# SVC-BASED MULTISOURCE STREAMING FOR ROBUST VIDEO TRANSMISSION IN MOBILE AD HOC NETWORKS

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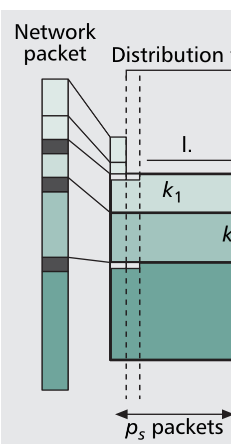


The authors present a multisource streaming approach to increase the robustness of real-time video transmission in MANETs. For that, video coding as well as channel coding techniques on the application layer are introduced.

## ABSTRACT

Emerging noninfrastructure-based network types like Mobile Ad-Hoc Networks (MANETs) are becoming suitable platforms for exchanging/sharing real-time video streams, because of recent progress in routing algorithms, throughput and transmission bit-rate. MANETs are characterized by highly dynamic behavior of the transmission routes and path outage probabilities. In this article a multisource streaming approach is presented to increase the robustness of real-time video transmission in MANETs. For that, video coding as well as channel coding techniques on the application layer are introduced, exploiting the multisource representation of the transferred media. Source coding is based on the scalable video coding (SVC) extension of H.264/MPEG4-AVC with different layers for assigning importance for transmission. Channel coding is based on a novel unequal packet loss protection (UPLP) scheme, which is based on Raptor forward error correction (FEC) codes. While in the presented approach, the reception of a single stream guarantees base quality only, the combined reception enables playback of video at full quality and/or lower error rates. Furthermore, an application layer protocol is introduced for supporting peer-to-peer based multisource streaming in MANETs.

## INTRODUCTION

Network types like wireless networks based on the Wireless LAN (WLAN) 802.11a, 802.11b, 802.11g specifications are becoming more popular and suitable for real-time transmission. These network types allow very high data-rates, but usually only if the distance between receivers and senders is sufficiently small. Longer ranges can, for example, be achieved by multihop transmission: In contrast to the inefficient case where many nodes are available in a certain area with a plain access point or hot-spot-based network structure, the ad hoc mode of WLAN nodes can increase coverage and may save costs for infrastructures. Mobile Ad Hoc Networks (MANETs) [1] make use of all nodes in a mobile network as routers in order to build up short-time dynamic network infrastructure. An example MANET is shown in Fig. 1 with one client and different established routes (dashed lines) to multiple server nodes.

The dynamic behavior of MANETs adds some challenges to high-quality video transmission: The high path-outage probability makes it impossible to reliably transmit a video stream by simply applying known techniques from the wired or cellular transmission environment. Therefore, this work proposes a robust multi-source video streaming protocol for reliable real-time video streaming, which mainly solves the route-loss problem in case of real-time streaming over MANETs. The basic approach is to enhance reliability by using different sources at the same time with different, independent representations of the media layers, resulting in a network multiple description approach. To realize this property the video stream is divided into different layers using SVC [2, 3]. SVC provides layers with different importance for the video reconstruction and different percentage of the complete stream bit-rate. On top of a novel unequal packet-loss protection (UPLP) scheme based on recently proposed Raptor forward error correction (FEC) codes [4] is used for protecting different layers with different importance.

The fountain property of Raptor codes enables the generation of a virtually infinite amount of independent encoding symbols (ESs) from a certain number of source symbols (SSs). Transmitting these ESs intelligently over different paths using different sources, strongly

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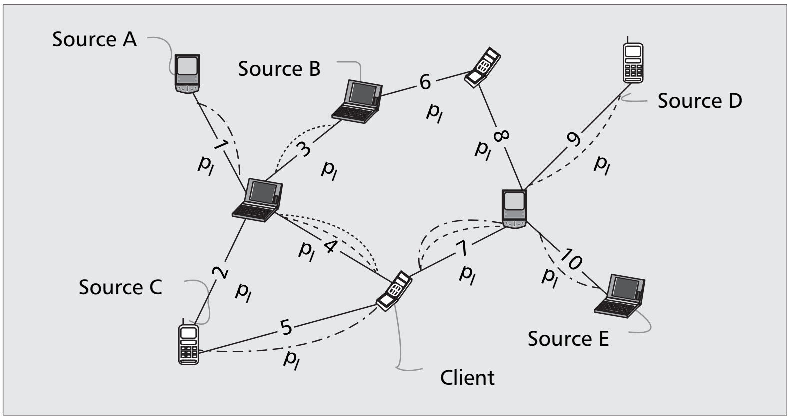
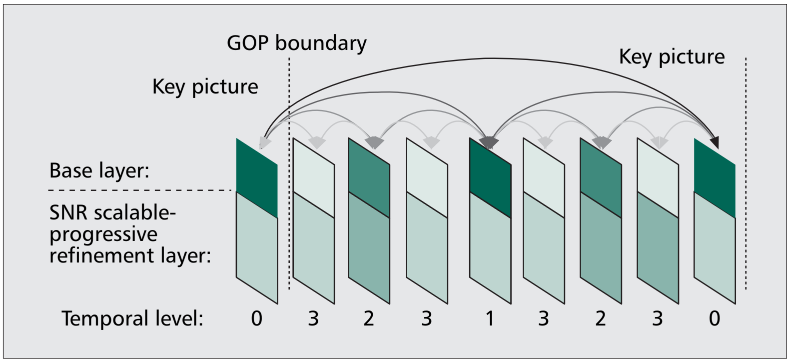
enhances the reliability of streaming sessions in MANETs. For the proposed scheme, the servers do not need to be connected in any way, since a random scheme provides the generation of independent ESs per source node.

