Issues on MPEG over Wireless Broadband Networks

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Abstract: *This paper discusses the transmission of MPEG video over wireless broadband networks. We look at the effect of wireless networks characteristics such as high error rate, mobility, limited processing capacity and bandwidth on the quality of the video at mobile terminals. We present results of transmission of MPEG flows over wireless links and show the applicability of a dynamic combination of Forward Error Correction (FEC) and Automatic Repeat Request (ARQ) in order to minimize the effects of transmission errors.*

1 Introduction:

Video traffic sources are expected to utilize a significant amount of resources in future integrated broadband networks [4]. Because of the high amount of bandwidth required by such sources, it is necessary to compress the original video stream in order to reduce the bit rate necessary to transmit each picture at a reasonable cost. The amount of data obtained after compression depends basically on the amount of activity in the scene and the compression protocol involved. For these reasons video sources generate a variable bit rate stream (VBR).

The Moving Picture Expert Group (MPEG) is becoming the standard video coding technique for most applications in the multimedia world. In these days we are beginning to see the first MPEG decoder VLSI chips imbedded in personal computers, and the network video market is expected to grow rapidly in near future. At the same time, the proliferation of portable computers with wireless capabilities together with the deployment of higher bandwidth wireless links is making it possible to transmit not only data, but also voice and video, over wireless networks. MPEG was originally designed for storage applications instead of transmission. However it has become de facto standard encoding protocol for most digital video applications. This issue, as we will show later, complicates the task of designing an efficient transmission protocol for MPEG video over both wired and wireless networks. The inherent characteristics of wireless links will introduce a complete new set of issues that need to be taken into account in order to provide a reasonable QoS.

In this paper we show the effect of wireless links characteristics on the displayed picture quality at a mobile terminal. Furthermore, we show the effect of simple error control schemes on the received picture quality under different error conditions.

The organization of this paper is the following: In section 2, we discuss the most important characteristics of the MPEG standard for transmission purposes. In section 3, we discuss the main issues when MPEG video is transmitted over wireless access broadband networks concentrating mainly on issues regarding the base-station. In section 4, we investigate how the high error rate characteristics of wireless links affect MPEG video streams. We show in detail how Forward Error Correction (FEC) and Automatic Repeat Request (ARQ) can be used to overcome these problems. In this section we also discuss issues related to mobility, limited bandwidth and limited processing capacity at the mobile terminal.

2 The MPEG standard:

The Moving Picture Expert Group (MPEG) video coding protocol was designed to achieve the maximum quality of video for a given bandwidth [1]. MPEG was designed to cover a wide range of applications, bit rates, resolutions, quality and services. MPEG performs a high compression ratio with still a good quality and supports other nice features such as random access, fast search and reverse playback.

In order to achieve a high compression ratio MPEG has to exploit the natural redundancy of video sequences as much as possible. There are basically two types of redundancies that are exploited:

- Temporal: This redundancy is due to the high correlation between pictures in a video sequence. It can be removed using temporal prediction with motion compensation (forwards, backwards or interpolate).

- Spatial: This redundancy appears due the high correlation among pixels of the same picture. This can be reduced by using an orthogonal transformation to set some of the coefficient to zeros.

MPEG defines three different types of pictures: I picture (intraframe encoded picture), P picture (predictive encoded picture) and B picture (bi-directional predictive encoded picture). The I pictures are encoded independently from any other picture, so the degree of compression is not very good since they do not take advantage of the high degree of temporal correlation between consecutive pictures. P pictures achieve a better compression ratio because they use motion compensation with respect to the previously encoded I or P pictures. B pictures are coded either by using motion compensation prediction with respect to the previous I or P picture or, by performing interpolation between I and P pictures which provides the highest degree of compression [3].

