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Enhancing real-time video streaming over mobile ad hoc networks using multipoint-to-point communication

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Abstract

Supporting video applications over mobile ad hoc networks is more challenging than over other best-effort networks due to the lack of a preinstalled infrastructure to provide centralized control for the entire networks. As mobile ad hoc networks emerge as a promising technology for untethered, ubiquitous service in future communication system, a solution to support increasingly popular video applications is essential. A considerable amount of research has been done to provide solution for video streaming over lossy networks, among which are the diversity techniques. Path and server diversities are proven feasible to guarantee reliable video delivery wired networks. Besides, using disjoint multipath to support video streaming over mobile ad hoc networks has also been widely studied. In this paper, we investigate the possibility of implementing server diversity over ad hoc networks. More specifically, we use multipoint-to-point transmission together with Multiple Description Coding to enhance the quality of video streaming. In order to discover maximally disjoint routes for each sender and to distribute the workload evenly within the network, an extension for video applications is added to the dynamic source routing (DSR) protocol. Simulation study is carried out using NS-2 to demonstrate the feasibility of the proposed mechanism and it shows that better quality of video streaming is achieved, in terms of fewer and shorter interruptions during the video session. Moreover, we also show that the number of transmission points should be limited because using too many transmission points burdens the network with unnecessary control overhead.

Keywords: Mobile ad hoc networks (MANETs); Multipoint-to-point communications; Real-time video streaming; Dynamic source routing (DSR)

1. Introduction

A mobile ad hoc network is formed when a group of wireless devices organize themselves dynamically in a mesh topology to relay packets from one node to any other node without the support of a preinstalled infrastructure. Initially, its target deployment is at critical environments where infrastructure is not readily available, such as during disaster recovery and battlefields. Nowadays, mobile ad hoc networks emerge as one of the most promising solution for providing untethered, ubiquitous service in future wireless system due to the increase in computing power and decrease in production cost of mobile devices [1–3]. Moreover, the development of new multiplexing techniques has made available higher bandwidth for wireless channel. For example, the deployment of orthogonal frequency division multiplexing (OFDM) has increased the data rate of IEEE 802.11a/g standard to 54 Mbps [4]. All these factors create a platform suitable for supporting bandwidth-constrained multimedia applications over wireless networks. Recently, supporting video applications over wireless networks has received a lot of attentions among researchers mainly due to the increasing demand on this service among users [5–9].

Video applications have rigid requirements on bandwidth, delay, and jitter. Much research has been done on supporting video applications on both wired and wireless lossy networks, and good performance has been

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demonstrated [10–13]. For example in [10,11], several routes are used to transmit video coded with multiple states coding in order to guarantee at least minimum quality of video streaming is achieved. Multiple state coding is a coding technique that encodes the video into a number of independent video streams, an example of this coding scheme is multiple description coding (MDC) scheme [14,15]. This technique ensures acceptable video quality at most of the time because the probability of congestion occurs simultaneously at all connections is very small. Besides, multipoint-to-point communication was proposed in [16] to provide a mean of transmission for real-time video over peer-to-peer networks. This work provided a framework for real-time video streaming because it allows users to start playing the video after a short waiting time. On the other hand, some research findings show that error-resilient video coding scheme is another technique to enhance video streaming [17–20]. These encoding schemes aim to reduce the distortion caused by frame lost at the receiver by dividing the video stream into several sub-streams or improving the compensation method at the decoder. All the methods explained above have been proven feasible and good performance is guaranteed over lossy networks, but it does not necessarily hold true for infrastructureless mobile ad hoc network. Without fixed infrastructure support, every change in the network topology is associated with a more complicated and time consuming recovery process. Besides, ad hoc networks rely on multihop transmission that involves several wireless links. This creates higher transmission loss due to higher probability of link breakages, errors and interferences.

Previous work on video streaming over mobile ad hoc networks concentrates on multipath transport [6,7]. Generally, the same concept used in the wired networks is implemented in mobile ad hoc networks. In [6], Mao and others proposed three techniques of video transmission using multipath transport and different video coding schemes, namely feedback based reference picture selection scheme, layered coding with selective Automatic Repeat Request scheme and multiple description motion compensation coding scheme. Basically, it is desired to transmit video coded with MDC scheme using optimally disjoint routes so that a single link failure affects only a minimum number of descriptions [7]; studies on finding multiple highly correlated disjoint routes have been studied in [21].

In this paper, we investigate the feasibility of implementing server diversity for video streaming over mobile ad hoc networks. Specifically, we proposed a method to provide continuous video transmission from several nodes to a single node, assuming the same video file is stored at more than one node within the network. This scenario is very common at present day due to the increasingly commonly used file-sharing software and the development of digital multimedia technologies. Besides, we also use MDC scheme for better error resilience. In order to accommodate the proposed algorithm, the routing protocol is extended to support multipoint-to-point video transmission and MDC scheme. Besides, we also examine the limitations of the proposed method considering the differences with wired networks. In wired networks, many concurrent connections are created to a single node to increase the network throughput. However, this does not hold true in ad hoc networks because the same transmission medium is shared by all members in the networks. Therefore, interference will become severe if too many connections are established nearby each other. Our proposal considers all these limitations to design a system with optimum performance.

The remainder of the paper is organized as follows. Our proposal is explained in detailed in Section 2, including the introduction of multiple description coding (MDC) scheme, the motivation of our proposal, and the extension added to the existing routing protocol to support our proposal. Section 3 presents the video evaluation framework used in this paper. Section 4 contains details of the simulation study and explanation on the evaluation parameters. This is followed by results and discussion in Section 5. Finally, conclusions and possible future work are given in Section 6.

