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# The Feedback Based Mechanism for Video Streaming Over Multipath Ad Hoc Networks

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## Abstract

Ad hoc networks are multi-hop wireless networks without a pre-installed infrastructure. Such networks are widely used in military applications and in emergency situations as they permit the establishment of a communication network at very short notice with a very low cost. Video is very sensitive for packet loss and wireless ad-hoc networks are error prone due to node mobility and weak links. High quality video transport under ad hoc network is a challenging task due to low bandwidth, high loss rate, unpredictable node mobility, and severe interference characteristics of such kind of network. End to end packet delivery often exhibits low performance due to one hop-by-hop routing failure and misinterpretation to such failures. Multi path routing has been used to increase the robustness for video over ad-hoc networks. The key idea of proposed schema is based on sending video packets over two disjoint paths beside using buffering technique in special nodes of network. In each of every path there is only one node selected as cache node. The main task of these nodes is to realizing different packet types, buffering some of important video packets, reducing forward traffic rate while detecting loss in network and local error management to overcome high loss rate of video packets. Selecting these nodes in network can be as an agreement between video sender and receiver or based on network topological status. The goal of this work is to reduce network end to end delay and increased quality of service over application layer.

**Keywords:** Ad Hoc networks; Buffering technique; Video cache nodes; Feedback mechanism

## Introduction

In coming years, mobile computing will keep flourishing, and an eventual seamless integration of Ad Hoc with other wireless networks, and the fixed Internet infrastructure, appears inevitable. An ad hoc wireless network is a collection of mobile nodes that can communicate with each other over radio channel in the absence of any infrastructure. If two nodes lie within the transmission range of each other, then they can communicate directly. Two nodes that cannot directly communicate, can do so in a multi-hop manner in which

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the other intermediate nodes function as routers. Such networks are used in military applications and in emergency situations as they permit the establishment of a communication network at very short notice time with a very low cost. However, these networks are limited by constraints in their bandwidth and power consumption. With their widespread deployment, ad hoc wireless networks now need to support applications that generate real-time traffic. Applications such as voice communication, video on-demand, video conferencing, and radio broadcasting require the network to provide guarantees on the Quality of Service (QoS) of the connection. Video communications over wireless networks often suffer from various errors. Wireless links have limited bandwidth and are prone to unexpected congestion and loss of connection, making end-to-end video transmission low delay and high quality hard to achieve. Generally in order to combat effects of errors, there are two methods: Error Resilient and Error Concealment methods.

The main goal of former methods is to make video resistant so at receiver side it is possible to detect and correct errors resulted in video transmission. In later methods, correcting distortion of video streams can be variable due to kind of Error Resilient methods. However comparing two mentioned methods we can divide them into two groups: first, methods have no attention on transmission and second, methods that can be selected supposing transmission conditions [1,2]. Wireless links are frequently broken and new links reestablished due to mobility. Furthermore, a wireless link has a high transmission error rate because of shadowing, fading, path loss, and interference from other transmitting users. In wireless environment, for efficient video transport, traditional error control techniques, including forward error correction [3,4] and automatic repeat request [5], should be adapted to take into consideration frequent link failures and high transmission errors. Different techniques offered for improving video quality in Ad hoc networks. Among various mechanisms, *multipath transport*, by which multiple paths are used to transfer data for an end-to-end session, is highly suitable for ad hoc networks, where a mesh topology implies the existence of multiple paths for any pair of source and destination nodes. These methods combined with appropriate source and/or channel coding and error control schemes, can significantly improve the media quality over traditional shortest-path routing-based schemes. This also inspired previous and ongoing standardization efforts for multipath transport protocols in the Internet Engineering Task Force [6,7]. In this Paper we presents effects of using buffering technique beside using local error management at special points of network that we call proxy (cache) nodes. The main goal of our work is to reduce end-to-end delay and increased quality of received video at receiver over application layer.

### *Multipath Multimedia Transport Architecture Overview*

The general architecture for multipath transport of video streams is depicted in Figure 1. At sender, raw video is first compressed by a video encoder into $M$ streams. When $M > 1$, we call the coder a *multistream coder*. Then the streams are partitioned and assigned to $K$ paths by a *traffic allocator*. These paths are maintained by a *multipath routing protocol*. When the flows arrive at the receiver, they are first put into a *resequencing buffer* to restore the original order. Finally, the video data is extracted from the resequencing buffer to be decoded and displayed. The video decoder is expected to perform appropriate error concealment if any part of a substream is lost.