In MPEG, the order in which the encoder introduces the different pictures in the output video stream is very flexible and depends of the requirements of applications in terms of video quality. MPEG defines this order in the group of pictures (GOP) specification. In a GOP, there is always at least one I picture, some P pictures and maybe some B pictures between I and P pictures. The I picture as well as the residual error images (picture predictor minus the picture that is being encoded) of the P and B pictures are divided into blocks of 8x8 pixels. On each of these blocks a two dimensional Discrete Cosine Transformation (DCT) concentrates the energy of each block into a few low order coefficients that are afterwards quantized. The compression is obtained because not all the coefficients survive after quantization and many of them become so small (zeros) making their transmission unnecessary.

Because of the high error rate characteristics of wireless links, the erroneous reception of MPEG packets will be a common event at the mobile terminal. For this reason and for further discussion in this paper is important to know the format in which the MPEG encoder transmits the video stream. Appendix 1 shows a diagram with the video sequence, GOP, picture and macroblock structures. The detail of this formats are beyond the scope of this paper, but it will be helpful, at least now, to note the difference between control and DCT information.

3 MPEG at the Wireless Access Point (Base-station)

In the past few years, the possibility of transmitting VBR video over wired networks (in particular ATM networks) has been one of the key areas of study in the research community. However, transmission of video over wireless access networks is a new area that is just recently gaining attention [6], [8].

MPEG presents two big disadvantages in wireless networks. These problems are basically due to the periodic peaks in its bit rate and because of the difference in image quality perception of different types of pictures. In other words, the MPEG video stream is not constant in a quantitative way (bit rate), and neither in qualitative way (picture type). In the remainder of this section we will look at how these issues will impact the quality of the video at a wireless terminal.

Quantitative disparity:

Figure 1 shows a MPEG picture size histogram over 1200 pictures of the film *Ben-Hur*. The histogram was obtained using a MPEG encoder developed at CTR (Center for Telecommunications Research at Columbia University). The group of pictures sequence used in this test is IBBPBBPBBPI.

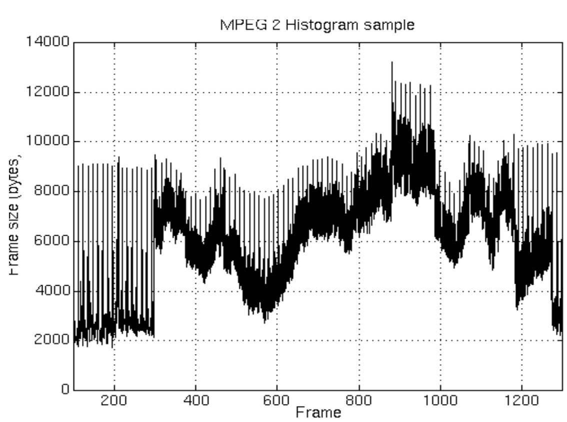


Figure 1. MPEG histogram example.

In figure 2 by observing fewer GOPs we will be able to appreciate in detail the size differences among pictures.

MPEG 2 Histogram sample

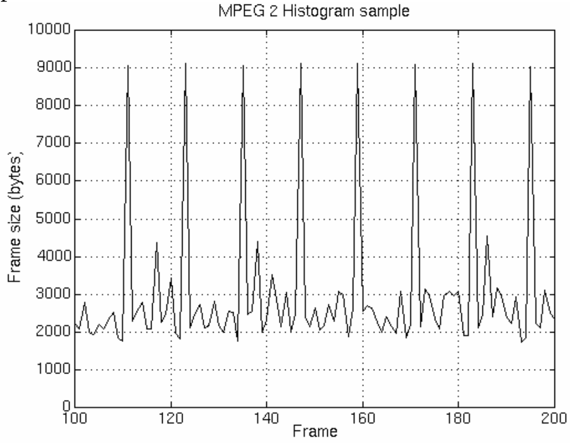


Figure 2. Few GOPs on a MPEG histogram.

Considering few GOPs we can easily observe that the size in bytes of I pictures is four to five times bigger than the size of P and B pictures (in our example scenario). It means that if the average picture size is around 3,000 bytes, we will have almost twice per second a peak of 10,000 bytes lasting only 42 milliseconds (if a picture rate of 24 pictures per second is being used).

