A Method for Detection/Deletion via Network Coding for Unequal Error Protection of Scalable Video over Error-Prone Networks

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Abstract. The development of universal systems for video streaming needs transmission strategies that exploit the characteristics of the transmission medium such as a wireless network. Scalable video coding allows partial decoding of the video for multiple demands or under severe reception conditions. Network coding increases the transmission rate and provides error control at network level. We propose a detection/deletion system for error reduction in presence of channel noise. We combine the error detection capabilities of the network code with erasure decoding and unequal error protection to improve the visual quality of the video.

Keywords: scalable video coding, network coding, network error correction, multicast, error detection, detection/deletion, erasure decoding.

1 Introduction

Due to the diffusion of connectivity and multimedia services, multimedia communication in the future will try to reach several kind of communication platforms. In universal video systems, a wide landscape of users are connected via different physical means to a service provider. On the one hand, a backbone infrastructure connects heterogeneous platforms to the streaming server. On the other hand, users display requirements, reception conditions and Quality of Service (QoS) demands can be deeply different between the platforms. Systems like Digital Video Broadcasting (DVB) differentiate the physical interface between terrestrial (DVB-T), satellite (DVB-S), and Handled (DVB-H) connected by a common backbone network.

Flexibility is required both from the backbone network and the coded data carried along. Scalable Video Coding (SVC) is a video coding paradigm that embraces Layered Coding (LC) applied to video, exploiting the variety of the transmission channels and users requirements [1]. SVC allows partial decoding of the video stream by extracting parts of the stream if needed. Decoding at reduced resolution, frame rate or quality from the same source bitstream is possible in case of different display requirements or impossibility to decode part of the stream due to channel errors.

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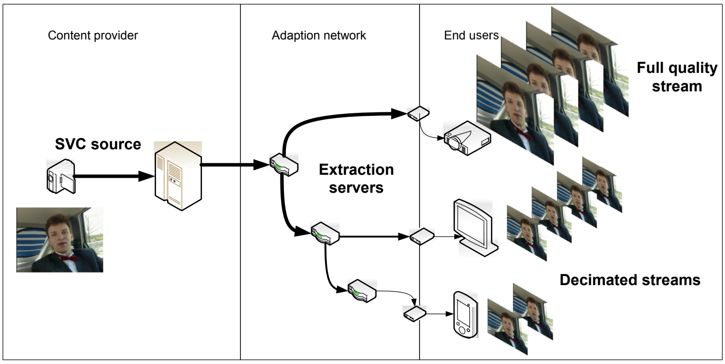

Fig. 1. An SVC application scenario with stream adaption for differentiated services

Multirate content delivery is possible in a structured scenario like the one in Fig. 1. Mobile devices with low computational power and small displays need a compact stream with reduced quality and resolution, whereas home users usually make use of big displays and a large bandwidth and big screens and demand a superior quality.

When it comes to transmission with channel noise, like in wireless links, SVC has an intrinsic mechanism to combat information loss and errors. Noise has different and independent impact on the layers. Errors occurring on the higher layers have less impact in the visual quality than those in the base layers. They are independently encoded and decoded, thus separating the impact of the error. Unequal Error Protection (UEP) increases the visual quality with respect to equal error protection and to the conventional non-scalable codecs [2,3]. Loss (e.g., in packet networks) or corruption of the information in the highest layers do not nullify the decoding of lower layers, thus offering a continuous service. Unequal Loss Protection (ULP) and Multiple Description Coding (MDC) techniques have been proposed to boost the performance in lossy packet networks [4,5].

In this paper we use an MDC-like coding technique based on ULP-FEC codes to protect against channel errors. We use an erasure decoding method with Detection/Deletion (D/D) via Network Coding. Network Coding (NC) was introduced by Ahlswede *et al.* in 2000 [6]. This paradigm allows intermediate nodes to decode the incoming packets and re-transmit to the other nodes a function of the received information. NC overpasses the traditional forwarding at intermediate nodes, which can be regarded as a special case of network coding. Conventional networking can not cope with bottlenecks, thus it is not always possible to serve all receivers at the same time in a multicast setting as in Fig. 2 (a). With network coding the information can be combined in the bottlenecks and communicated to all receivers, which under determinate conditions are able to decode the source information (Fig. 2 (b)). Network coding has found potential application possibilities in Content Delivery Networks (CDN) and Peer-to-Peer (P2P) networks [7]. NC increases the network throughput [6] and allows error control against link failures at network level thanks to Network Error Correction [8,20]. Further

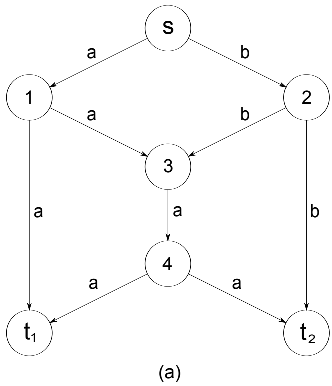

Fig. 2. The two-sources two-sinks butterfly network without (a) and with (b) network coding

benefits are found in using network coding in wireless networks, in which physical superposition of signals can be used opportunistically [9,10].