2. Multipoint-to-point (M2P) video transmission

2.1. Background

The emerging of peer-to-peer networking has brought drastic changes to the world of networking that provides users with the ability to accelerate the content retrieval from multiple sources. In other words, a client can download the same data simultaneously from multiple servers. This mechanism lies on the fact that the same data, especially multimedia data such as video and audio files, is available at multiple locations. Moreover, recent advances in file-sharing technology, cheap and high capacity storage at portable size, and more flexible multimedia encoding techniques have created a tailor-made platform for the implementation of multipoint-to-point multimedia transmission. There is a considerable amount of research showing that multimedia streaming could benefit from concurrent streaming [10–13,16,22].

Considering the similarities in topology between peer-to-peer networks and mobile ad hoc networks, we propose the implementation of multipoint-to-point video streaming over mobile ad hoc networks in this paper. Fig. 1 illustrates some possible implementations of the proposed mechanism for various video applications. Fig. 1(a) shows applications such as video-on-demand, live streaming and video conferencing over a hybrid network. In this case, if nodes 1, 2, and 3 can receive video directly from the original video source through an infrastructure network, and at the same time if they are part of the ad hoc network, they can become the secondary video source for other nodes in the network. Meanwhile, Fig. 1(b) demonstrates download-and-play application. This phenomenon is the same as file-sharing in peer-to-peer networks, but our objective is to allow users to start viewing the video after a short

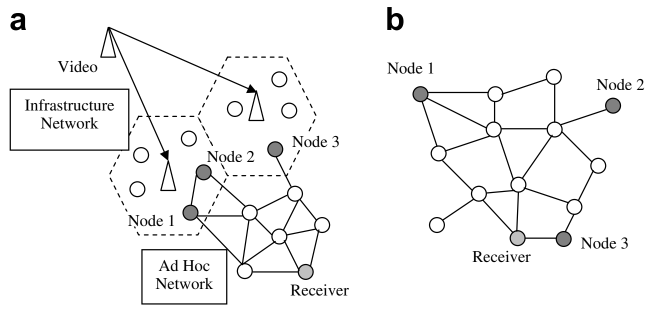

Fig. 1. Demonstration of multipoint-to-point transmission over (a) hybrid networks and (b) ad hoc networks.

waiting period, instead of having to wait until the download is completed.

In conventional peer-to-peer networks, a greedy mechanism is used to establish many concurrent connections to send duplicated video in order to achieve the highest possible data rate. It is obvious that this method increases the workload and bandwidth occupancy considerably, but the impact is manageable considering the high bandwidth available over wire-line channels. However, this does not hold true in wireless networks because the same transmission medium is shared by all nodes in the network; therefore, too many connections established to a single node may degrade the network performance notably. Besides, it is not cost effective to send duplicate copies of the same video due to the limited bandwidth available. With this in mind, the multiple description coding (MDC) scheme is used to generate several independent descriptions of video, and each sender is assigned to send only one description. This step can significantly reduce the network workload, yet maintaining the video quality at an acceptable level.

2.2. Multiple description coding (MDC) scheme

In this section, a brief overview of the video coding scheme used for video transmission over packet networks is given. Conventionally, video coding aims to encode the raw video into a single stream of video frames that is suitable for storage and transmission. Generally, this coding scheme uses motion-compensation prediction between frames, block-discreet cosine transform (DCT) for error prediction and entropy coding. One major problem of conventional video coding is the propagation of error where a single frame error may affect not only the current frame but its subsequent frames, until a frame with new prediction is received. In order to combat this problem, layered coding is introduced to divide the video stream into two layers: the base layer and the enhancement layer. In this case, the base layer alone is enough to reconstruct the original video but at lower quality. The enhancement layer is used to improve the video quality further. In normal circumstance, both layers of video are sent, but if any constraints exist, only the base layer is sent. This method is more robust but the same problem may occur if the base layer is lost and the enhancement layer becomes totally useless. Consequently, the MDC scheme is introduced. MDC is a coding technique that generates several video descriptions that are equally important and totally independent. Each description can be decoded independently to obtain the original video at low but acceptable quality and every additional description received improves the quality further. As mentioned in the previous section, MDC is more error-resilient than conventional methods for video transmission over lossy networks. Consequently, we also use this coding scheme in our proposal.

Fig. 2 shows a general form of MDC coding. Initially, the raw video stream is divided into several sub-streams prior to the encoder. One of the simplest ways to separate the video is the per frame allocation scheme as shown in Fig. 2(a). Next, the sub-streams are encoded independently to obtain several video descriptions. There is no constraint on the selection of the type of encoders; either the standard video encoder such as H.264 and MPEG4, or specially designed MD encoder with better error resilience [15]. Usually, higher bandwidth is required to accommodate the encoded video as compared to single video stream because of the larger difference between neighboring frames within each sub-stream. This causes lower compression efficiency and the resulted frame size is usually larger. In [23], a comprehensive analysis on video encoded with MDC scheme shows that the average frame size increases with increasing number of descriptions. The increment in average frame size depends on the contents of the video; normally, high motion video produces larger increment as the number of descriptions increases.