This article is related to peer-to-peer networks approaches like Avalanche [5], which is developed for file-sharing download services and also comprises multisource approaches, whereas our approach is targeting real-time streaming with the addition of being deployed in MANETs. Previous work in this area for wired Internet has for example been presented in [6], but our approach is targeting independently decodeable network streams.




■ **Figure 1.** *Mobile ad-hoc network.*




■ **Figure 2.** *Temporal structure of an SVC stream including PR.*

## Multisource Streaming Components

This section provides a brief overview of the employed components and their relationship to our proposed multisource streaming approach, which is presented subsequently.

### Real-Time Media Delivery in MANETs

In MANETs, each node operates either as a component of the mesh, as a duplex mobile node (forwarding), as a regular data-consuming node, or as a data-generating node, as well as a router. Especially, the router functionality is the main challenge and one of the major differences as compared to wired networks. Therefore, the issue of continuous routing has to be solved. In such a distributed system, each node does not have the full knowledge of the complete network's topology, that is, routing decisions made by nodes are in general suboptimal.

One of the key characteristics of MANETs is the excessive time-variant behavior due to mobility of the participating nodes and unreliable wireless transmission conditions. An aspect of time variance is the sporadic participation of individual mobile nodes in the network. Therefore, it is quite difficult to set up and maintain the shortest route to a specific destination within the MANET. Route changes as well as route break-downs occur quite frequently.

MANET routing algorithms can be classified into two categories: proactive and reactive [1]. In this work we exclusively rely on reactive routing algorithms, which initiate a routing query only in case a packet is to be transmitted to a destination for which it has no active entry in the routing table. Reactive algorithms reduce the routing overhead because of requesting routes only if required, but this strategy might also add some delay. Examples for reactive routing algorithms are ad hoc distance vector (AODV) [7] and dynamic MANET on-demand (DYMO) [8].

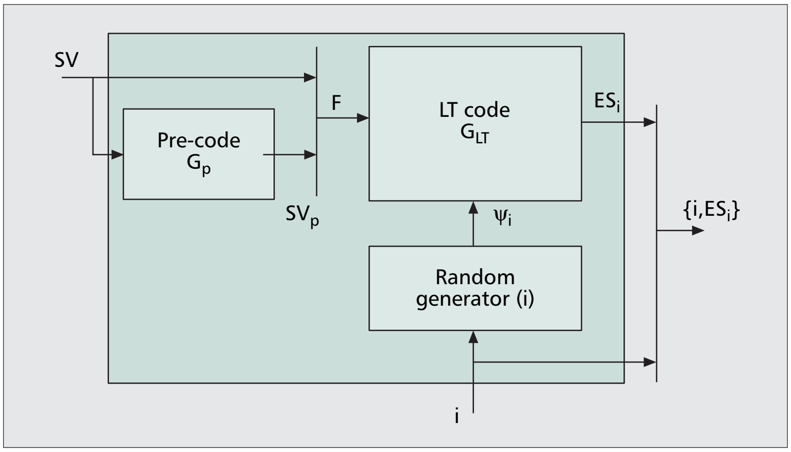
Figure 1 shows a mobile multihop ad hoc network in a client-server setup. Different established routes are indicated by dashed lines. $p_l$ gives the average link-loss probability per intermediate link for the ad hoc network. Further details are discussed as follows.

### Scalable Video Coding (SVC)

SVC [2, 3] is an extension to the H.264/MPEG4-AVC [9] video coding standard. The basic idea of SVC is to extend the hybrid video coding approach in a way that a wide range of spatio-temporal and quality scalability is achieved. Scalability within SVC is a functionality that allows the removal of parts of the bit-stream while achieving a reasonable coding efficiency of the decoded video at reduced temporal, SNR, or spatial resolution. An SVC bit-stream consists of a base layer and one or several nested enhancement layers. The base layer is a plain H.264/MPEG4-AVC bit-stream, ensuring backward compatibility to existing receivers.