We consider the multicast transmission of scalable video on error prone networks and we aim at decreasing the impact of channel errors on the visual quality. ULP techniques have been reproposed in various works to produce multiple descriptions of data [11,5]. Multiple Description Coding (MDC) produces a number of dependent representations of the source information and allows partial decoding with increasing quality as more chunks of coded data are recovered. We use a similar technique to combat errors instead of packet losses, by modifying the construction of FEC-based MDC. We produce data blocks coded with erasure codes with differentiated persistence via UEP. The streams are transmitted in independent paths through a network performing network coding and error detection is performed at the receivers. Errors are detected via Network Error Correction and data blocks with errors are deleted. Erasure correction on the temporal direction is performed on the remaining data blocks of the streams. The spatial dimension of error detection is separated from the temporal dimension of erasure correction. Orthogonal coding redundancy achieves superior error protection performance of the scalable video with respect to the cases of separated FEC coding and network error correction only.

This system benefits from increased throughput in multicast scenario thanks to the network coding. Although MDC-derived, our detection/deletion scheme is designed for robustness against channel errors, rather than packet losses, for the wireless radio channel. Packet losses can be treated with statistical decoding of NEC-coded packets.

The organization of the paper is as follows: Section 2 presents the scalable video coding paradigm and the traditional FEC-based MDC scheme. In Section 3 we discuss the implementation of the transmission system with network coding and we introduce the detection/deletion method of MDC derivation. In Section 4 the performance of the transmission system is evaluated. Section 5 concludes the paper.

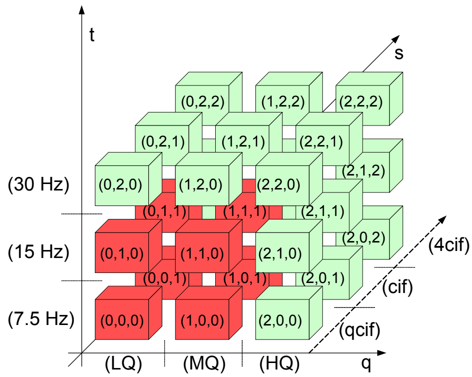

Fig. 3. Representation of atoms of an SVC stream with 3 levels of temporal, spatial and quality scalability [12]

2 Scalable Video Coding

Classic video codecs are designed to work under definite display requirements and channel rate. The variable nature of the wireless channel and the variety of transmission possibilities in large networks requires flexibility in terms of source coding. An Scalable Video Coding (SVC) approach supports multiple decoding configurations from the same embedded bitstream for different display and channel requirements, avoiding overloading the network and decreasing the coding computational load [1]. Full scalability allows reducing spatial resolution, frame rate and quality (SNR) if needed. The H.264/SVC extension of the video compression standard will support full scalability with a DCT block-based approach. Other alternative solutions have been proposed, like the W-SVC codec which has also full scalability but it is based on wavelet transforms [12].

The SVC approach to video coding produces an embedded bitstream of elementary units (atoms) containing the coded information of an enhancement level of time/space (T/S) resolution or quality for every Group of Pictures (GOP), as shown in Fig. 3. Atoms are organized in hierarchy where each layer always depends on the lower ones for decoding. Partial decoding can be performed by bitstream parsing according to a desired display configuration, by extracting and assembling in a reduced bitstream the required atoms to decode at a desired display configuration. SVC is designed to cope with differentiated demands as well as unpredictable channel conditions. When particularly adverse channel conditions nullify the decoding of parts of the stream, partial decoding at reduced time/space resolution or quality provides a continuous service.

Boosting the decoding performance is possible with proper channel coding of SVC. The hierarchy of the coding layers implies that lower layers have a higher impact on video quality. Also, higher layers are useless if the underneath layers are not decoded successfully. Unequal Error Protection (UEP) differentiates the redundant bits allocated to protect different layers under particular rate

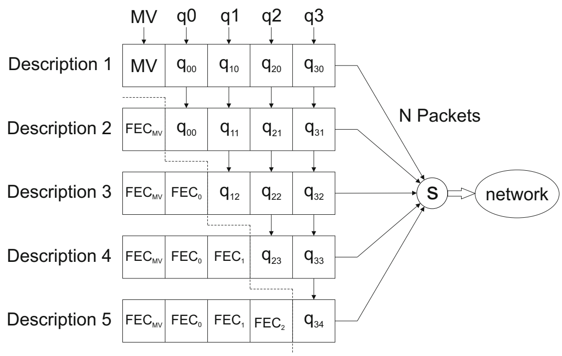

Fig. 4. Scheme for Multiple Description Coding via Forward Error Correction and for unequal packet-loss protection

constraints. Traditional channel coding techniques are applicable to differentiate FEC coding among layers, achieving superior quality and reducing the percentage of undecoded GOPs than equal protection or standard non-scalable video coding approaches [2,3].

