

## PAPER

# A Novel Approach to Supporting Multipoint-to-Point Video Transmission over Wireless Ad Hoc Networks

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**SUMMARY** It is predicted that there will be a high demand for video applications in future wireless networks including wireless ad hoc networks. However, supporting video applications over mobile ad hoc networks is more complicated than with other networks due to the lack of support from a preinstalled infrastructure. In this paper, we tackle this problem by adopting the concept of multipoint-to-point video transmission used in wire-line networks. A novel framework designed with features to accommodate the characteristics of ad hoc networks is presented. There are two key features in our proposal. First, Multiple Description Coding (MDC) scheme is used for video coding to reduce the redundancy by avoiding the transmission of duplicate video frames. Second, the routing protocol is expanded to include finding disjoint routes from video sources to the receiver so that a single link breakage or a single intermediate node failure affects transmission from only the minimum number of nodes. Furthermore, the use of disjoint routes also enables the workload to be distributed more evenly within the network. A simulation study was carried out using NS-2 to demonstrate the performance of the proposed mechanism. We show that the proposed mechanism outperforms conventional point-to-point transmission, especially under conditions of high mobility.

**key words:** *mobile ad hoc networks (MANETs), video transmission, multipoint-to-point (MP2P) transmission, multiple description coding (MDC), dynamic source routing (DSR)*

## 1. Introduction

A Mobile ad hoc network is defined as a group of wireless devices that are capable of organizing themselves in a mesh topology in order to find routes and relay packets from each node to any other node within the network without the support of a pre-installed infrastructure. The deployment of these networks is expected to take place in critical environments such as during disaster or on battlefields, where pre-installed infrastructure is not available. Recently, mobile ad hoc networks have emerged as a promising solution for providing ubiquitous communications [1], [2] and their ability to support various multimedia applications, including video, is becoming a topic of intense interest [3]–[5]. Video transport has rigid requirements on bandwidth, delay and jitter. The commonly used low bit rate video compression methods can achieve high compression efficiency and good quality reconstruction under error free or nearly error free transmission. However, the quality drops drastically when the error rate increases, which occurs regularly in lossy networks.

Many research activities have been conducted with the aim of providing good quality video streaming over lossy networks, and these include path diversity transmission [6]–[8], multipoint-to-point transmission [9], [10], and layered video transmission [11], [12] together with video coding schemes which are more error-resilient [13]–[17]. These techniques have been proven to be feasible in many types of lossy network but this may not hold true in mobile ad hoc networks. Video transmission over infrastructure-less ad hoc networks faces greater challenges due to node mobility, which causes continuous changes in topology. Furthermore, transmission loss due to attenuation, interference and fading, and the resultant error rate are usually higher in ad hoc networks than in conventional single-hop wireless networks because an end-to-end path in an ad hoc network is usually composed of several wireless connections.

To address the aforementioned problem, a number of solutions have been proposed for handling video transmission over mobile ad hoc networks. One of the solutions is to use multistream coding with multipath transport [5], [18]. The Multiple Description Coding (MDC) scheme is used to replace the conventional video coding scheme and generate several independent and equally important video streams, also known as video descriptions, in such a way that each description can be decoded independently to reconstitute the original video at low but acceptable quality, and any additional descriptions received increase the video quality above the minimum level [19]. Unfortunately, the strength of the MDC scheme is usually associated with a major drawback in that a larger bandwidth is needed to convey the video streams generated [20]. Therefore, using MDC alone does not necessarily achieve the objective of improving video quality, as demonstrated in [18]. In order to fully utilize the strength of the MDC scheme, these descriptions need to be transmitted over highly disjoint routes to ensure that a route failure affects only the minimum number of video descriptions. In this paper, we propose a framework that combines multipoint-to-point (MP2P) transmission with the MDC scheme to provide reliable video streaming over mobile ad hoc networks.

The rest of this paper is organized as follows. In Sect. 2, a literature review on the related work and a brief introduction to the concept of the MDC scheme are presented. A detailed explanation of the framework of multipoint-to-point video transmission is given in Sect. 3. An extended version of Dynamic Source Routing (DSR) protocol to support MP2P transmission is also presented in this section. Sec-

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tion 4 gives details of a simulation study using NS-2 and the evaluation parameters used. The simulation results are presented and discussed in Sect. 5. Finally, conclusions and possible future work are given in Sect. 6.

## 2. Background

### 2.1 Related Work

A considerable amount of work has been done in adopting the concept of diversity for video transmission over mobile ad hoc networks. Diversity can be achieved through multiflow transport, for example, through path diversity or server diversity. The concept of path diversity, commonly known as multipath transport, involves the splitting of the video into several parts and transmitting them via different paths [5], [18]. In [5], Mao et al. proposed three point-to-point video transport techniques that combined multistream video coding with multipath transport. These techniques are Feedback Based Reference Picture Selection (RPS), Layered Coding (LC) with selective Automatic Repeat Request (ARQ), and Multiple Description Motion Compensation Coding (MDMC). Their simulation demonstrated that the key factor for successful multipath transport is the use of disjoint routes to ensure that the losses experienced by each video stream are independent. For this purpose, they proposed an extended Dynamic Source Routing (DSR) protocol in order to ensure that two disjoint routes are maintained continuously between the video source and the receiver. In order to minimize packet lost due to topology changes, Wei and Zakhor proposed the Robust Multipath Source Routing Protocol (RMPSR), which builds a set of disjoint routes for a source-receiver pair instead of just maintaining only two disjoint routes [18]. In summary, both [5] and [18] have demonstrated the implementation of path diversity together with multistream coding schemes to improve the video quality.

