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Encoding Techniques for stable Video Streaming over heterogeneous wireless networks

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Abstract— The wireless channel used for communication in different network varies in their Bandwidth, supported Bit rate and Unequal Error Protection coding techniques. An important issue of supporting multi-user video streaming over wireless networks is how to intelligently utilize this available network resources while, at the same time, to meet video's QoS (Quality of Service) requirement. In this proposal we go throw various schemes in order to achieve the maximum number of paths from all found node-disjoint routing paths for maximizing multimedia streaming data transmission and guaranteeing the end to end transmission delay in wireless networks.

Index Terms—Video streaming, path diversity,vertical hanf off, Fountain code, LT code, Raptor code**.**

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1 INTRODUCTION

The concept of video streaming was on papers in the last decade, yet an exponential growth can be seen in this era. The advent of digital integrated circuits and Internet along with the increased bandwidth availability have significantly contributed towards the wide spread of huge amounts of multimedia content through the Internet. Traditionally the simplest way for video delivery is downloading, alike to a file download. Expressly, video download is alike to a document download, but it is a large file. However this approach of video delivery has a number of drawbacks. Multimedia data are generally of considerable large size, demanding long and sometimes unbearable transfer times and large storage spaces. These are significant real-world limitations. This limitation demands for a more flexible and effective approach to deliver multimedia content. The solution to this problem is video streaming. The video streaming facilitates users to get instant access and not have to delay until the document is finished downloading. For demonstrations, the importance of efficient multimedia streaming over the Internet is shown by its many submissions, each enforcing distinct Quality of Service (QoS) requirements. The sort of content that gets streamed ranges from TV and radio programmers to video conferences and meetings. Generally, multimedia streaming applies to entertainment, information, business and education. A large amount of Internet stations worldwide transmits live audio and video content including music, news, sports events, speeches, concerts, movies, and documentaries.

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2. WHAT IS VIDEO STREAMING?

Video delivery by video streaming endeavors to overcome the problems associated with document download, and also presents a significant amount of added capabilities. The rudimentary concept of video streaming is to divide the video into components, convey these components in succession and allow the receiver to translate and replay the video as these components are achieved, without having to wait for the whole video to be transported.

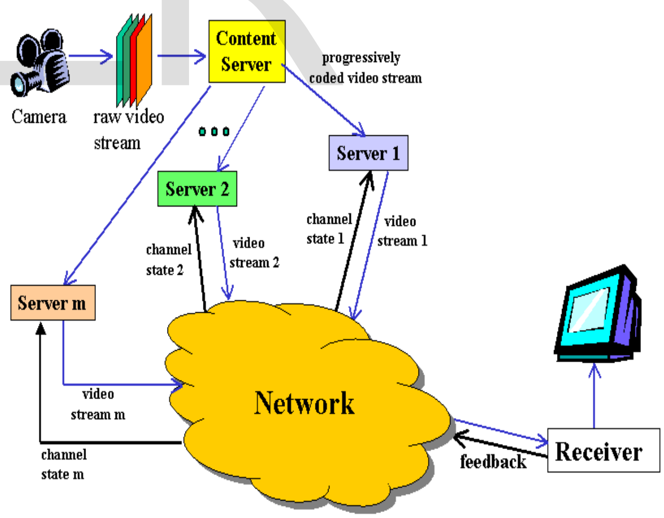

Figure.1 video streaming [37]

Video streaming can theoretically be supposed to comprise of the pursue steps:

1. Divide the compressed video into packets
2. Start delivery of these packets
3. Start decoding and playback at the receiver while the video is still being delivered

