A Multimedia Delivery System for Delay-/Disruption-Tolerant Networks

Christian Raffelsberger, Hermann Hellwagner
Institute of Information Technology
Alpen-Adria-Universität Klagenfurt
Klagenfurt, Austria
{christian.raffelsberger, hermann.hellwagner}@aau.at

***Abstract*—Multimedia delivery systems and protocols usually assume end-to-end connections and low delivery delays between multimedia sources and consumers. However, neither of these two properties can always be achieved in hastily formed networks for emergency response operations. In particular, disruptions may break end-to-end connections, which makes it impossible to deliver multimedia content instantly. This work presents a multimedia delivery system that can operate in disrupted networks and hence may help improve the situational awareness in emergency response operations. The multimedia delivery system is based on HTTP adaptive streaming (HAS) and uses a modified version of HTTP which is able to deliver data in partitioned networks. The multimedia delivery system is evaluated in a realistic emergency response scenario.**

I. Introduction

Current mobile devices have the processing power as well as the storage and wireless communication capabilities to perform many everyday computing tasks. Thus, such mobile devices may also be used by first responders in emergency response scenarios [1], [2]. One challenge of this application domain is that the networking infrastructure may not be available since it has been destroyed by the disaster itself or gets overloaded in the aftermath of a disaster. Furthermore, mobile ad-hoc networks (MANETs) that are created by first responders to replace fixed infrastructure may get partitioned [3]. Hence, it is important that routing algorithms and applications can adapt to the underlying network conditions and also take periods of disconnection into account.

One promising application of mobile computing in emergency response scenarios is the recording of videos that can be delivered to the incident command center in order to improve the situational awareness. Figure 1 shows an example scenario. Modern mobile devices allow first responders to record audiovisual content at the disaster scene and distribute them via wireless ad-hoc networks. Since the network may be partitioned, the mobility of first responders on the incident scene needs to be exploited in order to deliver the videos. Hence, a multimedia delivery system has to take such situations into account and provide appropriate delivery mechanisms. This work introduces such a disruption-tolerant mobile multimedia delivery system that can cope with the challenges that are imposed by emergency response operations. The system provides multimedia delivery in MANETs that provide end-to-end connections, as well as in delay-/disruption-tolerant networks (DTNs) where not all nodes are connected via end-to-end connections.

The remainder of the paper is structured as follows. Section II describes how multimedia services can be used in emergency response operations and state-of-the-art multimedia delivery protocols. Section III presents the design of our multimedia delivery system for delay-/disruption-tolerant networks (DT-MDS). The evaluation scenario and simulation setup are described in Section IV. Evaluation results are presented in Section V. Section VI presents related work, before Section VII concludes the paper and discusses possible future work.

II. Background

A. Multimedia Usage in Emergency Response

Studies with practitioners by Landgren et al. [4] have shown that mobile video can improve emergency response operations. In particular, the following use cases have been identified. One use case is to record videos while rescue units drive to the incident scene in order to provide on-route traffic situation updates. Another use of mobile video is for enhancing situation reporting on the incident scene. The third use for video recordings is for after-incident documentation where videos are useful to replay key phases of the response work. In this work we focus on the second use case.

There have also been studies that analyzed the consumption of videos during an emergency response. Bergstrand et al. [5] differentiate several patterns of use based on a temporal dimension. The first type of use is the *live use* where videos are consumed while they are broadcast. The *near live use* describes cases where a video is consumed a few minutes after it has been recorded, for instance, to check if the right amount of resources has been dispatched. The third use case is the *scheduled use* where videos are watched in scheduled meetings or conferences during the incident. Finally, the *post incident use* describes that videos can be beneficial after the incident, e.g., for completing incident reports or for providing additional information for investigators. Examples for all but the live use case could be found during a study of real-world emergency response operations [5]. The main reason for the lack of live use is that the involved persons usually have more important, time-critical tasks to perform which prevent them to immediately watch a live video stream.

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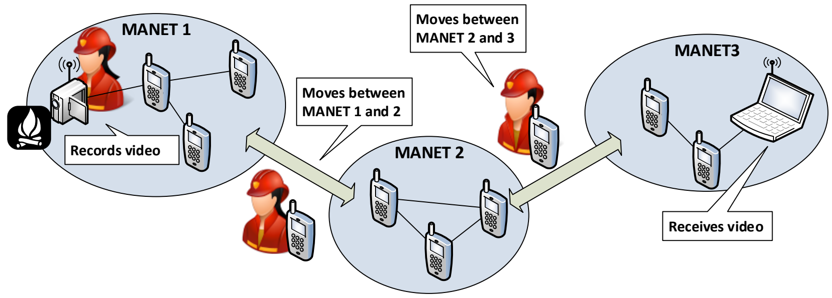

Fig. 1. Video delivery use case

From these studies two important conclusions can be drawn. First, video use for emergency response operations is delay-tolerant, which means that videos are usually not consumed while they are broadcast but used as a form of asynchronous communication [5]. Second, the videos usually have rather short durations (usually less than a minute and up to a few minutes in some cases) [4]. This is due to the fact that first responders neither have the time to record nor to consume long videos. Instead, professional first responders selectively record only important scenes which they think are useful for improving the situational awareness at the command center.