The temporal scaling functionality of SVC is typically based on a temporal decomposition using hierarchical B pictures, as shown in Fig. 2. Each B picture of a higher temporal enhancement level is encoded with a higher quantization parameter (QP) (cascaded QP assignment), thus the fidelity per picture is decreasing with the decreasing importance in terms of the number of succeeding references by other pictures. In Fig. 2 shows the structure of a SVC stream, which comprises a group of pictures (GOP) of size eight. GOPs can be independently decoded, if the preceding key picture is available and has random access properties.

Spatial scalability is achieved by different encoder loops with an oversampled pyramid for each resolution (e.g. QCIF, CIF, and 4CIF), including motion-compensated transform coding with independent prediction structures for each layer. In contrast to the encoder, the decoder can be operated in single loop, that is, for decoding interlayer dependencies it does not need to perform motion compensation in the lower layers on which it depends.




■ **Figure 3.** *Non-systematic raptor encoder.*

SNR scalability is based on a progressive refinement (PR) approach, where the extension layers contain refinement quality information of the base layer in a progressive way. Thus, cutting byte-wise from the end of a PR fragment is possible. A PR layer only contains refinements for the residual (texture) data, which is also used for prediction in next higher temporal levels. By coding and organizing the progressive refinement information in a cyclic and prioritized way for a video frame, the truncation property is realized. Up to three PR layers can exist in a stream. Typically, the quality of each layer is enhanced by a QP delta value of six. Finally, each layer of each slice is stored in a separate network abstraction-layer unit, which can be transported individually. More details are discussed below.

Within this work SNR scalability based on PR is used for differentiation between video layers. A certain bit-rate point is forming an operation point of the SNR scalable bit-stream.

### Raptor Forward Error Correction Codes

The Raptor code [4] (Fig. 3) is an error correction code mainly used in environments with packet losses. Furthermore, it is a fountain code, that is, a virtually infinite amount of encoding symbols can be produced from a vector of source symbols $SV$ of the length $k$. In average, the decoder is capable of reconstructing the source symbols from a number of $ES$ that is only slightly higher than the original length of the $SV$. For sufficiently long $k$, the Raptor code operates very closely to an ideal fountain code which would require only any $k$ encoding symbols for successful reconstruction.

The Raptor code can be viewed as a serial concatenation of a pre-code and a Luby Transform (LT) code. The inner LT code [4] is a realization of a fountain code. From an input vector $F$ an infinite number of encoding symbols $ES_i$ can be produced by XORing symbols in $F$ as indicated in vector $\mathbf{\Psi_i}$, (i.e., the vector $\mathbf{\Psi_i}$ consists of all 0s except for 1s at the positions to be XORed). The outer systematic fixed rate code $\mathbf{G_p}$ is the key for improved performance of Raptor codes over LT codes. It is important that the vectors $\mathbf{\Psi_i}$ are generated according to a random distribution whereby the degree distribution and the precode need to be appropriately selected for optimized performance.

For the Raptor decoder, a system of linear equations has to be solved to reconstruct the $k$ source symbols. More precisely, it is sufficient to reconstruct the vector F of intermediate symbols of length $k + s$, as the first $k$ symbols of $F$ correspond to the source symbols. Note that a symbol consists of $T$ bytes, with $\mathbf{1}_s$ being the identity matrix of size $s$, the received symbols vector $[ES_{i1},\ldots, ES_{ir}]$ having length $r$, and $\mathbf{G_{LT}}(i_1..i_r)$, the corresponding generator matrix, containing the vectors $\mathbf{\Psi_i}$ as the rows. For successful decoding, the following equation needs to be solved:

$$\begin{bmatrix} \mathbf{G}_p & 1_s \\ \mathbf{G}_{LT}(i_l..i_k) \end{bmatrix} \cdot \mathbf{F} = \begin{bmatrix} \mathbf{0}_S^T \, ES_{i_1}^T \, ..ES_{i_k}^T \end{bmatrix} \quad (1)$$

In general, the equation can be solved for $r$ being $k$ or only slightly larger. To obtain $\mathbf{G_{LT}}(i_1, \ldots, i_r)$ with each symbol $ES_i$, the symbols index $i$ to generate $\mathbf{\Psi_i}$ needs to be available at the receiver.

A systematic version of Raptor codes has been adopted by 3GPP [10]. For the systematic version of this code, an additional precoding matrix is used on the $SV$ of the length $k$ to ensure, that the first $k$ of $n$ $ES$s are equal to the $SV$. Within this work only statistical results produced by reference implementation for [10] are used for emulating the Raptor encoding and decoding process. A detailed description of systematic Raptor codes can be found in [4].

