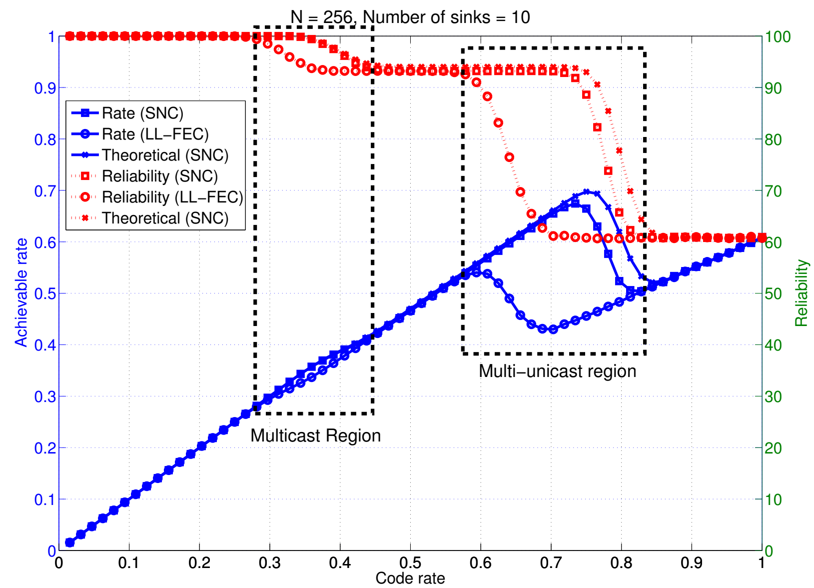


Figure. 5: LL-SNC multicast and multi-unicast region with two sinks.

