Performance issues of H.264 compressed video streams over IEEE 802.11b based MANETs

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***Abstract*— This paper addresses the problem of video streams transmission in IEEE 802.11b based Mobile Ad-hoc Networks (MANET). Through the study of the delivery of a live encoded video stream, we expose the characteristics of different routing protocols and the infeasibility to provide QoS. The analysis shows the impact of a MANET on H.264 real-time video flows in terms of packet loss, end-to-end delay, jitter and distortion, and the behavior of H.264 error resilience tools in order to determine their effectiveness on such network scenarios. The results show that video traffic has demands that are hard to be met by a standard MANET, and that improvements are required in terms of routing protocols and QoS provisioning either on the MAC layer or at IP level using traffic shaping tools. Also, most of H.264 error resilience tools are not so effective as expected with this kind of networks, being the random macroblock updating the most effective one we have tested.**

***Index Terms*— MANETs, IEEE 802.11b, H.264, performance, resilience**

I. Introduction

The increasing use of mobile devices and the demand for video oriented applications is leading companies and researchers to look for solutions in the field of mobile multimedia. Several improvements related to video compression technology were made in recent years resulting in the ISO MPEG-4 Part 2 [1] standard and ITU-T Recommendation H.263 [2]. The JVT H.264/MPEG-4 part 10 is a new standard that offers an enhanced video technology which provides superior compression performance and better error-resilience, as well as many other features as will be exposed in section III. Such improvements pave the way for ubiquitous human-to-human video communication, even when using low-bandwidth and error-prone network environments.

The widespread deployment of the IEEE 802.11b [3] technology, and the advances on other standards of the 802.11 family have increased the interest in MANETs that, although originally intended to cover military or disaster-related situations, are becoming more and more an alternative solution for the enterprise and home environments.

The Internet Engineering Task Force (IETF) has already presented a draft [4] intending to provide an integrated solution for Internet connectivity to MANETs, so that the Internet itself is seamlessly extended to unwired areas.

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Until now, most of the studies done about 802.11 MANETs and related performance issues have relied on overall statistics results regarding packet losses and other parameters of significance. In this paper we follow a different strategy in order to provide an accurate study of real-time video on 802.11b based MANETs. Our analysis focuses on a single H.264 video stream, so that the effects of different routing protocols and CSMA/CA radio technology are put into evidence in terms of packet losses, packet loss patterns, end-to-end delay and jitter. At the same time, we will be able to analyze the behavior of the H.264 error-resilience tools, evaluating their effectiveness in terms of perceived video quality distortion.

We shall measure the impact of several 802.11b MANET aspects over the final video quality perceived by the end user, like ad-hoc routing algorithms, mobility and traffic patterns, etc. We shall also evaluate current video error-resilience techniques.

Concerning the structure of this paper, in the next section we introduce some important aspects related to 802.11b based MANETs. Section III presents the H.264 video codec and the available error-resilience mechanisms, and in section IV we describe the simulation framework. Simulation results are presented in section V, and concluding remarks are made in section VI, along with some guidelines about future work.

II. Issues concerning 802.11b based MANETs

IEEE's 802.11b standard is being increasingly used throughout corporations worldwide due to its good balance of cost, range, bandwidth and flexibility. The bandwidths set by the standard range from 1 to 11 Mbps, but other standards in the same family aim at higher bandwidths. The 802.11 standard offers operation modes named Point Coordination Function (PCF) and Distributed Coordination Function (DCF). PCF is used in infrastructure mode, where Access Points are responsible for coordinating the transmissions from nodes. DCF, on the other hand, is a distributed mechanism through which each node has the responsibility of sensing the medium, to avoid and react to collisions. The medium access technique (CSMA/CA) is currently being enhanced by the IEEE P802.11 task group E in order to provide a framework for QoS [5]. Our analysis is focused on 802.11b networks with Distributed Coordination Function.

When operating in this mode each unicast packet is optionally preceded by a RTS/CTS sequence, followed by a mandatory acknowledge packet. The RTS/CTS process aims at eliminating the well-known hidden node problem [6], while the acknowledgment assures that the packet, when delivered, is free of bit errors. However, the Link Layer for 802.11 is not connection-oriented, which means that after a standard/user-defined number of failed transmission attempts, a packet is dropped. Also, the 802.11 link level frames contain error-detection data, which means that data sent to higher-level layers are free of bit errors. Therefore, the only kind of losses present in such environment are packet losses.

The 802.11's DCF mode enables the formation of mobile ad-hoc wireless networks through node cooperation using one of the available routing protocols designed for these networks. Protocols such as AODV [7], DSR [8], TORA [9], DSDV [10], OLSR [11] and ZRP [12] are well known ad-hoc routing protocols.

Due to their nature, MANETs are very unstable due to frequent route changes caused by node movement, node on/off activity or noise. In an ideal situation, messages associated with routing on MANETs should be given a high priority since, due to mobility, route changes are very frequent. Also, mobile nodes are usually battery bounded, which means that sending data though invalid paths should be avoided whenever possible.

However, in order to sense the unpredictable neighborhood, MANET nodes typically make use of broadcasting. Broadcast packets, unlike unicast packets, are not acknowledged and also do not benefit from the RTS/CTS mechanism. This means that they are transmitted only once, and so there is no assurance that the packet will be correctly received by any surrounding node, suffering therefore of interference, collisions, or time-varying channel effects. Moreover, the fact that they are sent at the lowest possible rate (1 Mbps) increases the transmission time, which also increases the probability that these frames collide. Despite of these drawbacks, most routing protocols broadcast packets in many occasions, such as to advertise themselves through "Hello" messages, request a route to the neighbor nodes, or both.

Another problem associated with message broadcasting is the exploration of unidirectional links. While the acknowledgment mechanism assures bidirectional unicast communications, packet broadcasting, which lacks of such a mechanism, is prone to explore invalid routes.