The decoding process is given in Fig. 2(b). Each description is decoded separately and the reconstructed substreams of video are merged for display purpose. At this stage, a compensation scheme can be introduced to replace the missing frame with the one previously displayed. An example is shown in Fig. 3. If frame P_5 is lost during transmission, both P_7 and P_9 become undecodable too because they depend on the previous frame for motion-compensation. The error is propagated until I_{11} is received because

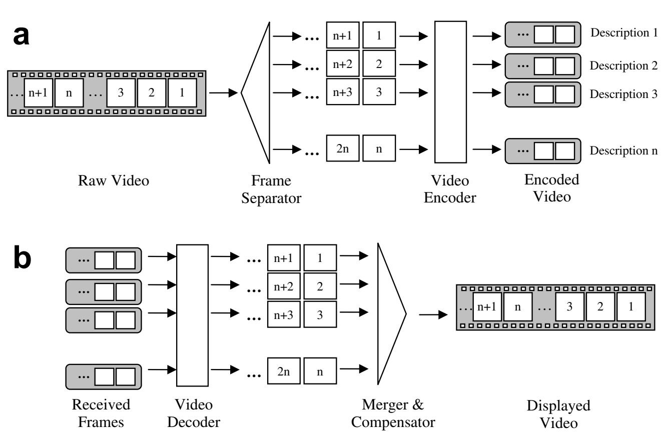

Fig. 2. Multiple description coding (MDC) scheme: (a) the encoding part and (b) the decoding part.

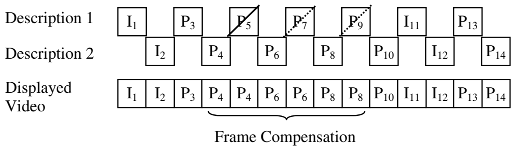

Fig. 3. Frame compensation scheme with the present of frame lost.

I-frame can be decoded independently without referring to the previous frame. If only one video description is used (in conventional coding scheme), this error causes a distortion in the video streaming. By using MDC, the undecodable frames are replaced with decodable frames from the other sub-stream, this can avoid the occurrence of obvious distortion.

2.3. Novel architecture

The basic idea of our proposal lies on implementing concurrent streaming of video from several nodes to a single receiver. Each node sends only a portion of the video coded with MDC scheme, assuming the same video can be retrieved from more than one node in the network, as explained in the previous section. Fig. 4 shows the novel architecture proposed for this mechanism. There are three main stages involved:

(i) *Source searching stage* involves the searching of the potential video sources. This stage involves request for information sent out by the requestor (video receiver), and response by the video sources (video senders). In our proposed mechanism, the receiver decides the number of senders involved and which description is sent by each sender. This information is fed back to the selected video sources so that the video is encoded and sent out accordingly. Most of the file-sharing software has effective and efficient searching mechanism which can be used in our proposal. In [24], two mechanisms are introduced for information searching using PULL and PUSH methods. This stage is not covered in this paper because this paper focuses mainly on the routing mechanism.

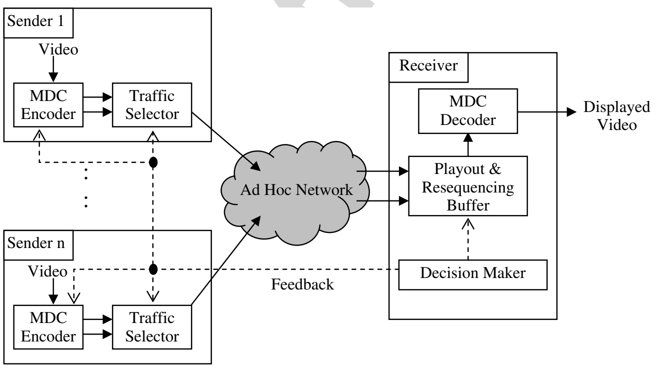

Fig. 4. Novel architecture of the proposed system using multipoint-to-point communication.

(ii) *Route establishment stage* aims to obtain routes connecting the senders and the receiver. Since we are dealing with the transmission of a number of video descriptions coded with MDC scheme, it is desired to have optimal disjoint routes for each description so that a single link breakage affects only a minimum number of descriptions. An extension is added to the original routing protocol (we use Dynamic Source Routing Protocol in this paper) to discover disjoint routes for each sender. This paper focuses on this stage and simulation study is carried out to demonstrate the effectiveness of the extension added. Detailed explanation on the extended DSR protocol is given in the next sub-section.

(iii) *Video streaming stage* is started once a valid route is obtained. The feedback information obtained during the source searching stage is used for video encoding at senders and video decoding at receiver. At the sender, the MDC encoder generates the number of descriptions as requested by the receiver. Then, the traffic selector chooses only one description to be sent out, as assigned by the receiver. At the receiver, we assume no synchronization among senders is needed and a playout/resequencing buffer is used to absorb the frames jitter and to organize the frames into proper sequence prior to the decoding process. The frame compensation scheme is carried out by the MDC decoder as well.

2.4. Extended dynamic source routing for multipoint-to-point transmission

We select the dynamic source routing (DSR) protocol as the routing protocol in our approach [25]. DSR is suitable because it allows the discovery of multiple paths and can easily fit into our proposal with minimum modifications. An extension is added to the original DSR protocol in order to support multipoint-to-point video transmission; we call the extended version eDSR protocol. This extension is only used by the video applications and other services should follow the original protocol. The main objective of this extension is twofold. First, to encourage the senders to use maximally disjoint routes; and second, to provide better protection for video traffic.