B. Multimedia Delivery Protocols

Traditionally, the Realtime Transport Protocol (RTP) has been considered as suited best for multimedia delivery. Usually, RTP runs on top of UDP since its simplicity and low delivery delays have been seen as beneficial over TCP's reliability. However, in recent years there has been a paradigm change and multimedia delivery is now often performed via HTTP.

HTTP adaptive streaming (HAS) is a technique that supports downloading multimedia via standard HTTP but also supports reacting to situations where the available resources (e.g., bandwidth) do not meet the needed resources. To provide this adaptability, HAS divides the multimedia content into segments which are downloaded separately. Segments are made available in different representations that can differ in resolution, frame rate and/or encoding quality and hence have different bit rates. The available representations are described in manifest files. Clients can select which representation to download based on the current context (screen size, current bandwidth, etc.) and also switch between different representations if the context changes. Usually, such switches are caused by variations of the available download bandwidth.

We believe that HAS-based multimedia delivery is very suitable for DTN scenarios. In particular, it offers several advantages compared to other protocols (such as RTP) in DTN scenarios. First, in HAS the multimedia content is separated into segments that contain video and/or audio data with a certain duration (e.g., two seconds). Every segment is a semantically meaningful unit for a decoder (i.e., it can be decoded right after delivery). Thus, segments fit the notion of DTN bundles [6] which should encapsulate merged application data that proceed the application state. Another advantage of HAS is that the segments and the representation descriptions (i.e., the manifest files) can be reused in a DTN scenario. Hence, the same content can be streamed without any modifications in traditional end-to-end scenarios as well as in DTN scenarios. Another advantage of HTTP for multimedia delivery in DTNs is that HTTP utilizes the entire available bandwidth for transmitting data, while RTP usually adjusts the sending rate to the media bit rate. This behavior of HTTP is beneficial in networks with intermittent connectivity, where links are not always available or their performance changes dramatically over time. Hence, it is important to exchange as much data as possible if two nodes are in contact. Another disadvantage of RTP-based delivery is its control overhead and complexity. In particular, RTP assumes a rather complex session setup compared to HAS and an RTP-based delivery system constantly exchanges information about the multimedia session between the server and its clients in the form of RTCP sender and receiver reports. Although this provides an accurate way of controlling the quality of service of the multimedia session, it is hard to ensure that these periodic reports are delivered on time in a DTN environment. This may prevent to establish a well-working control loop. Compared to that, HAS has no complex control loops involved and usually the client decides which representation to download based on information that is locally available (e.g., current bandwidth or buffer fill state). However, this adaptation decision could also be performed at the server or in the network.

An open issue of HAS in DTNs is that it is usually not possible to establish an end-to-end TCP connection between sender and receiver. However, HTTP itself does not presume such an end-to-end connection and thus it is possible to use HTTP in DTNs. The next section describes how HTTP and in particular an HAS-based multimedia delivery system can work in delay-/disruption-tolerant networks.

III. Disruption-Tolerant Multimedia Delivery System

The Disruption-Tolerant Multimedia Delivery System (DT-MDS) is based on adaptive bit rate streaming over HTTP, in particular MPEG-DASH [7]. The multimedia content is partitioned into segments with a certain duration. The segments contain all modalities of the multimedia content such as a video stream and an audio stream. Additionally, segments are self-contained, which means that they can be decoded independently of each other. The multimedia content is made available in different qualities. These versions of the content are referred to as representations. Manifest files describe which representations are available and how they can be accessed (e.g., via a URL).