III. H.264 RELATED ISSUES

The recent video coding standard H.264 [13], part of an activity on-going since 1997 named H.26L, was developed by the Joint Video Team (JVT), an alliance formed by the former ITU-T VCEG and ISO MPEG-4 groups. This new standard is not application-specific, and performs significantly better than the available ISO MPEG-4 Part 2 standard [1] and ITU-T Recommendation H.263 [2] in terms of compression, network adaptation and error robustness.

With the H.264 standard there is a back to the basics approach, where a simple design using well known block-coding schemes is used. In the design of this codec, the Video Coding Layer was separated from the Network Adaptation Layer in order to enable a modular development of each of its components. Due to its general purpose nature, some mechanisms were included on both encoder and decoder envisioning enhanced performance in lossy environments, such as wireless networks or the Internet. By tuning certain parameters, the user can obtain a trade-off between compression rate and error resilience.

The most commonly used methods to stop temporal propagation of errors when no feedback channel is available are the random intra macroblock updating and the insertion of intra-coded pictures (I-frames). While intra frames reset the prediction process, avoiding error propagation, their use has a generally high bandwidth cost, causing also severe bit rate variations. The use of random intra macroblock updating is more effective than I-frames because it not only aids in generating streams with more constant bit-rate, but can also provide better results by statistically resetting the error for each of the macroblocks.

Another method which is sometimes used is called Flexible Macroblock Ordering (FMO), whereby the sender can transmit macroblocks in non-scan order. This method, although similar to slice interleaving, provides greater flexibility and can be tuned to be more effective in terms of error resilience because of increased fine grain control on macroblock ordering. It aims essentially at dealing with packet loss bursts by spreading errors throughout the frame, a process which eases the decoder's error-concealment task.

Multi-frame prediction is another tool targeting to increase both compression performance and error-resilience. This is achieved by using more than one reference frame in the prediction process. As exposed in [14], this technique is particularly useful after the loss of a full frame when some of the previous reference frames are available, enabling partial motion compensation.

Concerning the decoder, it plays a fundamental role in error resilience since it is responsible for error concealment tasks. With that purpose, it keeps a status map for macroblocks which indicates, for each frame being decoded, weather a certain macroblock has been correctly received, lost or already concealed. The methods used vary between intra and inter frames. For intra frames, the task mainly consists of performing a weighted pixel averaging on each lost block in order to turn it into a concealed one. For inter frames, there is a process of guessing the adequate motion vector for lost macroblocks, although intra-style methods can also be used. For a more complete description of such methods please refer to [15].

IV. SIMULATION FRAMEWORK

In a previous work [16] we performed a detailed analysis of different tuning parameters and behaviors integrated in the H.264 codec. The performance of the H.264 codec was evaluated using the reference software JM3.9a.

Taking into account the results from that work, the Hadamard transform, CABAC and Rate Distortion Optimization were used since they offered the best results. The use of adaptive block transforms for inter and intra blocks was set to the fully flexible mode. Concerning the error-resilience issues, the best options were: enabling random intra macroblock updates (set to 1/3 of frame size) and applying FMO reordering.

In this work we adopt the same strategy though without applying FMO reordering since it is not supported by the current H.264 codec.

The chosen test sequence is the well-known QCIF Foreman sequence with a resolution of 176x144 pixels, which we considered adequate for display in current PDAs and other mobile devices displays.

The frame rate is set to 10 frames per second, and the stream's bit-rate will be tuned in section V-A. In order to perform the desired evaluations we used the discrete event network simulator ns-2 [17] version 2.1b9a. The physical layer for the simulation uses two-ray ground reflection as the radio propagation model. The link layer is implemented using IEEE 802.11 Distributed Coordination Function (DCF), and the Media Access Control Protocol (MAC) is CSMA/CA - Carrier Sense Multiple Access with Collision Avoidance. This module was modified in order to correctly update the *contention window size* and the *short retry count*.

The transmission range for each of the mobile nodes is set to 250m and the bandwidth to 11Mbps (full rate).

To evaluate the desired video streams, the RTP output from the H.264 encoder was converted to the NS-2's native input format. This allows to stress the network with real-life video traffic instead of relying on CBR flows.

Though the sequence is only 10 seconds long, NS-2 automatically re-reads its input so that the sequence automatically restarts. Our evaluation is done over 100 simulated seconds, achieved by averaging the video distortion on each 10 second interval. Moreover, all results presented are average results from 20 random simulation processes. A settling period was introduced at the beginning of each simulation in order to allow routing protocols to converge, and also to start the background traffic, so that the video stream is evaluated on an almost steady-state situation.

After the NS-2's simulation process ends, we process the output results in order to determine the reconstructed video sequence according to the packet loss pattern.

This method aims at performing evaluations as real as possible, in order to predict the effects of MANET networks and video codecs on the video quality perceived by the final user.

V. Performance Results

In this section we start by tuning the H.264 video codec in terms of packets per frame, followed by a preliminary evaluation of several ad-hoc routing algorithms in order to determine their average re-routing times. If re-routing times are long, MANETs will have problems to deliver compressed video streams.

We then measure the impact of node mobility in a typical scenario, followed by an analysis of the final delivered video quality under network congestion.

Mobility and congestion are two different aspects that may affect the video quality performance at different degrees. For that reason we also test node mobility and network congestion independently. Finally, we will test the behavior of the H.264 video codec at different network congestion levels in order to analyze the effectiveness of its error-resilience tools on MANETs.

A. Packetization of video data

The H.264 codec offers the possibility of generating its output in RTP packet format, so that the integration with packet networks is straightforward. Moreover, it allows setting the desired level of video packetization granularity, so that the user can split each frame into the desired number of packets.

By increasing the number of packets per frame, the video overhead is slightly increased due to the need for more headers at the video stream level. However, this effect is only slight when compared to the overhead introduced at lower layers, as exposed in figure 1.