2.1 FEC-Based Multiple Description Coding for Scalable Data

We now describe a FEC-based Multiple Description Coding (MDC) technique, whose structure is adopted in our Deletion/Detection method to protect against channel errors. MDC has been proposed to allow partial decoding of source data under random loss of information, e.g. in packet networks [13]. MDC can exploit the independence of the coding layers of SVC and yield to opportunistic recovery of information under severe channel conditions. The traditional method for generating multiple descriptions of a scalable stream is resumed in Fig. 4 [14]. In case of quality scalability, the quality layers $Q_i, i = 1, 2...$, are grouped in non overlapping sets and organized in independent streams, as:

$$\begin{aligned} Stream_0 &= \{Q_0, Q_1, \ldots, Q_{n_1}\}, \\ Stream_1 &= \{Q_{n_1+1}, \ldots, Q_{n_2}\}, \\ &\ldots \\ Stream_N &= \{Q_{n_{i-1}+1}, \ldots, Q\}, \end{aligned}$$

and with source rates (without error-control coding):

$$R_{stream_i} = \sum_{i=n_{i-1}+1}^{n_i} R_i. \qquad (1)$$

where R_i is the rate of the i-th video layer.

Coded blocks of equal length are obtained by applying erasure codes with variable rate to each stream. Portions of k_i symbols are taken from each i-th layer and organized into FEC-coded packets of $n = k_i r_{FEC}^{(i)}$ symbols, in an

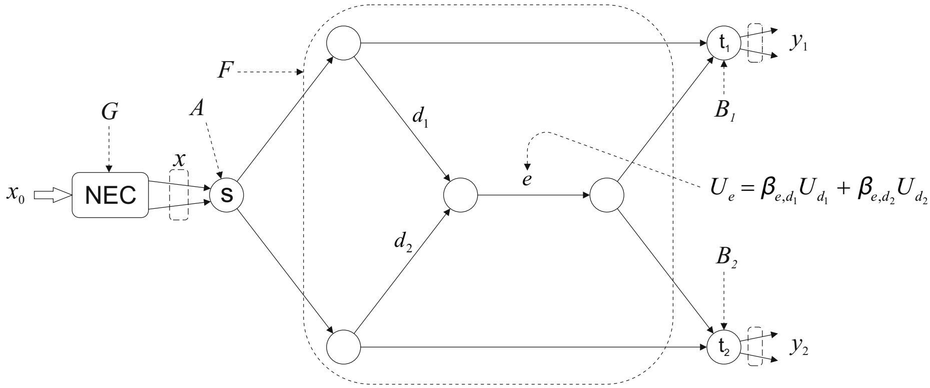

Fig. 5. Algebraic modeling of network coding with local linear encoding

interleaving-like manner, as show in Fig. 4. Smaller messages from the base layers are encoded with a stronger erasure code, with rate $r^{(0)}_{FEC} < r^{(1)}_{FEC} < \ldots < r^{(N)}_{FEC}$, so that

$$R_{stream_0}/r^{(0)}_{FEC} = R_{stream_1}/r^{(1)}_{FEC} = \ldots = R_{stream_N}/r^{(N)}_{FEC}. \quad (2)$$

The loss of l packets, with $n - k_i \geq l > n - k_j, i < j$, allows erasure decoding of the streams up to the i-th, from the $n - l$ remaining packets. The recovered packets being not enough to decode the other layers, partial decoding of the video is possible, with increasing quality as more packets are received.

This scheme is useful in transmission affected by packet losses. We use this criterion to perform opportunistic deletion of coded symbols across the packets upon detection of errors via network coding. Our scheme protects against channel errors occurring in an error-prone network, as explained in Sec. 3.

3 Network Coding for Error Detection

Network coding transmission has the property of modifying the information like a coding operator. The motivation for employing network coding is the possibility of reaching the network capacity in multicast settings and the possibility of performing error correction/detection on the spatial dimension at network level. We consider the single-source multicast transmission on an acyclic network, i.e., without closed cycles, where linear coding and decoding operations in a vector space over a finite field are implemented at the nodes. [15].

Consider a network model as a directed graph $\mathcal{G} = (V, E)$, where V is a set of vertices and E is a set of edges. A vertex is designated as the source node s and the receivers are grouped in the nodes set T. Every edge has unitary capacity. Links with capacity multiple of the unit are treated as multiple parallel edges. Given a non-source node i with input edges $d \in In(i)$ and output edges $e \in Out(i)$, then the message U_e transmitted on an edge e can be expressed as:

$$U_e = \sum_{d \in In(e)} \beta_{d,e} U_d. \tag{3}$$

where $\beta_{d,e}$ constitute the *local coding kernels* of the edges $e, \forall e \in E$. Blocks of m bits are considered as symbols in a finite field $\mathbb{F}$ of size q, e.g., a Galois Field of size $q = 2^m$.