Supporting multiflow transport not only involves issue at the network layer, but also at the higher layers, especially the application and transport layers. The existing real-time transport protocol (RTP) may not be able to support this framework efficiently due to the rapid topology changes and lack of reliable real-time connections in a mobile ad hoc network. In order to tackle this problem, Mao et al. proposed a multiflow real-time transport protocol (MRTP), which is an extension of RTP, to facilitate multiflow real-time video transmission over ad hoc networks [21]. MRTP is implemented at the application layer to provide essential support including session and flow management, data partitioning and reassembly, traffic dispersion, data framing, timing and quality-of-service feedback. The authors suggested three application scenarios for this protocol, including point-to-point multipath transmission, multipoint-to-point communication and multicast.

Apart from path diversity, the concept of server diversity, more commonly known as multipoint-to-point communication, is also a feasible solution for improving video qual-

ity. In [9] and [10], server diversity is employed to improve video quality over packet-lossy wire-line networks. However, for ad-hoc networks, only path diversity has been employed to improve video quality, as elaborated above, with respect to [5] and [18]. We believe that applying the server diversity concept to mobile ad hoc networks offers good potential, for two main reasons. First, the mesh topology of mobile ad hoc networks, where each node is connected virtually to other nodes within its transmission range, is itself a readily available topology for multipoint-to-point communication (server diversity). Second, it has been demonstrated that the introduction of MRTP provides a suitable platform at the application and transport layers for real-time multiflow transport [21]. However, in order to achieve optimum performance of the mobile ad hoc network, an efficient routing protocol at the network layer is required. Therefore, in this paper we propose a framework for server diversity in a mobile ad-hoc network, in order to tackle the routing issue at the network layer. Specifically, we introduce a routing protocol at the network layer for mobile ad-hoc networks, for multipoint-to-point communication. This work, which concentrates on routing at the network layer, is complementary to MRTP investigation on multiflow transport at the transport and application layers, and serves as an alternative to path diversity for video transmission.

### 2.2 Multiple Description Coding (MDC)

Conventionally, video coding aims to encode the raw video into a single stream of video frames suitable for storage and transmission. Generally, these coding schemes use motion-compensation prediction between frames, block-Discrete Cosine Transform (DCT) for error prediction, and entropy coding. One major problem of conventional video coding is the propagation of errors when a single frame error affects not only the current frame but also all subsequent frames, until a frame with new prediction is received. In order to combat this problem, layered coding is introduced to encode the video stream into two layers, i.e. a base layer and an 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 under a heavy traffic load, only the base layer is sent. This method is more robust but the same problem may occur if the base layer is lost, since the enhancement layer is totally useless by itself. Consequently, with the MDC scheme it is proposed to generate several equally important and totally independent video descriptions. 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 [19]. MDC is more error-resilient than the conventional methods and it has been proven to outperform conventional coding schemes in most lossy networks [22].

Figure 1 shows the general form of MDC coding. Initially, the raw video stream is divided into several sub-

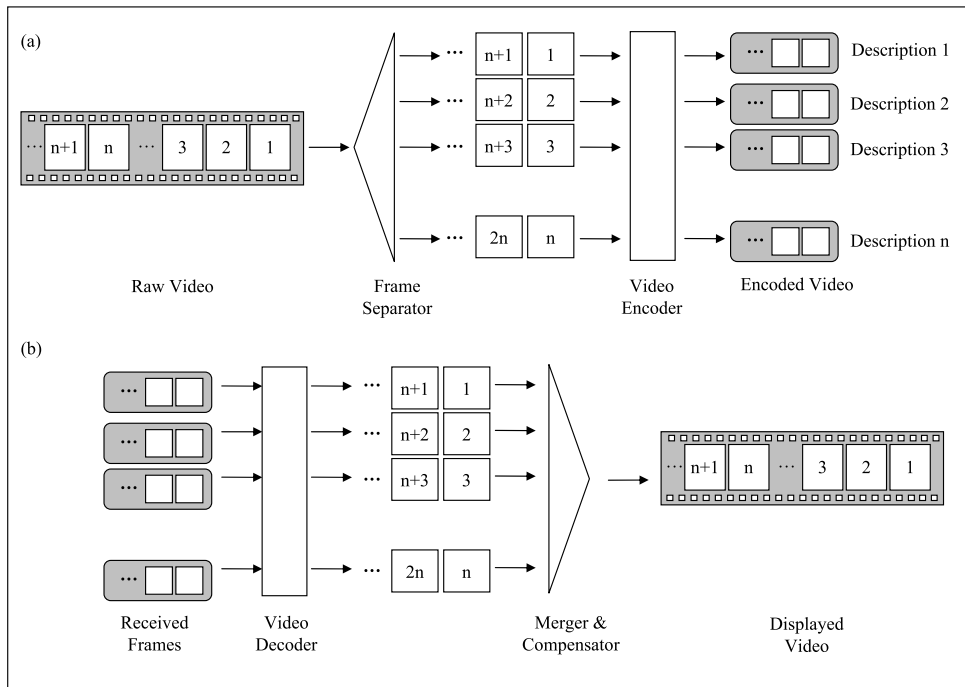


Fig. 1 Multiple Description Coding (MDC): (a) Encoding process; (b) Decoding process.