In the original DSR protocol, route discovery process is carried out using route request packet (RREQ), identified with a duplex that contains sender ID and sequence number. The intermediate nodes forward a RREQ only if the sequence number is newer that the one previously forwarded. Otherwise, the RREQ is discarded. Besides, the intermediate nodes can create a reply for the RREQ if they have a valid route to the destination. At the receiver, all RREQs are replied by generating route reply packet (RREP) that carries the reversed route traveled by the RREQ. This RREP is sent unicastly back to the sender. From Fig. 5(a), we can observe that this mechanism has

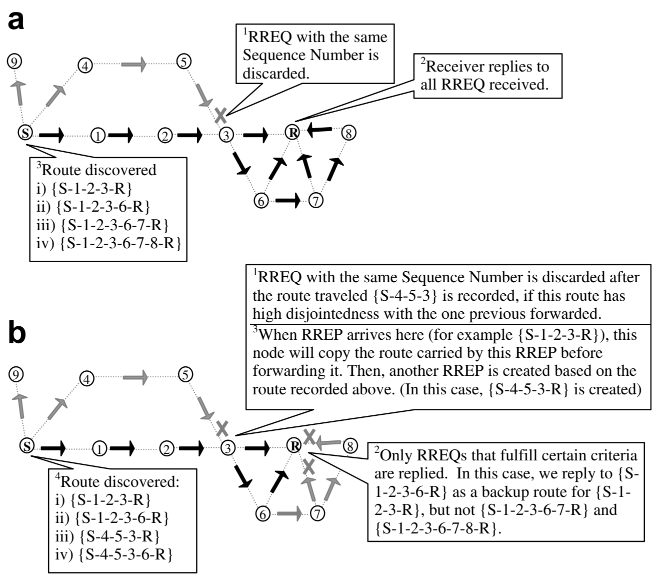

Fig. 5. Illustration of the route discovery mechanism in (a) original DSR and (b) extended version.

```
1. if (RREQ)
   1.1 if (intermediate node)
       1.1.1 r0 = route traveled;
       1.1.2 if (fresh RREQ)
           1.1.2.1 rebroadcast the RREQ;
           1.1.2.2 r1 = r0;
           1.1.2.3 reset b1;
       1.1.3 if (duplicated RREQ)
           1.1.3.1 discard the RREQ;
           1.1.3.2 if (r1.length <= r0.length) and (shared-nodes-ratio(r0,r1) < Th1)
               1.1.3.2.1 if (b1 is empty) or (shared-nodes-ratio(b1,r1) > shared-nodes-ratio(r0,r1))
                   b1 = r0;

   1.2 if (receiver)
       1.2.1 if (can_reply)
           1.2.1.1 find_Priority;
           1.2.1.2 create RREP and send out;

2. if (RREP)
   2.1 if (intermediate node)
       2.1.1 if (b1 is not empty)
           2.1.1.1 create backup RREP.
       2.1.2 send out the RREP.
       2.1.3 send out backup RREP, if exist.
   2.2 if (sender)
       2.2.1 store the route in route cache;
```

Note:
i. Sender is the node that initiates the route discovery mechanism.
ii. Th1 is a threshold values.
iii. r0 is the route traveled by the current RREQ; r1 is the route traveled by the first RREQ; and b1 is the backup route at intermediate node.

Fig. 6. Route discovery mechanism for video applications.

the tendency to discover routes that use most of the nodes nearby the receiver but not the sender. The weakness of this method is that a single link breakage nearby the sender may causes all routes becoming invalid. More importantly, it is difficult to discover disjoint routes for different senders using this mechanism. In order to avoid this problem, RREQs sent by video senders follow a different route discovery mechanism as shown in Fig. 5(b). To explain our proposed extension, we need to introduce Eq. (1) first, which is used to calculate the ratio of shared nodes between two routes. This ratio indicates the level of disjointedness of two routes, which have the same destination. For example if given Route 1 is {X–A–B–C–D–Z} and Route 2 is {Y–U–C–D–Z}, to calculate the shared-node ratio for these routes at node Z, these routes have two shared-nodes, *i.e.*, {C, D} (node Z is excluded because it is a common destination); and since Route 2 is shorter than Route 1, the length of the shorter route is four (again, node Z is excluded); by using Eq. (1), we obtain the shared-nodes ratio equals to 0.5.

$$\text{Shared-nodes ratio} = \frac{\text{Number of shared nodes}}{\text{Length of the shorter route}} \qquad (1)$$

The algorithm of the extension added is given in Fig. 6. Basically, the following changes are introduced:

(i) At the intermediate nodes, the RREQ is processed by following step (1.1). The fresh RREQ is rebroadcasted as usual as in (1.1.2) but the duplicated RREQ is processed as in (1.1.3) before it is discarded. This step aims to obtain a backup route, which will be used in (2.1) to create a backup RREP. For example in Fig. 5(b), if route {S-1-2-3} is the route traveled by the first RREQ, then route {S-4-5-3} is stored as the backup route.

(ii) At the receiver, not all RREQs are replied. In step (1.2.1), the route is compared with the routes replied previously to the same sender. If the shared-nodes ratio is above certain value, the RREQ is not replied. Next, the priority of the route is determined in step (1.2.1.1). This is done by comparing the current route with routes replied previously to other senders. If the shared-nodes ratio is below certain value, it is given high priority. Else, low priority is assigned. In this paper, we arbitrarily set a strict threshold for the shared-node ratio, which is equal to 0.8. The selection of this threshold depends on the density of the networks. If the network has high density, it is likely that highly disjoint routes can be discovered easily, therefore, we can set a strict threshold for the shared-nodes ratio. On the contrary, the discovery of disjoint routes will be more difficult if the network has low density and we should allow a looser requirement on this parameter. As mentioned earlier, the priority indicates the disjointedness of the route with routes used by other senders. Therefore, the senders always use the shortest high priority route for video transmission. This step guarantees shortest possible delay and encourages the use of disjoint routes.