Generation-based transmission is considered, where each nodes stores in a buffer all the packets belonging to the same generation before transmitting their linear combination. Unitary coding vectors are put before the payload in the packets so that the receivers can deduce the network code [16]. This corresponds to a transmission delay equal to the delay of the longest path.

For modeling the network transmission, an algebraic model can be employed, where the network transform characteristic can be modeled by a transfer matrix M_t from the source to the receiver t [17]. The source data is arranged in vectors $\mathbf{x} = [x_1, x_2, \ldots, x_h]$. If the network supports a rate of h symbols through h edge-disjoint paths, the sink t receives a vector $\mathbf{y}_t = [y_1, y_2, \ldots, y_h]$ obtained by the network transformation as:

$$\mathbf{y}_t = \mathbf{x} M_t + \mathbf{z} F_t \tag{4}$$

which also models random errors and erasures by means of a $1 \times |E|$ additive error vector $\mathbf{z}$.

3.1 Error Control via Network Error Correction

Consider a network with a max-flow h to each destination. With Network Error Correction (NEC), the set of source messages $\mathbf{x}$ belongs to an ω-dimensional vector space $\mathcal{C} \subseteq \mathbb{F}^h$. The codebook $\mathcal{C}$ generated by means of traditional block codes with $\omega \times h$ generation matrix G, is a Minimum Distance Separable (MDS) code with minimum distance $d_{min,s} = h - \omega + 1$. Every network codeword is generated by means of

$$\mathbf{x} = \mathbf{x}_0 G, \tag{5}$$

where $\mathbf{x}_0 = [x_{0,1}, x_{0,2}, \ldots, x_{0,\omega}]$. The coding space at the receivers $t \in T$ can have by construction minimum distance $d_{t,min} = h - \omega + 1$ and the redundant linear combinations can be used at the receiver for error control purposes [8]. The receivers are able to correct up to $\lfloor \frac{d_{min,t}-1}{2} \rfloor$ and detect up to $d_{min,t} - 1$ errors.

A distributed approach for the construction of the network code can be assumed. A randomized choice of the coding kernels achieves an MDS code with correct minimum distance with increasing probability as the field size grows. For codebooks with low redundancy ($d_{min} = 0, \ldots, 2$), a reasonable choice of field size is $q = 2^8, 2^{12}, 2^{16}$, which achieve success probability above 90% [18]. Packetized transmission is necessary to communicate the chosen kernels to the receiver. This approach is more suitable in wireless ad-hoc networks, where the nodes occasionally join a mesh network and take part to the transmission.

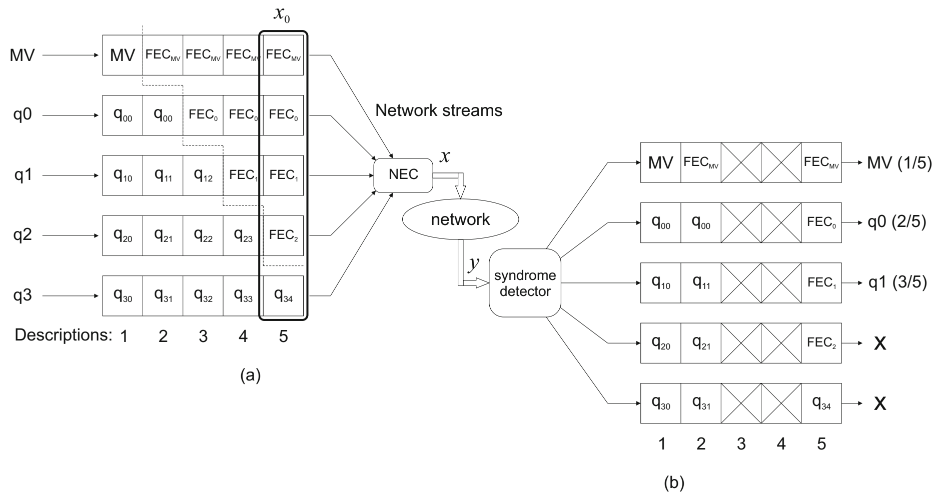

Fig. 6. (a) Scheme for Multiple Description Coding via Forward Error Correction and modified transmission criterion for detection via Network Error Correction, deletion and erasure decoding. (b) An example of partial decoding of MDC in case of erroneous decoding of network codewords 3 and 4.

3.2 The Detection/Deletion Scheme

The detection/deletion (D/D) system for robustness against random errors has packetization scheme transversal to the MDC scheme.