streams prior to the encoding stage. One of the simplest ways to do this is the per-frame allocation scheme as shown in Fig. 1(a) [23], [24]. Next, these sub-streams are encoded independently to obtain several video descriptions. There is no constraint on the selection of the type of encoder used; either a standard video encoder such as H.264 [25] and MPEG4, or a specifically designed MD encoder [23] may be used. As compared to a single video stream, a higher bandwidth is required to accommodate the MD encoded video due to the larger difference between neighboring frames within each sub-stream. This results in a lower compression efficiency and the resulting data size is larger. In [20], a comprehensive analysis of video encoded by the MDC scheme shows that the average frame size increases as the number of descriptions increases. The size of this increase depends on the content of the video; normally, high motion video produces a larger increase as the number of descriptions increases. The decoding process is illustrated in Fig. 1(b). Each description is decoded separately and the reconstructed video sub-streams are merged for viewing purpose. At this stage, a compensation scheme may be used to replace any missing frame with the one displayed previously. This compensation scheme can reduce distortion during display of the displaying.

### 3. The Proposed Solution

#### 3.1 Objective

A mobile ad hoc network has a mesh topology where each node is connected virtually to all other nodes within its transmission range. Consequently, every node can obtain

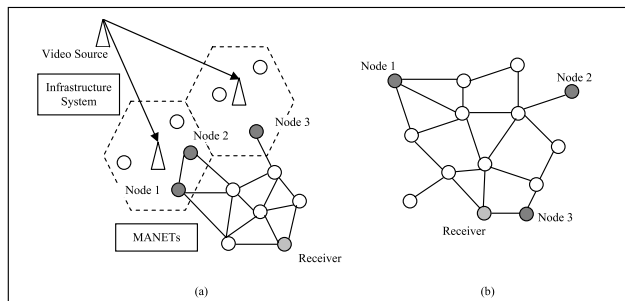


Fig. 2 MP2P video transmission over MANETs.

data freely from any other node as long as a valid route can be established between them. In addition, if the same information is available from several nodes within the network, a node can retrieve the information at a higher rate by downloading it simultaneously from several nodes, which is similar to the concept used in peer-to-peer networks. In light of this, we propose multipoint-to-point (MP2P) transmission in this paper. Figure 2 illustrates the concept of the proposed mechanism for different types of video application over mobile ad hoc networks. Figure 2(a) shows video applications such as video-on-demand, live streaming and video conferencing. Figure 2(b) demonstrates download-and-play video using file-sharing concept of peer-to-peer networks. On top of MP2P transmission, the MDC scheme is used for video coding and each video source sends different video descriptions to minimize the workload of the network. In order to fully utilize the strength of the MDC scheme, disjoint routes should be used to send these video descriptions. This leads to the need to provide the routing protocol with the ability

to obtain disjoint routes for these video sources.

On the other hand, we must limit the number of concurrent connections to a receiver. In wired peer-to-peer networks, a greedy mechanism is used where the receiver tends to establish many concurrent connections, in order to achieve the highest possible data rate. This greedy behavior should be avoided in ad hoc networks because the same transmission medium is shared among all the mobile nodes, and so too many concurrent connections to a single node can notably degrade the overall performance of the network. Besides, it is not cost effective to generate many video descriptions considering the excessive redundancy on account of the increased average frame size mentioned earlier in the coded video. Studies have shown that the use of two video descriptions can maintain the compression efficiency and at the same time increase the error resilience. Therefore, in this paper we use only two video sources and two video descriptions. Our preliminary work showed that MP2P improved the quality of video transmission over ad hoc network by increasing the overall video quality and decreasing the occurrence of interruptions during video [26].

### 3.2 Novel Architecture

The basic idea behind our proposal is to implement concurrent video streaming from several video sources to a single receiver using disjoint routes. Figure 3 shows the novel architecture of our proposal. In this, the receiver is given the responsibility of selecting the video sources and instructing them to send different video descriptions. In addition, the receiver also needs to ensure that the received video frames are decoded correctly for viewing purpose. In general, the MP2P framework can be divided into three stages and the timeline of packets exchanged at each stage is given in Fig. 4.

#### 3.2.1 Source Searching Stage

Both source-initiated and receiver-initiated searching mechanism can be used to locate the video sources but the receiver-initiated approach is used in this paper in order to synchronize the starting time of the subsequent stages and to avoid large difference in the starting time of the video transmission of the different video sources. In Fig. 4, the receiver initiates the source searching by broadcasting a video request packet (VREQ) to the networks and a timer is activated. Nodes with the desired video may reply by using a video reply packet (VREP), which can be sent out broadcast or unicast using the reversed route. Since the DSR protocol is used in this paper, we allow the VREQ packets to record the route traversed and therefore, the VREP is returned unicast along the reverse route. Upon the timeout of the waiting period, the receiver selects two most suitable nodes to become the video sources; in our case, we select the video sources based on the arrival time of the VREP. We name these nodes S1 and S2, where the reply from S1 arrives before S2. If only one node has replied to this request be-

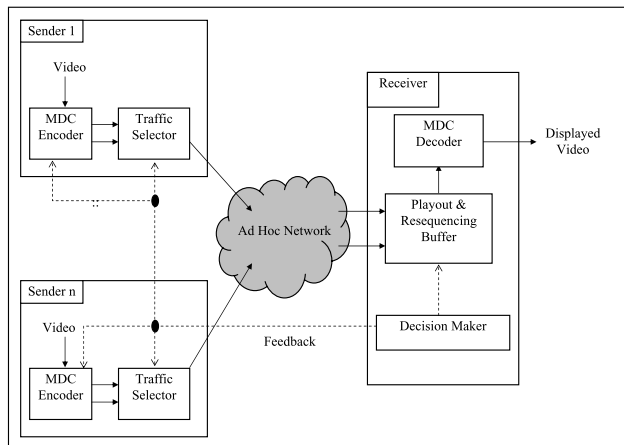


Fig. 3 Novel architecture of MP2P video transmission.

