Predictable Reliability and Packet Loss Domain Separation for IP Media Delivery

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Abstract—Internet protocol (IP) based media delivery enables applications such as Voice over IP or, most recently, stereoscopic HD-TV. Suchlike transport of typically time-constrained media between two end-points has a number of important characteristics, both on the transport and content side. We focus on live streams of audio-visual data that are assumed as time-constrained and loss-tolerant. We aim at optimizing IP transport over heterogeneous networks by hybrid and loss domain separated error correction with residual error. Hybrid error correction (HEC) approaches the channel capacity dynamically and achieves predictable reliability and predictable delay (PRPD) when residual loss is tolerable, while loss domain separation applies optimally chosen HEC parameters to individual network segments. In this paper, we apply an atomic HEC coding unit to individual segments of network links between source and sink, and show by analysis and example how redundancy is distributed optimally.

Index Terms—transport layer; channel coding; loss tolerance; predictable reliability; PRPD; hybrid error correction; 3D-IPTV

I. INTRODUCTION

From a communications engineering point of view, Internet transport is exchange of information via heterogeneous communication channels that are either cascaded, parallel, or both, when the application consists of multiple streams. Establishing a reliable communications channel by the end points via IP has been a design paradigm, but in IP delivery of media streams as e.g. VoIP or 3D-IPTV packet loss may be tolerated to a certain extent. At the same time, suchlike services may not tolerate transport delays above a certain time limit. Everything over IP describes how digital communication has evolved: systems like telephony and broadcast have been dedicated, i.e. specifically designed for their purpose and predictable in their performance, while IP-based delivery is provided via heterogeneous, IP-enabled infrastructures available to a larger audience and nearly everywhere. But still, transport of such content over the Internet is mostly done using UDP/IP or TCP/IP. The latter typically provides reliability, yet does not take into account that links between the end points may have largely different characteristics. Thus, streaming applications using TCP may stall when used over a wireless link. What is more, positive acknowledgments obviate scalable one-to-many delivery. This is why most current streaming applications make use of UDP, which leaves the implementation of functionality to the application.

Recent developments on RTP1,2 and DVB (e.g. [1]) provide application layer error coding techniques. But even the most recent (including those actually within the standardization phase) do provide only a limited adaptivity. Turbo-, [2], Fountain- [3] and Raptor-Codes [4], [5] have revolutionized forward error coding (FEC) as they approach the Shannon-bound very closely (depending on the channel characteristic down to a few tenth of a dB). The way to achieve this, however, is to overcome the prohibitive increase in complexity when designing long codes (the complexity for RS-codes grows with a small power of the block length while that of one of Raptor-codes approximately increases linearly). Hence these codes are efficient because they are long. This makes those codes very appropriate for lower layer error protection (Turbo- and LDPC-Codes on the physical layer of broadcast systems) but leads to very long transmission delays when applied to packet erasures on higher layers.

In our earlier and recent work [6], [7] we have optimized a general hybrid error correction (HEC) architecture in a way that—under a given delay constraint and an allowed residual error—the Shannon-bound is approached dynamically, as opposed to similar work [8]. We could prove that due to the fast changing channel capacity especially in wireless networks the efficiency of this adaptive error coding is far higher than that of purely FEC or retransmission based systems. With the increasing demand for Internet delivery of high throughput, delay constrained media streams (video alone is expected to exceed 91% of global consumer traffic in 2014 [9]) and the limitations of existing approaches, an increase in efficiency and flexibility of Internet transport will be required.

In this paper we apply above mentioned atomic HEC coding unit to individual segments of IP network links according to the PRPD principle: predictable reliability and predictable delay. PRPD is achieved by each individual HEC unit adaptively and globally when combined. Optimized media delivery as proposed herein follows the end-to-end principle [10], [11] with support of intermediate packet relays that are configured by the end points in order to globally optimize transport over multiple media streams on heterogeneous links.

The paper is structured as follows: Sec. I-A introduces PRPD via IP. Sec. I-B provides an overview of related work. In II we state the analysis problem and introduce a model of calculating the channel capacity for loss tolerant erasure channels, which is extended to multiple hops in III. Sec. IV deals with the optimization of parameters for hybrid error correction and provides a 3D-IPTV example over multiple hops and links. Finally, a conclusion is given in Sec. V.