The main difference to other HAS-based system is that DT-MDS uses a modified version of HTTP in order to work in a DTN. In particular, we adopt the ideas of Wood et al. [8] who suggest to introduce additional *Content-** HTTP headers and forward HTTP GET and PUT requests in a hop-by-hop manner. This modified version of HTTP can be used as a DTN overlay protocol and is referred to as HTTP-DTN. The *Content-** headers are used to identify the original source of the request and its final destination and hence can be used to route the requests in the network. Every HTTP-DTN node contains a storage for storing the received files and some metadata which is used for routing and error detection (e.g., *Content-Source*, *Content-Destination*, MIME type, checksum). Thus, intermediate nodes can store the HTTP-DTN requests and the accompanying payload, which allows them to bridge partitions and deliver content despite the lack of end-to-end connectivity (i.e., perform story-carry-forward routing). For example, when two nodes come into contact, they can issue PUT and GET requests for the files that have to be exchanged. The decision to exchange a file is based on the final destination of the file, which is stored in the *Content-Destination* header. Similarly, GET requests for files can be forwarded by intermediate nodes to the node that is identified in the *Content-Destination* header. In order to identify the original source of a file, the *Content-Source* header is used. Both headers may contain an IP address, DNS name or any other textual information to identify a node in the network.

Every HTTP-DTN node contains a neighbor discovery module. This module is used to advertise a node's existence to other nodes in the local network and provide information about how to exchange data. The discovery module regularly broadcasts beacons to direct neighbors. The beacons contain a sequence number, the identifier of the node and information about available transport protocols (e.g., a TCP socket address). When the neighbor discovery module finds a new node, it triggers a handshake process between the two nodes. The information that is exchanged in this handshake process depends on the routing protocol. For example, it could include information about which messages the nodes have buffered or about their meeting probabilities with other nodes. If the discovery module of a node does not receive several consecutive beacons from a previously available node, it marks the other node as disconnected.

DT-MDS uses a simple naming scheme for nodes which supports addressing nodes in different IP networks without relying on DNS servers. This naming system makes it possible to address nodes independently of their current IP address. Additionally, nodes can advertise different transport protocols that can be used underneath HTTP-DTN to transfer files. This is similar to the concept of convergence layers in the Bundle Protocol [6]. These advertisements are sent periodically and are part of the discovery process.

Since each segment only contains a certain duration of audiovisual data, a recording can only be fully presented if all of its segments are received. There are cases where not all stored segments can be delivered or forwarded, for example, because the transmission bandwidth does not suffice. An HTTP-DTN node can use different forwarding strategies to decide the order in which segments are forwarded. In order to keep the number of partially transmitted contents low, it is important to deliver many segments from the same video, instead of segments from different videos. In particular, we use the following multimedia forwarding strategy: Manifest files have the highest priority and are exchanged before segments. Segments are prioritized based on the representation. In particular, segments from lower representations are sent before segments that offer a higher quality. Additionally, the recording time is used to prioritize segments from older videos. The idea behind this strategy is to increase the chance to deliver the content at least in a basic quality and only forward content in higher qualities when there are enough transmission resources available. Evaluation results (see Section V) show that this strategy can increase the performance of the system compared to a standard first-in, first-out (FIFO) forwarding strategy.

IV. Scenario Description and Evaluation Setup

The scenario that is used for evaluating the multimedia delivery system models an emergency response operation after an explosion in a chemical plant. The scenario has been adopted from previous work [3]. It includes wireless obstacles that attenuate signals between nodes and also obstacles that restrict the mobility of nodes. It consists of several first responder teams that search and rescue victims from two buildings at the incident site. In total, the scenario includes 25 nodes which represent first responders that move according to the Disaster Area Mobility Model [9] with a speed between 1 m/s and 2 m/s. These nodes are assigned to tactical areas as illustrated in Figure 2. Two nodes that move between the Incident Locations (ILs) and the Patients Waiting For Treatment Area (PWFTA) and a node that is located in the PWFTA itself, record videos that need to be delivered to a node in the Technical Operational Command (TOC) area. The videos could either be consumed by an incident commander at the TOC or sent to an off-site command center via a network gateway (e.g., via a satellite uplink).

Each video is made available in two representations with an average bit rate of 500 kbit/s and 2.5 Mbit/s, respectively. The length of the videos is randomly distributed between 15 s and 60 s and videos are recorded in a randomly chosen interval between 120 s and 360 s. Videos are created from 500 s to 3500 s and the simulation is run for 4500 s to give the routing protocols some additional time to deliver the videos. The message buffers of all nodes are unlimited (i.e., each node can buffer all generated segments). All simulations are performed in the ONE simulator [10] and are repeated 23 times using different random seeds.

The ONE simulator is widely used for simulating DTNs and includes well-tested implementations of several DTN routing protocols. However, it does not include models for the physical and MAC layers. Compared to the ONE simulator, OMNeT++ [11] supports to model the physical characteristics and the MAC layer of a wireless communication system. In particular, we use the OMNeT++ IEEE 802.11 MAC layer implementation on top of the free-space path loss propagation model and a wireless obstacle model [12]. Due to space constraints we cannot present configuration details but refer to previous work [3], where these details are given. We used OMNeT++ to prepare connectivity traces for different wireless transmission ranges that were imported into the ONE simulator to also include the aforementioned effects of wireless obstacles.