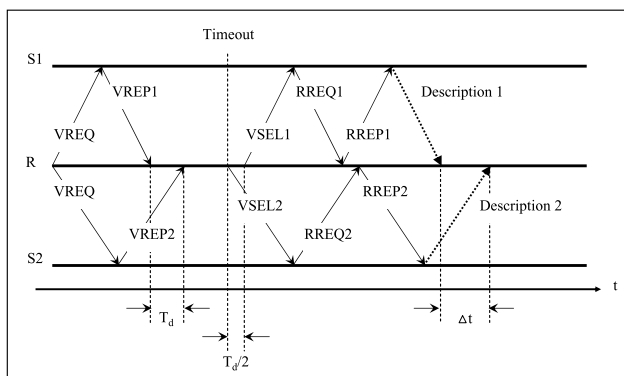


Fig. 4 Exchange of control packets in MP2P video transmission.

fore the timeout, normal point-to-point video transmission is used. Finally, the selected video sources are informed using the video select packets (VSEL). Each VSEL carries information used for MD coding, i.e. the number of video descriptions to be created and which description is to be sent. For point-to-point transmission, both values are set to one. In order to help in the discovery of disjoint routes in the next stage, the identity of the other video source (for example, the IP address) should be carried by the VSEL packets as well.

The sending of VSEL packets is associated with a simple mechanism to synchronize the starting time of the route discovery stage at both video sources. VSEL to S1 is sent out  $T_d/2$  after VSEL to S2 is sent out, where  $T_d$  is the difference between arrival times of VREP1 and VREP2. The objective of this mechanism is to ensure that both video sources start their route discovery at virtually the same time and consequently, the starting of video streaming will also be virtually concurrent, as shown in Fig. 4. We provide only a loose synchronization here because a more precise synchronization would involve a greater number of packet exchanges and would subsequently increase the control overhead and result in higher delay in the video streaming. Besides, our mechanism together with a small playout buffer at the receiver is sufficient to provide best-effort video at good

quality, as demonstrated in our simulation study.

### 3.2.2 Route Discovery Stage

The aim of this stage is to obtain routes connecting the video sources and receiver. These routes should be optimally disjoint in order to fully utilize the strengths of the MDC scheme as mentioned in the previous section. For this purpose, we have extended the DSR protocol accordingly, to provide it with the ability to discover disjoint routes for nodes sending the same video to the same receiver; we call it the MP2P-DSR protocol. We have chosen the DSR protocol because it makes the discovery of disjoint routes relatively easy and allows the senders to determine the route traveled by each data packet [27]. Furthermore, this routing protocol has been proven to be efficient and good performance can be achieved under different network conditions [28]. A detailed description of the extended DSR protocol is given in the next subsection.

### 3.2.3 Video Streaming Stage

We assume the same MDC codec is installed in every node. The video sources encode the video frames and send out only one video description based on the instruction carried by the VSEL. The video streaming is started once a valid route to the receiver is available and no further synchronization is performed here. Instead, we use a playout buffer at the receiver to absorb the difference in arrival time of video packets, and for re-sequencing purposes. The playout buffer is activated after the first video packet arrives at the receiver. Packets which arrive after their deadline are discarded because in this paper we are dealing with best-effort services. However, the size of the playout buffer should be carefully designed by taking into considering the maximum number of hops can be handled by the DSR protocol (MAX\_SR\_LEN). Generally, it is safe to design the playout buffer by considering the worst case, in which S1 is one hop away from the receiver and S2 is MAX\_SR\_LEN hops away from the receiver. From Fig. 4 and by assuming the route discovery stage is started at the same time at both video sources, we obtain

$$\Delta t_{max} = 3 \times (\text{MAX\_SR\_LEN} - 1) \times \text{NODE\_TRAVERSAL\_TIME} \quad (1)$$

where NODE\_TRAVERSAL\_TIME is a conservative estimate of the average one-hop traversal time for a packet including queuing delays, interrupt processing times and transfer times; a common value for this parameter is 40 ms. The buffer size should be larger than  $\Delta t_{max}$ , to also take account of the time for frames re-sequencing.

## 3.3 Extended DSR Protocol for MP2P Transmission (MP2P-DSR)

The DSR protocol is an on-demand routing protocol for

multi-hops wireless ad hoc networks in which the end-to-end route from source to destination is carried in the packet header. In order to accommodate our proposal, we extend the routing protocol to find disjoint routes for different video sources. It is important to note that this extension is only applied to video applications and non-video applications should follow the original DSR protocol. Basically, the routing protocol starts with a route discovery mechanism initiated by the requestor (video source) to find valid routes to a destination (receiver). This mechanism involves the exchange of a route request packet (RREQ) and route reply packet (RREP). Readers are referred to [27] for a complete explanation of the DSR protocol. As explained in the previous section, both requestors are synchronized to start the route discovery at almost the same time. Generally, our proposal consists of two mechanisms, i.e. preventive and corrective mechanisms.