2.1 Review of video streaming

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The first video transmission was done by the band Severe Tire Damage on June 24, 1993 [38]. The occasion was seen live in Australia and other places over the Internet. There are numerous preceding studies associated to video streaming. Video streaming with MRC has been revised in [1, 2]. Scalable video can be used to provide video streaming reliably to a heterogeneous set of receivers with distinct subscription grades. However, the performance of such designs can be highly influenced by scheduling constraints and unreliable response. Network coding schemes, on the other hand, has decreased level of scheduling and prioritization complications and carried out well in wireless scenarios with perfect feedback. Steluta Gheorghiu et al. proposed a system to overcome the problem with the idea that when the server sends a generation of coded packets (i.e. coded packets related to a GOP), then it starts the loss recovery process, in order to recover lost packets. Realistic feedback is sent back from receiver nodes to the server in order to minimize the amount of unnecessary transmissions for recovering lost packets [1]. Other studies use video streaming with MDC [3, 4]. For example, J. Kim et al. employs MDC using an optimized rate allocation algorithm to minimize overall distortion. The work uses set partitioning in the hierarchical trees (SPIHT) algorithm to generate a convenient video stream that uses adaptive bit-rate according to the network conditions. Any rate changes in the network can be accommodated by dropping unnecessary packets from the generated stream [3]. J. Kim proposed an MDC extends a quality-scalable H.264/AVC video coding procedure to produce two independent descriptions. The two descriptions are transferred over different tracks to a receiver in order to improve the effect of unstable channel circumstances of wireless ad-hoc networks. If one description is vanishing because of transmission errors, then the properly received description is used to determine the lost information of the degraded description [4]. Many other studies, like T. Yoo et al. study video streaming in wireless environments; the approach focuses on minimizing the congestion experienced by video stream by jointly allocating link capacity and traffic flow. The work uses a cross-layer design framework that aims to support maximum data rates and yields minimum end-to-end delay [5].

2.2 Path diversity in video streaming

Diversity methods have been studied for several years inorder to understand the circumstances of wireless communication. They were introduced in order to exploit the large variability in terms of channel quality when multiple channels are considered for simultaneous transmission. A number of studies have shown that there is an analogous situation in Internet communication: Stefan Savage et al. conclude in that in 30-80% of the situation there is an alternate path that performs significantly better than the default path between two hosts [6]. Performance is measured in terms of loss rate, bandwidth and round-trip-time. These studies have motivated the introduction of packet path diversity for video streaming in [7], where J. Apostolopoulos proposed to send complementary descriptions of a multiple description (MD) coder through two different Internet paths. The presented investigation outcomes show the potential benefits of the proposed system. Since then a number of studies have appeared that exploit the concept of packet diversity in media communication. In [8] J. Apostolopoulos at al. employ path diversity in the context of video communication using unbalanced MD coding, the video is coded into a number of independently decodable streams, each with its own prediction procedure and state information. By having multiple streams, if one stream is corrupted the other streams remain accurate, and can still be accurately decoded to produce usable video, and most importantly can be used to recover the corrupted state of the damaged stream, to accommodate the fact that different paths might have different bandwidth constraints. The unbalanced descriptions are created by adjusting the frame rate of a description sent over a particular path.

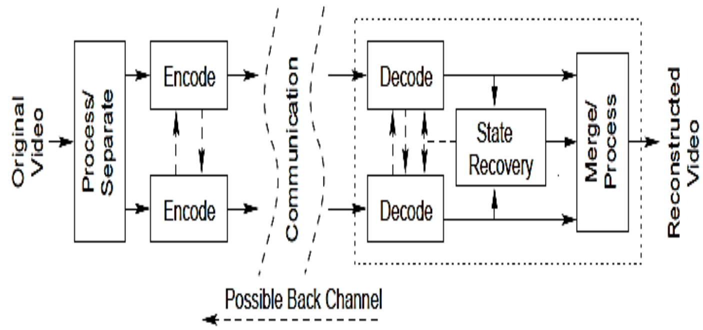

Figure.2 General two-state video communication system [7]