In a set of evaluations we show how DT-MDS performs using different routing algorithms. In particular five routing protocols are included in the evaluation. Epidemic routing [13] is a flooding-based protocol which forwards each message to all nodes that do not already have buffered the message. PROPHET [14] is another flooding-based protocol. It uses the so called delivery predictability metric to reduce the number of message replications by only forwarding messages to nodes which have a higher predicted chance to deliver the message to its destination. Spray and Wait (SaW) [15] routing limits the number of message copies. Hence, SaW provides lower overheads compared to the aforementioned flooding based schemes. Additionally, we also evaluate a MANET routing protocol that is enhanced by a packet buffering mechanism in order to cope with disruptions in the network. We refer to this approach as MANET-SaF. It works like a traditional MANET routing protocol in connected parts of the network and stores messages when no end-to-end path can be found in the routing table. In previous work we showed that such a hybrid MANET/DTN approach is beneficial in emergency response scenarios [16]. One drawback of this hybrid approach is that packet buffering is not able to cope with permanent disruptions where the sender and the receiver are never in the same connected component. Thus, we also evaluate another combined MANET/DTN routing approach called CoMANDR [17] that enhances MANET routing by packet buffering and a utility based forwarding scheme and hence is also able to deliver messages if the sender and the receiver are never in the same connected component.

To simulate networks with different connectivity characteristics, the transmission range is varied between 20 m and 60 m. Varying the transmission range allows us to evaluate how the multimedia delivery system performs in different scenarios from well-connected networks to sparse networks. Additionally, we also evaluate the multimedia forwarding strategy that has been described in Section III.

It is important to note that the evaluations focus on rather short video sequences since they are more frequently found in emergency response scenarios (see Section II-A). One could argue that for this type of use a ‘download and play’ solution suffices and segmenting the videos is not needed. However, segmenting the videos into smaller parts is still beneficial for the performance of the system, since it reduces the number of partially transmitted messages in the presence of link disruptions. Thus, we also evaluate the effects of using different segment sizes and the case where the videos are not segmented at all. Additionally, the streaming capability of DT-MDS may be useful in other application scenarios.

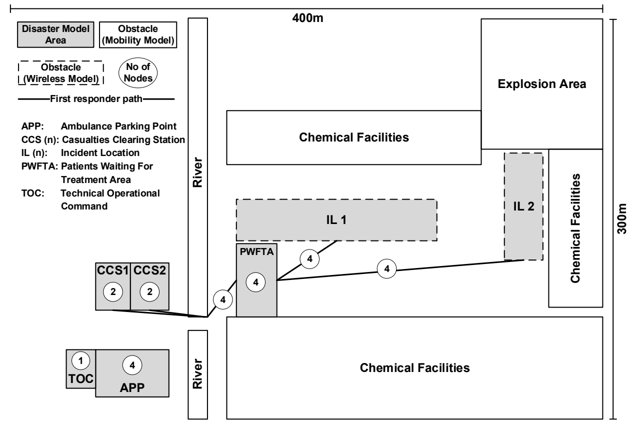

Fig. 2. Evaluation scenario

TABLE I
SIMULATION PARAMETERS FOR THE ROUTING PROTOCOLS.

Parameters for PROPHET/CoMANDR	
$P_{init}(=\alpha)$	0.9
β	0.7
γ	0.995
Parameters for Spray and Wait	
No. of copies	8
Spraying scheme	binary

The first metric that is used in the evaluation is the ratio between created and received videos (video delivery ratio). A video is considered received only if all of its segments could be received. Additionally, the average bit rate of the received videos is evaluated by calculating the average of all received segments of a video. If a segment has not been received, the bit rate for this segment is set to 0, otherwise it is set to the bit rate of the highest received representation. Hence, the average bit rate is a measure for the received quality, since high quality representations also have a higher bit rate. Finally, we also evaluate the delivery latency which denotes the time that is needed to receive a recorded video at the destination.

V. RESULTS

This section presents the evaluation results. All figures include arithmetic means and the error bars indicate the 95% confidence interval.