The preventive mechanism aims to avoid the convergence of both video streams at the same intermediate node by encouraging the discovery of disjoint routes for the different video sources during the route discovery stage. For this purpose, the following modifications have been introduced:

- i) Each duplicated RREQ is not discarded immediately at an intermediate node. Instead, the route it traversed is analyzed to provide a backup route if it is highly disjoint with the route traversed by the first RREQ, and better than the existing backup route, if available. This backup route is used to provide another alternative route for the requestor, as shown in Fig. 5(a). At intermediate node X, RREP2 is created after RREP1 is received and forwarded. The route carried by RREP2 is Route 2 and the remaining route from node X to node D as recorded in RREP1. This method provides more disjoint routes for the requestor.
- ii) The original DSR protocol allows the destination node to reply to all RREQs that arrive, and this creates a scenario where most or all immediate (one-hop) neighbors of the destination are involved in the route created for one requestor. Consequently, route discovered by the other requestor may have low disjointness. In order to overcome this problem, selective reply is introduced in our proposal. If a route has high ratio of shared-node with route which replies have already been sent to the same requestor, this RREQ is discarded, as shown in Fig. 5(b).
- iii) Each route for which the destination sends a reply is given a 'priority.' The priority is determined by comparing the route traversed by the RREQ with routes for which replies have previously been sent to the other requestor (video source). If the shared-node ratio is below certain threshold, it means that this route is highly disjoint with routes used by the other video source; therefore, it is given high priority. This priority is carried by RREP back to the requestor. The video sources always use the shortest high-priority route for video transmission if available.
- iv) Video sources cannot use route that includes the other video source in order to avoid an excessive workload at one video source.

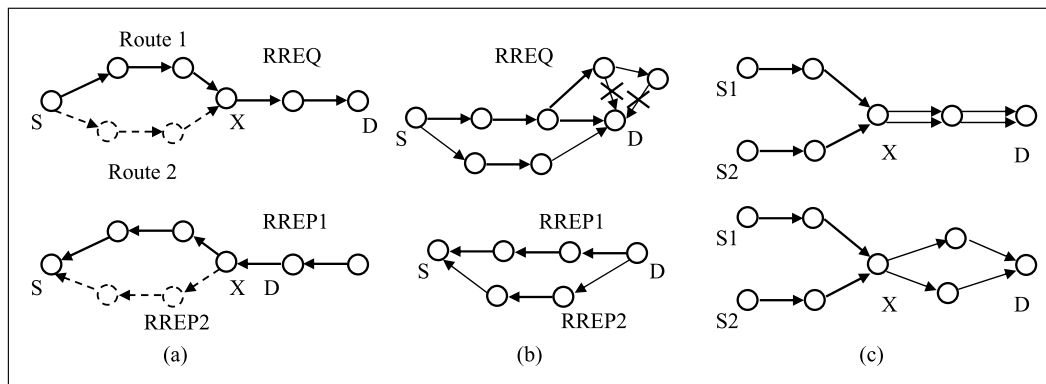


Fig. 5 Modifications introduced in the MP2P-DSR protocol.

Clearly, the preventive mechanism encourages node disjointness in video transmission, but with no guarantee because the topology may be such that no fully node-disjoint route exists. With this in mind, we propose a corrective mechanism to encourage link disjointness in the event that preventive mechanism fails to provide this. As shown in Fig. 5(c), the corrective mechanism is activated when both video streams are transmitted via the same intermediate node X, which is not an immediate neighbor of the destination, and would continue their flows over the same route to the destination. When this happens, node X should find an alternative route to the destination from its route cache and transmit one of the video streams using this route. In order to avoid unnecessary delay in video transmission, the length of the alternative route should not be longer than the original one.

#### 4. Simulation Study

##### 4.1 Simulation Scenario

A simulation study was carried out using NS-2 [29] with CMU wireless extension [30] to compare the following three cases: (C1) Point-to-point transmission using DSR, (C2) Multipoint-to-point transmission using DSR, and (C3) Multipoint-to-point transmission using MP2P-DSR. The simulator uses 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 Mb/s and the transmission range is 250 meters.

The simulation topology contained 50 nodes within an area of 1500 meters by 800 meters. The node movement was modeled using the enhanced Random Waypoint Model [31]. The minimum traveling speed was set to 0.1 m/s and the maximum speed was varied between 2.5 m/s and 15.0 m/s to model different levels of node mobility. For each level of mobility, 30 scenarios were generated prior to the simulation so that identical scenarios could be re-used for each case to ensure fairness in the comparison study. In each

Table 1 Statistical results of the scenarios used.

Max. Speed (m/s)	2.5	5.0	7.5	10.0	12.5	15.0
AND	7.5	7.8	7.6	8.1	8.0	7.9
RLC	0.3	0.6	0.9	1.0	1.3	1.5
LSP	10.3	10.1	10.7	10.7	10.9	11.2

scenario, each node is randomly assigned with an initial location, a destination and a traveling speed, the speeds being uniformly distributed between the given minimum and maximum speeds. During the simulation, these nodes are assumed to 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, and the movement continues. This is repeated until the end of the simulation and only continuous mobility with no pause is considered. Table 1 shows statistical analysis for the scenarios generated for references in the next session. Average Node Degree (AND) is the number of one-hop neighbors of a node, and is averaged over all nodes and over the duration of the simulation. Rate of Link Changes (RLC) is the number of times the connection between any two nodes is formed and broken divided by the simulation duration. Length of Longest Shortest-Path (LSP) is the distance, in number of hops, of the longest shortest-path encountered by any two nodes within the simulation duration. These values are the average values for all 30 scenarios. It can be observed that the AND and the LSP are independent of node mobility, and as expected, the RLC increases with node mobility. To represent the background traffic, five constant bit rate (CBR) connections of 18 kb/s each were introduced randomly in the networks. UDP was used in the transport layer.