In [9] N. Gogate et al. study image and video transmission in a Multihop mobile radio network. It is shown that the combination of MD coding and multiple path transport in such a setting provides higher bandwidth and robustness to end-to-end connections. In [10] Thinh Nguyen et al. proposed a novel method to overcome the drawback of FEC that it results in bandwidth extension and hence decreases the amount of available bandwidth for the actual video bit stream, by combining path diversification and FEC, the bursty loss behavior in the Internet can be combated more efficiently. Specifically the distinction between the network characteristic for a single route streaming and FEC level becomes more relaxed in distributed streaming uses due to the extra redundancy introduced by multiple routes. Receiver-driven streaming of video from multiple servers to a single client is also studied by A. Majumdar et al. in [11]. The authors propose a network friendly streaming algorithm based on distributed MD coding of video. Because of the diversity effect of multiple servers, the procedure is resistant to single points of failure and provides natural load balancing of the servers. In [12] Y.J. Liang et al. proposed a framework for video transmission over the Internet, based on rate-distortion and path diversity optimized reference picture selection. Here, based on feedback, packet dependencies are adapted to channel conditions in order to minimize the distortion at the receiving end, while taking advantage of path diversity. In [13] J. Apostolopoulos et al. study the performance of path diversity and MD coding in Content Delivery Networks (CDN). 20-40% reduction in distortion is reported over conventional CDNs for the network conditions and topologies under consideration.

3 DIFFERENT METHODS OF VIDEO STREAMING

With the advances in multimedia coding standards like MPEG-4, the claim for bandwidth famished multimedia content over the Internet has bigger. Regardless of these improvements, streaming video data over the Internet is still a challenging difficulty due to a number of reasons, comprising of heavy bandwidth requirements and the sensitivity of video data to packet loss, delay and jitter. Some authors have characterized the current methods to solve these problems in the protocol centric, channel coding centric, network and source coding centric methodologies [14]. Automatic repeat request (ARQ) method came into existence to overcome random packet loss that happens throughout transmission of video over networks [15]. Forward error correction (FEC) recommended to lessen the delay because of retransmission, at the expense of bandwidth expansion [16] [17]. Since ARQ requires a strict delay constraint but on the other hand FEC increases the reliability by transmitting data with some additional information. However, these fixed-rate FEC, where for a given source block the channel code rate is fixed or updated according to a prediction based on past observations of the packet loss rate. Unfortunately, the packet loss rate on the Internet and packet-based wireless networks are hard to predict and can rapidly change over time. Thus, the performance of fixed-rate FEC schemes may be poor because of the unavoidable mismatch between the actual packet loss rate and the predicted one. The extended form of the FEC is fountain code is utilized to overcome the problem. fountain code is AL-FEC (Application layer Forward error correction) scheme. In [18] fountain coding is used to achieve path diversity.

3.1 Fountain Coding

The idea of Fountain coding is different from the original FEC idea where channel encoding is performed for a fixed channel rate and all encoded packets are generated prior to transmission. The Fountain encoder is an imaginary fountain of a limitless supply of water drops (output symbols). Any person who wants to reconstruct all of the input symbols has to wait to fill their bucket with slightly more water drops than the number of input symbols. Thus, the main idea behind Fountain coding is to produce as much output symbols as needed on-the-fly. This property gave another name to fountain codes, rateless codes.

The principle of Fountain codes is illustrated in Figure 3. The encoder generates potentially limitless output symbols (water drops) from the input data. The decoder (bins) try to collect enough number of output symbols to complete decoding and reconstruct the input symbols. During the transmission (collecting water drops), output symbols may get lost due to the channel conditions. In such a case, decoders do not send any retransmission request messages back to the encoder. They just wait to receive enough number of output symbols to complete the decoding.