The periodic peaks in the MPEG video stream complicate the task of designing an efficient transmission protocol for such type of sources. These peaks may suddenly congest at the base-station. This is due to the immense difference between wireless and wired networks in terms of transmission capacity as well as the scarcity of the wireless bandwidth.

Let us first provide a quick review of MPEG streams over packet switched networks. For real time applications to work properly, the network has to reserve resources along the flow path to guarantee that packets will be delivered correctly according to delay and BER bounds defined at set up time. For MPEG flows, if the network reserve resources based on I picture requirements (peak bandwidth allocation), it can be guaranteed that the switches may not become congested, however this allocation will be inefficient since P pictures do not require as much bandwidth to be transmitted. In the other case, if the bandwidth allocation is between average and peak rate, I pictures of many sources might suddenly appear at the same time at one switch generating congestion.

The encoder will try to retransmit a lost packet, however, the extra delay involved and the necessary amount of buffer in the receiver may not make re-transmissions suitable. Even without retransmissions, a link shared for several users will introduce variations in the delay at which each packet is delivered (jitter delay) to the final user. This make necessary to buffer the packets in the receiver for some time before they are played (decoded and shown on the screen in this case) to compensate for delay delivery variations.

For wireless access broadband first and second generation networks this problem will be even more complicated. In wireless networks a fix amount of bandwidth is assigned per user connection (using either FDMA, TDMA or CDMA air interfaces). However as we noted before, the variable bit rate nature of MPEG video does not seem to fit efficiently in a fixed bandwidth allocation environment unless peak rate allocation is assigned.

In recent years there has been a lot of work to propose new types of air interface protocols for third generation wireless networks that support both fix and variable bit rate applications [9], [11]. The bit rate distribution is not the only concern since these protocols have to deal also with applications having different QoS requirements in terms of maximum delay and packet loss rate. Using these protocols, however, does not solve the congestion a the base-station completely. Namely, if at the base station peaks of MPEG streams appear at the same time they can generate congestion as well. During congestion the base-station may, depending of the situation, either drop video packets, or deteriorate the QoS of other active connections in order to obtain extra bandwidth. In both cases the solutions will not be desired. The issue is that if the variability of the bit rate generated by MPEG can be reduced, it will immediately improve the performance of the transmission protocol across both the wired and wireless paths. It is important to kwon that, as the number of MPEG connection increases, the statistical multiplexing gain of the switch (base-station) increases as well making the overall QoS performance of each user less susceptible to individual variations. However, when the multiplexing gain is not high, one effective solution will be to reduce the variability of the MPEG stream at the encoder itself. Frame spreading [2] is a technique that may be applied not only to MPEG video but to any other VBR video source resulting in a reduction of its burstiness without modifying the quality of the pictures. In Frame Spreading, small size pictures (such as P or B pictures) are transmitted faster than the frame period which allows the following big picture (I picture) to use that remaining time fraction plus its own frame period to transmit its data slower, reducing its original transmission rate. In [2], results of frame spreading over MPEG-1 streams showed that there is an immediate reduction in the burstiness of the source traffic as frame spreading is increased from 1 to 2 pictures. The way MPEG transmits the DCT data can also be used advantageously in the network to overcome congestion situations. In MPEG the DCT data is transmitted in a hierarchical way, it means that low frequency DCT coefficients (basic quality) are transmitted before high frequency DCT coefficients (details), so routers along the path may drop, if necessary, high frequency DCT coefficients without a big level of damage in the quality of the video at the mobile terminal.

However increasing the frame spreading beyond three or four frames does not reduce noticeably the burstiness of the source. For MPEG over wireless links these two schemes, statistical multiplexing and some "burstiness reduction " techniques will be the key point to truly support highly changing resource-demanding video applications.