The video sources and receiver were chosen randomly among the 50 nodes and recorded so that the same nodes were used for each case to maintain the fairness of the comparison study. The video sequence used was *highway* in quarter common intermediate format (qcif) [176 × 144 pixels/frame]. The first 1800 frames were used and they were encoded at 12 frames per second (f/s) using the H.26L video codec for both the SDC and MDC schemes [25]. The Group of Pictures (GOP) size was 12 and the default coding parameters were used for both SDC and MDC

schemes to maintain the fairness of the comparison study. For MDC scheme, we used the per-frame separator in [24] to divide the raw video frames into two groups prior to implementing video coding as explained in Sect. 2.2, and the EvalVid toolset was used to obtain the video trace for the simulation [32]. The average peak-signal-to-noise ratio for the encoded video frame was 43.27 dB, and the average frame sizes were 4143 and 4657 bits for SDC and MDC, respectively. Clearly, the MDC scheme adds redundancy to the encoded frames as explained in [20]. By using MAX\_SR\_LEN and NODE\_TRAVERSAL\_TIME equal to 20 hops and 40 ms, respectively, we obtain  $\Delta t_{max}$  as equal to 2.28 s using Eq. (1). In order to accommodate time elapsed for frame re-sequencing, which should be of moderate level [5], we conservatively set the buffer size to 5 s, a practical waiting time for real-time video. Each simulation ran for 200 s and the results were averaged over the 30 scenarios.

## 4.2 Performance Metrics

The following parameters were used:

(a) **Average Peak Signal-to-Noise Ratio (PSNR)** is calculated by averaging the PSNR of the decoded frames, which is calculated by comparing the luminance value of each reconstructed frame with its corresponding original frame. This parameter represents the quality of the decoded video as a whole.

(b) **Normalized Routing Load (NRL)** is the ratio of the total number of routing packets forwarded at each node to the total number of data packets received, expressed as a percentage. This parameter measures the efficiency of the routing protocols. In the calculation, the data packets include both video and non-video packets because the routing information is used by both types of traffic, so they cannot be differentiated. This calculation does not affect the fairness of our comparison study because the number of data packets is the same for all cases.

(c) **Normalized Control Overhead (NCO)** is the ratio of the total control overhead created during the source searching and route discovery stages to the total number of data packets received. This parameter is also expressed as a percentage and it represents the efficiency of the proposed framework as a whole.

(d) **Total Time elapsed for source searching ( $T_e$ )** measures the amount of time spent on source searching.

(e) **Average End-to-End Delay** is the average end-to-end delay experienced by the video packets. This parameter gives an indication on the additional delay caused by our proposed routing protocol.

(f) **Throughput of Non-Video Traffic** is also measured, to observe the influence of our proposal on non-video applications, *i.e.* the background traffic.

(g) **Number of Bad Frames** specifies the quantity of decoded bad quality frames. For this purpose, we perform a PSNR to Mean Opinion Score (MOS) mapping for the video frames as recommended by ITU-T, and given in Table 2 [33]. We further simplify the classification into two

**Table 2** The PSNR to MOS mapping.

MOS	1	2	3	4	5
PSNR	< 20	20–25	25–31	31–37	> 37
Quality	Bad	Poor	Fair	Good	Excellent

categories only; a frame is considered as a bad frame if its MOS is below 4.

(h) **Number of Interruptions** measures how frequent distortions occur during the video viewing. An interruption is observed when a number of bad frames are decoded consecutively. Naturally, the human visual system cannot notice the distortion if the interruption lasts for a very short period of time. In this paper, we assume an interruption is noticeable only if it lasts for at least 0.5 s.

## 5. Results and Discussion

Figure 6 shows the average PSNR, which represents the overall video quality. As may be seen, MP2P (C2 and C3) provides a better overall performance than P2P transmission (C1). Further improvement is achieved by using a routing protocol that uses disjoint routes for video transmission, as demonstrated in C3, where the improvement over C1 is between 0.2 to 0.95 dB. In addition, it can be observed that C3 provides a more stable performance over various levels of mobility. The fluctuation is only 0.35 dB for C3, and 0.5 dB for C2, but as high as 0.9 dB in C1. This also leads to the observation that our proposal provides larger improvement under cases of high mobility. Since the video quality is directly proportional to the quantity of video frames successfully received, there are two factors that contribute to this improvement. First, the route discovery mechanism is enhanced to obtain more disjoint routes for the video sources, and using disjoint routes can achieve better workload balancing within the network, which leads to less congestion. Second, MD coded video has better error resilience and the impact of frames lost is generally smaller. The combination of these two factors contributes to the improvement of the overall video quality.