(iii) When a high priority RREP reaches an intermediate node while it is unicastly sent back to the sender, algorithm given in step (2.1) is used to create a backup route for the sender. This is done by combining the backup route obtained from step (i) above with the route carried by the current RREP. In our example, if high-priority RREP carrying route {S-1-2-3-R} reaches node 3, another RREP is created using the backup route {S-4-5-3}. In this case, the backup RREP carrying route {S-4-5-3-R}.

In our mechanism, the receiver is given the full responsibility in determining the optimal disjoint routes for each video sender. Therefore, the intermediate nodes are not allowed to reply to RREQs even they have information to the destination. Since this method generates more control load, only RREQs sent by video senders are allowed to use this mechanism. On the other hand, intermediate nodes are required to record the route carried by all RREP in their route cache. This step can increase the successful rate of salvaging, which is a mechanism to rescue a packet when the next hop recorded in its source route is broken. This is done by replacing the broken route with another valid route to the destination from in the route cache.

3. Video evaluation

Video evaluation must be based on the perception of real humans viewing the received video because providing good quality video for humans viewing is the ultimate objective of video transmission. In order to standardize the subjective evaluation, ITU-R has recommended a subjective quality metric called mean opinion score (MOS) on a scare of 1 (bad) – 5 (excellent) [26]. However, having real humans for video evaluation is very expensive and time-consuming. Therefore, it is desired to have an automatic and systematic evaluation system for video that emulates human visual system. The most commonly used metric is the peak-signal-to-noise ratio (PSNR) calculated using Eq. (2) [27].

$$\mathrm{PSNR} = 20\log_{10}\left[\frac{V_{\mathrm{peak}}}{\sqrt{(1/N)\sum_{i}\sum_{j}(Y_{\mathrm{ref}}(i,j) - Y_{\mathrm{prc}}(i,j))^{2}}}\right] \tag{2}$$

where $Y_{\mathrm{ref}}(i,j)$ and $Y_{\mathrm{prc}}(i,j)$ are the pixel value of the reference and reconstructed frames, respectively; N is the total number of pixels in a frame; and V_{peak} equals to 255 for picture coded with eight-bit resolution. A possible mapping between PSNR and MOS is given in Table 1.

A video stream consists of many frames and usually the average PSNR (APSNR) is calculated to represent the video quality. This parameter gives an overview of the video streaming as a whole but it does not take into account every distortion that occurs during the video viewing. For example, it can be noted that both reconstructed sequences in Fig. 7 have APSNR above 37 dB, which is of excellent quality. However, the viewers may not agree because there is a serious distortion occurs between frames 15–25 in case (a). Besides, it is difficult to compare the performance of these video sequences because their average PSNR is very close.

Table 1
PSNR to MOS mapping

MOS	PSNR (dB)	Comments
1	dB < 20	Bad
2	20 ⩽ dB < 25	Poor
3	25 ⩽ dB < 31	Fair
4	31 ⩽ dB < 37	Good
5	dB ⩾ 37	Excellent

In this paper, we introduce a performance metric called interruption. An interruption is observed when one or more consecutive frames with MOS below an accepted value are displayed. In this paper, we assume MOS below 4 is unacceptable. Naturally, it is very difficult for a viewer to notice distortion if only a small amount of consecutive unacceptable frames are displayed due to the nature of human visual system. When the number of consecutive unacceptable frames increases beyond certain value, the distortion can be noticed. The severeness of an interruption depends on how long the interruption occurs. With this in mind, we introduce the following measurement for the severeness of an interruption: unnoticeable, minor and major interruptions. We assume an interruption is unnoticeable if it lasts shorter than 0.5 s; a minor interruption lasts between 0.5 and 1 s, and a major one lasts more than 1 s. Besides, it is important to measure how frequent the interruptions occur as well.

4. Simulation framework

Simulation study has been carried out using NS-2 [28] with CMU wireless extension [29]. The objective of our simulation study is twofold: first, to demonstrate the effectiveness of multipoint-to-point transmission as compared to point to point transmission and second, to investigate the limitations of multipoint-to-point communications. The simulator used contains the IEEE802.11 protocol in the MAC layer working in the distributed coordination function (DCF) mode, a form of carrier sense multiple access with collision avoidance (CSMA/CA). Its physical layer features are not modeled. The channel bandwidth is 11 Mbps and the transmission range is 250 ms. The simulation topology contains 50 nodes within an area of 1500 m × 800 m. A rectangular shape area is used to create more connection breaks during simulation as mentioned in [7]. UDP is used in the transport layer. The node mobility is modeled using the enhanced Random Waypoint Model [30]. The minimum traveling speed is set to 0.1 m/s and the maximum speed varies between 2.5 and 15 m/s to represent different levels of mobility. For each maximum speed, 30 scenarios are generated to obtain the average performance. In each scenario, every node is assigned randomly with an initial location, a destination and a traveling speed, which is uniformly distributed between the minimum and maximum speeds. These nodes travel from their initial location to the destination at the assigned speed. After they reach the destination, a new destination and traveling speed are assigned. This is repeated until the simulation ends. Also, we consider continuous nodal mobility with zero pause time. Five cross-traffics of

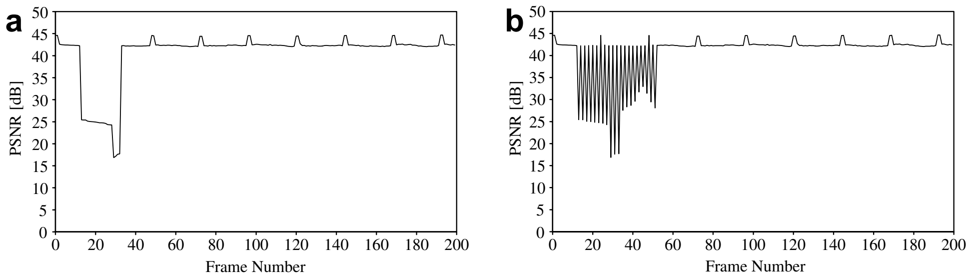

Fig. 7. PSNR for two video streams with distortion: (a) APSNR = 40.5 dB; (b) APSNR = 40.8 dB.