In a network coding transmission system, the loss of a packet in one link is not perceived from the receiver as with traditional networking. A packet loss is perceived as an alteration of the coding space, by $\mathbf{z}Ft$ as in Eq. (4), rather than an actual loss of information. This impedes identification of losses and thus traditional erasure correction, because losses are analogous to channel errors.

The D/D system detects error by means of the NEC code and applies erasure decoding on the remaining symbols. The system described in the following is not formally an MDC system, but achieves superior resistance to random channel errors which impair wireless transmissions.

Coding. The coding scheme is shown in Fig. 6 (a). As for traditional MDC, ω streams are build from the scalable data. On the temporal dimension every stream is independently erasure-encoded. FEC coding rates are such that the resulting datablocks have length equal to the network packet lenght, as in Eq. (2). The source streams are arranged in vectors $\mathbf{x}_0 = [x_{0,1}, x_{0,2}, \dots, x_{0,\omega}]$, where every $\mathbb{F}$ symbol from the i-th stream stays at the corresponding position $x_{0,i}$ of $\mathbf{x}_0$. Then the streams are NEC coded with the chosen generation matrix and sent in parallel on the network paths (Fig. 6).

Detection/Deletion and Decoding. All symbols in a packet share the same coding at the intermediate network nodes. On the other hand random errors affect network codewords $\mathbf{y}_t$ independently. Traditional decoding of network coding is based on complete decoding, i.e., $\lfloor \frac{d_{min,t}-1}{2} \rfloor$ errors are corrected at network level.

At the receiver the network codewords are generated by means of a system of linear equations with core M_t. Since the number of equations h is more than the number of variables ω, if the system is undetermined the codewords are detected as not belonging to the target codebook at the receiver. A syndrome detector can be used for this purpose [19].

The received codeword is detected as erroneous if such word has a minimum distance from the rest of the receiver's codewords higher than the number of errors occurred during the transmission slot. All the symbols at the corresponding position in the ω streams are flagged as erroneous and discarded (Fig. 6). Of all the symbols of a packet, those that are not recognized as erroneous can be used for erasure decoding. If l network codewords belonging to the same data block are flagged, with $n - k_i \geq l > n - k_j, i < j$, the blocks of the streams up to the i-th are correctly decoded from the remaining symbols.

This technique decouples the errors happening on the temporal dimension with a detector on the spatial dimension that detects up to $d_{min,t} - 1$ errors on the same time slot. E.g., when using a $(h-1, h, 1)$ code $(d_{min,t} = 1)$ error patters of weight equal to 1 (i.e., $|\mathbf{z}| \leq 1$) are always detected. Patterns with higher number of errors are partially detectable if the base field is large enough (Fig. 7). In a multirate setting, the sinks can receive differentiated flows, with differentiated detection capabilities.

Code construction in decentralized setting is performed in a randomized manner. The intermediate nodes retransmit random linear combinations of the packets that they have in the buffer. The coding distance of the codewords at the receiver $d_{min} = h - \omega$ is random. The probability of having a minimum coding distance for all codewords is higher for small distances and decreases for values close to the coding limit [18]. Small coding distances are required for the task of only detecting errors, i.e., a distance $d_{min,t} = 2, t \in T$ is required for detection, which is more probable than a distance $d_{min,t} = 3, t \in T$ for complete NEC decoding. In a randomized network code the distance between the codewords can be irregular. Codewords are more probable to have small distance (usable for detection purposes) among them. Detection performance of a random code is shown in Fig. 7 and commented in the next section. In order to cope with packet losses, a traditional method for statistical decoding can be used [20]. Even with small coding distances, e.g. $d_{min,t} = 2$, due to the fact that all the symbols in a packet share the same network code, a possible erasure pattern and error vector can be found for all the slots in a packet with high probability, and the packet loss corrected.

We perform network detection at receivers by means of a syndrome-based detector [19]. Consistently with classic syndrome decoding, a parity check matrix H_t is built from the codebook generator matrix G and the system matrix M_t. If the systematic form of the generator matrix of the code at the receiver t is