The connectivity characteristics of the network for different transmission ranges are presented in Table II. The node degree shows the arithmetic mean of the number of 1-hop neighbors. The largest connected component denotes the number of nodes

TABLE II
CONNECTIVITY CHARACTERISTICS

Transmission range (in m)	Connectivity degree CD (avg)	Largest connected component (avg)	Avg. node degree
20	0.15	7.49	2.04
30	0.35	12.79	3.48
40	0.57	17.76	4.85
60	0.75	21.67	8.54

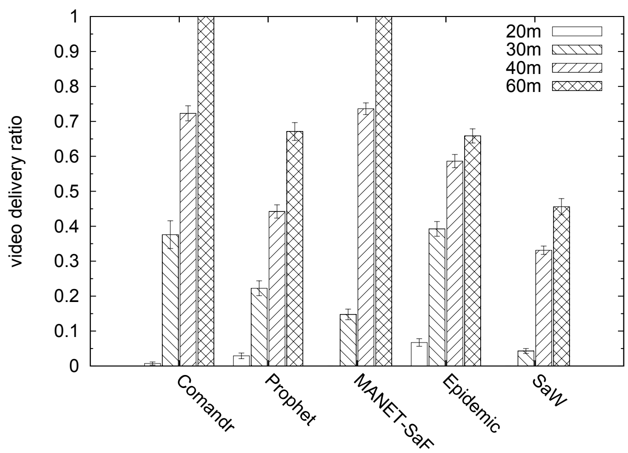

Fig. 3. Video delivery ratio for different transmission ranges, using FIFO forwarding strategy (transmission bandwidth 2 Mbit/s, 2 s segments)

that are in the largest partition and can communicate via end-to-end paths. Finally, the connectivity degree (CD) [17] denotes the probability that two randomly selected nodes are in the same connected component (i.e., an end-to-end path between the two nodes exists). The CD is 0 if all nodes are isolated and 1 if all nodes are connected. According to the connectivity characteristics presented in Table II, it can be seen that the evaluation includes different scenarios from well-connected to sparsely connected ones.

The video delivery ratio of the system for different routing protocols and transmission ranges is shown in Figure 3. In this experiment FIFO was used as forwarding strategy (i.e., the segment buffered for the longest time is forwarded first). In well-connected scenarios (i.e., transmission range of 60 m) all videos can be delivered using the hybrid MANET/DTN routing protocols, namely CoMANDR and MANET-SaF. The flooding based protocols PROPHET and Epidemic introduce too much overhead and hence can only deliver about 65% of the videos. In the best connected scenario, Spray and Wait can deliver about 45% of the videos. The main reason for its poor performance compared to the other protocols is that the available message copies are often only distributed between nodes close to each other (e.g., the nodes in the PWFTA) which never get into contact with the destination node located in the command center.

In sparsely connected networks (i.e., transmission range of 20 m) less than 10% of the packets can be delivered by any protocol. This is due to the fact that only a few other nodes in the network have contact opportunities with the node

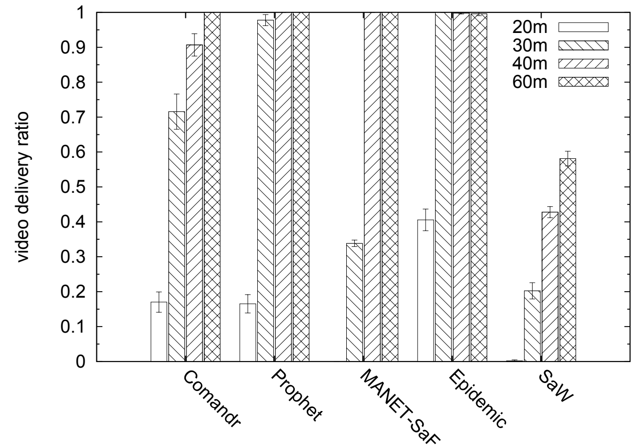

Fig. 4. Video delivery ratio for different transmission ranges, using multimedia forwarding strategy (transmission bandwidth 2 Mbit/s, 2 s segments)

in the command center and hence there are only very few delivery opportunities. MANET-SaF cannot deliver any video in this case since the senders and the receiver are never in the same connected component. Similarly, Spray and Wait cannot deliver any video since the available message copies are never forwarded to nodes that get in contact with the node in the command center.

The multimedia forwarding strategy described in Section III prioritizes lower representations in order to increase the chance that a video is at least received in a basic quality. Figure 4 shows the video delivery ratio using the multimedia forwarding strategy. It can be seen that this strategy significantly improves the number of delivered videos by better utilizing the available bandwidth in order to deliver more segments of a video. For instance, even in the least connected scenario, about 40% of the videos can be delivered using Epidemic routing, compared to the 7% that were achieved using FIFO strategy. These results show the importance of the forwarding stategy and that it is useful to take the semantic of the messages into account when deciding the order in which to transfer buffered segments.