Quality Disparity:

The difference in image quality perception of different type of pictures also complicates the transmission protocol for these types of sources over wireless links. In fact, as we explained earlier and will show later, the loss of packets belonging to I pictures has an immediate detrimental effect on the overall quality of the video. For this reason it will be very useful if each packet transmitted is marked with a flag identifying if the packet belongs to an I, P or B picture. By using and appropriate scheduling mechanism that looks into packet type, packets belonging to I pictures can be transmitted at the cost of discarding P and/or B pictures at the time of congestion. Now, by receiving I frames at least some level of picture quality is preserved.

In fact, a flag (one bit) has already been included in every ATM packet for that purpose so that, in case of congestion, routers can selectively discard packets. Another solution will be to transmit only intraframes (I) pictures. However, as we mentioned in section I, I pictures require too many a bits to be encoded, resulting in a prohibited average bit rate. For this reason mixed intra/interframe encoders are expected to be widely used. In ATM networks, some researchers have proposed to transmit different types of coded MPEG pictures on separate links (VC) as a way to minimize the variability of the bit rate. Transmitting I and P pictures on different links reduces the variability of the bit rate in each link compared when both types of pictures are transmitted together decreasing the overall packet loss rate [6]. However, in wireless access networks will be very complex.

Another dimension of the problem has to do with compressed video sources itself. For this sources it is common that the loose of few key packets belonging to a particular picture, make the transmission of the remaining packets useless because they can not be decoded correctly. This has to do with packets that contain very important control information such as synchronization, quantization, codewords, motion compensation, etc. (refer to Appendix 1).

In this paper we refer to such packets as control packets and we refer to packets containing only DCT information a video packet. Now since the reception of control packets is absolutely critical and without them picture can not be decoded correctly, the base-station must send such control packets with a higher priority compared to video packets. This can be done by an appropriate tagging of packets by the encoder and a scheduler at the base-station.

5 MPEG video over wireless links.

In addition to the issues arising at the base-station, several other issues need to be considered when the video stream is transmitted to mobile/wireless terminals. The issues are:

- Error rate characteristics
- Mobility
- Processing capacity limitations at the mobile

These issues, as we will show later, have an immediate detrimental impact on MPEG quality at the mobile terminal. We now discuss each issue in more detail.

4.1 Error rate characteristics.

So far, transmission of MPEG video streams has been studied over protocols that are designed with links of high reliability and where packets losses are mostly due to congestion. However in wireless links packet loss is mainly a result of high bit error rate of the wireless medium and this will drastically deteriorate received picture quality

In order to test the effect of wireless link errors on MPEG streams we used a very simple model that introduces random errors representing a fast fading environment. This is represented in figure 3.

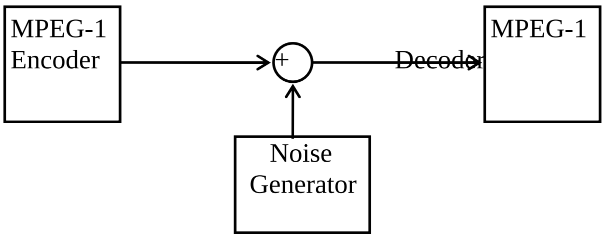


Figure 3. Wireless link error simulation test-bed

In our test-bed, data is packetized into ATM cells (48 bytes). We assume that no contention for access the link occurs and the necessary bandwidth in the network is always available to the video stream. By doing this we isolate only the error behavior of the system. The video source we use is a MPEG-1 sequence of the film *Ben-Hur* with a GOP sequence IBBPBBPBB. The bit rate of the video is 500 Kbps. Figures 4, 5 and 6 We show the first (I Picture), third (B Picture) and fourth (P picture) of a sequence (without errors) in order to be able to make comparisons later.

Sequence 1 (original, no errors)

Figure 4. Picture 1 (I Picture)

Figure 5. Picture 2 (B Picture)

Figure 6. Picture 4 (P Picture)

Fast fading (Multipath fading). Fast fading causes the strength of the local mean signal to change rapidly due to reflections of the main signal in natural and human-made objects, movement of the transmitter and receiver or movement of objects between them (Doppler effect). For fast fading we assumed that the packets will be delivered with only single random errors.