A. Motivation

Media streaming protocols have a broad variety of applications. The transport and coding parameter space for those applications, however, is huge: performance of Internet channels with respect to packet loss and round-trip time is highly dynamic. In addition, different media have different requirements, and they may differ in orders of magnitude: A VoIP codec is typically designed for unreliable transport and thus may tolerate up to 10% packet loss but end-to-end delay of only 50–80 ms. IPTV on the other hand may tolerate several hundred milliseconds transport delay but only very few losses before clearly visible or audible artifacts occur. Furthermore, services comprising multiple streams, such as 3D-IPTV—or hierarchically coded video in general—where a base layer and...
an enhancement layer (delivering increased resolution or an additional view) are transmitted in parallel, may have identical delay but unequal error protection requirements.

We therefore are convinced that transport protocols for the Future Media Internet have to be adaptive to the applications’ requirements as well as the characteristics of those channels over which media is transmitted. We conclude this behavior under the term Predictable Reliability under Predictable Delay (PRPD): We avoid to formulate QoS guarantees but believe that careful analytical parametrization of transport layer protocols based on statistical channel models is able to meet the application constraints - an approach that seems feasible and suitable to be applied to the unmanaged Internet.

B. Related Work

Driven by audiovisual applications, partial reliability has become a wide area of research. SCTP [footnote] is a novel protocol designed to enable multi-homing and multi-streaming by aggregation of several TCP-like connections. It supports a fixed number of retransmission approaches in order to limit the transport delay. However, since SCTP inherits the connection-oriented characteristics as well as the flow control from TCP, it is not suitable for scalable media distribution.

Exposito et al. at LAAS Toulouse have developed a comprehensive framework on partial order and partial reliability [12]. They make use of available protocols and QoS negotiation mechanisms on different OSI layers to offer the optimal transport according to the application’s QoS requirements.

Several solutions have been proposed to increase reliability especially in wireless IP networks. Similarly to our approach [8] applies H-ARQ but without explicit time and reliability constraint. Since TCP generally operates suboptimally under the presence of distributed corruption losses, several Decode and Forward policies [13] using H-ARQ have been evaluated on lower protocol layers in order to provide increased reliability to transport layer.

II. CHANNEL CODING UNDER PRPD

Packet-switching networks differ significantly from classical communication channels: Instead of sending short coded symbols at a high frequency, long datagrams consisting of multiple symbols are transmitted asynchronously. Furthermore, a nondeterministic time division multiplexing of several connections over shared resources leads to unpredictable delivery time. Several approaches have been developed in order to establish guaranteed end-to-end QoS over dedicated network links. On the other hand, there is a significant trend towards the usage of unmanaged Internet transport for large-scale media delivery. In fact, for most audio-visual content statistically predictable residual packet loss rate and transport delay are sufficient and more feasible than strict QoS guarantees. Consequently, we introduce loss-tolerant erasure channels and channel coding under time constraints in the following.

A. Loss-tolerant Erasure Channel Introduction

In order to calculate the channel capacity while allowing a residual error rate, we virtually cascade two erasure channels, as depicted in figure 1, in which the combined error rate of the cascaded channels is equal to the error rate of the original channel. Hence the second link provides the residual error $P_r$, which can be regarded as a target loss rate $P_r$. For the overall loss rate $P$ we have $P = 1 - (1 - P_v)(1 - P_r)$ and solving this for $P_v$ yields

$$P_v = \begin{cases} \frac{P - P_r}{1 - P_r} & P \geq P_r \\ 0 & \text{else} \end{cases}$$

Replacing the original error rate of the channel by $P_v$, we pretend that the channel has a virtual error rate $P_v$, as depicted in figure 1, which is less than its original error rate for a residual error rate $P_r > 0$. Assuming $P \geq P_r$ and an ideal code, then, by definition the residual link capacity is

$$C_r = 1 - P_v = \frac{1 - P}{1 - P_r} \quad (1)$$

and hence the required residual amount of redundancy information is

$$RI_r = \frac{P_v}{1 - P_r} = \frac{P - P_r}{1 - P} \quad (2)$$

Eq. 1 & 2 demonstrate the channel capacity respectively the required minimal redundancy information for a loss-tolerant erasure channel that allows a residual packet error of $P_r$.

B. Channel Coding under a Time Constraint

The classical information theory does not explicitly include a measure of time. Time consumption of error correcting codes is implicitly fixed by the symbol rate of the communication channel and the code word length. ARQ schemes even introduce a dynamic coding delay due to their variable number of request and retransmit cycles.