In the last section we argued that segmenting the multimedia content is beneficial for the performance of the system. To support this claim, the effect of the segment length on the video delivery ratio is shown in Figure 5. It can be seen that the video delivery ratio decreases for all protocols when no segments are used (i.e., for each video only one segment with the duration of the video is created). We retrieved similar results for other transmission ranges and bandwidths. We also evaluated the effects of the segment length when using the multimedia forwarding strategy. Here the segment size has a lower impact on the video delivery ratio and in many cases we could not find statistically significant differences. However, if the received video bit rate is considered, segmenting is still beneficial as it improves the overall quality of the received videos (see Figure 6).

The latency for receiving a video is another important metric for evaluating the video delivery system. The overall latency

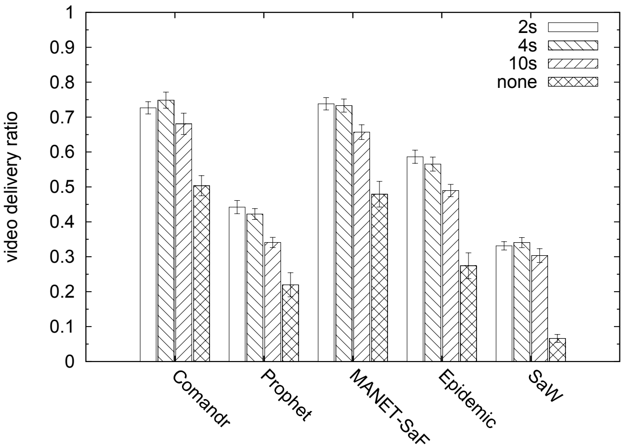

Fig. 5. Video delivery ratio for different segment sizes (FIFO forwarding strategy, transmission bandwidth 2 Mbit/s, transmission range 40 m)

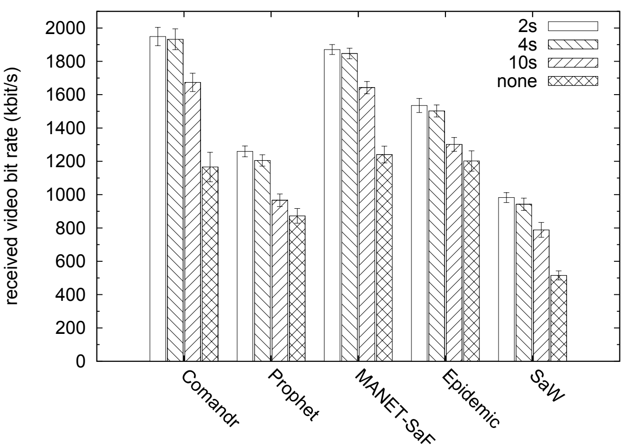

Fig. 6. Received video bit rate for different segment sizes (multimedia forwarding strategy, transmission bandwidth 2 Mbit/s, transmission range 40 m)

for receiving a video is calculated by taking the average of the delivery delays of all segments of a video. The delay of a segment is calculated by measuring the time between segment creation and segment reception. For missing segments (i.e., segments that could not be delivered in any representation) the receiving time is set to the end of the simulation. Thus, missing segments affect the delay. Figure 7 shows the results for a transmission range of 60 m using the multimedia forwarding strategy, where most evaluated protocols could deliver all videos (cf. Figure 4). Since all videos are made available in two representations, the delivered quality of a video may change over time, while segments providing a higher quality are received. Thus, two latency values are calculated. First, the latency for receiving the videos in a basic quality. That means if both representations for a segment are received, the latency is calculated based on which representation has been received first. Second, the delay for receiving the best representation. The hybrid protocols MANET-SaF and CoMANDR achieve the lowest delivery latency. On average, all segments of a video are received in less than two minutes and it takes about five

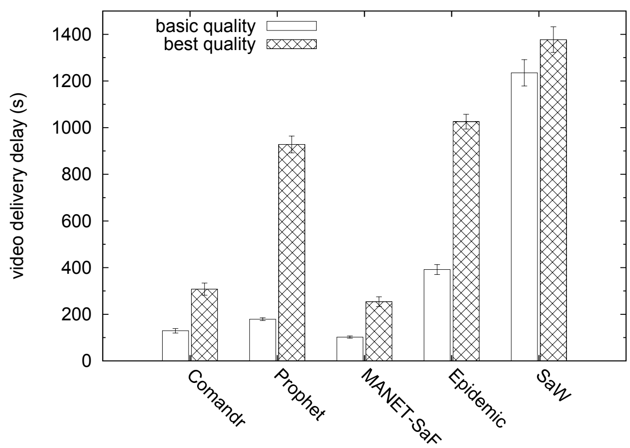

Fig. 7. Video delivery latency for receiving the videos in a basic quality and for receiving the videos in the best available quality (multimedia forwarding strategy, transmission bandwidth 2 Mbit/s, transmission range 60 m)

minutes until the quality of the video is not increased anymore. For PROPHET and Epidemic routing the difference between receiving videos in a basic quality and receiving them in the best quality is relatively large. These differences are due to the fact that both protocols introduce large transmission overheads by flooding the network. Thus, it takes more time until the higher quality representations can be delivered. The latency of Spray and Wait is the highest of all evaluated protocols, since it is not able to deliver all videos and missing segments negatively affect the latency.