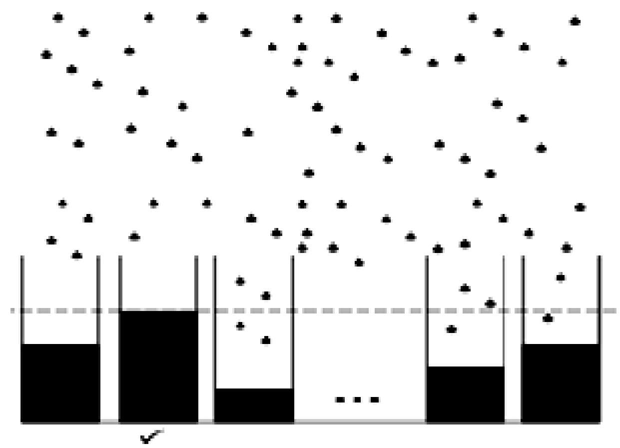

Figure 3: Overview of Fountain coding: The decoders (bins) try to collect a sufficient number of output symbols (water drops)

Algorithm

Raouf Hamzaoui et al proposed Digital fountain code, which are based on bipartite graphs, that is, graphs with two disjoint sets of vertices such that two vertices in the same set are not connected by an edge. The first set of vertices contains the source symbols, while the second set contains the encoded symbols. The symbols are binary vectors, and arithmetic on symbols is defined modulo 2. In particular, ⊕ denotes modulo 2 addition. If the number of source symbols is k, the degree of an encoded symbol is given by a degree distribution $\Omega\ (x) = \sum_{i=0}^{k} \Omega_i x^i$ on $\{1, \ldots, k\}$ where Ω_i is the probability that degree i is chosen. For example, suppose that

$$\Omega_i = \begin{cases} 0 & if\ i = 0 \\ \dfrac{1}{k} & if\ i = 1 \\ \dfrac{1}{i(i-1)} & otherwise \end{cases}$$

Then $\Omega\ (x)$ is called the ideal soliton distribution. A more practical distribution is the robust soliton distribution $\Delta\ (x)$ given by $\Delta_i = \frac{\Omega_i + \Gamma_i}{d}$ where $\Omega\ (x)$ is an ideal soliton distribution, $\Gamma(x)$ is given by

$$\Gamma_i = \begin{cases} \dfrac{s}{ki} & if\ i = 1, \ldots. \dfrac{k}{s} - 1 \\ \dfrac{s}{k} \ln(s/\delta) & if\ i = k/s \\ 0 & otherwise \end{cases}$$

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$d = \sum_{i=1}^{k} \Omega_i + \Gamma_i$ and $s = C ln \frac{k}{\delta} \sqrt{k}$. Here C and δ are parameters. Any distribution $\Omega(x)$ on $\{1,\dots,k\}$ induces a distribution on F_2^k the set of binary vectors of length k, by $prob(v) = \frac{\Omega_{w(v)}}{\binom{k}{w(v)}}$ where $v \in F_2^k$ and $w(v)$ is the weight of v (that is, the number of nonzero components of v).

A reliable algorithm for a digital fountain code is an algorithm that ensures that all k source symbols can be recovered with probability $1 - 1/k_c$ for a positive constant c [19].

The Digital Fountain approach is first described in 1998 as a novel technique for reliable distribution of bulk data [20]. In the same year, a company named Digital Fountain Inc. is founded by Charlie Oppenheimer and Dr. Michael Luby in CA, U.S. with the aim of commercializing and standardizing the fountain coding approach. In 1999, the first patent on fountain coding appeared [21]. In 2002, Luby Transform (LT) codes, named after Michael Luby, are published in [22] that described the coding scheme in the patents. The coding scheme attracted significant interest and various papers on LT codes have been published since then.
Raptor coding is proposed as a multi-stage extension of LT coding. They are invented in 2000 and patented [23]. They are still one of the most advanced fountain coding scheme, and similar to LT coding, attracted a wide interest.