Figure 7 shows the normalized routing load for the three cases. As expected, our proposal has the highest routing load due to the additional routing packets created during the route discovery mechanism to find disjoint routes. The additional normalized routing load is about 2–4%. It is also of interest to observe the overall control overhead which involves the source searching stage as well as the route discovery, as presented in Fig. 8. It can be seen that Figs. 7 and 8 exhibit the same pattern with small difference, which is only about 0.5%. The additional control overhead created during source searching stage is relatively small because the source searching stage is executed only once at the beginning of the simulation but the route discovery and route re-establishment are carried out whenever the topology changes. From Figs. 6, 7 and 8, we observe a tradeoff between control overhead and throughput; the higher control overhead in our proposal has led to better video performance. However, it is important to note that excessive con-

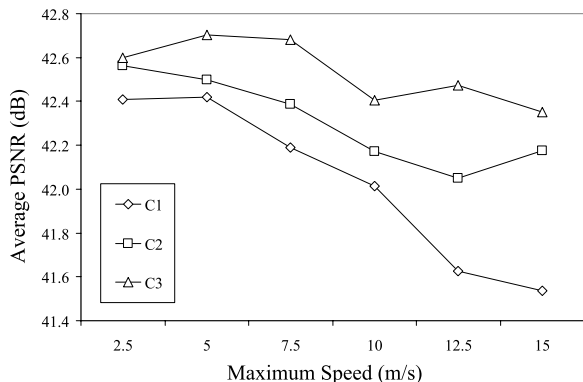


Fig. 6 Simulation results: Average PSNR.

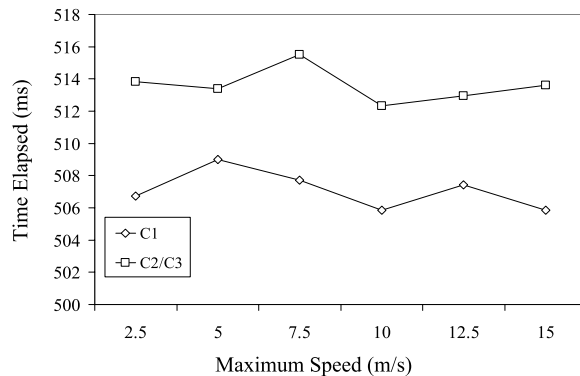


Fig. 9 Simulation results: Time elapsed for the source searching stage.

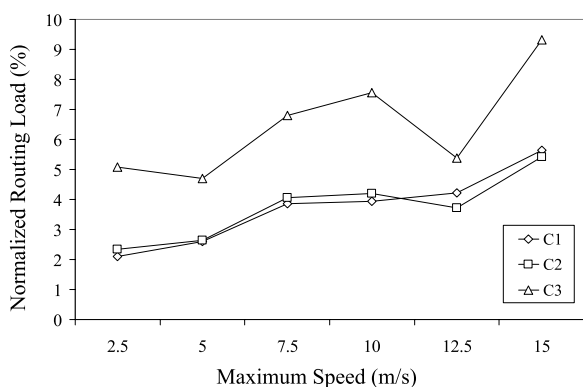


Fig. 7 Simulation results: Normalized routing load.

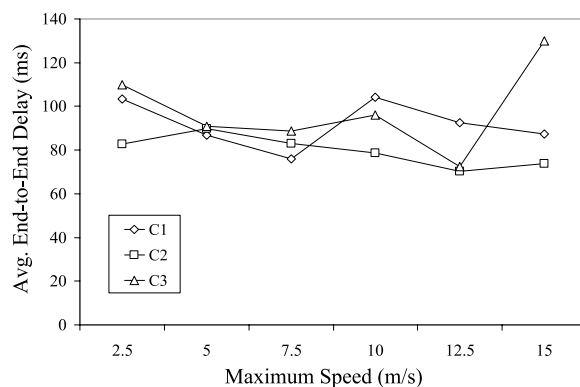


Fig. 10 Simulation results: Average end-to-end delay.

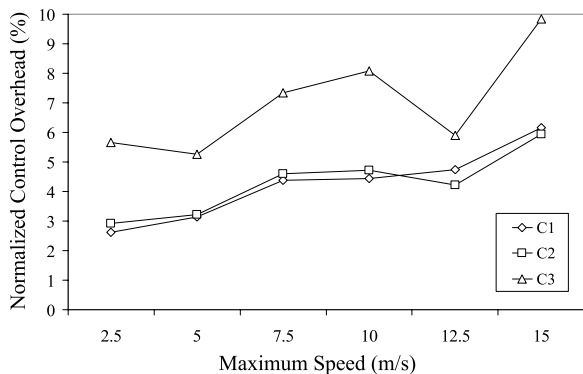


Fig. 8 Simulation results: Normalized control overhead.

control overhead may lead to a drastic drop in the overall network performance. Therefore, we limit the number of video sources in our proposal in order to limit the control overhead. Clearly, using two source nodes is a practical compromise considering the tradeoff between control overhead and throughput.

Figure 9 shows the total time elapsed for source searching ( $T_e$ ). Since cases C2 and C3 use the same source searching mechanism, they have the same value for this parameter. Source discovery for the multipoint-to-point framework introduces a slight increase in delay of about 10 ms. This delay is due to the need to locate two video sources and

for synchronization purposes as discussed in Sect. 3.2.1. In order to consider the delay caused by the modified routing protocol and the video transmission, the average end-to-end delay for video packets is shown in Fig. 10. For most levels of mobility, C3 has the highest delay, which is about 20 ms higher than the other cases for all levels of mobility, except for the highest speed of 15.0 m/s, where the increase rises to 60 ms. In summary, our proposal introduces higher overall delay in the system, but fortunately, the worst case of this additional delay is only about 70 ms (60 ms + 10 ms) and this only occurs for the maximum speed of 15.0 m/s. We believe the impact of this delay is small considering the size of the playout buffer, which is almost ten times greater.