VI. Related Work

Klaghstan et al. [18] study video delivery in DTNs using scalable video coding (SVC) where the content is divided into several layers. Based on the importance of the layer, the degree of redundancy is changed. The authors conclude that SVC is better suited for DTNs than single layer codecs, since it allows to receive the content in a lower quality and then gradually improve viewing quality while more layers are received. In [19] the same authors improve the performance by segmenting the layers into smaller chunks in order to adapt the delivery to available contact times. In DT-MDS the content dissemination is already based on segments. However, since SVC can also be used in an HAS system [20], SVC may be an option to increase the performance of DT-MDS.

The Bundle Streaming Service (BSS) [21] adds streaming support to the Bundle Protocol. Although BSS is mainly intended for inter-terrestrial communication, it may also be used for scenarios such as emergency responses. Any BSS-capable node has to provide at least one best-effort and one reliable delivery protocol. To reduce delivery delays, bundles are first sent via the best effort protocol. Since all bundles need to be acknowledged, BSS can detect transmission failures and then switch to reliable delivery. In contrast to BSS, our system uses HTTP-DTN instead of the Bundle Protocol for delivery.

Cabrero et al. [22] suggest a temporal video adaptation technique for DTNs where the quality of a video is adapted by reducing the frame rate. The goal of this adaptation is to provide a constant frame rate that depends on the available network resources. The adaptation technique also changes the order in which stored frames are forwarded in order to cover a bigger span of the recorded video. This adaptation technique is mainly useful for continuous recordings which have not been the focus of this work.

VII. Conclusions and Future Work

In this paper we presented a multimedia delivery system called DT-MDS for delay-/disruption-tolerant networks which is based on HTTP adaptive streaming. This design choice has been made since HAS offers several characteristics that are beneficial in DTNs. In particular, the segmentation of videos into self-contained pieces and the simple control flows allow HAS to work well in DTNs. DT-MDS uses a modified version of HTTP in order to support store-carry-forward routing and bridge network partitions. Evaluation results show that the system works well in a realistic emergency response scenario, especially in combination with hybrid MANET/DTN routing strategies.

This paper presented some initial evaluation results of DT-MDS. Future work includes evaluating the system in more scenarios also from other domains. For instance, evaluating streaming scenarios that include longer video sequences where more intelligent adaptation techniques are needed. One particular open topic for such scenarios is to find adaptation techniques that offer a good trade off between delivery delay and the quality of the delivered content.

Currently, we are working on a proof-of-concept prototype of DT-MDS for Android devices. Initial results are promising and show that current mobile devices have the processing power and transmission resources to record and distribute videos in ad-hoc networks. For future work we plan to improve the implementation in order to test our system in emergency exercises, which would give us valuable feedback by practitioners on how to enhance the system.

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References

[1] A. Monares, S. F. Ochoa, J. A. Pino, V. Herskovic, J. Rodriguez-Covili, and A. Neyem, "Mobile computing in urban emergency situations: Improving the support to firefighters in the field," *Expert Systems with Applications*, vol. 38, no. 2, pp. 1255–1267, Feb. 2011.

[2] A. Al-Akkad, L. Ramirez, A. Boden, D. Randall, and A. Zimmermann, "Help beacons: Design and evaluation of an ad-hoc lightweight S.O.S. system for smartphones," in *Proc. ACM Conf. on Human Factors in Computing Systems (CHI '14)*. ACM, 2014, pp. 1485–1494.

[3] C. Raffelsberger and H. Hellwagner, "Evaluation of MANET routing protocols in a realistic emergency response scenario," in *Proc. Workshop Intelligent Solutions in Embedded Systems (WISES '12)*. IEEE, 2012, pp. 88–92.

[4] J. Landgren and F. Bergstrand, "Mobile live video in emergency response: Its use and consequences," *Bulletin of the American Society for Information Science and Technology*, vol. 36, no. 5, pp. 27–29, 2010.

[5] F. Bergstrand and J. Landgren, "Information sharing using live video in emergency response work," in *Proc. Int. Conf. Information Systems for Crisis Response and Management (ISCRAM '09)*, 2009, pp. 1–5.