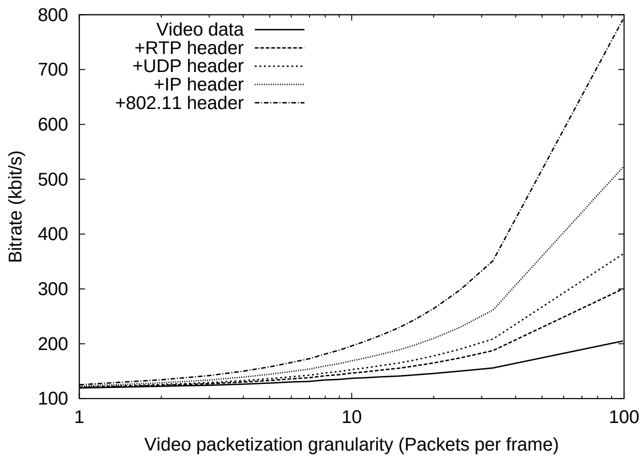


Fig. 1. Overhead at different layers for different levels of video packetization granularity

In terms of end-to-end delay and, more important, in terms of video error resilience, increasing packetization granularity achieves better results. When the number of packets per frame reaches 9, though, the bit-rate value at 802.11 level compared to the single packet per frame solution already reaches an increase of 50%, so we consider that for this video sequence we should not use more than 9 packets per frame in order to achieve efficiency.

Besides bit-rate increase, higher levels of packetization granularity have other negative effects on 802.11b based networks due to time overhead involved in the channel access technique. In order to assess this factor we create a simple scenario with two nodes, where one node is the source of both a video and a FTP flow, and the destination is the other node. This scenario allows us to measure the effective decrease in available bandwidth for the TCP flow by increasing the video packetization granularity. Results of this experiment are presented in figure 2.

As it can be seen, the decrease in TCP throughput is much faster than the increase in video throughput.

Taking into account the previous observations, we set the H.264 codec's packetization granularity to 7 packets per frame,

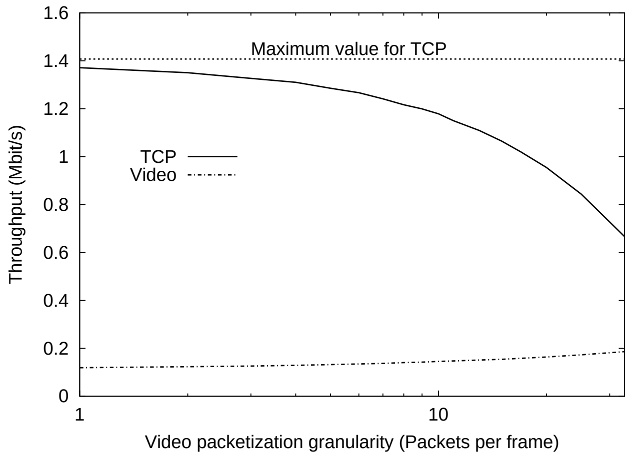


Fig. 2. TCP throughput degradation with increasing levels of video packetization granularity

value that is maintained from now on. The average bit-rate for the sequence at 802.11 level is 172.84 kbit/s.

B. Preliminary evaluation of routing protocols

Protocols used for routing in MANETs are usually divided into two main categories: reactive and proactive. Moreover, another division can be made according to the way in which they detect link failures. While the method of sending "Hello" messages is commonly used, IEEE 802.11 enables the use of a more effective and efficient method to detect link breaks by using the information it provides. Awareness of the link layer allows nodes to react to broken links more quickly, avoiding sending packets to nowhere.

Broken links are the main cause of long packets-loss bursts in MANETs. In fact, long packets-loss bursts can be a major source of problems for video streams. This problem is more evident when "Hello" packets are used to detect broken links. Typical "Hello" intervals range from 1 to 2 seconds [7], [11], and so re-routing times can be as high as 6 seconds or more - a connection is considered lost usually after 3 missing "Hello"s. Since such failures are too long to be handled even by the most versatile video codec, we recommend enabling protocols with Link Level awareness in order to perform re-routing tasks as soon as possible.

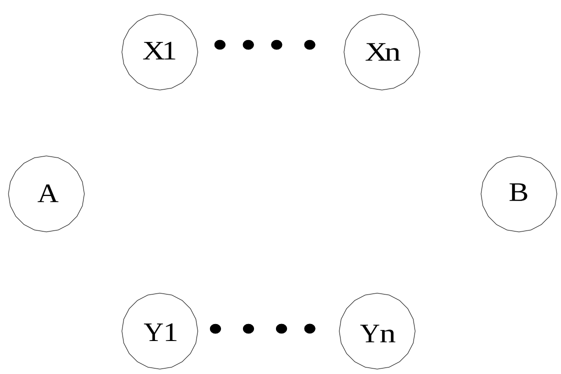


Fig. 3. Simple scenario for re-routing evaluation

In this preliminary evaluation a simple scenario was devised, as presented in figure 3. The purpose is to evaluate the re-routing times for different protocols by setting a CBR flow from node A to node B through path {A,X1,...,Xn,B} and enforce a re-routing process using path {A,Y1,...,Yn,B}. To achieve that, the last intermediate node from the upper path (Xn) moves quickly away making that route unusable; just before that Y1 moves into the range of A. Choosing the departure of the last node (Xn) aims at achieving worst scenario results.