3.2 LT code

Michael Schier et al. proposed an encoding design of the LT-codes use two probability distributions: a so-called Robust Soliton distribution which is utilized to get the degrees of encoded symbols and a uniform distribution responsible for the selection of addends contributing to the encoded symbol generation. By replacing the latter with a distribution with $P_\psi(s_i) = \frac{\alpha + (1-\alpha)\psi(s_i)}{\sum_j^k \alpha + (1-\alpha)\psi(s_j)}$ which reflects the perceptual relevance of input symbols s_i of the current source block $\{s_1, \dots, s_k\}$, the probabilities of successfully decoding perceptually more important input symbols can be raised. In this connection, the appropriate choice of $\alpha \epsilon [0;1]$ is crucial: a too low value might cause significant transmission overhead due to certain input symbols being ignored by the encoder whereas a too high value may almost eliminate the benefits of perceptual relevance estimation. For decoding, they use a modified version of a binary Gaussian elimination algorithm with a matrix of non-fixed size as data structure increase and decrease of its dimension is caused by incoming encoded symbols and resolved (covered) input or redundant (released) encoded symbols correspondingly. [24]
Shakeel Ahmad et al. proposed a system for unequal error protection with a Fountain code. When the data were divided into two protection modules (the least important and the most important), the proposed system required a smaller transmission bit budget to attain low bit error rates as compared to the two state of the art methods. The LT code proposed in this paper, extend the set of information symbols by duplicating it. This idea has similarities with the sliding window (SW) technique. The recommended approach can be described as follows,
Consider a source block $S = i_0 * \dots\dots * i_{k-1}$ consisting of k information symbols $i_0 \dots\dots i_{k-1}$. Let $\Omega(x)$ be the degree distribution of an LT code on $\{1, \dots\dots k\}$. The source block S is expanded by repeatedly appending the same k information symbols at the end of the block. The new (virtual) source block can be written $\underbrace{S * S \dots\dots * S}_{EF}$ where the expanding factor EF denotes the number of times the original source block occurs in the new source block. This new source block has a length of $EF \times k$ and its information symbols have indices ranging from 0 to $EF \times k - 1$ (Figure 4). Next, the original degree distribution $\Omega(x)$ is expanded from $\{1, \dots\dots k\}$ to $\{1, \dots\dots EF \times k\}$ and use a standard LT encoder with this new degree distribution to generate the encoded symbols. For the robust soliton distribution, this is done by replacing k by $EF \times k$. An encoding graph using the original k information symbols $i_0 \dots\dots i_{k-1}$is obtained by replacing the index $j \in \{0, \dots\dots, EF \times k\}$ of a selected information symbol by j mod k.

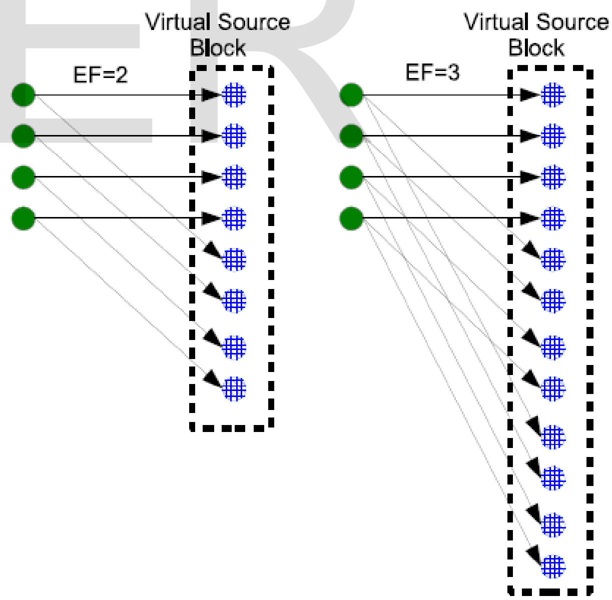

Figure.4 Virtual increase of the source block size for k = 4, (left) EF = 2 and (right) EF = 3 [25].