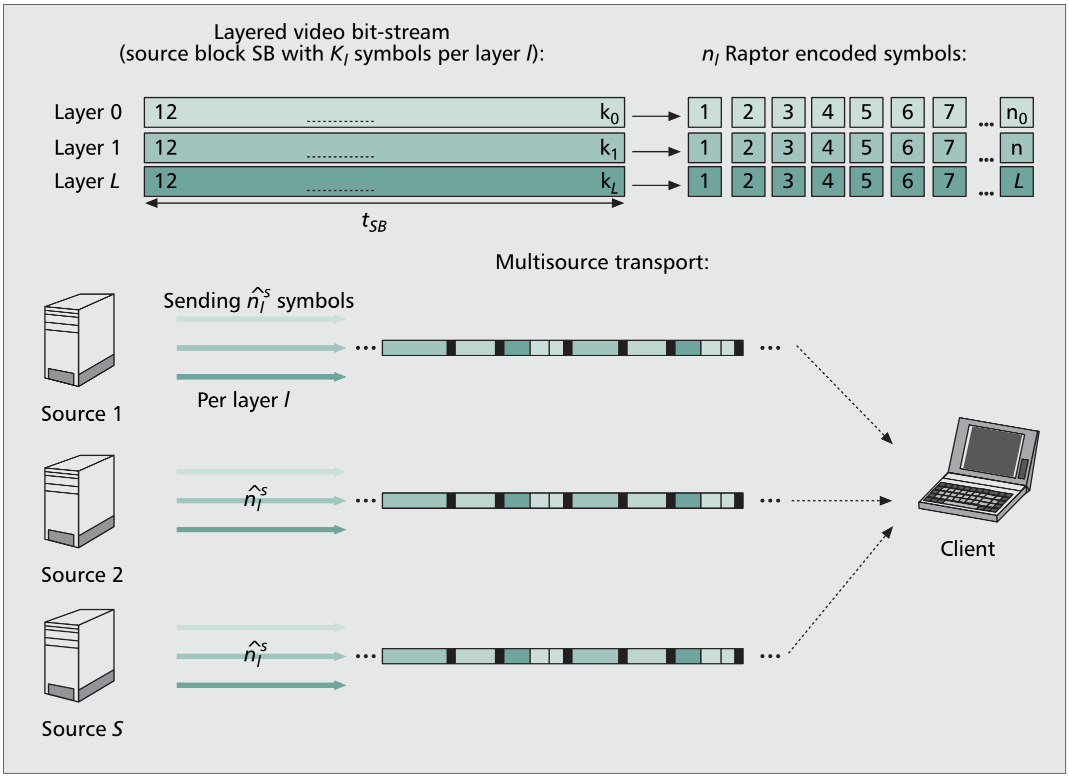
## Multisource Streaming in MANETs

This section introduces the proposed multisource streaming approach by appropriately combining the components introduced earlier. The basic idea is to use multiple sources for guaranteeing reliability by redundancy in sources as well as in network paths. Nevertheless the selected network paths are mainly determined by the underlying MANET routing protocol. The proposed approach of this work exclusively relies on application layer techniques and does not need any special cross-layer interfaces.

### Media and Channel Coding

By combining layered video coding with a Raptor code-based UPLP, a distribution of linearly independent representations of the video stream across servers can be achieved. Figure 4 illustrates the behavior of the proposed multisource streaming approach using the Raptor code for generating $S$ linear independent network streams from $L$ layers from $S$ sources.

***Source/Sender Behavior*** — The idea is that each source generates linear independent Raptor ESs for data corresponding to a certain amount of media time $t_{SB}$. This data, referred to as source block (SB), is divided in a certain number of symbols of a of each media layer $l$, that is, each source $s$ generates and emits $\hat{n}_l^s$ ESs from $k_l$ source symbols of layer $l$ for a media time interval $t_{SB}$. That means a source $s$ sends the sum of all $\hat{n}_l^s$ symbols per layer $l$ and per interval $t_{SB}$.




■ **Figure 4.** *Multi-source transport of layered media.*

With a symbol size $T_l$ in bytes for layer $l$, we obtain the transmission rate $\hat{r}_{tl}^s$ for layer $l$ at source $s$ in terms of bytes per second as follows:

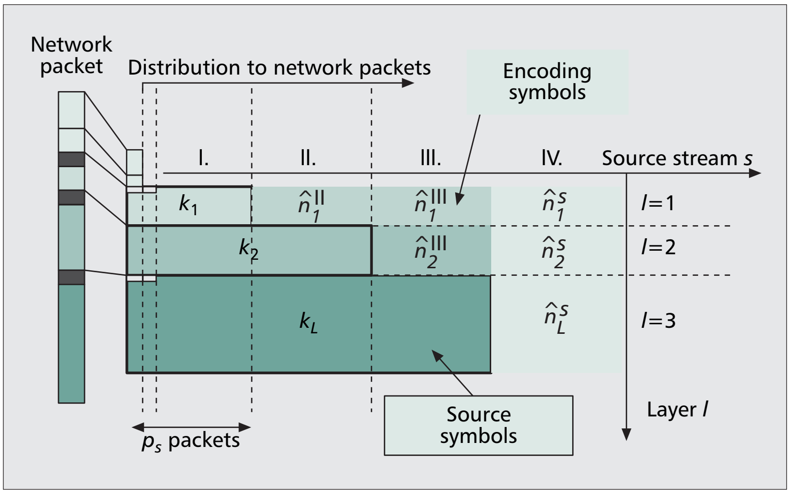
$$\hat{r}_{tl}^s = \frac{\hat{n}_l^s \cdot T_l}{t_{\text{SB}}} \tag{2}$$

Assuming constrained overall transmission rate $\hat{R}_t^s$ for source $s$, the sum over all $L$ layers of transmitted symbols $\hat{r}_{tl}^s$ within time interval $t_{\text{SB}}$ must be equal or smaller than $\hat{R}_t^s$.

***Sink/Client Behavior*** — Assume now that a client receives for a time interval $t_{\text{SB}}$ a number of $\tilde{N}_l^s$ linearly independent symbols from $S$ sources for some layer $l$. If the number of received symbols $\tilde{N}_l^s$ is equal to or slightly higher than $k_l$, decoding will be successful. Assuming an average, layer-independent loss rate $\lambda_s$ per source $s$, the expected number of received symbols for layer $l$ is the sum $\tilde{N}_l^s$ of $(1 - \lambda_s)\hat{n}_l^s$ received symbols over all sources $S$. That means a client can influence the number of received symbols per layer, by selecting the number $S$ of sources. All other parameters are basically determined by the transmission scheme.

***Parameter Options and Selection for Sender and UPLP*** — At this point, we still need to select transmission parameters. In this work, we apply some heuristic settings that provide reasonable results rather than looking for detailed optimization which is subject of ongoing work.

At first, the characteristic of the layered media is defined in terms of byte-rate $r_l$ for each layer $l$. With a constant media rate per SB, the media rate $R_L$ is the sum over all $L$ layers of layer byte-rates:




■ **Figure 5.** *Symbol distribution in an encoding block (EB).*

$$r_l = \frac{1}{t_{SB}} k_l \cdot T_l$$

(compare Eq. 2).

The symbol arrangement for the UPLP approach per source is according to the example shown in Fig. 5. The interleaved combination of $S$ source streams is forming the encoding block (EB), which contains the ESs representing the media for each source block lasting some time $t_{SB}$.

The quotient of source symbols $k_l$ and encoding symbols $\hat{n}_l^s$ is given by the code-rate $r_{cl}$ for a layer $l$ and source $s$.

For simplification we assume that the stream-

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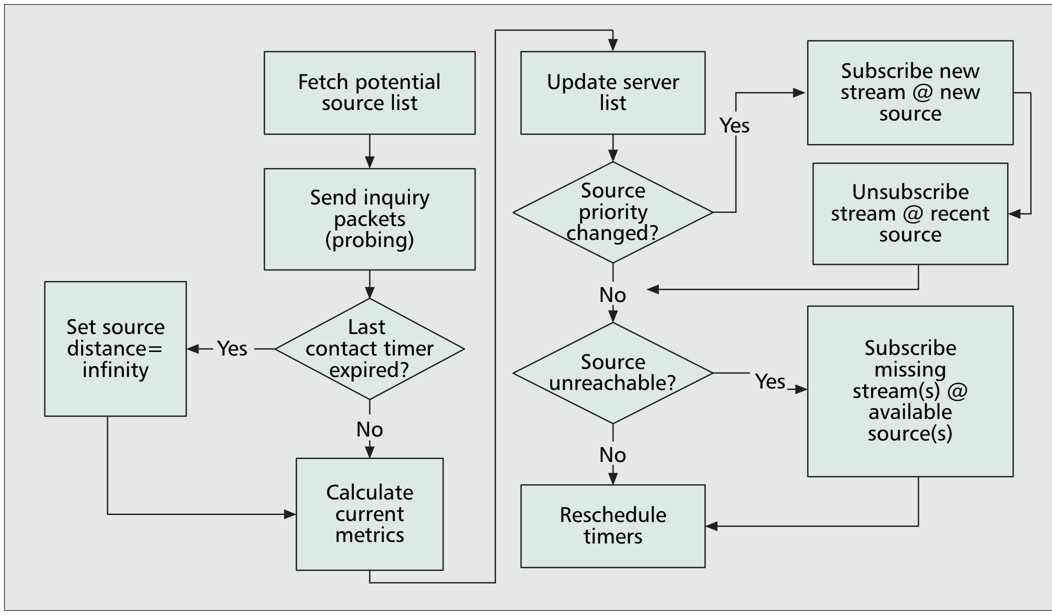