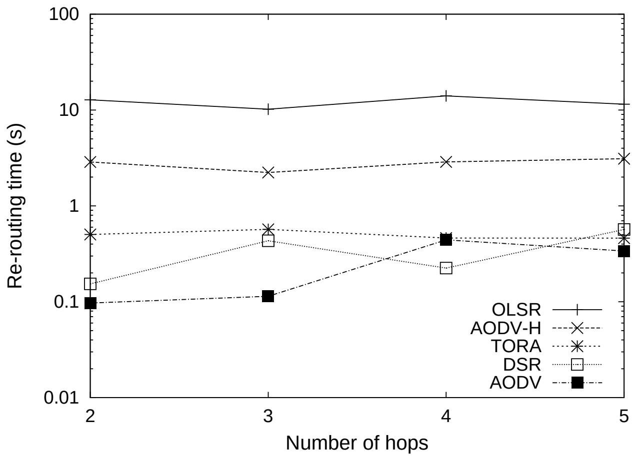


Fig. 4. Re-routing times for different protocols in the simple test scenario

The results for the evaluation under this scenario are presented in figure 4. "Hello" based protocols such as OLSR and "Hello" enabled AODV (AODV-H) perform significantly worse than link aware protocols as expected. Moreover, re-routing time for "Hello" based protocols depends essentially on the "Hello" period and on the number of missed "Hello"s until the link is considered lost. In OLSR the "Hello" period is 2 seconds, twice that in AODV; both consider the link is lost after 3 failed "Hello"s. This explains the difference between both. This implementation of OLSR also requires that a node receives 3 "Hello"s from a neighbor before the link between both can be used, which explains why this value (worst case) is twice the one in a normal situation.

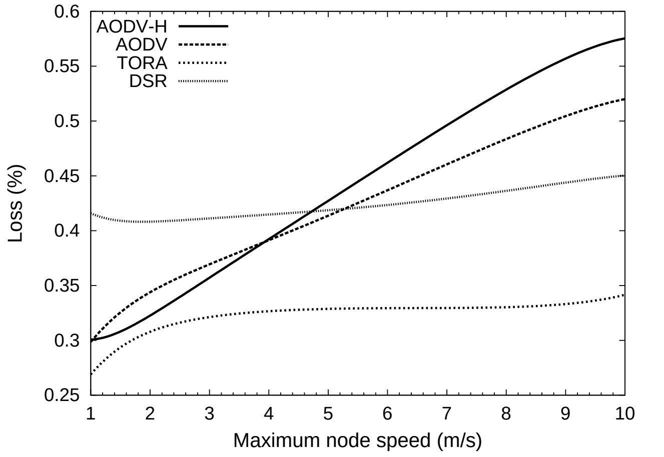
To proceed with our work we drop the OLSR protocol and keep AODV-H simply as a basis for comparison with AODV.

C. Mobility evaluation in a typical scenario

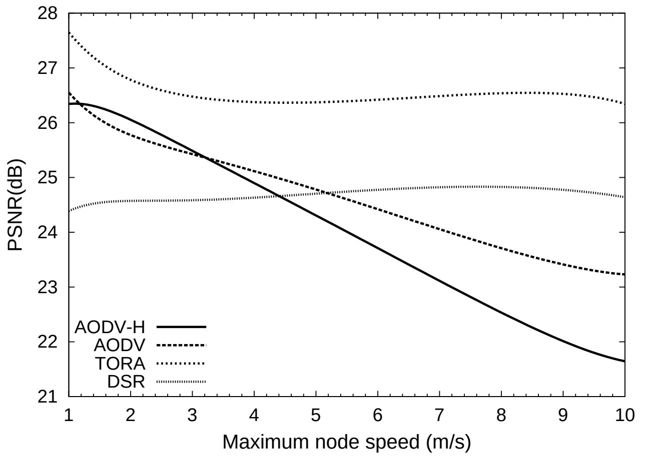
After this initial evaluation, we devised a scenario with 30 nodes in a 670×670 area. Mobility was generated through the random waypoint model available in the NS tool with maximum allowed node speed varying between 1 and 10 m/s. We also set a node wait time of 5 seconds before starting each movement. In addition to the video flow, 5 background FTP flows are also set (1 every 6 nodes).

Figure 5 shows the results achieved by using different routing protocols, with this scenario, in terms of distortion and packet loss rate.

"Hello" based AODV performs relatively well in situations of low mobility because route changes do not occur so often.



a)



b)

Fig. 5. Evaluation of different routing protocols for varying mobility in terms of a) packet losses and b) perceived PSNR for the video test sequence

Also, there are less chances that background congestion causes one link to be considered lost (3 consecutive "Hello"s have to be lost). TORA shows the best overall behavior under this scenario, showing good distortion levels at all speeds and good ability to maintain the packet loss rate with high mobility. DSR is also able to maintain steady levels of distortion and packet loss rate, although not so efficiently as TORA.

This analysis does not pretend to evaluate the goodness of different routing protocols, but rather to evaluate the video performance achieved on a congested network using different routing methods. Please refer to works such as [18] for a more general study on the performance of different routing protocols.

D. Performance under congestion

After the mobility evaluation we chose both TORA and the AODV protocols to proceed with our analysis. We evaluate their performance when submitted to different levels of congestion at user mobility levels (maximum speed of 2 m/s). These results were achieved using the same 30 node square scenario described in the previous subsection.

Figure 6 allows us to compare the performance of TORA

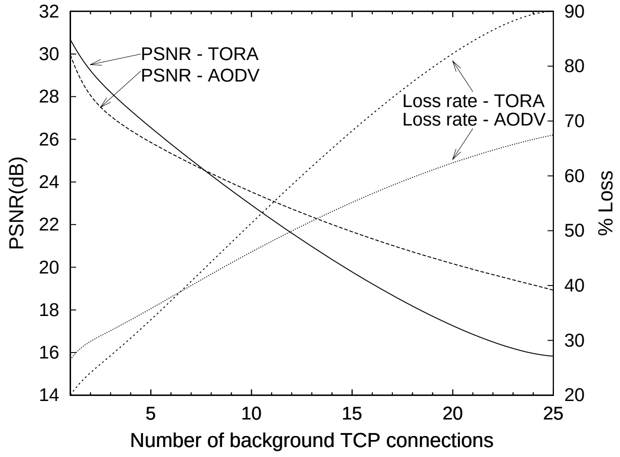


Fig. 6. PSNR and packet loss rate performance for a variable number of background TCP connections

and AODV with a variable number of TCP connections in the background. We can see from that figure that acceptable distortion levels cannot be reached with more than 10 background connections using either TORA or AODV. TORA is, therefore, the best choice for this range and, even though AODV performs significantly better under critical levels of congestion, the results in terms of distortion are almost at noise levels.