$$GM_t = [I_\omega, P], \tag{6}$$

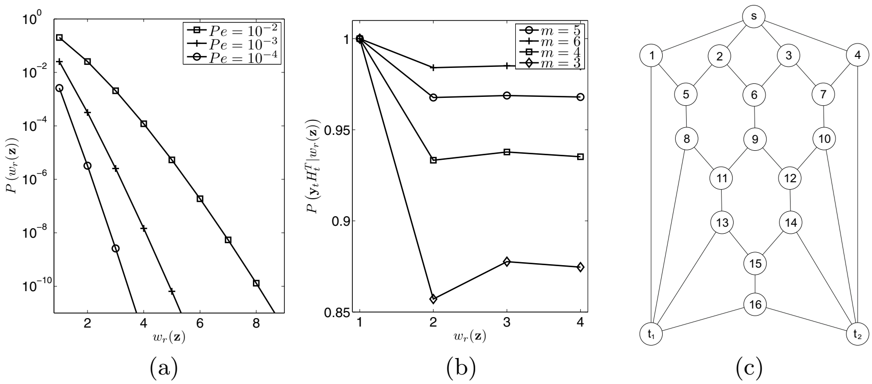

Fig. 7. (a) Probability of successful detection of errors with weight $w_r(\mathbf{z})$ (field size $q = 2^m$). Network code is $(\omega, h, d_{min,t}) = (3, 4, 2), tinT$ and the network is (c). (b) Probability of occurrence of error patterns with weight $w_r(\mathbf{z})$. (c) Network used for graphics in (a) and (b).

then the parity check matrix for the source code projected at receiver t is

$$H_t = \left[P^T, I_{h-\omega}\right]. \tag{7}$$

For all network codewords transmitted without errors:

$$\mathbf{x}M_t \cdot H_t^T = \mathbf{0}, \quad \mathbf{x} \in \mathcal{C} \tag{8}$$

Codewords with syndrome not equal to $\mathbf{0}$ are flagged as erroneous.

In the next section we present the simulation results of the detection/deletion system with scalable video transmission.

4 Simulation Results

When the network allows, the network code can be constructed deterministically to achieve the desired minimum distance at each receiver $d_{min,t} = h - \omega + 1, t \in T$. This is in general not easy to implement in wireless networks unless a centralized authority controls and manages the connections. Randomized code construction can be performed in decentralized manner. The packets are mixed at the intermediate nodes with random coefficients which are attached to the packet header.

We consider the detection capabilities of the network code. A network code with minimum distance $d_{min,t}$, is designed to detect all errors with pattern weight:

$$w_r(\mathbf{z}) = \text{rank}(\rho_{\mathbf{z}}) = d_{min,t} - 1, \tag{9}$$

where the error pattern $\rho_{\mathbf{z}}$ is a $1 \times E$ vector with unitary components in the positions corresponding to the non-zero components of $\mathbf{z}$. Its rank is the number

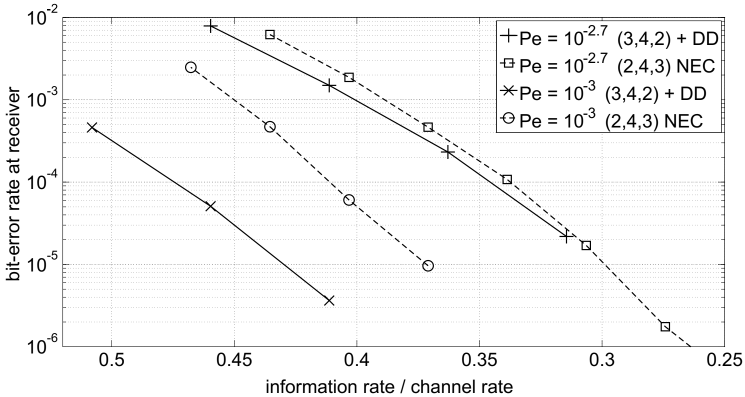

Fig. 8. Comparison of bit-error rate at the receivers at different FEC coding rates with traditional network error correction and with the detection/deletion method (D/D) under bit-error probability at the intermediate links is 10^{-3} and $10^{-2.7}$. The field size is $q = 2^6$ and block size for symbol-based FEC code $N = 186$ bits.

of non-zero components. Although a network code with distance $d_{min,t}$ allows detection of up to $d_{min,t} - 1$ link errors per time slot of a symbol at receiver t, the following considerations motivate the use of a code with source redundancy $\delta = h - \omega = 1$. Such codewords can achieve a distance $d = 2, t \in T$ with higher probability than a code with higher redundancy [18]. Especially for small field sizes, whose choice benefits the computational loads at the intermediate nodes and at the decoder, small coding distances among the codewords (even if the minimum distance is smaller) are more probable than higher distances, e.g., $d_{min,t} = 3, t \in T$ for detecting errors with $w_r = 2$ or correcting $w_r = 1$ [18]. Error patterns with $w_r > 1$ are often dominated by patterns with $w_r = 1$. Such capability of detecting error patterns beyond the limit is shown in Fig. 7 (a) for the network in Fig. 7 (c). The probability of detecting error patterns with weights beyond > 1 with a code with $d_{min,t} = 1$ is over 95% when using a code in a Galois Field with size $q = 2^m$ with m higher than $m = 6$. Additionally, error patterns with increasing weight appear with decreasing probability as shown by Fig. 7 (b). Such results suggest that the missed detection of errors happens for a small percentage of patterns and thus with small probability. In order to cope with packet losses, a traditional method for statistical decoding can be used [20].