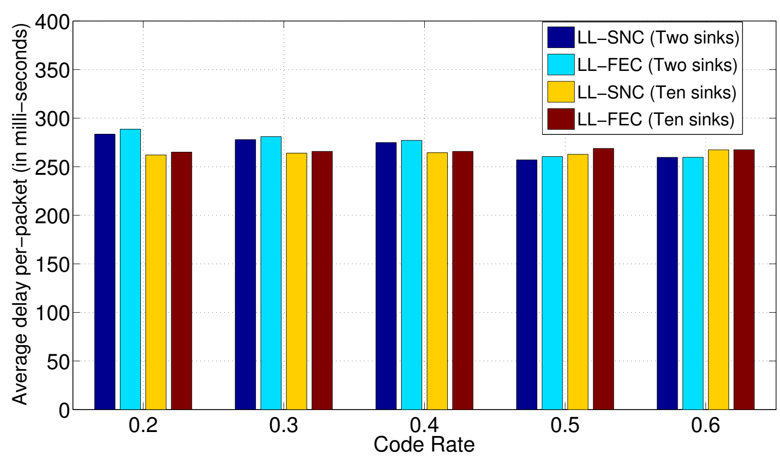


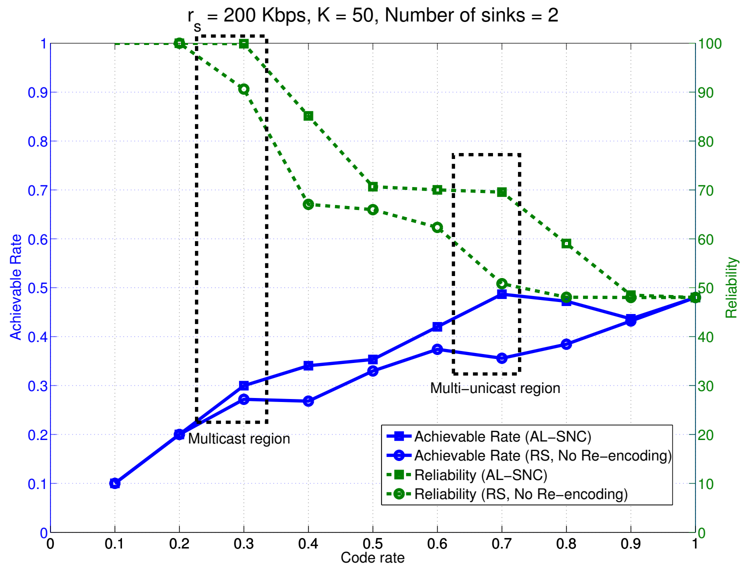
Figure. 6: Average delay per-packet

B. Results for the implementation at link layer

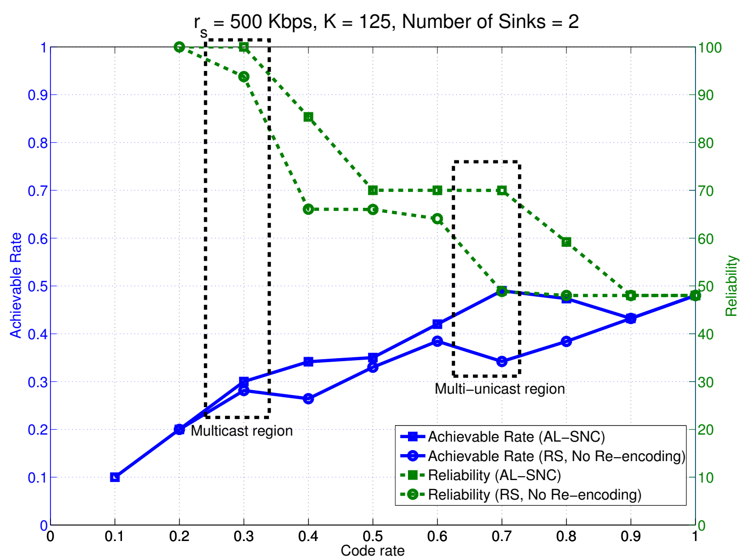
1) Simulation setup: In our simulation setup, first we consider network coding implementation at the link layer. We consider realistic satellite transmission scenarios with links having light rainfall (erasure rate of 0.2) and/or heavy rainfall (erasure rate of 0.6) [29]. These erasures correspond to the loss of LL-PDUs. In each case, we compare LL-SNC with LL-FEC. We assume IP PDUs of length 1500 bytes. Each IP PDU is mapped to a column of the ADTs of consecutive LL-SNC frames. Two LL-SNC frame lengths, $N \in \{50, 256\}$ and several values of code rates are considered for comparison. The size of ADT, i.e., K changes with the code rate. We set the physical layer symbol rate of $B_s = 27.5$ Mbaud/s, $\varsigma = 2$ as the modulation constellation and $r_{phy} = 1/2$ as the physical coding rate such that the bit rate is $B_s \varsigma r_{phy} = 27.5\ Mbps$. The transmission delay is set to be 250 ms. In each case, we average over 1000 experiments for every performance metric. The number of erasures per-frame varies (according to the random erasure rate) between 1000 experiments.

2) Results: In Figure 4, we show the results on achievable rates and reliability when there are two sinks in the network. The multicast capacity of the network is limited by one of the links joining the intermediate node to the sink node. We consider the following erasure probabilities: $\epsilon_1 = 0.2$, $\epsilon_2 = 0.2$ and $\epsilon_3 = 0.6$. In Figure 5, we show the results on achievable rates and reliability when there are ten sinks in the network. The multicast capacity of the network is limited by one of the links joining the intermediate node to the sink node. We consider the following erasure probabilities: $\epsilon_1 = 0.2$, $\epsilon_j = 0.2, j = 1, 2, ..., 9$ and $\epsilon_{10} = 0.6$. Furthermore, we present the maximum achievable rates for both multicast and multi-unicast in Table I. Following are the key conclusions from these results:

- We have identified two regions in these graphs: one for



(a) AL-SNC multicast and multi-unicast regions with video codec rate $r_s = 200$ Kbps.