In general, the quality of the paths may change over time. We assume that the system receives feedback about network QoS parameters. Although not necessary, such feedback can be used to adapt the coder and transport mechanisms to network conditions (e.g., the encoder could perform rate control based on feedback information, in order to avoid congestion in the network). The number of available paths, as well as their bandwidths, may vary over time due to network topology changes and congestion. The point-to-point architecture in Figure 1 can be used for two-way conversational services as well as one-way streaming services. For the latter case, it can be extended to more general cases. Effect of using multipath transmission against single path is exhibited in Figure 2.

### *Issues and Challenges*

Multipath Transport (MPT) has been studied in the past in wireline networks for (i) increased aggregate capacity, (ii) better load balancing, and (iii) path redundancy for failure recovery [8-10]. The research effort on MPT can be roughly divided into the following two categories:

1) Multi-path Routing, which focuses on finding multiple routes for a source-destination pair, and on how to select a maximally disjoint set of routes from the multiple routes found [11-14].

2) Traffic Dispersion, which focuses on how to allocate traffic to multiple end-to-end routes [15,16]. Generally traffic dispersion can be performed with different granularities. Ref. [17] is an excellent survey

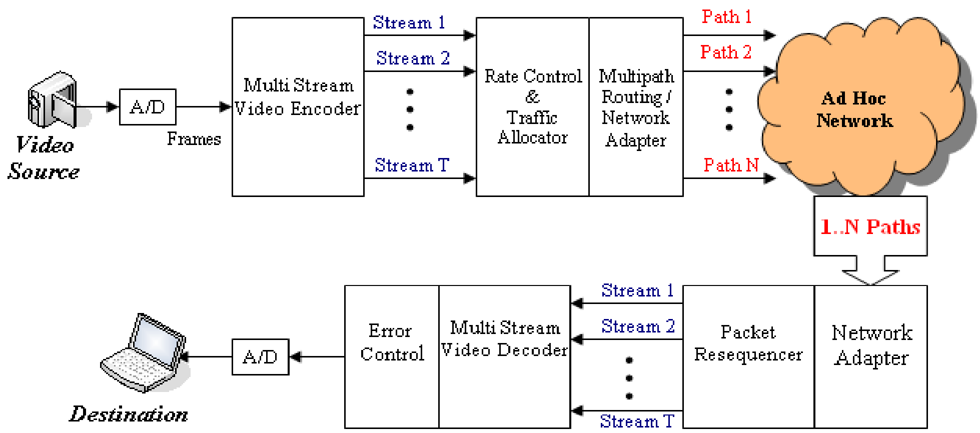


**Figure 1.** The general architecture for the multipath transport of real time multimedia applications.


**Figure 2.** Comparison of transferring video over different paths. a(1 path 80 kbps, PSNR 32.5 dB), b(3 paths 187 kbps, PSNR 36.2 dB ), c(6 paths 278 kbps, PSNR 38 dB).

on this topic.

The particular communication environment of wireless ad hoc networks makes MPT very appealing. In ad hoc networks, (i) Individual links may not have adequate capacity to support a high bandwidth service1; (ii) A high loss rate is typical; and (iii) Links are unreliable. MPT can provide larger aggregate bandwidth and load balancing for ad hoc video applications. In addition, the path diversity inherent in MPT can provide better error resilience performance.

Furthermore, many of the ad hoc network routing protocols, e.g., DSR [18], AODV [19], and ZRP [20], are able to return multiple paths in response to a route query. Multipath routing can be implemented by extending these protocols with limited additional complexity.

There are many challenges in supporting MPT in ad hoc networks. First, from multiple paths returned by a route query, the routing process should select a set of maximally disjoint paths. Shared or nearby links of the paths could make the loss processes of the substreams correlated, which reduces the benefit of using MPT [21]. Algorithms for finding disjoint paths are presented in [11,12]. Second, finding and maintaining multiple paths requires higher complexity and may cause additional overhead on traffic load (e.g., more route replies received). However, caching multiple routes to any destination allows prompt reaction to route changes. If a backup path is found in the cache, there is no need to send new route queries. Rerouting delay and routing overhead may be reduced in this case. These problems should be addressed carefully in the design of a multipath transport protocol to balance its benefits. Third, a problem inherent in MPT is the additional delay and complexity in packet resequencing. Previous work shows that resequencing delay and buffer requirement are moderate if the traffic allocator in Figure 1 is carefully designed [15,22,23].