18 kps each are introduced randomly as background traffic in the network.

We use a high motion video sequence in quarter common intermediate format (qcif) format called *highway*, which consists of 1800 frames [31]. The video application is encoded at 12 frames per second (fps). The encoder explained in Section 2.2 is used to generate the number of video descriptions required in each case [23,32]. Table 2 shows the average frame size for video encoded using MDC scheme with different number of descriptions. Obviously, the average frame size increases with the number of descriptions generated as explained in Section 2.2. A 5-s buffer is used at the receiver as playout buffer and to absorb the frame jitter. We assume the frames received is decoded and displayed directly after the 5 s waiting time to simulate real-time video applications. Therefore, packet arrives after the playout deadline is discarded.

Our simulation study is divided into two parts. First, point-to-point video transmission is compared with multipoint-to-point transmission; we vary the number of concurrent connections from two to four. DSR protocol is used in point-to-point transmission while eDSR is used in multipoint-to-point transmission. Second, we compare the performance of eDSR with the original DSR by observing video transmission from two nodes. The video senders and receiver are chosen randomly in each simulation. The following parameters are used for performance evaluation:

(i) Average peak signal-to-noise ratio (APSNR).
(ii) The number of minor interruptions and total length of minor interruptions.
(iii) The number of major interruptions and total length of major interruptions.
(iv) Normalized routing load is the ratio of total number of routing packets forwarded at each node to the total number of data packets received at the destinations, as given by Eq. (3). The data packets include both video and non-video applications because some information carried by the routing packets are used for both traffic, therefore, they cannot be differentiated.

Table 2
Average frame size for video encoded with different number of video descriptions (n)

Number of video descriptions (n)	1	2	3	4
Average frame size (bytes)	4142.93	4656.52	4977.62	5220.21

$$\text{Normalized routing load} = \frac{\text{Total routing packets propagated at every node}}{\text{Total data packets received}} \tag{3}$$

5. Results and discussion

First, we examine the overall video quality in terms of average peak signal-to-noise ratio, as given in Fig. 8. As seen, video transmission using multipoint has better overall performance than using single point transmission (c1_dsr); the improvement ranges between 0.3 and 1.0 dB. Larger improvement is observed at high mobility because at low mobility, the network is stable and link breakages occur less frequently; therefore, using a single node can provide video at good quality and using multipoint transmission has less impact. As mobility increases, the connection between video sender(s) and receiver breaks frequently, and with multipoint-to-point transmission, the probability of losing connection to all video senders simultaneous is very small. When comparing the original DSR and eDSR with only two transmission nodes, the extended DSR

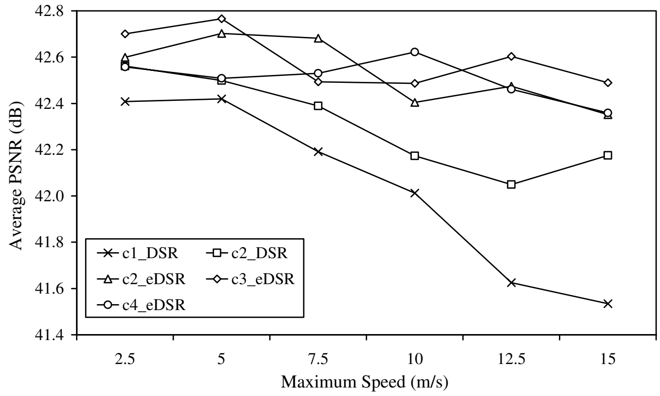

Fig. 8. Average PSNR versus node speed for the comparison study.

(c2_eDSR) gives better performance. This shows that using disjoint routes is one of the major factors that determines the success of our proposal.

From the same graph, we also notice that the video quality does not improve linearly with the increase in the number of transmission nodes. In most cases, using three transmission nodes (c3_eDSR) gives the highest average PSNR; c3_eDSR is better than c2_eDSR for about 0.1dB in most cases, except at 7.5 m/s. As compared to e4_eDSR, the average PSNR for case c3_eDSR is at least 0.1 dB higher except at 7.5 and 10 m/s. We believe the cause of this phenomenon is the combination effects of many factors, including the characteristics of the random waypoint model used for scenarios generation, the randomness in the selection of video sources and receiver, and the effects of mobility. Generally, it is fair to mention that using three video sources give the best performance at most levels of mobility and increase the number of video sources further brings negative impact to the system. This observation matches our previous prediction that it is not worthwhile using too many video sources and descriptions because it not only makes the system more complicated, but the unnecessary workload also reduces the network performance. From Table 2, we can see that using two descriptions of video only increases the video size by 12%, but an increase of 26% if four video descriptions are generated. Besides, it is more difficult to discover the disjoint routes for each video source if too many transmission nodes are used. Based on our simulation results, it is fair mentioning that using three descriptions is optimum.