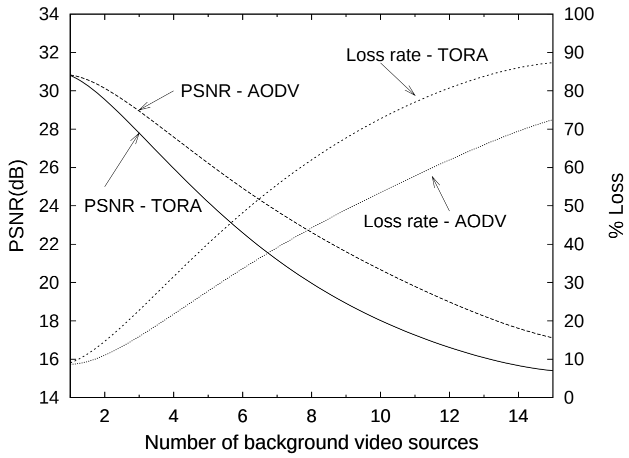


Fig. 7. PSNR and packet loss rate performance for a variable number of background video connections

Figure 7 shows a similar analysis, but now all the background traffic is composed of video flows identical to the one under evaluation. In this scenario AODV always performs better than TORA and, in overall, we consider AODV to be an adequate choice to support video flows as reliably and uninterruptedly as possible.

E. Results on the effects of re-routing and background traffic

To complete our analysis, we change the scenario shape, keeping the same number of nodes and area size. Now the scenario is made rectangular (1500×300 meters) to increase the average number of hops. Envisaging a differentiated analysis of mobility and congestion, we started with a situation having neither background traffic nor mobility. We then analyzed separately the effect of allowing high mobility to all nodes (maximum speed of 10 m/s and no background traffic) and the effect of congesting the network by setting all the nodes to transmit a moderated amount of CBR traffic (no movement). In all situations, the average (or exact) number of hops was three; the routing protocol used was AODV.

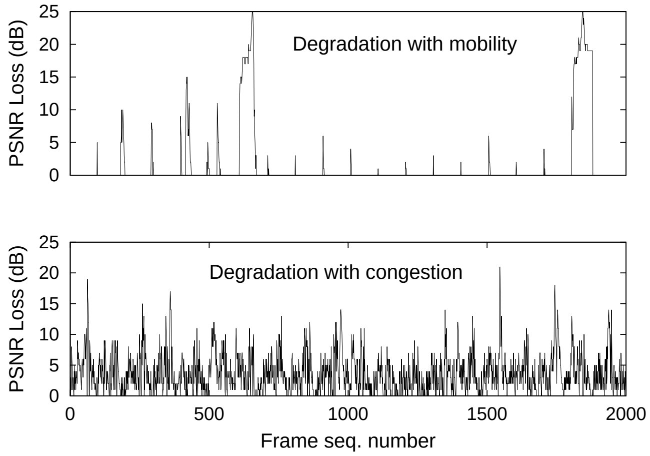


Fig. 8. Effect of congestion and mobility on user perceived PSNR

Figure 8 shows the effects of mobility and congestion on user perceived video distortion. As it can be seen, mobility affects distortion in a bursty fashion, typically causing the loss of multiple frames and consequently freezing the image. On the other hand, traffic congestion causes packets to be lost in a more random fashion, so that distortion variation is smoother though more frequent.

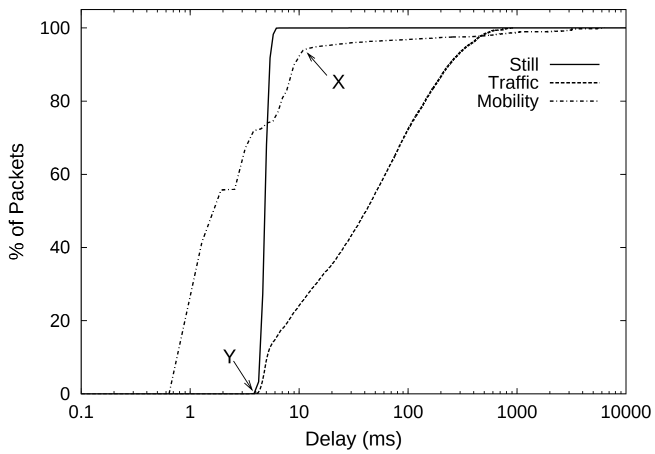


Fig. 9. Delay effects of congestion and mobility

The delay analysis also evidences the nature of both kinds of losses, as presented in figure 9.

In the reference situation (still), more than 99,9% percent of the packets arrive before 7 ms; with high mobility, 92% of the packets arrive in less than 10 ms. Point X is the frontier of two distinct regions: the one on the right where a very small number of packets have very high delays (as much as 6 seconds), and the one on the left where packet forwarding is uninterrupted.

In the “mobility” scenario, although the average number of hops is 3, this value varies throughout the simulation, explaining why some of the packets arrive earlier than those in the reference scenario and others arrive later (before X). The phenomena whereby some packets arrive with very high delays (after X) is expected since AODV causes packets to wait in a queue when re-routing tasks are being performed.

Congestion causes a very different behavior, so that all packets that arrive at the destination do so in less than 1 second, though the delay between consecutive packets can vary greatly. The start point (Y) for both reference and congestion scenarios is common because the destination is 3 hops away on both.