Assume a packet-switching network where a message requires a maximum round trip time RTT between a fixed source and a fixed set of sinks. Datagrams are transmitted at a period of $T_s$. Let an arbitrary packet-level error correcting code operating between network source and sink be an atomic PRPD coding unit as depicted in figure 2 that has an input erasure rate $P_v$. Under an upper-bounded coding delay $D_t$ it requires redundancy information $RI$ in order to achieve a residual erasure rate $P_r$ with high probability.

The actual amount of $RI$ depends on the error correction scheme applied within the PRPD coding unit. Optimally, $RI$ would be slightly less than $RTT$. The tighter the delay constraint, the more $RI$ deviates from $RTT$, in order to maintain the residual erasure rate $P_r$. For instance, an ARQ may have to shift more redundancy on earlier transmission cycles in order to meet the reliability constraint within less delay.

\[\text{Figure 1. A sequence of two erasure channels. The second one provides the residual error.}\]

\[\text{Figure 2. PRPD coding unit achieving a target delay } D_t \text{ at a residual error rate } P_r < P \text{ with a certain redundancy } RI\]
III. MULTI-HOP CODING

The characteristics of IP based delivery paths are increasingly heterogeneous due to the variety of underlying architectures. Especially wired and wireless networks differ significantly in reliability. Ignoring the network segment heterogeneity is suboptimal since redundancy and feedback generated for unreliable network segments share resources on reliable segments.

A. Problem Statement

Concerning a packet-based transmission from a sender $S$ to a receiver $R$ via one or more intermediate network segments as depicted in figure 3, the path $(S, ..., R)$ is regarded as a packet erasure channel. Thus, we assume that a packet is either forwarded to the next hop or it is lost. Following the principle of decode-and-forward architectures, the proposed PRPD coding unit (cf. figure 2 on the preceding page) may operate between any two hops in a serial policy. In order to meet overall delay and reliability requirements for the virtual end-to-end link between source and sink, each coding unit is allowed to contribute a defined portion of predictable residual loss and predictable delay. Thus, the question is how to dimension the amount of redundancy information (RI) on each individual segment between the end points. Here we refer to RI as the amount of information that is necessary to offer a sufficiently reliable end-to-end transmission at an upper bounded packet loss rate (PLR) within an upper bounded period of time.

It should be noted that at this level of consideration the algorithms to provide a sufficient and efficient amount of RI are not further specified but dynamically chosen by the PRPD coding unit. We assume that the best choice for each individual segment will depend on the respective PLR and delay behavior of the segment and the scenario, i.e. the target requirements, the number of sources and the interdependency of segments and links to the sources [6], [7].

In order to optimize link-level coding parameters with global efficiency, the scheme evaluates inherent knowledge about the entire network path, which is obtained by a deeper packet inspection. We are aware of the fact that the introduction of segment-wise, globally optimized packet recovery interferes with the OSI layer model. However, the proposed multi-hop error correction relies on an evolutionary deployment, i.e. the fall-back to a pure end-to-end transport layer connection is always possible whereas the efficiency increases significantly with the introduction of smart decode-and-forward nodes at sensibly chosen positions.

As depicted in figure 3, several scenarios are possible. The information may, as depicted in figure 3 a), be transmitted via a single link. In figure 3 b) and c) the situation is more complicated since we are controlling two or more intermediate segments, assumed to be completely independent with respect to their PLR and delay behavior. The next step in complexity is the consideration of multiple sources transmitting via independent paths, each consisting of independent segments, as depicted in figure 3 d). For the latter case, we will only give an exemplary parameterization for a particular scenario in section IV.

B. Individual Residual Error for two Segments

The residual error rate as defined above and adopted to this case is

$$P_r = 1 - (1 - P_r[1])(1 - P_r[2])$$

(3)

Solving this for $P_r[2]$ yields

$$P_r[2] = \begin{cases} \frac{P_r[1] - P_r[1]}{1 - P_r[1]} & P_r \geq P_r[1] \\ 0 & \text{else} \end{cases}$$