Unequal error protection

The concept of virtually increasing the size of the source block by duplicating information symbols has a natural application to UEP. Suppose that a source block $S = i_0 * \dots\dots * i_{k-1}$ is partitioned into L adjacent blocks $S_1 \dots\dots S_L$ such that the first block S_1 consists of the most important bits, the next block S_2

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consists of the next most important bits and so on. Now the different level of protection can be applied to this block by duplicating them according to a sequence of repeat factors $RF_i, i = 1 \dots L$. That is, a (virtual) source block is obtained.

$$\underbrace{S_1 * S_1 \dots\dots * S_1}_{RF_1} * \underbrace{S_2 * S_2 \dots\dots * S_2}_{RF_2} * \dots * \underbrace{S_L * S_L \dots\dots * S_L}_{RF_L}$$

Whose information symbols have indices ranging from 0 to $\sum_{i=1}^{L} RF_i\ |S_i| - 1$. Now the degree distribution of the LT code is expanded from $\{1, \dots\dots k\}$ to $\{1, \dots\dots \sum_{i=1}^{L} RF_i\ |S_i| - 1\ \}$. To generate an encoded symbol, they find its degree d using the new degree distribution and then select d information symbols from the virtual source block. An encoding graph using the original k information symbols $i_0 * \dots\dots * i_{k-1}$ is obtained by replacing the index $j \in \{0, \dots\dots, \sum_{i=1}^{L} RF_i\ |S_i| - 1\ \}$ of a selected information symbol by an index l as follows [25]:

$$l = \begin{cases} j \bmod |S_1| & if\ 0 \le j \le RF_1\ |S_1| - 1 \\ [(j - RF_1\ |S_1|) \bmod |S_2|] + |S_1| & if\ RF_1\ |S_1| \le j \le RF_1\ |S_1| + RF_2\ |S_2| - 1 \\ \left[\left(j - \sum_{i=1}^{L-1} RF_i\ |S_i|\right) \bmod |S_L|\right] + |S_{L-1}| + \dots + |S_1| & if \sum_{i=1}^{L-1} RF_i\ |S_i| \le j \le \sum_{i=1}^{L} RF_i\ |S_i| - 1 \end{cases}$$

3.2 Raptor code

Raptor Codes belong to a class of codes called ‘fountain codes’. The rate of such codes can be made arbitrarily close to zero. Hence, they are also called ‘rateless codes’.

Rateless Codes

An ideal rateless code allows the generation of an unlimited number of code symbols from a single source block of finite size, say k, ideally, any k of these packets can be used by the decoder to obtain the original source block. Erasure channels drop packets at random, but provide error-free transmission of the remaining packets. This is shown in Figure 5. Since a fountain code encoded block of length k can be recovered from the first k symbols received, which is the minimum amount of information needed, they are ideal for such channels.

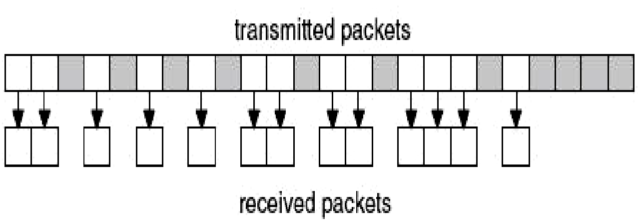

Figure.5 Erasure channel [26]

Raptor codes are fountain codes which are based on LT codes. An LT decoder may, however, in some cases be unable to decode even if sufficient data are available. Raptor codes add a pre-coding stage before LT encoding. They use a fixed length systematic code in the pre-coding stage. This increases robustness, with very little increase in redundancy. This is shown in Figure 2. [26]