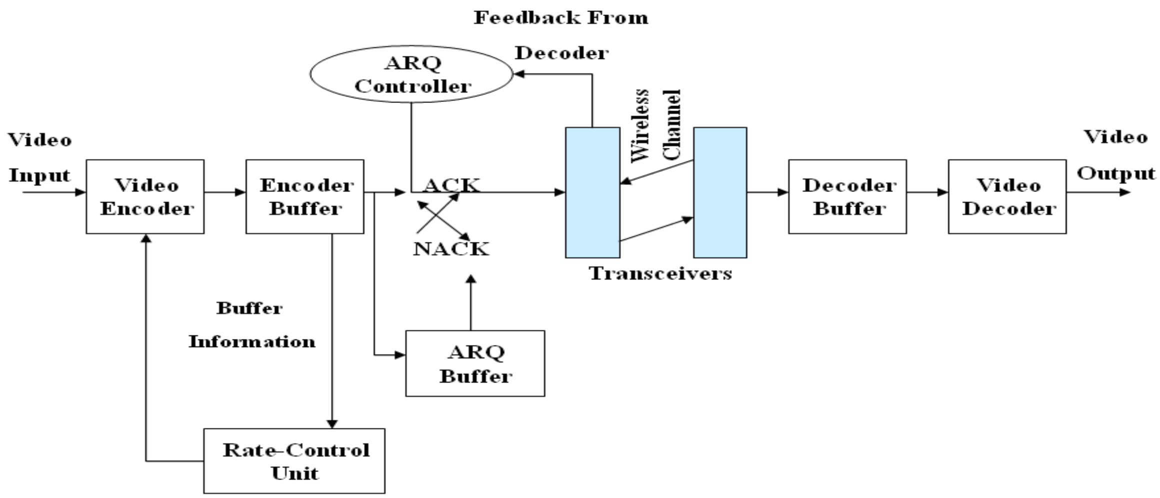


**Figure 3.** Block diagram of a retransmission-based wireless video system [7].

A block diagram of the retransmission based system is shown in Figure 3. The encoder buffer is used to smooth out the video bit-rate to prevent the bits from being discarded when the instantaneous video bit-rate exceeds the channel bandwidth. The transmitted packets are kept in the ARQ buffer until they are acknowledged being received correctly at the decoder. In Selective Repeat ARQ, the receiver sends a positive Acknowledgement (ACK) or a Negative Acknowledgement (NACK) to the transmitter, depending on whether the packet is received correctly or not. The transmitter retransmits the corresponding packet in the ARQ buffer when it receives a NACK from the receiver. From the video transmission point of view, the wireless channel has time-varying capacity due to the retransmissions.

***Related Works***

In [24] *reference picture selection* (RPS) technique has been presented. However, a more network-aware coding method is used, which selects the reference picture based on feedback and estimated path status. In this method the decoder will send a negative acknowledgment (NACK) for a frame if it is damaged or lost, and a positive one (ACK) otherwise The encoder can then estimate the status of the paths and infer which of the previous frames are damaged. Based on the estimation, for a picture to be coded, the closest picture for which itself as well as its reference pictures have been transmitted on the better path is selected as the reference picture. The RPS scheme offers a good trade-off between coding efficiency and error resilience. The RPS scheme is only applicable for online coding, because it adapts the encoding operation based on channel feedback.

In *layered coding* technique, a video frame is coded into a base layer and one or more enhancement layers. Reception of the base layer can provide low but acceptable quality, while reception of the enhancement layer(s) can further improve the quality over the base layer alone, but the enhancement layers cannot be decoded without the base layer. When the layered video is transmitted over multiple paths (e.g., two paths), the traffic allocator sends the base layer packets on one path and the enhancement layer packets on the other one. The path with a lower packet loss rate is used for the base layer if the two paths have different qualities. The receiver returns selective ARQ requests to the sender to report base layer packet losses. When the sender receives such a request, it retransmits the requested base layer packet on the enhancement layer path. The transmission bit rate for the enhancement layer will be reduced correspondingly according to the bandwidth reallocated for base layer retransmissions. This schema denoted as LC with ARQ [24-26]. If there is a base layer packet loss, the base layer path is likely to be experiencing a packet loss burst. Therefore, base layer retransmission using the same path is likely to be unsuccessful. Moreover, if the loss was caused by congestion at an intermediate node, using the base layer path for retransmission may intensify the congestion condition. When disjoint paths are used, the loss processes of the paths may not be totally correlated. Therefore, base layer packet retransmission using the enhancement layer path could have higher success probability and lower delay. In my previous work I had an improvement to this technique that instead of sending base layer frames to sender, and resending it to the second path, responsible node on first path could decrease forwarding packet rate due to network congestion conditions [27]. The third technique is to use *multiple description coding*(MDC). MDC is a technique that generates multiple *equally important* descriptions. The decoder reconstructs the video from *any* subset of received descriptions, yielding a quality commensurate with the number of received descriptions. In [24] a multiple description (MD) coder known as multiple description motion compensation (MDMC) is employed. With this coder, two descriptions are generated by sending even pictures as one description and odd pictures as the other. When coding a picture, say picture $n$, the encoder uses two kinds of predictions:

1) The prediction from a linear superposition of two previously coded frames, pictures $n-1$ and $n-2$, called the *central prediction.*

2) The prediction from previously coded picture in the same description, picture $n-2$, named *side prediction.*

Then the encoder codes two signals for picture $n$; that is, the central prediction error (the difference between picture $n$ and the central prediction) and the reference mismatch signal (essentially the difference between the central and side predictions). Description one includes central prediction errors and the reference mismatch signals for even pictures, and description two includes those for odd pictures. When both descriptions are received, the decoder can reproduce the central prediction and will reconstruct a picture by adding the central prediction error to the central prediction. When only one description is received, the decoder can only generate the side prediction, and a picture is decoded by using both the central prediction error and the mismatch signal. Compared to layered coding, MDMC does not require the network or channel coder to provide different levels of protection. Nor does it require any receiver feedback. Acceptable quality can be achieved even when both descriptions are subject to relatively frequent packet losses, as long as the losses on the two paths do not occur simultaneously and sufficient amount of redundancy is added by appropriately choosing the predictor coefficient and mismatch signal quantizer [28-30]. There are different other work that evaluate video transferring over multipath Ad hoc networks. Many of these works are based on merging previous works and new ideas. Some of them like [31] are trying to improve route selection algorithms. Some others works like [32] are looking for supporting video on demand services over wireless mesh networks. In [33] an algorithm for calculating the loss compensation is presented. The other work in this field is developing a model which captures the impact of quantization and packet loss on the overall video quality [34].

## Materials and Methods

My proposed schema will improve QOS parameters supposing available parameters in application layer (video encoder/decoder). This structure uses cross layer techniques for increasing QOS. In fact this structure will not warranty quality of service but presenting a new structure that is consistent to available structure, QOS in application layer will be increased.

The main assumption of the plan is based on the fact that in a network with a long chain of nodes always some parts of network are in a good communication conditions and other parts are in bad conditions thus using some nodes called video proxy nodes in suitable situations of network due to their duty for realizing video stream status and detecting importance of frames/packets in order to undertaking some encoder/decoder needed operations like sending ARQ will result in lower end to end delay for key video frames. In this plan delivering safely feedback messages is done using RTP protocol. Since in an Ad Hoc network there are a lot of routes and streams may passes from a node, video proxy (cache) nodes should able to realize favorite stream and recognize it from other streams. Beside these nodes should have capability for realizing stream structure and different parts of them to have favorite work on important frames/packets. Also these nodes must use cross layer technique for achieving various informations in different layers for example video informations that relates to application layer, recognizing stream structure in transport layer and etc. In our plan first of all two disjoint paths will be found using AOMDV [35,36] protocol then using some nodes in each of two paths as video proxy nodes, transmitting packets between sender and receiver begins. Since here we have two disjoint paths, important frames/packets like I frames will be sent through path1 and other packet/frames like B and P frames will be transferred using path 2. Selecting video proxy nodes can be based on agreement between sender and receiver or other factors like network traffic load.

The proposed schema is specially suitable for scenarios that there are many hops between sender and receiver that result in longer Round-Trip delay. We will call video proxy nodes as key nodes in this paper. The duty of these nodes is detecting video streams and buffering some of important frames that upon packet loss in network and receiving ARQ messages from receiver, instead of sending ARQ to sender , start to send lost frames itself and decreasing forward traffic rate if receiving ARQ. Also if possible these nodes can detect lost frames before receiver and start to send ARQ messages to the sender and after this step receive lost frames and forward themes. The key idea of this schema is that there is unequal probability of fault in all parts of network and therefore after having congestion between sender and receiver, these nodes can manage connection locally regarding adjacency to sender.