We set the target loss rate $P_r = P_T$ to a fixed value as specified by the application. Then $P_r[1]$ is a free parameter allowed to be within $0 \leq P_r[1] \leq P_T$. For the residual amounts of redundancy $RL$, we obtain

$$RL[1] = \begin{cases} \frac{P_r[1] - P_r[1]}{1 - P_r[1]} P_r[1] \geq P_r[1] \\ 0 & \text{else} \end{cases}$$

$$RL[2] = \begin{cases} \frac{P_r[2] - P_r[2]}{1 - P_r[2]} P_r[2] \geq P_r[2] \\ 0 & \text{else} \end{cases}$$

$$RL = \frac{RL[1] + RL[2]}{2}$$

The average residual amount of redundancy per segment is thus

$$RI_s = RL$$

and the average residual amount of redundancy per segment for end-to-end is

$$RI_{2e} = P_r - P_r[1]$$

while the ratio of both becomes

$$\frac{RL}{RI_{2e}} = \frac{1}{2} \left( \frac{1 - P_r}{P_r - P_r[1]} \right) \left( \frac{P_r[1] - P_r[1]}{1 - P_r[1]} + \frac{P_r[2] - P_r[1]}{1 - P_r[2]} \right)$$

This term is mainly dominated by factor $1/2$.

The relative redundancy information for each segment $m1, m2$ compared to the end-to-end case is

$$RI_{m1}[m2] = \frac{(P[m1] - P[m1])(P[m2] - 1)}{P[m1][P[m2] - P[m1] - P[m2] - P_r]}$$

C. Multi Hop Residual Error

In the following we will distinguish two cases of how to cascade multiple hops with residual error. The case we also refer to as combined residual error is describing a residual error when treating the individual segments’ error rates as a single, combined one. Secondly, we consider residual error rates for each segment individually, and thus transmission’s residual error is a combination of the individual ones.
This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE ICC 2011 proceedings.

Figure 4. Combined residual error, where the loss rates of the individual erasure channels are combined.

a) The last two of M erasure channels.
b) The first erasure channel represents the combination of the individual erasure channels from a). The second one provides the residual error.

Figure 5. Individual residual error. Here, the last two of M residual erasure channels are depicted. The combination of M residual erasure channels provides the residual error.

1) Combined Residual Error: Here the individual error rates denoted by $P[m]$ for each segment $m$ of $M$ are combined to be

$$P_c = 1 - \frac{M}{m=1} (1 - P[m])$$

as depicted in figure 4a). Therefore, in analogy to the single hop residual error, the residual amount of redundancy information is

$$RI_{c2e} = \frac{P_c}{1 - P_c} = \frac{P_c - P_e}{1 - P_e}$$

(4)

It is the same on every segment, therefore the average per-segment is $RI_{c2e}$. The target loss rate in this case is $P_t = P_c$ as depicted in figure 4b).

2) Individual Residual Error: In this case we introduce individual residual error rates for each segment. The overall residual error rate is accordingly

$$P_r = 1 - \frac{M}{m=1} (1 - P_r[m])$$

as depicted in figure 5, whereas the target loss rate $P_t = P_r$. For each segment we find the redundancy information depending on the individual original error rates as well as the residual error rates to be

$$RI[m] = \frac{P_r[m]}{1 - P_r[m]} = \frac{P[m] - P_r[m]}{1 - P[m]}$$

(5)

The average residual amount of redundancy per segment is then

$$RI_s = \frac{1}{M} \sum_{m=1}^{M} RI[m]$$

While the average amount of redundancy per segment is rather a measure for the overall network load, the individual RI load per segment is, relative to the end-to-end case, given by $RI_s / RI_{c2e}$.

IV. PARAMETER OPTIMIZATION

A. Adaptive Hybrid Error Correction

The performance of Adaptive Hybrid Error Correction (AHEC) over packet erasure channels was already shown by Tan [14]. Those schemes include Forward Error Correction (FEC) and Automatic Repeat reQuest (ARQ) offering scalability for large receiver groups and timely adaptation of the code rate. Given the numerous degrees of freedom a complex analysis is required for the design of coding parameters under given channel conditions. For convenience and to exemplify our following thoughts, we build upon an ordinary Type II HEC in the upcoming sections.

Type II HEC is well known for the deployment in incremental redundancy or chase combining strategies [15] established in 3G networks. Generally, those schemes are based on Maximum Distance Separable (MDS) block codes, such as the Vandermonde codes [16]. They apply different puncturing patterns to the same coded message block within several transmission cycles, whereas the individual cycles are triggered upon receiver request. Receivers accumulate the code symbols conveyed during all cycles to reassemble the coded block.

A perfect MDS code generates $n$ code symbols out of $k < n$ information symbols. Via virtual interleaving as it is deployed in DVB IP datacast [17], systematic block coding on packet level is possible with just a one-sided delay that corresponds to the time required to collect $k$ data packets from a real-time source. The properties of perfect maximum distance separable (MDS) codes enable the recovery of all data packets as soon as arbitrary $k$ packets from the same block are available at the receiver.