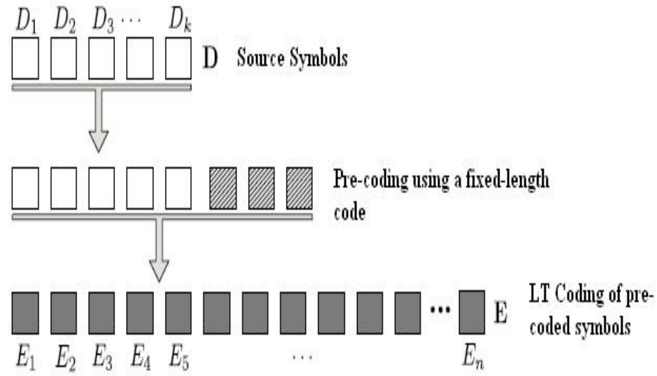

Figure.6 Raptor coding scheme [26]

Christos Bouras et al. explore the impact of the Raptor FEC parameters on the AL-FEC system robustness. Raptor FEC is the only method dedicated to the MBMS reliability enhancement. The use of Raptor codes in the application layer of MBMS has been applied to 3GPP by Digital Fountain. In preparation for the encoding, a certain allowance of facts and figures is assembled inside an FEC source block. The data of a source block are further split up into k source symbols of a fixed symbol size. The decoder is adept to retrieve the entire source block from any set of FEC encoding symbols only somewhat more in number as compared to the source symbols. The Raptor code specified for MBMS is a systematic fountain code producing n encoding symbols E from $k < n$ source symbols C. This code can be considered as the concatenation of several codes. The most-inner code is a nonsystematic Luby-Transform (LT) code with l input symbols F, which provides the fountain property of the Raptor codes. This nonsystematic Raptor code cannot be constructed by simply encoding the source symbols with the LT code, but by encoding the intermediate symbols generated by some outer high-rate block code. This means that the outer high-rate block code generates the F intermediate symbols using k input symbols D. Finally, a systematic realization of the code is obtained by applying some preprocessing to the k source symbols C such that the input symbols D to the nonsystematic Raptor code are obtained.

Raptor codes have a performance very close to ideal, that is, the failure probability of the code is such that in case that only somewhat more than k encoding symbols are received, the received code are sufficient to reconstruct the source block. In fact, for $k > 200$ the small inefficiency of the Raptor code can accurately be modeled by (1)

$$P_f(m,k) = \begin{cases} 1 & if\ m < k \\ 0.85 \times 0.567^{m-k} & if\ m \ge k \end{cases}$$

In (1), $P_f(m,k)$ denotes the failure probability of the code with k source symbols if m symbols have been received. It has been observed that for different k, the equation almost perfectly emulates the code performance. Although an ideal fountain code would decode with zero failure probability when $m = k$, the failure for Raptor code is still about 85%. However, the failure probability decreases exponentially when the number of received encoding symbols increases [27].

Table 1Fountain coding

Shakeel Ahmad et al. "Unequal Error Protection using Fountain Codes with Applications to Video Communication"	The proposed method for unequal error protection with a Fountain code showed an improvement up to 13 dB in PSNR for unicast video transmission as compared to other UEP methods. For multicast video transmission results showed a better average PSNR performance than the best previous methods [25].
Mattia C.O. Bogino et al. "Sliding-Window Digital Fountain Codes for Streaming of Multimedia Contents"	The recommended method shows an enhancement in terms of trustworthiness. In fact, keeping the overhead unchanging, SF approach permits to accomplish an undecoded symbol rate lower of 10^5 than old models. Furthermore, the SF encoding method is less convoluted. Eventually, a SF system needs a small amount of memory with respect of a traditional one [28].
Sang-Chun Han et al. "Path Virtualization using Fountain Code for Video Streaming over Heterogeneous Networks"	In [29] author proposed a fountain code. The whole system is developed in java. By the experiment in the real wireless environment, it is concluded that the system supports higher bandwidth, lower delay, and lower BLR than traditional approaches.