The same process is about having congestion on receiver side of proxy node. For achieving this goal, these nodes should realize video streams at beginning of session. If there are more than one proxy nodes in each of transmission paths only one of them will select as proxy node. Proxy nodes pseudo codes has been presented in Figure 5.

The general structure of suggested video transferring path schema is shown in Figure 4 and proposed schema is presented at Figure 6. This plan includes two Disjoint paths, basic sender, and receiver as source and destination and selective Video proxy nodes per each of routing paths. We use AOMDV routing algorithm for finding disjoint paths. Video packets will be send to receiver through two disjoint paths simultaneously. In this plan the paths composed of intermediate nodes between sender and key nodes are known as sender side network and also the paths between key nodes and receiver are receiver side network. At sender encoded video packets are sent through UDP packets to each of two paths.

This network can have a lot of nodes that may create high or low latency. After receiving and processing video packets by key nodes, having failure or packet loss in frames/packets, key node will send feedback number 1 to sender. Key nodes will send received packets to receiver side network with lower forward transfer rate. In other hand receiver starts to receive and decoding video packets. If it was failure in packets or connection, feedback number 2 will be sent to sender. Upon receiving feedback number 2 and having process on it by key node, it will refer to it's video frames cache, if possible will answer to feedback by sending needed frames/packets otherwise it will send feedback number 3 to sender. Likewise sender after receiving feedbacks numbers 1 and 3 if possible will use needed mechanism to correct the fault. Here feedback messages are same as ARQ ones. Decoder not only can send ARQ, feedback number 2, but it can try to correct fault and tracking consequences of errors up to receiving needed frames/packets. Schema of nodes connections along with feedbacks and their sides are shown in Figure 6.

In this schema key node is waiting for a packet, after receiving a packet, key node will analyze kind of packet. If packet was not belong to video frames/packet, key node will switch back to waiting mode, otherwise it will scrutiny whether previous frames/packets that this frame/packet depends on them are received correctly or not. If yes, key node will switch back to waiting mode and otherwise it will wait a period of time for receiving packets then after this step the status of key node will set to waiting mode. If the packet didn't receive by key node, it will send feedback 1 to sender but if packet is kind of feedback, referring to it's frames caches if possible key node will send requested frame/packet and otherwise will send feedback 3.

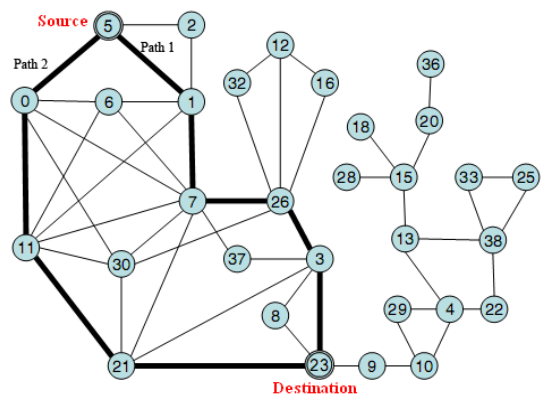


**Figure 4**. Transmitting video path between sender and receiver using two disjoint paths.

```
Start
{- For Each of Disjoint Paths Permanently check for
probable loss of desired frames/packets and send
"Feedback 1" to sender upon loss occurrence}

1. Wait for a packet;
2. Switch packet :
            Case "Desired Video Stream":
                        Goto 3.1;
            Case "Feedback 2":
                        Goto 3.2;
            Case "Other Streams":
                        Goto 1;
3.1. If the frame/packet belongs to desired category
            Copy the packet in the buffer (cache);
      Goto 1;
3.2. If required frame/packet is in the buffer (cache)
            Reduce Forward Transmit Rate
            Retransmit frame/packet;
      Else
            Send feedback 3 to sender;
      Goto 1;
End.
```