(b) AL-SNC multicast and multi-unicast regions with video codec rate $r_s = 200$ Kbps.

Figure. 7: AL-SNC multicast and multi-unicast regions with different video codec rates.

multicast, and another for multi-unicast (represented by dashed boxes). The region for multicast is corresponding to the case when the average reliability approaches 100%. This would mean that all the sinks in the network are able to recover all the data packets. We have also identified the multi-unicast region where the sinks with better channel recover all the data packets and the sink(s) with bad channel still suffer from some losses. The benefit of multi-unicast over multicast is that one can achieve overall higher transmission rates by not sacrificing the rate due to the bottleneck sink (link with higher erasure rate). Hence, based on the requirements of the users, our results provide optimal usage of available bandwidth for transmission.

- The multicast is feasible only when the code rate is smaller than the capacity of the network, which is $\min_j(1-\epsilon_j) = 0.4$. However, when the code rate is higher than the multicast capacity, multi-unicast is feasible. This is because the capacity of the sinks with good channel is different and higher than the capacity of the network (in our example it is 0.8). Therefore, when the code rate is smaller than 0.8, the sinks with good channel can recover all the data packets making multi-unicast feasible.
- LL-SNC provides higher transmission rates and higher reliability than LL-FEC in all the cases. When the number of sink increases, LL-SNC can provide close to 100% reliability in the multi-unicast region itself. This is because there is only one bottleneck link in the network and only

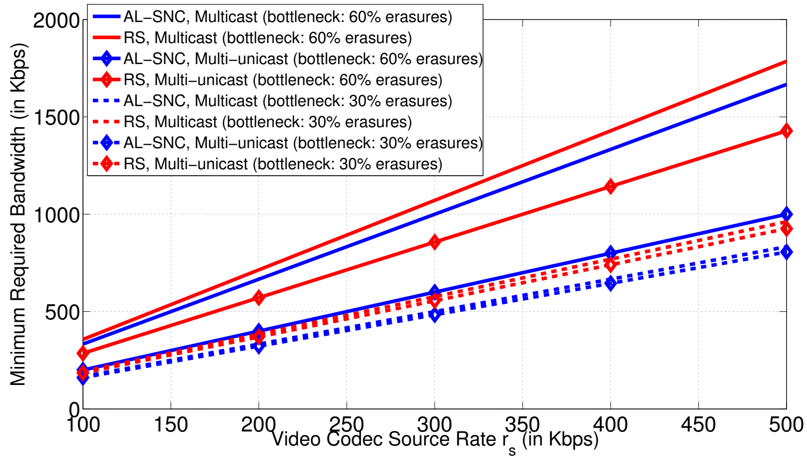


Figure. 8: Minimum network bandwidth requirement for multicast and multi-unicast with AL-SNC.

one sink suffers from the bad channel. In this case, it would be efficient to transmit in the multi-unicast region such that the higher transmission rates are achieved and almost all the sinks (except the one with the bad channel) are able to recover all the data packets. Furthermore, LL-SNC also provides higher maximum achievable rates than LL-FEC for both multicast and multi-unicast. The maximum achievable rate increases as the frame length increases or the number of sinks increases. Our results (Table I) show that LL-SNC can achieve up to 26.90% and 24.26% higher maximum achievable rates than LL-FEC for multicast and multi-unicast respectively.

In Figure 6, our simulation results also show that LL-SNC provides smaller average delay per-packet than LL-FEC. This is because of the following two reasons. First, the progressive decoding in LL-SNC allows the sinks to start decoding and recovering as soon as it receives the first packet. Second, the re-encoding in LL-SNC helps the sink to receive K degrees of freedom and complete the decoding process in fewer time slots than LL-FEC. The overall delay includes the inherent transmission delay of 250 ms of the satellite systems. Note that the average per-packet delay for LL-SNC and LL-FEC are very close. This is due to the fact that the transmission delay majorly contributes to the overall delay and per-packet delay is smaller due to the higher rates. However, at the lower rates, when per-packet delay is higher, LL-SNC is expected to provide higher advantage than LL-FEC.