The HEC scheme splits the $n - k$ parity packets of the systematic code into portions of size $n_{p,w}$, to be transmitted in transmission cycle $w$, respectively. Let $\bar{N}_p = (n_{p,1}, n_{p,2}, \ldots, n_{p,r})$ be a vector defining the portions of parity for overall $r$ cycles, where $w = 1$ refers to the initial transmission immediately after the $k$ data packets. The portions for $1 < w \leq r$ are sent upon receiving a negative acknowledgment (NACK) from the receiver, indicating that additional parity is required to recover lost data packets. Obviously, the minimum distance of the MDS code has to be chosen such that at least $n - k \geq \sum_{w=1}^{r} n_{p,w}$ parity packets are generated. The coding scheme is explicitly determined by the FEC information length $k$ and the parity distribution vector $\bar{N}_p$.

B. Performance Evaluation

Predictive coding parameter design requires statistical performance evaluation on estimated channel characteristics such as the current packet erasure rate $P_e$ as well as the Round Trip Time $RTT$. We propose a simplified performance evaluation and refer the reader to the analysis of the General Architecture on Hybrid Error Correction [14] for more details.

Predictable, upper-bounded coding delay $D_t$ is achieved by adjusting $k$ and the number of transmission cycles $r = \text{dim}(\bar{N}_p)$ with respect to the real-time stream’s average packet interval $T_s$ and the measured $RTT$. For a chosen $r$ and including the initial transmission delay we obtain:

$$k(r) = \left\lfloor D_t - (\frac{r}{2} + r) \cdot RTT \right\rfloor$$

Tan [14] applies sequence analysis on the Gilbert-Elliot [18] model representing a two-state Markov chain in order to estimate the residual erasure rate of a $C(n,k)$ MDS code over a channel with erasure rate $P_e$ and correlation coefficient $\rho$ for subsequent state transitions. We choose the binomial distribution, which is equivalent to the Gilbert-Elliot model with zero correlation coefficient, to obtain the probability $P_{\text{dist}}(d,n)$ of $d$ packet erasures in a sequence of length $n$:

$$P_{\text{dist}}(d,n) = \binom{n}{d} P_e^d (1 - P_e)^{n-d}$$

Given the above channel model, the residual loss rate of a $C(n,k)$ MDS code corresponds to the expected number of unrecovered data
various fractions of the overall budgets. Partial solutions are forwarded overall parameter search incorporating the local channel quality under knowledge about the network quality. In a stepwise approach each programming such that the source is not required to maintain a global parameterization consists in the set of local AHEC parameters that certainly contribute to the overall redundancy sum: 

\[ P_d(n, k, i, j) = \binom{k}{i} \binom{n-k}{j} \]

\[ P_{res} = \frac{1}{k} \sum_{i=1}^{n-k} \sum_{j=max(0, n-k+i)}^{n-k+i} i \cdot P_d(n, k, i, j) \cdot P_{dist}(j, n) \]

Obviously, \( n \) has to be chosen sufficiently large to drop the residual erasure rate below the PLR constraint \( P_l \).

The redundancy information that finally appears on the channel depends on the distribution of the parity packets among the \( r \) transmission cycles. In cycle \( 1 \leq w \leq r \) the sender has sent \( \pi(w) = k + \sum_{j=1}^{w} \bar{N}_p[s] \) packets. The parity for cycle \( w \) is triggered by the receiver's notification about insufficient parity after the previous cycle, which happens with the residual erasure rate of round \( w - 1 \). Let \( R_w \) be a random variable indicating the number of packets aggregated at the receiver after cycle \( w \), which is less than \( k \) with probability:

\[ P_r(R_w < k) = \sum_{j=\pi(w)-k+1}^{\pi(w)} P(j, \pi(w)) \]

Since \( \bar{N}_p[1] \) parity packets are sent without receiver request, they certainly contribute to the overall redundancy sum:

\[ RI = \frac{1}{K} \left( \bar{N}_p[1] + \sum_{w=2}^{r} P_r(R_{w-1} < k) \cdot \bar{N}_p[w] \right) \]

An AHEC coding parameter set \((k, \bar{N}_p)\) might be optimized by iterative adjustment between \( k \) and \( r = dim(\bar{N}_p) \) while finding the optimal distribution of the parity among the available cycles. Besides this naive full search [14] proposes a greedy algorithm to solve the optimization problem.