**Figure 5.** Proxy Nodes Pseudo Code.

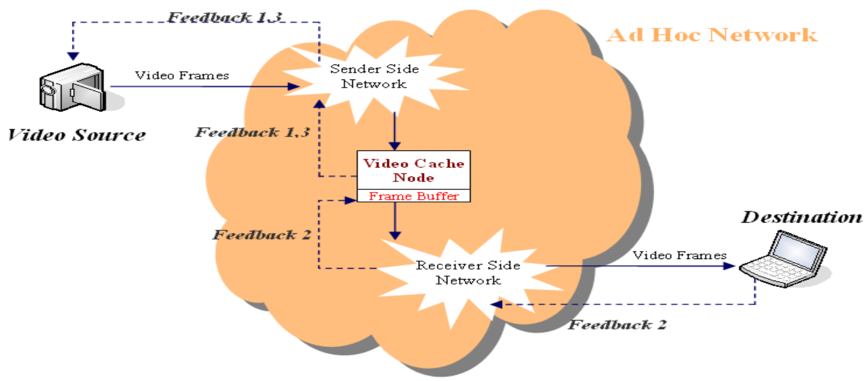


**Figure 6.** Video Transmission proposed Schema.

***System Cost***

In this schema H.264 [37] is used for encoding video stream. Video caches contains reference frames at sender and receiver and also there are key nodes that contain some recent reference frames. Suitable number of existing frames in buffer can select due to the power of mobile node. In order to evaluate this schema we use Evalvid[38], that is a framework and tool-set for evaluation of the quality of video transmitted over a real or simulated communication network. Besides measuring QOS parameters of the underlying network, like loss rates, delays, and jitter, a subjective video quality evaluation of the received video is provided using Evalvid. In this framework we use ffmpeg [39] encoder. We used peak signal-to-noise ratio (PSNR) factor along with MOS to measure our video quality. PSNR is defined as below:

**Definition 1.**

$$PSNR\left(n\right)_{dB} = 20\log_{10} {}^{*}V_{peak} \,/\, \frac{1}{\sqrt{\frac{1}{N_{col}N_{row}}\sum_{i=0}^{N_{col}}\sum_{j=0}^{N_{row}}\left[Y_s\left(n,i,j\right)-Y_D\left(n,i,j\right)\right]^2}}$$

Here $V_{peak} = 2^k-1$ and k = number of bits per pixel (luminance component). MOS (Mean Opinion Score) is a subjective metric to measure digital video quality at the application level. This metric of the human quality impression is usually given on a scale that ranges from 1 (worst) to 5 (best) that has been shown at Table 1.

Simulation is done using NS-2.32 [40]. From memory point of view if we suppose video pattern as QCIF (176*144) and there are 10 frames at buffer, mandatory memory for each video node include sender, receiver and key node will equal to 15*(176*144 + 2*88*72) = 570240 bytes, that is a reasonable memory for today's advanced devices. System calculation cost can be teeny because in this plan there is no need to online encoding therefore video encoding can be in any form. Receiver calculation cost will not change because there is no change in receiver operation.

## Results

Simulation study has been carried out using NS-2.32. The objective of our simulation study is twofold: first, to demonstrate the effectiveness of multipath as compared to single path transmission and second, comparing results of using video proxy nodes against the network general status beside other objects like end to end delay. UDP is used in the transport layer. 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.

We use movie Foreman's sequences encoded from quarter common intermediate format (QCIF) for our test. This video segment comprises 400 frames, each with 176x144 pixels and they are encoded into MPEG-4 [41] video with frame rate at 30 frames per second. The MAC layer used in our simulation is IEEE MAC 802.11b. The maximum size for packet is defined as 1050 bytes. 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.

In this section we used different scenarios. Simulation is done in a 700*700 area that respectively former is X-dimension and later is Y-dimension of simulation environment. Figure 7 shows the difference between transferring video in a single path against proposed multipath. In some of these scenarios like Figure 8 and Figure 9, nodes coordinates selected manually by X and Y. Routing were done using AOMDV and a random position based traffic like TCP or File Transferring, imposed to network. As you can see in Figure 10, transmitting video over random positioned nodes without sending extra traffic has been exhibited. In some other scenarios, we used NS2 random generator positioning tool to create random coordinating X and Y of nodes then like previous scenario we imposed random based positioned traffic to network. As shown in Figure 11, we expected to have lower PSNR in contrast to the case we transmitted packet without imposed traffic to network but our simulation results showed that due to decrease in hop count between sender and receiver, PSNR not only decreased but in some cases increased, as pointed this is because of random based nature of nodes coordinates. And finally we tried to control position of imposed traffic at sender and receiver and both sides. For this reason in order to scrutiny sender side traffic effects we used 60 nodes that proxy nodes in paths 1 and 2 respectively situated at 29th and 30th nodes. Imposed traffic was generated between 11th and 12th nodes in first path of video transmitting and for second path of video transmitting this is done between 9th and 10th nodes. The results of our experiments are shown at Table 2. Likewise for testing receiver side traffic effects, traffic generated between 40th and 41th nodes from the first path and 38th and 39th nodes from the second path. Result are shown in Table 3. Finally same work repeated for comparing both sender and receiver side traffic on PSNR and proxy nodes performance as shown in Table 4. The Reconstructed frames of Foreman sequence at receiver for video multipath transferring in disjoint paths (a) against our proposed schema (b) has been shown at Figure 12. As mentioned before the main goal of proposed schema was to decrease end to end delay between sender and receiver that is shown at Figure 13. And finally as shown at Figure 14, proposed schema has better reciver side video quality in contrast to no proxy nodes.