The erroneous codewords detected via network coding are flagged as erroneous and erasure decoding is performed at the receiver based on the symbols received correctly (Fig.6). Symbol-based erasure codes are used, such as Reed-Solomon codes, with generator polynomial in a field of dimension 2^m. We test the transmission on the network in Fig. 7(c), with random coding at intermediate nodes for every generation of packets, i.e., every h packets generated at the source at the same time are synchronously coded throughout the network. Detection is performed by means of a $(\omega, h, d_{min,t}) = (3, 4, 2)$ network code $t = 1, 2$. We compare the detection/deletion (D/D) transmission technique with a

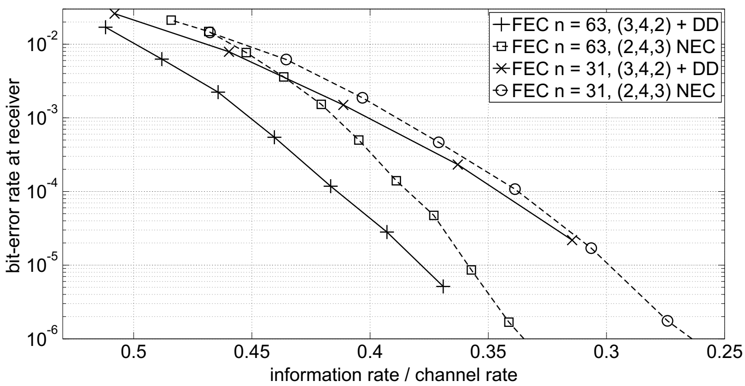

Fig. 9. Comparison of bit-error rate at the receivers at different FEC coding rates with traditional network error correction and with the D/D method and block sizes for symbol-based FEC code $N_1 = 31 \times m$, $N_2 = 63 \times m$. Bit-error probability at the intermediate links is $Pe = 10^{-2.7}$, field size is $q = 2^6$.

transmission with a network error correction code $(\omega, h, d_{min,t}) = (2, 4, 3)$. This code can correct all single errors at network level. Since the information rate is 2/3 of the other case, error-control codes with lower protection are used. Differentiated local FEC codes are applied at the source, but globally the allocation of the error protection is less flexible because the NEC code protects equally the video layers.

Fig.8 shows the error protection performance of the D/D method and the normal NEC transmission under two different link error rates. For low error rates the performance of the D/D method is always higher than the traditional NEC for all chosen coding rates. The ratio between information rate and channel rate is calculated considering the rate of the erasure codes and the NEC code rate.

The choice of large block length for the erasure code can sensibly reduce the impact of link errors on the bitstream decoded by the receiver. This is shown in Fig. 9. The error protection performance of the D/D method is higher than NEC with the tested block lengths. As the block length increases the NEC system performs similarly. The use of small block sizes can be motivated by the fact that an erroneous decoding of a whole datablock propagates the errors to a large part of the bit stream and may lead to undecodability of a GOP.

Fig. 10 shows the error protection performance of the D/D method and the normal NEC transmission with different field size. The distance between the codewords has a higher probability of being respected with a larger field size. The increase in the field size increases more the performance of the traditional NEC rather than the detection deletion method. On the other hand the datablock size is reduced in order to accommodate the same length in terms of number of bits, thus reducing the correction capability of the erasure code and yielding, for the tested setting, to a higher error rate at the receiver.

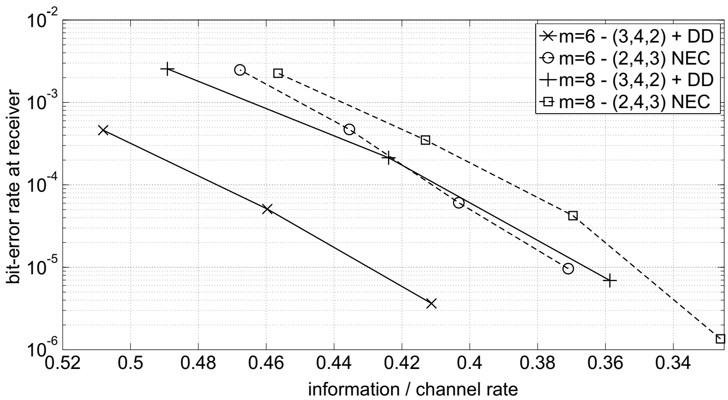

Fig. 10. Comparison of bit-error rate at the receivers at different FEC coding rates with traditional network error correction and with the D/D method and field sizes $q_1 = 2^6$ and $q_2 = 2^8$. Bit-error probability at the intermediate links is $Pe = 10^{-3}$ and block size for symbol-based FEC code $N = 378$ bits.