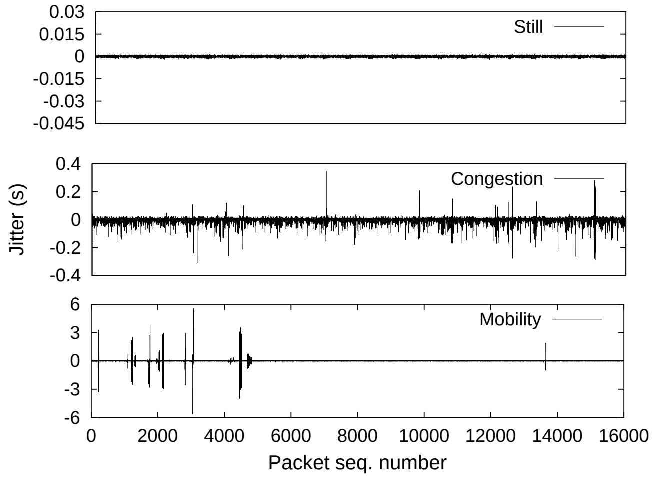


Fig. 10. Jitter due to congestion and mobility

The jitter analysis of figure 10 also aids at visualizing the behavioral difference between both. Even though the jitter peaks occur rather infrequently, they are an order of magnitude superior than those caused by congestion. We conclude that jitter peaks usually translate into a change of route when using reactive protocols.

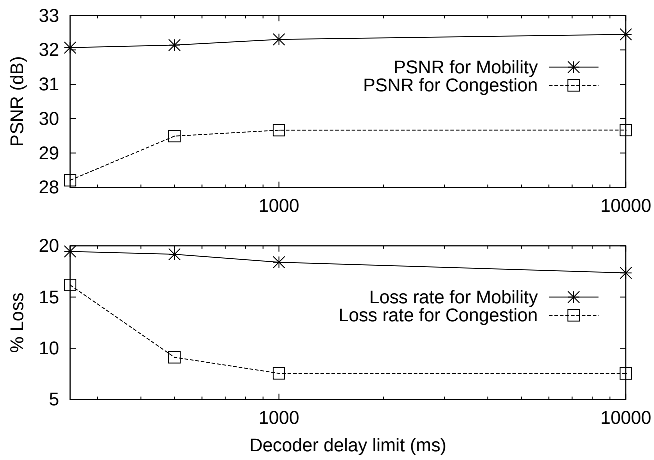


Fig. 11. PSNR and Packet Loss Rate variation for different delay thresholds

Real-time video tightens the limits of end-to-end delay and jitter. Depending on the decoding strategy and buffer size, different degrees of flexibility can be achieved. The results presented in figure 11 show the variation in terms of distortion and packet loss rate when applying different delay thresholds to the video stream. As it can be seen, the effects of congestion are apparently more negative since worse distortion values are achieved at smaller loss rates. It should be noticed, however, that long packet burst losses result in image freezing. This effect, though much more annoying to a human viewer, does not translate entirely in terms of PSNR due to the fact that the minimum value achieved is around 13 dB, and not zero. Random packet losses, on the other hand, cause more uniform distortion levels, being therefore more suitable from a human viewer point of view.

As it could be inferred from previous results, tightening the limits on packet delay causes more negative effects in high-congestion scenarios than in high-mobility ones. However, these effects can be countered by QoS policies at either the MAC or higher levels. Transmission breaks due to mobility are much more difficult to counter and are more critical. Solutions to this problem could be introduced at the MAC level itself by assigning routing traffic a higher priority as proposed in the developing standard IEEE 802.11e [5]. Due to the nature of the wireless channel, though, we are not able to provide a 100% delivery guarantee even to a single surrounding node.

F. Evaluation of video codec choices

Our evaluation concerning the video codec parameters focuses on two topics: the number of reference frames for motion estimation and the best method for intra-macroblock updating. The evaluation relative to the number of reference frames was done using the heavy congestion scenario presented in the previous subsection.

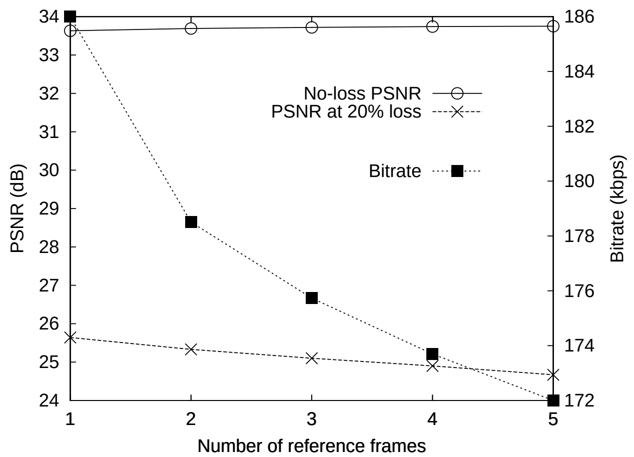


Fig. 12. Performance under high congestion of a variable number of reference frames

In figure 12 we present the distortion achieved in this scenario. As it can be seen in that figure, the use of multiple reference frames increases compression and reduces the bit-rate slightly, being therefore the expected result. In terms of error-resilience there is a monotonous distortion decrease and a degradation of 1 dB is achieved by using 5 reference frames instead of just 1.

Since this result was unexpected according to [14], we completed our analysis by evaluating the performance of this parameter in the situation it was originally proposed for: entire frame losses. Instead of running a high mobility scenario (known to cause that kind of losses), we have directly tested the effects of loosing 1 to 5 consecutive frames, so that the error propagation effect was presented as clearly as possible.