From the same graph, a non-linear relationship between video quality and mobility is also observed. This is mainly due to the randomness in the scenarios created using Random Waypoint Displacement method and the settings of the routing protocol. Besides, the performance of our proposal is steadier. As mobility increases, the drop in average PSNR is about 0.8 and 0.6 dB for c1_DSR and c2_DSR, respectively. For other cases that use eDSR, the drop is only about 0.3 dB. Again, this is the benefit of using disjoint routes because using disjoint routes can guarantee video streaming from at least one of the video sources. This promised continuous video viewing at the receiver.

Next, the normalized routing load is inspected in Fig. 9. As expected, the proposed eDSR has higher routing load as compared to the original DSR, the increase is between 2% and 6%. This is mainly caused by the modified route discovery mechanism, in which the RREQ is replied by the destination node only. Besides, the mechanism to provide backup route by creating RREP at intermediate nodes also contributes to the additional routing load. Therefore, there is a tradeoff between routing load and efficiency, which should be given careful consideration while designing the video transmission framework. We can also observe that the routing load increases as the number of transmission nodes increases because more connections between senders and receiver are involved. Therefore, it is generally not practical to use too many transmission nodes because the

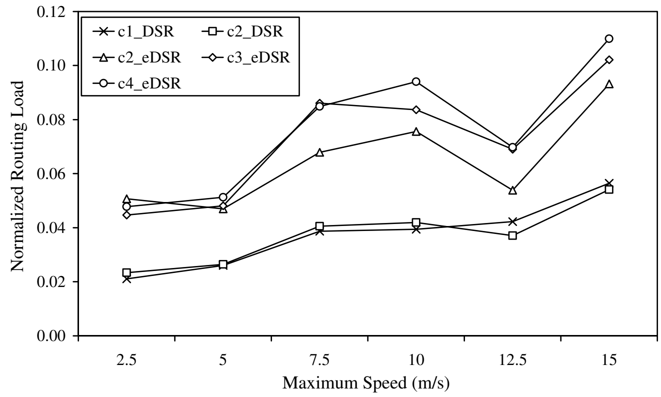

Fig. 9. Normalized routing load versus node speed for the comparison study.

network becomes overloaded and the improvement is less significant. In order to have a closer look on the video quality, the occurrence of interruptions is observed. Figs. 10 and 11 show the video quality in terms of minor and major interruptions occur during the video viewing. Three important observations are obtained.

(i) For both parameters, multipoint-to-point transmission performs much better than point-to-point transmission. The improvement is very significant and

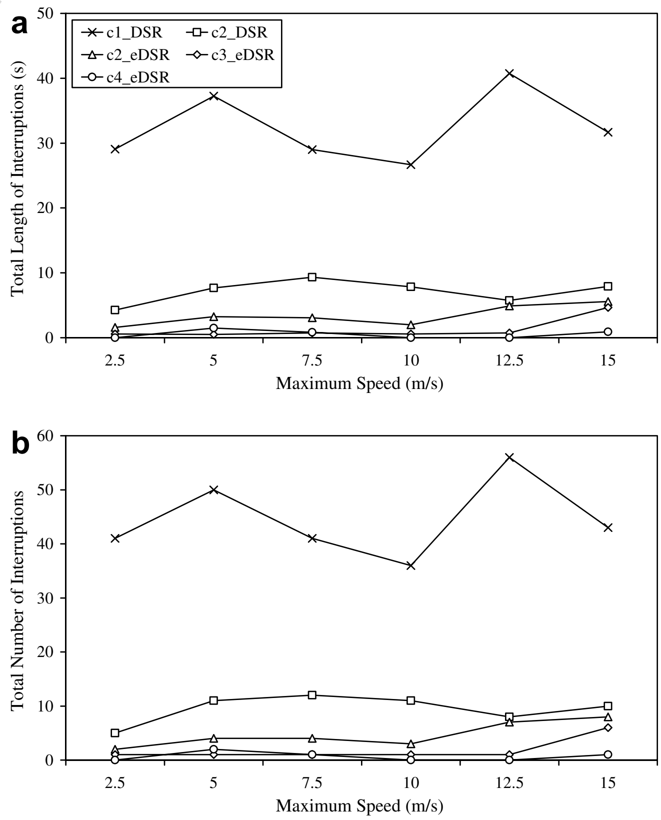

Fig. 10. Evaluation on minor interruptions occur during the video streaming at different levels of mobility: (a) total length of interruptions and (b) total number of interruptions.