■ **Figure 6.** *Simplified client scheme for frequent server evaluation.*

ing server transmission rate $\hat{R}^s_{tl}$ is fixed for all $S$ sources and, with that, also the number of transmitted symbols. Proportionally to the code rates $r_{cl}$ per layer $l$, the symbols $\hat{n}^s_l$ are distributed into UPLP packets, as shown in Fig. 5.

A reasonable setting for code rates $r_{cl}$ is proposed as follows:

$$r_{cl} = \frac{L-l+1}{L} \tag{3}$$

With that constraint, it is guaranteed that with each additional received source stream $s + 1$ the next layer $l + 1$ becomes decodable at the client with very high probability and the rationales of applying decreasing code rates for a more important layer with lower layer indices $l$ is employed. With the constraints mentioned above, the proposed transmission rate (not considering packet loss) per source $s$ is given as the sum of products $(r_{cl} \cdot r_l)$ over all $L$ layers.

The overall multisource coded stream rate as a fraction of $R_L$ (compared to the rate of the original layered media) resulting from this approach is mainly determined by the rate proportions of the layered media itself. The overhead fraction $x$ is shown in Eq. 4. Optimizations will follow in later work based on [6].

$$x = \frac{1}{R_L}\sum_{l=1}^{L}(L-l)r_l \tag{4}$$

The source block size $k_l$ and symbol size $T_l$ for each layer $l$ is chosen according to the constraints in [4, 10], such that the Raptor coding process for each layer $l$ can be assumed as being almost identical to ideal coding, that is, decoding is successful if the number of source symbols per layer $l$ used for encoding is equal or higher than the minimal number of symbols $k_{min}$ specified in [10].

A key feature of the approach is to guarantee that the different sources are using different random seeds $i$ for generating $\Psi_{\mathbf{i}}$ for the Raptor encoding process, thus ensuring the generation of independent ESs for each network/source stream. This allows a huge number of sources for generating equivalent but linear independent network streams. Therefore, the sources do not need coordination. A source is selecting randomly a value from a set $I$ of valid values of $i$ for Raptor encoding. $I$ is known to all encoders and decoders and should be of reasonable size guaranteeing independence of ES with high probability.

## Application Layer (APP) Protocol for Multisource Media Delivery

In this subsection a protocol for requesting media and connection-quality evaluation is presented.

The basic idea of the multisource media coding approach and protocol is to reduce failures in video transmission caused by route losses on the transmission path. Route losses are the main problem when comparing ad hoc networks with networks using fixed infrastructures.

If assuming an average link-loss probability of $p_l$ per intermediate node in an ad hoc network (Fig. 1), the resulting route-loss probability for a transmission path going via $M$ intermediate links is given as $P_r = 1 - (1 - p_i)^M$.

The proposed concept is to increase the number of used sources for enhancing reliability in server availability while keeping the overall used network transmission rate/bandwidth as small as possible. We assume that nodes are not running in congestion state at any time, that is, the transmission rate at an intermediate, source, or client node is not higher than the available transmission rate provided by the air interface.

For simplification, we assume further the overall probability of having a route from at least one source $s$ out of $S$ sources to the client being $P_c$ with having independent network paths per source and having the same loss probability $P_r$ per source route. $P_c$ is given by