We test the transmission of video coded with the W-SVC codec with 3 levels of quality scalability, at the bitrates of 288 kbps for the base layer and 480 kbps and 800 kbps for the refinement layers. Following the D/D scheme in Fig. 6 we apply unequal error correction (UEP) of the video layers by means Reed-Solomon erasure codes. We compare the video quality by means of Peak Signal-to-Noise Ratio (PSNR) of the decoded video, shown in Fig. 11 under variable error rate at the intermediate links. The error-rates of the same streams are compared in Fig. 12. The error correction performance and the video quality are higher with the D/D system in the shown configuration. The error rate with the D/D method is equal to zero up to a certain threshold. This characteristic allows to perform perfect decoding of the video up to this value. Even if NEC allows a gentle increase of the error, even few errors often nullify the decoding of a GOP, thus it's preferable to have zero errors for a wider range of link-error rates. The nature of the scalable video, coupled with the UEP, allows to decide to drop a video layer when this becomes uncorrectable due to the undecodable errors. This happens for an error rate at the receiver of around 10^{-4}, which is reached earlier by the NEC method. This in general interests first the enhancement layers and then the lower ones as the link-error rate increase. The NEC system overcomes the D/D system for higher error rates. This error rates are though not acceptable for decoding the video. The D/D method performs better at low or controllable error rates, which makes necessary some kind of link-based error control system. It also performs better with small field sizes which also yield to less computational load for the intermediate nodes. Much less complexity weights on the receiver, due to the simple detection method (rather than the statistical solution of overranked linear systems) and the need of a smaller field size.

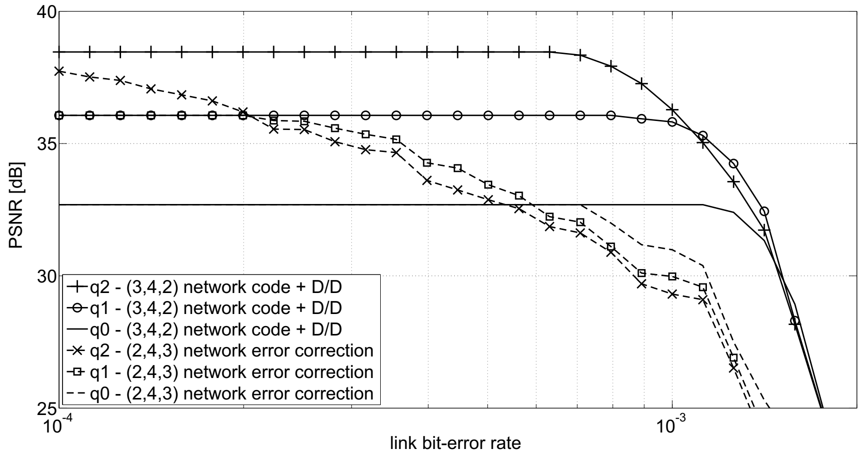

Fig. 11. Average Peak Signal-to-Noise ratio for *city* sequence at CIF resolution on the network in Fig. 7(c). Detection/Deletion by means of a network code $(\omega, h, d_{min,t}) = (3, 4, 2)$ is compared with a traditional $(2, 4, 3)$ NEC code.

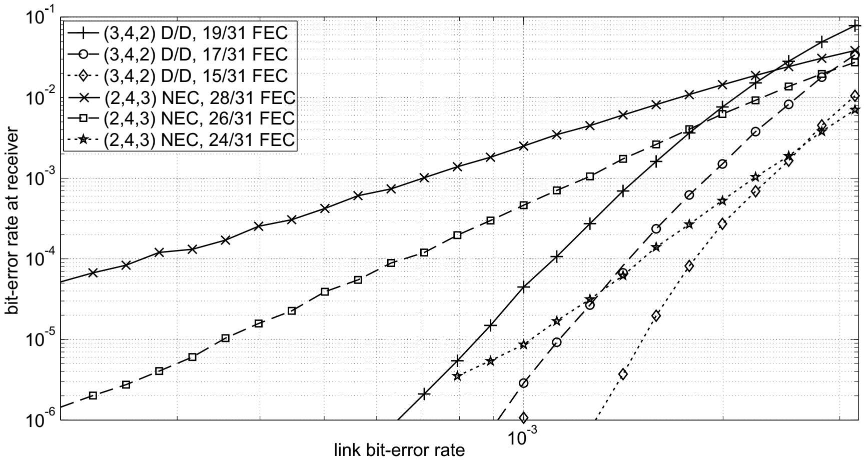

Fig. 12. Bit-error rates for the three video streams of the example in Fig. 11

5 Conclusions

We presented a system for error detection and deletion via Network Error Correction codes and erasure correction for scalable data. This technique allows for better use of transmission rate for error control purposes. Detection and deletion of errors allows zero-error decoding of the video up to higher levels of noise than traditional network error correction coding and normal forward error correction codes, exploiting the scalable nature of the video for higher error rates and yielding to an untouched video quality and decodability.

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