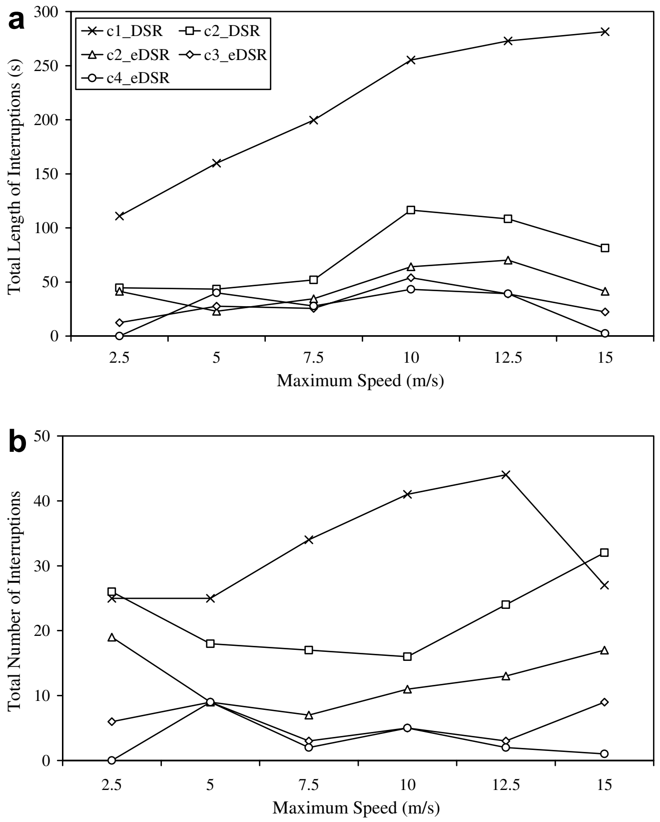

Fig. 11. Evaluation on major interruptions occur during the video streaming at different levels of mobility: (a) total length of interruptions and (b) total number of interruptions.

shows clearly from the graphs. Multipoint-to-point transmission can significantly reduce the occurrence of interruptions as well as shortening the total length of interruptions because the probability of getting service from at least one of the video sources is higher.

(ii) Considering only two nodes are used as video sources, we can observe that the extension added to the original DSR protocol increases the video quality. The improvement is mainly contributed by the modified route discovery mechanism and the mechanism of encouraging the video senders to use disjoint routes. One interesting observation is observed in Fig. 10(b) when the maximum speeds are 2.5 and 15 m/s. We can observe that more interruptions occur in c2_DSR than c1_DSR. This shows that using multipoint transmission with MDC does not necessarily improve the overall performance. We must ensure these video descriptions are sent out using disjoint routes to obtain the desired output.

(iii) As the number of transmission nodes increases, better performance is observed. However, the improvement is less significant when too many transmission nodes are used. For example, c2_eDSR performs much better than c1_dsr, but c3_eDSR performs just slightly better than c2_eDS; and c4_eDSR does not guarantee an improvement in most of the cases. Generally, increasing the number of transmission nodes also increases the number of video descriptions used and the system complexity. All these factors must be taken into consideration to design the optimum system for video transmission. In this case, we believe using two transmission nodes is sufficient, although using three transmission nodes gives the optimum results.

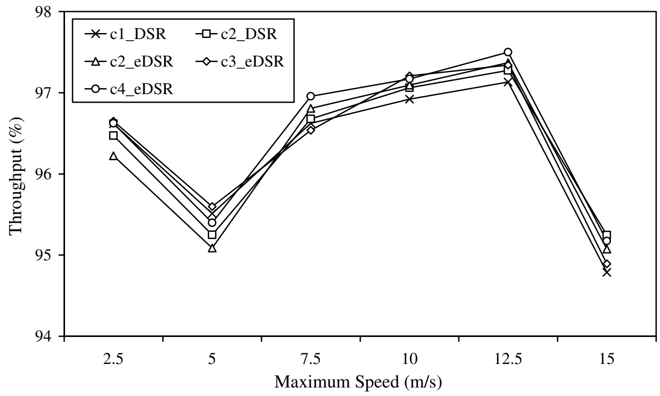

Fig. 12. Evaluation on non-video throughput at various levels of mobility.

As mentioned earlier, the extension added is only used for video traffic but it is possible that the extra routing overhead created may affect the non-video traffic as well. In order to inspect this criterion, we measure the throughput of the background traffics. From Fig. 12, we can see that the difference in throughput for non-video traffic is very small for each case, which is less than 0.5%. Generally, the extra overhead created is of very small potion and it does not take too much of the processing power of each node. Therefore, the influence is very small. Besides, our proposal tends to distribute the video traffic evenly within the network to avoid local congestion. This indirectly also reduces the losses of non-video packets. Therefore, the negative and positive impacts are offset and it is fair to conclude that our proposal has insignificant influence on the non-video traffics.

6. Conclusions and future work

There are two main contributions of our paper. First, we have proposed the use of multipoint-to-point as an alternative to conventional point-to-point transmission to increase the quality of real-time video streaming over mobile ad hoc networks. Our work is supported by simulation study using real video trace and comprehensive analysis has been carried out to observe the occurrence of every interruption during the video viewing. We have shown that using multipoint-to-point transmission not only improves the overall video quality, but also reduces the number of interruptions and shortens the length of the interruptions. This improvement is achieved by the combination of the extended DSR protocol proposed in this paper and the use of MDC coding. It is important to note that using MDC coding alone does not necessarily give better video quality because if the extra workload created may degrade the network performance if it is not handle carefully. Consequently, we have extended the DSR proposal to find disjoint paths for these transmission points to reduce the negative influence of the extra workload.

Second, our simulation study shows that using three video sources gives the optimum results. However, the largest difference is achieved when the number of transmission nodes is increased from one to two. Meanwhile, using four transmission nodes does not perform better than using three nodes. This is in contrast with multipoint-to-point transmission over wired networks, in which many concurrent connections help in achieving higher data rate. With this in mind, we suggest that using two or three transmission nodes is practical depending on the requirements; however, the tradeoff between system complexity and performance should be taken into consideration.

Possible future work includes supporting multicast video transmission using multiple video sources. As commonly known, it is not cost effective to send the same data unicastly to several receivers; therefore, we will investigate the possibility of developing multipoint-to-multipoint transmission over mobile ad hoc networks using disjoint trees. In this work, the most challenging part is the establishment of multiple disjoint trees. On the other hand, the extended routing protocol can be further enhanced by reducing the routing load to guarantee better delivery ratio, for example, by using cross-layer design.

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