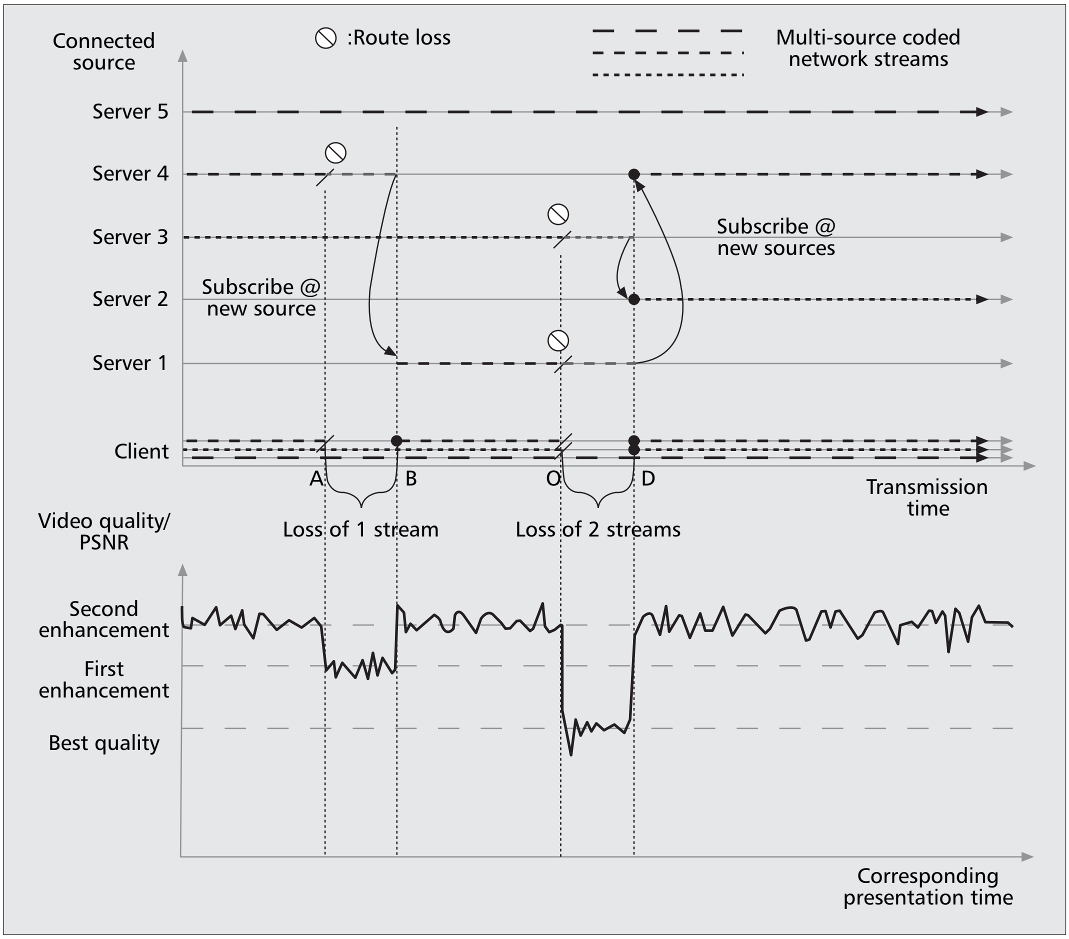



**Figure 7.** *Stream management with resulting layered video quality.*

The multi-source coded network streams are requested from (subscribed at) nodes with best the metric. If metric information changes for the access able nodes or if additional sources become access able during transmission, the subscription process of streams at source nodes is initiated.

$$P_c = \sum_{s=1}^{S} \binom{S}{s} (1-P_r)^S (P_r)^{S-s} = 1-(P_r)^S \qquad (5)$$

The equation for $P_c$ gives a motivation for increasing the number of sources for enhancing the probability of source availability (i.e., having at least one intact route between client and source nodes). The multisource coding approach proposed above allows for receiving multiple streams for increasing media play-out quality, while maintaining base quality if receiving exactly one stream.

The protocol for source monitoring and selection probes available sources cyclically. The assumption is that the addresses of source nodes available in the ad hoc network area are introduced by an external instance, which is not considered within this work.

The monitoring of sources is achieved by sending probing packets (*inquiry packets*) to all known sources for collecting path quality information per source. To keep a consistent status about the network condition and accessible sources, sources are probed continuously during media transmission. All required information for link-quality evaluation is transferred on the application layer via RTCP packets and as header extension to RTP packets.

The link/route quality information collected by the inquiry process is called metric information. By using the metric information the decision on source selection is made. The multi-source coded network streams are requested from (*subscribed at*) nodes with best the metric. If metric information changes for the access able nodes or if additional sources become access able during transmission, the subscription process of streams at source nodes is initiated.

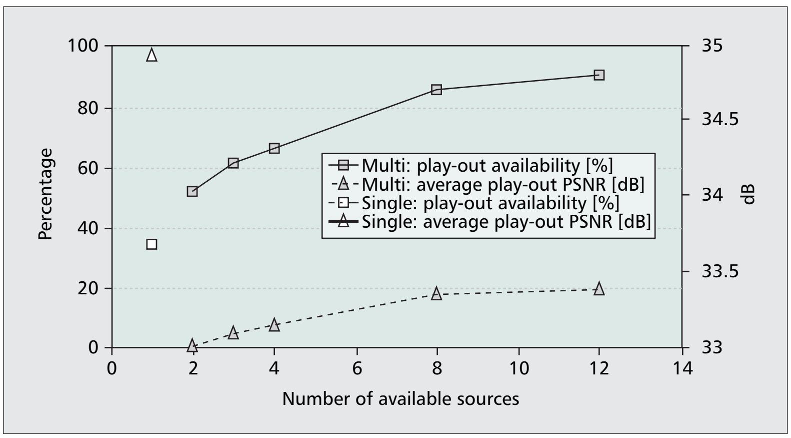
The metric used is the distance from client to the source node in terms of hops. This metric is motivated by equation for $P_r$ while assuming an average link loss probability per intermediate node, that is, the more nodes are used within a path, the higher the probability is that the route between source and client node may break down. This assumption is, in general, a simplification, since in random waypoint movement scenarios the equation for $P_r$ will not comprehensively cover the link characteristics, but gives a simplified view on the route properties in ad hoc Networks.