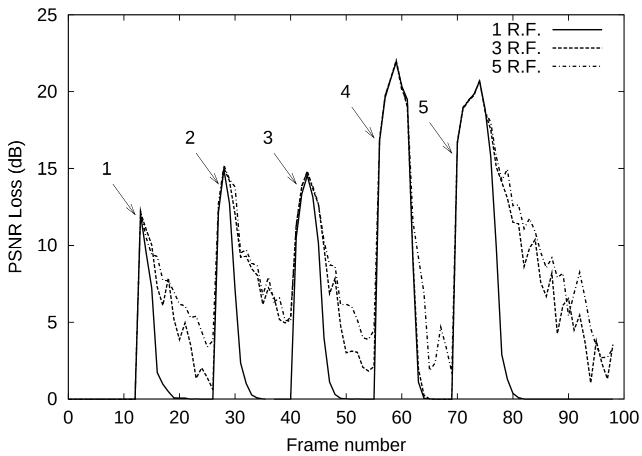


Fig. 13. Analysis of error propagation by simulating an increasing number of entirely lost frames

Results shown in figure 13 evidence the appreciated behavior when using 1, 3, and 5 reference frames. Number/arrow pairs refer to how many frames were lost. From figure 13 we can conclude that using a single reference frame is the most effective choice to reduce temporal error propagation, being therefore an adecuate tuning choice for H.264 in MANETs; demands in terms of memory on both encoder and decoder are also reduced by this setup.

Concerning intra-updating of macroblocks, H.264 provides several choices to the user. We have evaluated the main available choices in the reference software, which are: use of I frames, intra update a pre-defined number of macroblocks randomly and intra update a whole macroblock line randomly chosen for each frame.

In this process, all test files are encoded at the same bit-rate by varying the global quantization values. Such process is required because the H.264 software codec does not currently provide a loop-back mechanism for bit-rate control. This allows a fair comparison between different parameter choices, paving the way for more meaningful conclusions.

The scenario is the same one used in previous subsections. It is considered as an example of high congestion, with a packet loss ratio of 20%. Besides this scenario, we also created one with low congestion (4% loss) to provide a more consistent and general evaluation.

Figure 14 shows the difference between the two most effective choices for intra updating, along with a reference solution with no intra updates. As it can be seen, for a same average number of intra macroblock updates, the random solution presents better and more stable results. Table I presents the

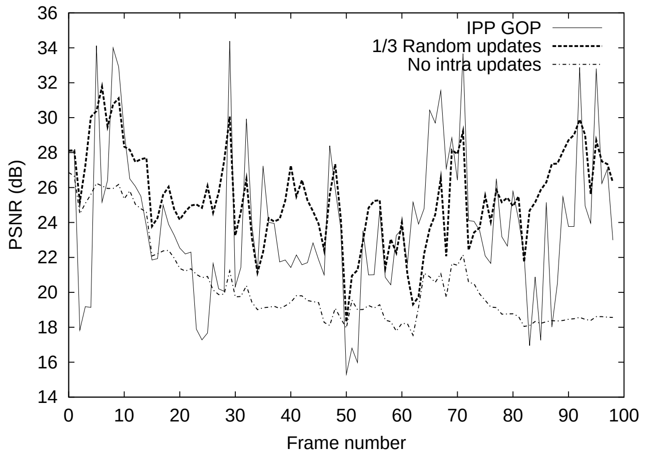


Fig. 14. Evaluation of the most effective techniques for intra macroblock-updating under high congestion

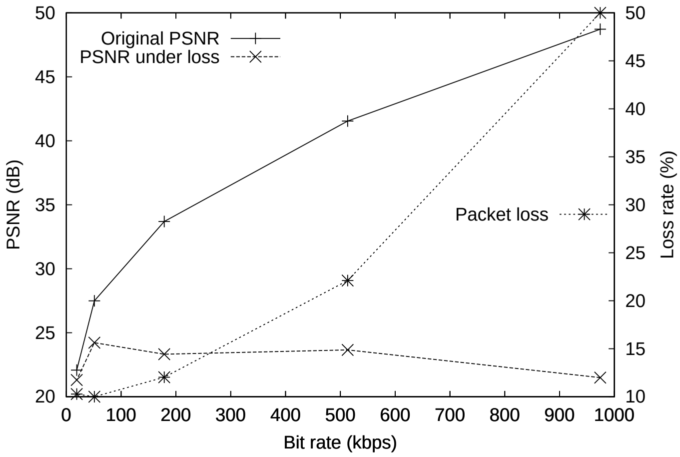


Fig. 15. Behavior experienced by varying the sequence's original distortion

average distortion values for this scenario for low and high congestion levels. The use of random macroblock updates proves to be the best option in terms of error-resilience, showing its effectiveness with respect to no updating (around 5 dB of difference).

TABLE I
AVERAGE PSNR RESULTS EVALUATING STRATEGIES FOR INTRA MB UPDATING

Updating method	Mean PSNR (20% loss)	Mean PSNR (4% loss)
1/3 random MB updates	25,58	30,63
IPP GOP sequence	24,01	30,19
IPPPPP GOP sequence	23,35	29,32
Random line intra update	22,79	28,93
No intra MB updates	20,62	25,84

The process of random intra-macroblock updating could be tuned to adapt to network congestion interactively. This process would require a feedback channel, which would also increase network congestion. Therefore, we didn't consider it a priority, though certainly a possibility.

G. Source distortion tuning

To achieve the results presented in previous sections we used an average bit-rate value by setting the quantization parameter to a mid-scale position. The current analysis focuses on the results achieved by using different quantization values, which produce distinct video distortion and bit-rate values at the source. The scenario used was the high congestion one referred to in the previous subsection, 1500x300 meters in size.

Figure 15 presents the results of our evaluation. An interesting effect can be noticed in that figure, whereby improving the original distortion does not translate into better PSNR at reception. This effect happens because the PSNR growth is surpassed by the packet loss increase. The distortion experienced by the user is, therefore, almost constant, with a maximum around 50 kbps.

By looking at table II we observe that high bit-rate values provoke another drawback since, not only congestion is increased - resulting in higher PSNR decays - but the standard deviation is also much higher. The user will, consequently, perceive less stability on the quality of the video stream.

TABLE II
PSNR DECAY AT DIFFERENT BIT-RATES

Bitrate (kbps)	Mean PSNR decay (dB)	Standard deviation for PSNR
19.08	0.77	0,81
51.15	3.27	1,64
178.64	10.37	2,55
513.35	17.88	4,22
974.18	27,23	4,08

VI. CONCLUSIONS AND FUTURE WORK

We presented the main issues related to 802.11b based MANETs, taking into account the requirements of real-time video. The results from previous works related to H.264 were used to tune the video flows to achieve good error-resilience under severe losses. A preliminary analysis, focused on typical re-routing times of MANET routing protocols, evidenced the effectiveness of link-level aware routing protocols in re-routing tasks.