C. Results for the implementation at application layer

1) Simulation setup: For the case of network coding implementation at application layer, we also consider different cases with erasure rates of 0.2 and 0.6. These erasures correspond to the loss of RTP packets. We assume that the source generates packets of length $M = 1500$ symbols over Galois Field with size $q = 2^8$. We consider state-of-the-art video codecs with video frames grouped into Groups of Pictures (GoPs). Each GoP contains several packets. The codec outputs each GoP in a fixed time T_{GoP} such that the source block is $K = \left\lfloor \frac{r_s \times T_{GoP}}{M \times 8} \right\rfloor$ where r_s is the video codec source rate. Several values of code rates are considered for comparison. The size of coded block, i.e., N varies with the change in code rate. In each case, we average over 1000 experiments for every performance metrics.

2) Results: Figure 7 shows the achievable rates for the application of network coding at application layer. Two configurations of codecs are considered with $r_s = 200$ kbps and $r_s = 500$ kbps with $T_{GOP} = 3$ seconds. The source blocks corresponding to these configurations are $K = 50$ and $K = 125$, respectively. We show the multicast and multi-unicast region of transmission in these figures.

The main difference between the simulation of link and application layer network coding implementation is the variation of block sizes and frame sizes. In case of link layer implementation, the frame size (N) is constant (K changes with the code rate). In case of application layer implementation, the block size (K) is constant (N changes with the code rate). Here, the block size depends on the codec rate, GoP time and packet sizes. In this paper, we have presented results with K changing with codec rates. However, the conclusions remain the same with the variations in packet sizes and GoP time.

Finally, we also show the minimum network bandwidth required for enabling multicasting and multi-unicasting at application layer. The minimum network bandwidth required is calculated as the ratio of video codec source rate and achievable rate. We consider several video codec source rates ranging from 100 Kbps to 500 Kbps. We consider two erasure rates for the bottleneck (worst) link: 30% erasures and 60% erasures. It is shown that to achieve 500 Kbps video codec rate, it is required to have at least 1.8 Mbps network bandwidth for multicasting (100% average reliability). Note that 500 Kbps is used for the video code source rate, and the rest of the available network bandwidth is used for the network coding rate to couteract packet losses and to guarantee 100% reliability. In addition, with the help of SNC, the required network bandwidth can be reduced from 1.786 Mbps to 1.667 Mbps in case of multicasting. This is equivalent to save around 6.66% in network bandwidth when SNC is used instead of RS coding. In case of multi-unicasting, the gain is even higher where SNC helps to reduce the network bandwidth requirement from 1.429

Mbps to 1 Mbps (around 30.02% save in bandwidth).

VIII. Conclusions

In this paper, we have focussed on the use of SNC for multicast and multi-unicast over satellite. We have identified the transmission regions for multicast and multi-unicast over satellite by characterizing the reliability and achievable rates offered by SNC in these two different regions. We have derived the theoretical expressions for the average reliability and the average achievable rate of the considered topology. Our theoretical and simulation analysis present the benefits of SNC over end-to-end coding for both multicast and multi-unicast. Our results have shown that a higher rate is achievable for the multi-unicast however, not all the users in multi-unicast can recover all the data packets. Therefore, based on the requirements from different users, the transmission region can be chosen for the optimal usage of available bandwidth. Finally, we have explored the benefits of network coding at different layers of the satellite network protocol stack. We have shown the encapsulation and architecture feasibility of network coding application in the satellite-dependent layers and satellite-independent layers. Future work includes the investigation of SNC on more complex networks such as network with multiple sources. Several other factors like bursty erasures, video codec characteristics, processing complexity, etc should be taken into account while analyzing the performance of network coding schemes for multicasting.

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