In Fig. 6 the basic control scheme of the protocol is shown. The protocol has been implemented and executed in ns-2.

Figure 7 depicts the basic stream management of the protocol and the resulting video presentation quality as peak signal-to-noise ratio [11] in terms of dB. It is shown how the protocol reacts on detected link losses. Figure 7 further shows the results of resubscribing streams at different/new source nodes (*source switching*). For

| Operation points: | Bit-rate [kb/s] | PSNR (dB) |
|---|---|---|
| 1 | 93.03 | 27.55 |
| 2 | 173.01 | 30.76 |
| 3 | 318.54 | 33.65 |
| Single layer | 313.90 | 34.94 |

■ **Table 1.** *Operation points of SVC/single-layer media stream.*




■ **Figure 8.** *Average results for single-source and multisource modes.*

simplification, buffering and transmission delays are not considered in Fig. 7. In the shown example, five server nodes are available.

## Simulation Results

All network simulations have been carried out with ns-2. The video streams have been encoded with version JSVM5 of the SVC reference software. A repeated Paris sequence (288 frames) with 8640 frames (285 sec), 30 frames per second, CIF resolution, GOP size 32, and random access property is used as version with three SNR scalable operation points and as a single-layer version, as shown in Table 1.

Multisource coded network streams with three protection classes are generated and at maximum three streams are received simultaneously. Each of the streams contains the full base-layer, thus a base quality can be played out by receiving only one of the network streams (Fig. 5). Each protection class has been encoded within an emulated and independent Raptor encoding process; thus, the resulting streams are decodable independently. For Raptor encoding, we used the 3GPP-recommended preconditions [10]. A prebuffering for network jitter compensation of 5 s is assumed. DYMO [8] is used as underlying routing protocol in combination with an IEEE 802.11b adapter. In an area of 1000 × 600 m, 40 nodes are moving in random waypoint patterns in the presence of 1 client and 1, 2, 3, 4, 8, and 12 available and randomly selected source nodes. Each simulation per value of available source nodes is repeated 60 times in independent random waypoint scenarios, which is an overall simulation time of 4.75 h per value of available source nodes. The average FEC stream rate is approximately 594 kb/s. Due to packetization overhead, the effective FEC protection rate is 84.19 percent for the multisource coded stream and 86.30 percent for the single-layer stream.

Figure 8 gives the overall results in terms of playout quality for the conducted experiments. The average PSNR as indicator for video playout quality is measured only for simulation time with available playout (frame freezes are not included). SNR scalability (multisource case) is predominately used in case of switching sources, thus only minor scalability can be denoted within the average results. If not enough sources for all network streams are available, more than one network stream is requested from a certain source in order to maintain high playout quality. For more SNR scalability results we refer to [12]. A reduced quality of the video must be generally accepted if using scalability, but scalability further allows for graceful degradation in case of network errors and throughput problems.

The more important indicator is the available playout time in terms of percentage of simulation time. This indicator represents the ability of playing out media without interrupts. The results obviously show that using multiple sources drastically enhances the ability of frequent media playout. Interruptions in playout can be reduced. A higher number of sources increase the playout availability.

## Conclusion

We have presented an extended unequal packet-loss protection (UPLP) scheme based on Raptor FEC using different sources for reliable media streaming in MANETs. We showed the benefits of using linear independent FEC streams with unequal loss protection for multisource streaming in scenarios with high route-loss probability, as is present in MANETs. This approach has been evaluated with a new protocol for media tracking and delivery in MANETs, which exclusively relies on applicationlayer techniques.

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## BIOGRAPHIES

THOMAS SCHIERL (schierl@hhi.fhg.de) received a Dipl.-Ing. degree in technical computer science from the Berlin University of Technology, Germany in December 2003. He has been with Fraunhofer HHI since 2003. He has done different research works on reliable real-time transmission of H.264/MPEG-4 AVC and scalable video coding (SVC) in mobile point-to-point, multicast, and broadcast environments. He has submitted different inputs on real-time streaming to standardization committees like 3GPP, ISMA, MPEG, and IETF. His current work mainly focuses on developing new real-time streaming techniques for Mobile Ad Hoc Networks. Further research interests are reliable transmission of real-time media in mobile networks and joint source channel coding.

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