We proceeded with a mobility evaluation under average congestion, where TORA offers the best distortion results to the video stream. Variable congestion tests followed using TORA and AODV. Using TCP as background traffic, TORA has only provided slightly better results with less than 10 connections, with AODV offering a better overall performance. In fact, up to four extra video connections can be achieved with AODV relative to TORA maintaining the same level of distortion.

The obtained results evidenced that even though routing protocols detect broken links in milliseconds, they are not able to perform re-routing tasks as quickly as it would be desired. This phenomena occurs because, due to collisions, they are not always able to successfully broadcast routing packets, causing long transmission breaks. In fact, increasing background traffic intensifies this problem, causing routing tasks to become more and more unfeasible.

An analysis of delay and jitter followed, showing the effects of congestion and mobility on video streams separately. Here, the ON/OFF behavior with high mobility can cause the loss of communication during long time periods (i.e., 10 seconds or more), being therefore prone to cause annoyance to the receptor. This point will require special consideration in further enhancements.

Concerning the H.264 video codec, we have also showed that the tuning performed was effectively resilient in terms of macroblock updating. The use of more than one reference frame, though effective in reducing bit-rate, increases the temporal error propagation and it should be avoided, except for situations where the media is reliable (CD, DVD, or hard-disk).

Finally, we analyzed the effect of varying the sequence's bit-rate. Data showed that under high congestion no distortion improvement can be achieved by increasing bitrate. In fact, the optimum value found was around 50 kbps, a very low one.

Future work will focus on finding techniques suitable for offering good QoS to video streams by differentiating traffic flows, as well as by making routing related communication more reliable.

REFERENCES

[1] "ISO/IEC IS,Coding of Audio-Visual Objects, part 2: Visual (MPEG-4)," Information Technology, November 2001.

[2] "Video coding for low bitrate communication," ITU-T Recommendation H.263, 1995.

[3] I. S. 802.11, *Wireless LAN Medium Access Control(MAC) and Physical Layer (PHY) specifications*. The Institute of Electrical and Electronics Engineers, Inc., August 1999.

[4] R. Wakikawa et al., "Global Connectivity for IPv6 Mobile Ad Hoc Networks," Internet Engineering Task Force, Internet Draft, Work in progress, November 2002.

[5] "IEEE P802.11 - Task Group E, MAC Enhancements for Quality of Service," http://grouper.ieee.org/groups/802/11/, 2002.

[6] F. A. Tobagi and L. Kleinrock, "Packet switching in radio channels: Part-II - the hidden terminal problem in carrier sense multiple-access models and the busy-tone solution," *IEEE Transactions in Communications*, no. COM-23(12):1417143, 1975.

[7] C. E. Perkins and E. M. Royer, "Ad hoc On-Demand Distance Vector Routing," in *Proceedings of the 2nd IEEE Workshop on Mobile Computing Systems and Applications, New Orleans, LA*, February 1999, pp. 90–100.

[8] D. B. Johnson and D. A. Maltz, "Dynamic source routing in ad hoc wireless networks," in *Mobile Computing*, Imielinski and Korth, Eds. Kluwer Academic Publishers, 1996, vol. 353. [Online]. Available: citeseer.nj.nec.com/johnson96dynamic.html

[9] V. Park and S. Corson, "Temporally-ordered routing algorithm (TORA) version 1 - functional specification," Internet Draft, MANET Working Group, draft-ietf-manet-tora-spec-03.txt, November 2000, work in progress.

[10] C. E. Perkins and P. Bhagwat, "Highly dynamic destination-sequenced distance-vector routing (DSDV) for mobile computers," *ACM Computer Communication Review*, vol. 24, no. 2, pp. 234–244, October 1994.

[11] T. Clausen, P. Jacquet, A. Laouiti, P. Muhlethaler, A. Qayyum, and L. Viennot, "Optimized link state routing protocol," *International Multi Topic Conference, Pakistan*, 2001.

[12] Z. Haas and M. Pearlman, "The zone routing protocol (ZRP) for ad hoc networks," Internet Draft, MANET Working Group, draft-ietf-manet-zone-zrp-02.txt, June 1999, work in progress.

[13] "Draft ITU-T Recommendation and Final Draft International Standard of Joint Video Specification (ITU-T Rec. H.264 — ISO/IEC 14496-10 AVC)," http://bs.hhi.de/ wiegand/JVT.html, March 2003.

[14] M. Budagavi and J. D. Gibson, "Error Propagation in Motion Compensated Video over Wireless Channels," *Proceedings of the IEEE International Conference on Image Processing, Santa Barbara, USA*, pp. 89–92, October 1997.

[15] Ye-Kui Wang, Miska Hannuksela, Viktor Varsa, Ari Hourunranta, and Moncef Gabbouj, "The error concealment feature in the H.26L test model," *IEEE 2002 International Conference on Image Processing (ICIP'2002), Rochester, New York, USA*, September 2002.

[16] Carlos T. Calafate and M. P. Malumbres, "Testing the H.264 Error-Resilience on Wireless Ad-hoc Networks," *4th EURASIP Conference focused on Video / Image Processing and Multimedia Communications, Zagreb, Croatia*, July 2003.

[17] K. Fall and K. Varadhan, "NS notes and documentation," The VINT project, UC Berkeley, LBL USC/ISI, and Xerox PARC, November 1997.

[18] S.-J. Lee, C.-K. Toh, and M. Gerla, "Performance Evaluation of Table-Driven and On-Demand Ad Hoc Routing Protocols," in *Proceedings of IEEE PIMRC'99, Osaka, Japan*, September 1999, pp. 297–301.

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