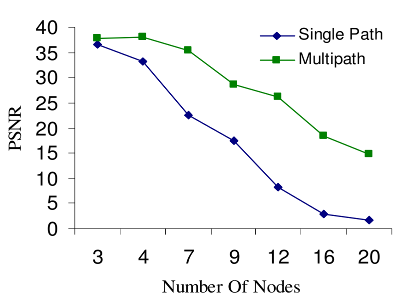


**Figure 7.** PSNR for Single path against Multi path.

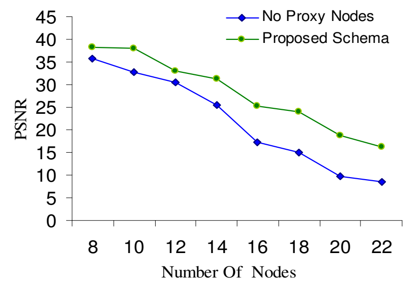


**Figure 8.** Transferring video over Multiple path Without sending extra Traffic imposed by some other application.

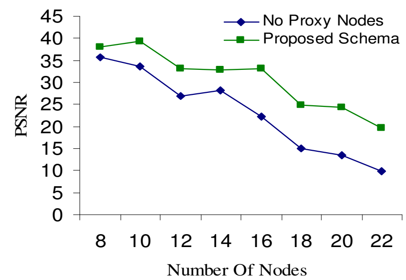


**Figure 9.** Transferring video over Multiple path using extra Traffic imposed by some other application like file transferring.

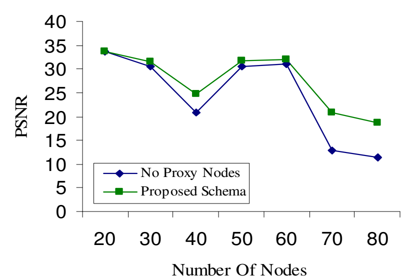


**Figure 10.** Transferring video over Random Positioned nodes Coordinates Without sending extra Traffic.

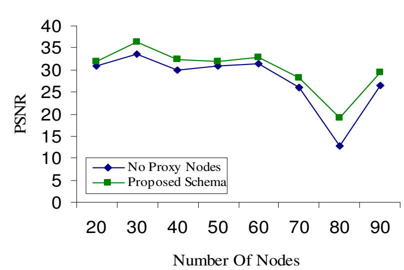


**Figure 11.** Transferring video over Random Positioned nodes Coordinates sending extra traffic like TCP Traffic.


**Figure 12.** Reconstructed frames of Foreman sequence. (a) transmitting video packets over three disjoint paths; (b) Same scenario using proposed schema.

**Table 1.** Possible PSNR to MOS conversion [38]

| PSNR[dB] | MOS |
|---|---|
| >37 | 5 (Excellent) |
| 31-37 | 4 (Good) |
| 25-31 | 3 (Fair) |
| 20-25 | 2 (Poor) |
| <20 | 1 (Bad) |

**Table 2.** Sender side traffic effects

| PSNR Average | Scenario |
|---|---|
| 31.38 | Normal case without using traffic |
| 30.53 | Normal case using sender side traffic |
| 37.24 | Using proxy nodes having no traffic |
| 35.72 | Using proxy nodes having traffic on sender side |

**Table 3.** Receiver side traffic effects

| PSNR Average | Scenario |
|---|---|
| 31.38 | Normal case without using traffic |
| 28.41 | Normal case using receiver side traffic |
| 37.24 | Using proxy nodes having no traffic |
| 35.24 | Using proxy nodes having traffic on receiver side |

**Table 4.** Both sender and receiver sides traffic effects for disjoint paths

| PSNR Average | Scenario |
|---|---|
| 31.38 | Normal case without using traffic |
| 22.58 | Normal case using sender & receiver side traffic |
| 37.24 | Using proxy nodes having no traffic |
| 28.64 | Using proxy nodes having traffic on sender & receiver side |

As you can see in Figure 15, using 90 to 100 frames is suitable for transferring video over multiple path. Choosing less than 50 frames for proxy nodes is not desirable. This results illustrated that video cache retransmitted packets are useable for decoding if only appropriate amount of buffer is used at cache node.