[6] K. Scott and S. Burleigh, "Bundle protocol specification," IETF RFC 5050, 2007.

[7] I. Sodagar, "The MPEG-DASH Standard for Multimedia Streaming Over the Internet," *IEEE MultiMedia*, vol. 18, no. 4, pp. 62–67, 2011.

[8] L. Wood and P. Holliday, "Using HTTP for delivery in delay/disruption-tolerant networks," Internet Draft draft-wood-dtnrg-http-dtn-delivery-09, Jun. 2014.

[9] N. Aschenbruck, E. Gerhards-Padilla, M. Gerharz, M. Frank, and P. Martini, "Modelling mobility in disaster area scenarios," in *Proc. ACM Symp. Modeling, Analysis, and Simulation of Wireless and Mobile Systems (MSWiM '07)*. ACM, 2007, pp. 4–12.

[10] A. Keränen, J. Ott, and T. Kärkkäinen, "The ONE simulator for DTN protocol evaluation," in *Proc. Int. Conf. Simulation Tools and Techniques (Simutools '09)*. ICST, 2009, pp. 55:1–55:10.

[11] A. Varga and R. Hornig, "An overview of the OMNeT++ simulation environment," in *Proc. Int. Conf. on Simulation Tools and Techniques for Communications, Networks and Systems & Workshops (Simutools '08)*, 2008, pp. 60:1–60:10.

[12] C. Sommer, D. Eckhoff, R. German, and F. Dressler, "A computationally inexpensive empirical model of IEEE 802.11p radio shadowing in urban environments," in *Proc. Int. Conf. Wireless On-Demand Network Systems and Services (WONS '11)*. IEEE, 2011, pp. 84–90.

[13] A. Vahdat and D. Becker, "Epidemic routing for partially-connected ad hoc networks," Duke University, Tech. Rep. CS-2000-06, Jul. 2000.

[14] A. Lindgren, A. Doria, and O. Schelén, "Probabilistic routing in intermittently connected networks," *SIGMOBILE Mob. Comput. Commun. Rev.*, vol. 7, pp. 19–20, July 2003.

[15] T. Spyropoulos, K. Psounis, and C. S. Raghavendra, "Spray and wait: An efficient routing scheme for intermittently connected mobile networks," in *Proc. ACM Workshop Delay-tolerant Networking (WDTN '05)*. ACM, 2005, pp. 252–259.

[16] C. Raffelsberger and H. Hellwagner, "A hybrid MANET-DTN routing scheme for emergency response scenarios," in *Proc. IEEE Int. Conf. Pervasive Computing and Communications Workshops (PERCOM Workshops '13)*. IEEE, 2013, pp. 505–510.

[17] ——, "Combined mobile ad-hoc and delay/disruption-tolerant routing," in *Proc. Int. Conf. Ad-hoc, Mobile, and Wireless Networks (ADHOC-NOW '14)*. Springer, 2014, pp. 1–14.

[18] M. Klaghstan, D. Coquil, N. Bennani, H. Kosch, and L. Brunie, "Enhancing video viewing-experience in opportunistic networks based on SVC, an experimental study," in *Proc. IEEE Int. Symp. Personal Indoor and Mobile Radio Communications (PIMRC '13)*. IEEE, 2013, pp. 3563–3567.

[19] M. Klaghstan, N. Bennani, D. Coquil, H. Kosch, and L. Brunie, "Contact-based adaptive granularity for scalable video transmission in opportunistic networks," in *Proc. Int. Wireless Communications and Mobile Computing Conf. (IWCMC '14)*. IEEE, 2014, pp. 773–778.

[20] M. Grafl, C. Timmerer, H. Hellwagner, W. Cherif, and A. Ksentini, "Evaluation of hybrid scalable video coding for HTTP-based adaptive media streaming with high-definition content," in *Proc. Int. Symp. and Workshops World of Wireless, Mobile and Multimedia Networks (WoWMoM '13),*. IEEE, 2013, pp. 1–7.

[21] S.-A. Lenas, S. Burleigh, and V. Tsaoussidis, "Reliable data streaming over delay tolerant networks," in *Proc. Int. Conf. Wired/Wireless Internet Communications (WWIC '12)*. Springer, 2012, pp. 358–365.

[22] S. Cabrero, X. Paneda, R. Garcia, D. Melendi, and T. Plagemann, "Dynamic temporal scalability: Video adaptation in sparse mobile ad-hoc networks," in *Proc. Int. Conf. Wireless and Mobile Computing, Networking and Communications (WiMob '12)*. IEEE, 2012, pp. 349–356.