C. Multi-hop Optimization

In the multi-hop usage the AHEC analysis is applied locally for each link. In order to provide end-to-end PRPD over heterogeneous networks, the AHEC stack is deployed individually to each link segment as an atomic protection unit. Each AHEC unit has to be configured with respect to the overall delay and reliability constraints, i.e. the local coding delays have to sum up to less than \( D_s \), whereas the individual residual loss rates combine according to a multiplicative term (cf. section III-C1). The globally optimum parameterization consists in the set of local AHEC parameters that minimizes the overall amount of redundancy on the entire path. This problem has been formulated as a Budget Distribution Problem (BDP) with respect to the application’s time and reliability budget, which can be reduced to a two-dimensional Multiple-Choice Knapsack Problem (MCKP) [19].

The MCKP allows for a distributed solution via dynamic programming such that the source is not required to maintain a global knowledge about the network quality. In a stepwise approach each intermediate node contributes a set of tentative, partial solutions to the overall parameter search incorporating the local channel quality under various fractions of the overall budgets. Partial solutions are forwarded along the network path towards the source. Finally, the source is able to solve the global optimization problem by combining partially optimum solutions according to Bellman’s principle of optimality. The source has to disseminate the link-individual AHEC parameters corresponding to the globally optimum solution.

D. 3D-IPTV Delivery Example

We believe that 3D content delivery is going to be the most bandwidth intensive application in the Future Internet. Hierarchical distribution, e.g. via MVC [20], is essential since network bandwidth and consumer preferences might require a backward compatible 2D representation. In our example we model the most general case where base and extension layer of the hierarchically coded 3D-IPTV signal origin from different sources, i.e. the delivery network is multi-hop as well as multi-path. The setup reflects a nearly reliable first segment (e.g. unmanaged wired Internet) connected to a lossy but fast second segment (e.g. in-home wireless LAN).

In order to fulfill the requirements of IPTV, the overall time constraint is set to 100 ms at a residual packet loss rate of not more than \( 2 \cdot 10^{-6} \). According to equation 3, in a two-hop decode and forward policy, each segment has to keep a residual loss rate of \( P_l < 10^{-6} \). In figure 6 the upper link (S1+S2) is assumed to deliver a 10 Mbit/s base layer HDTV stream whereas the lower segments (S3+S4) contribute a 4 Mbit/s MVC enhancement view.

In table I we provide the analytical RI values required by the optimal parameter set of the PRPD coding unit in either end-to-end or dual-hop configuration with respect to application constraints and channel state. For comparison we added the corresponding theoretical optimum RI under loss tolerance (equation 2), Whereas optimal end-to-end AHEC parameter sets require more than twice the theoretical end-to-end RI, multi-hop configurations perform very closely to the theoretical optimum at the individual segments. Note that the RI in a practical setup will be slightly greater than the analytical value since it depends on the header and message overhead as well as on the error detection method of the actual protocol implementation.

<table>
<thead>
<tr>
<th>Source 1</th>
<th>Source 2</th>
<th>S3</th>
<th>S4</th>
</tr>
</thead>
<tbody>
<tr>
<td>50ms / 0.1%</td>
<td>5ms / 8%</td>
<td>70ms / 0.03%</td>
<td></td>
</tr>
<tr>
<td>S1</td>
<td>S2</td>
<td>S3</td>
<td>S4</td>
</tr>
<tr>
<td>S1+S2</td>
<td>S3+S4</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 1

END-TO-END VS. MULTI-HOP REDUNDANCY INFORMATION

V. Conclusion

Optimizing the transport of streaming media has been shown to have significant effect in scenarios like 3D-IPTV delivery. Including time and loss tolerance into the design of transport protocols may deeply influence the current research in a variety of fields:
multipath-routing, network coding, multicast group management etc.. Considering the fact that most streaming media are loss tolerant, a virtual erasure channel has been introduced for calculating the Shannon-bound for loss tolerant applications. The simplified model presented herein, however does not cover the whole parameter space. An existing algorithm for optimizing an end-to-end media transport has been applied to a general hierarchically coded IPTV delivery scenario. Thus, in this paper we have shown how an optimized PRPD AHEC coding unit for streaming media can be applied to multiple network segments individually, separating loss domains. The required overhead is reduced while the applications’ delivery time requirements are met. Future work is required on optimally distributing the delivery time budget as well as the residual error rates amongst arbitrarily interconnected delivery segments.

REFERENCES