## Discussions

The opportunity and importance of ad hoc networks is being increasingly recognized by both the research and industry community, as evidenced by the flood of research activities, as well as the almost exponential growth in the Wireless LANs and Bluetooth sectors. The wireless links in an ad hoc network are highly error prone and can go down frequently because of node mobility, interference, channel fading, and the lack of infrastructure. Multi-Way video communications over a low bit-rate channel is suitable for support of several applications such as videophone and video

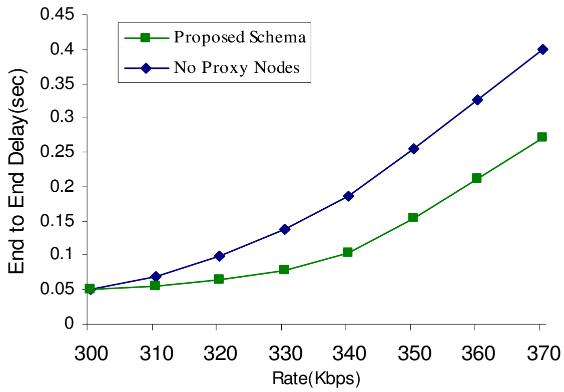


**Figure 13.** End to End Delay Comparison Between proposed schema and multipath network general status.

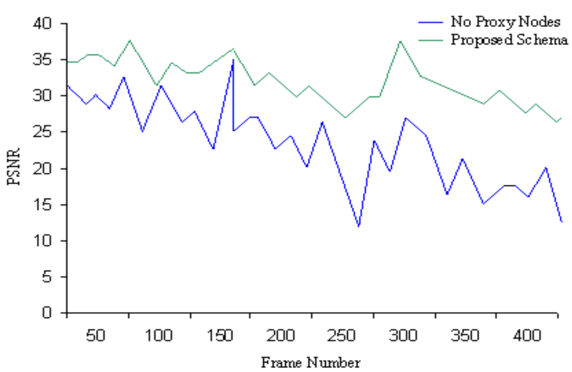


**Figure 14.** comparison of Received video quality at receiver.

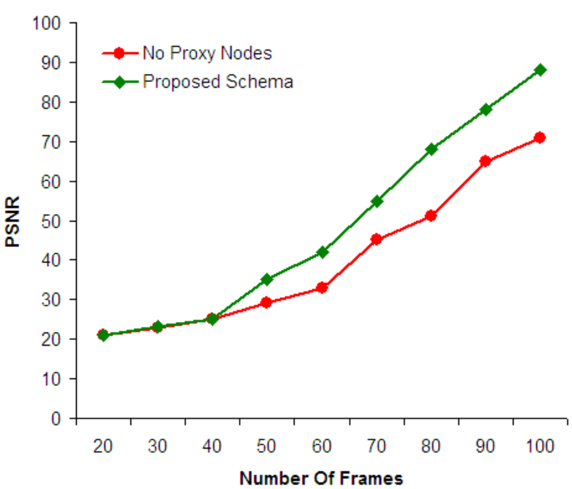


**Figure 15.** Video cache node buffer changes.

conferencing. Indeed, multipath transport provides an extra degree of freedom in designing error resilient video coding and transport schemes. From the technical standpoint, as shown in this article, despite the large volume of research activities and rapid progress made in the AD HOC technologies in the past few years, almost all research areas (from enabling technologies to applications) still harbor many open issues. However, using several paths in an ad hoc network can be exploited as an effective means to battle transmission errors. In this paper an optimized plan has been introduced for transferring video packets over multiple ad hoc paths in order to improve quality of received video over application layer.

Using Evalvid and doing simulation over disjoint paths for several number of ad hoc nodes, results shows that using suggested structure, the quality of received video at destination is better than similar cases. While there is request for retransmitting of lost video packets in an error prone channel that is result in longer Round-Trip delay, using this schema can be useful. I should note that further improvements could be made for each video cache nodes in the proposed framework. Increasing capabilities of key nodes in analyzing and reencoding video streams, we can achieve more advantages. Another work could be about video codec parameters that further tuned and optimized in the rate distortion sense, given the path conditions at proxy nodes. Expanding this plan for more than three paths, selecting key nodes due to dynamic mapped traffic of network, and increasing amount of buffer for proxy nodes could be good works for future.

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