Table 2 LT coding

Ali Talari et al. "Optimized cross-layer forward error correction coding for H.264 AVC video transmission over wireless channels"	In this paper the joint optimization of cross-layer is considered, UEP LT coding at the AL and RCPC coding at the PL for robust H.264 video transmission over wireless channels. The experimental results conclude that the proposed method outperformed as compared to the other FEC methods that use either UEP coding at the PL alone or EEP FEC schemes at the AL. Further the recommended method optimization works well for different H.264-encoded video sequences, which have generally different characteristics [30].
Shakeel Ahmad et al. "Video multicast using unequal error protection with Luby Transform codes"	The author proposed a scheme for unequal error protection with Luby Transform codes simulations showed that the recommended UEP scheme has a better average PSNR performance when the transmission overhead is large and a worse performance when the overhead is low [31].
Andrea Magnetto et al. "P2P Streaming with LT Codes: a Prototype Experimentation"	The major finding of this work is that the efficient yet simple LT codes yield a very high CI for a streaming rates up to 1 Mbps both in stable and dynamic scenarios with limited bandwidth resources [32].

Table 3 Raptor coding

Philipp M. Eittenberger et al. "Raptor Codes for P2P Streaming"	This paper introduces the application of Raptor codes for P2P streaming. The results obtained by this approach are promising the code require only a small overhead rate for large block sizes and are able to support the necessary streaming rates [33].
Nikolaos Thomos et al. "Collaborative Video Streaming With Raptor Network Coding"	In [34] author proposed a source and channel rate allocation algorithm for collaborative streaming with Raptor network coding. The experimental evaluation demonstrates that the rate allocation algorithm performance is better than previous one.
Rohit Watve et al. "Comparison of Raptor Codes with ARQ for Video Streaming on	In this project the author modeled two schemes for error resilience for video streaming in a peer-to-peer network, FEC and

Peer-to-Peer Networks”	ARQ. The models considered a variation of loss probability with delay. The results showed that raptor codes are better compared to the single retransmission scheme for lower node disconnection probabilities. Retransmission scheme with more than one request always performs better as compared to the scheme that uses raptor codes, which was verified by models as well as experiments [26].
Philipp M. Eittenberger “Raptor Stream: Boosting Mobile Peer-to-Peer Streaming with Raptor Codes”	The author modeled Raptor codes as AL-FEC to exploit the upload capacity of mobile devices and to reduce the complexity of P2P chunk scheduling. The results show that the Raptor codes require only a small overhead rate and are able to support the necessary streaming rates [35].
Pasquale Cataldi et al. “Sliding-window Raptor codes for Efficient Scalable Wireless Video Broadcasting with Unequal Loss Protection”	The author proposed a new class of DF codes, called the SW-Raptor codes. The experimental results show that the proposed encoding scheme achieves very low decoding failure probability, also when the actual packet loss rate is different from the nominal value [36].

4 CONCLUSION

Nowadays, various wireless networks with different characteristics are available to support diverse user applications. Cooperation among this network is very important to provide seamless data service. Recently, fast vertical handoff and path diversity technologies have obtained a lot of interest for solving the aforementioned cooperation problem. Vertical handoff is a switching technology between two different networks to support seamless service. This approach needs sophisticated architecture, mutual agreement among various network service providers, and additional implementation costs and time. In path diversity one mobile node can establish multiple paths over multiple wireless access networks. Still there is a problem in transmission of multimedia data transmission over different paths as different path has different channel parameters, bandwidth various methods had been modeled to overcome this problem like ARQ and FEC. The ARQ requires a strict delay constraint on the other hand FEC requited extra data to enhance reliability the transmission. The extended form of the FEC is a fountain code, used to achieve path diversity, is utilized to overcome the drawbacks of previous methods. We have studied the literature related to fountain coding, LT coding and the Raptor coding. Raptor Codes and LT codes belong to a class of codes called ‘fountain codes’. Raptor codes are a significant theoretical and practical improvement over LT codes. From this study we concluded that the raptor code best as compared to previous methods in the field of complexity, rate allocation and packet loss.

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Different methods of video streaming

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