As mentioned earlier, the MP2P extension added to the DSR protocol is only used by the video traffic but it is still possible that the extra routing overhead created, as discussed in the previous paragraph, may affect the non-video traffic as well. In order to check this possibility, the throughput of the background traffic is also plotted. As seen in Fig. 11, the difference in throughput for non-video traffic is very small, less than 0.5%. Generally, the extra overhead created takes only a small portion of the processing power at each node. Therefore, the influence is small. Besides, our proposal tends to distribute the video traffic evenly within the network to avoid local congestion. This directly reduces the loss of non-video packets due to congestion. Therefore, the negative and positive impacts are offset and it is fair to

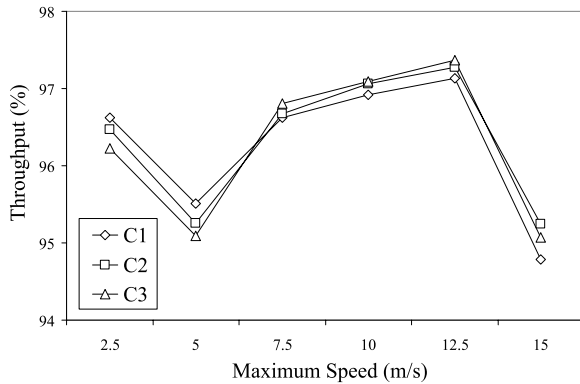


Fig. 11 Simulation results: Non-video throughput.

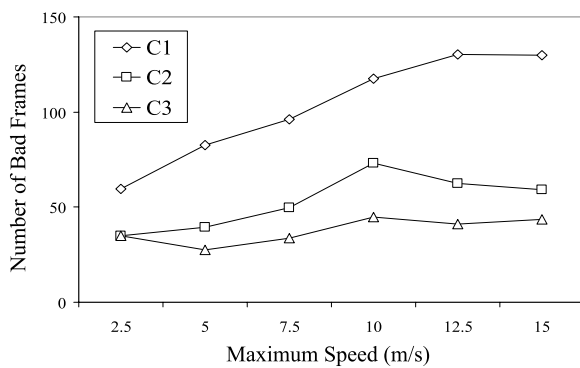


Fig. 12 Simulation results: Number of bad frames.

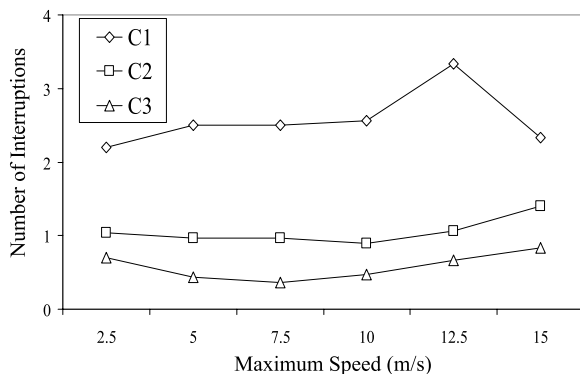


Fig. 13 Simulation results: Number of interruptions.

conclude that our proposal has insignificant influence on the non-video traffic.

Next, we inspect the video quality from a more detailed perspective. Figure 12 shows the total number of bad frames received. For C3, the number of bad frames is always below 50, which is less than 3% of the total number of frames. For C2, the number is slightly higher but the performance is still quite stable with respect to speed of mobility. Once again, C1 shows a large variation in performance with the bad frame ratio growing to almost 8% under the highest mobility. Figure 13 shows the number of interruptions, and from these results it is fair to say that our proposal gives almost

perfect video viewing because the average number of interruptions is below one. C2 resulted in an average of about one interruption per video session while C1 resulted in more than two interruptions per video session. These observations can be explained by checking on the rate of link changes (RLC) in the topology, as given in Table 1. At low mobility, the connection between video source and receiver is stable and using a single node is enough to provide video viewing with good quality. As mobility increases, the connection between the video source and receiver breaks frequently, but with MP2P transmission, the probability of losing connection to both video sources simultaneously is smaller than with point-to-point transmission, and thus a smaller number of interruptions is recorded. Furthermore, the use of disjoint routes in C3 further increases the probability of receiving at least one video description, and this further reduces the occurrence of bad frames and interruptions.

## 6. Conclusions and Future Work

The issue of video streaming over mobile ad hoc networks has been studied in this paper. We have adapted the concept of multipoint-to-point video transmission used in wire-line networks to wireless ad hoc networks with some modifications to take account of the differences between them. The contribution of this paper is twofold. First, we presented a novel architecture for multipoint-to-point video transmission over mobile ad hoc networks by considering three discrete stages at the network layer. Second, we extended the routing protocol to handle video transmission from multipoint to a single receiver. The main objective of this extension is to use disjoint routes for the video transmission, so that the strengths of the MDC scheme are fully utilized. Moreover, our routing protocol takes into consideration both node-disjointness and route-disjointness to achieve the best possible workload distribution within the network. To verify the feasibility of our proposal, a simulation study using real video sequence was carried out. The simulation study has shown that the proposed multipoint-to-point transmission outperforms point-to-point transmission in terms of the overall video quality and the number of interruptions during the viewing stage, especially under high mobility.

Future work includes studying multicast video transmission using multiple video sources. As well known, it is not cost effective to send the same data unicast to several receivers; therefore, we will investigate the possibility of developing multipoint-to-multipoint transmission over mobile ad hoc networks.

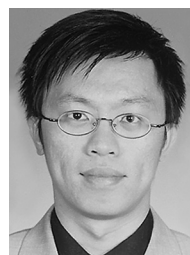
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