Test (1). We introduced errors but just in the first bytes of the video stream. When this is done the hole session crashes and nothing appears on screen. The errors detected in the decoder happen to be related with information in the header of the video stream which we refer to as control packets or control fields (refer to Appendix 1). The header contains the most important parameters for the MPEG decoder to work properly. These parameters include among others: Number of pictures in the group of pictures, Structure of the group of pictures, Intra quant matrix, Non intra quant matrix, size of the pictures, etc. As it is easy to understand control fields of the MPEG header should be strongly protected against link errors, and the strongest error correction care should be used here. In case that still some of the control bytes were corrupted the decoder should ask for a retransmission.

Test (2). For a bit error rate of 10^{-5}, there is not a noticeable effect on the quality of the video at the receiver. Since most part of the encoded video stream is a DCT data, a DCT coefficient in error has not big impact on the overall quality of that particular group of picture (GOP).

Test (3). For an error rate of 10^{-4} the situation changes drastically. For this error rate many control fields were corrupted by errors. These errors are enough to completely distort about 10 to 20 percent of the macroblocks per picture (refer to figures 7, 8 and 9). However, for a viewer used to clean video quality, the presence of these few distorted macroblocks will have a much stronger impact on image distortion.

Sequence 2. (BER = 10^{-4}, no errors in the header of the GOP).

Figure 7. Picture 1 (I picture)

Figure 8. Picture 2 (B Picture)

Figure 9. Picture 4 (P Frame)

Test (4). For an error rate of 10^{-3} among 30 to 60 percent of the macroblocks on the screen are distorted.

Results of these tests clearly suggest that it is the type of field in error which for the most part decides the level of damage in the quality of video rather than simply the raw bit error rate. The problem now is that we need to protect control packets against transmission errors much more than others in order to achieve a reasonable quality of video. The final goal is to hide the high error rate from the higher layers without a large overhead so protocols and software already in use can still work properly.

Different schemes can be used in order to guarantee that packets containing important control fields will be received correctly at the mobile terminal. In the following we discuss the effect of a number of such schemes.

FEC and ARQ

Forward error correction (FEC) and automatic repeat request (ARQ) have been widely used to combat errors in wired and wireless links [14]. In the ARQ approach a packet is retransmitted until it is correctly received. This technique is specially suited for links where the size of the packets is much bigger compared with the round trip propagation delay (small delay-bandwidth product). In FEC, redundant information is added to the packets before transmission so in the presence of random link errors, the original data can still be recovered at the receiver. Since it is true that FEC can reduce the error rate it can be impossible to guarantee an error-free reception with FEC only. This means that a small probability that errors will go to the higher layers remains.

For example, for ATM packets and a Satellite link with propagation delays in the order of 100-300 milliseconds running at 10 Mbps, ARQ will not be suitable. For this particular example an FEC protocol appears more appropriate. In a micro-cellular network where the propagation delays are in the order of microseconds, ARQ or/and FEC can be the most appropriate choices.

For MPEG over wireless access broadband networks the delay bandwidth product will be large since every request for retransmission means to go all the way back towards the encoder, making ARQ inefficient. Some authors [12] have proposed new mechanisms to compensate for this problem in order to make ARQ applicable. These proposals are optimized and tuned for data transmission purposes using IP networks, but the concepts can be easily extrapolated to other broadband wireless networks. This protocols basically propose to buffer the incoming video packets at the base station so in the case that one packet becomes damaged in the air, it can still be retransmitted on time using ARQ locally. By doing error-recovery at the base-station, the round trip propagation delay of a packet retransmitted may be short enough to still use that packet at the decoder on time.

The other alternative is to intercept packets at the base-station to increase the error correction capability of FEC to protect even more important fields according to channel conditions. However, if we selectively change the FEC error correction capability we may need more complex encoder/decoder circuits at the mobile terminal and base station in order to perform this task.

Slow fading (shadowing): In the presence of shadowing the strength of the signal changes slowly with respect to time. These variations may be due to the obstruction of the main signal by buildings and others high structures, or also because of the attenuation of the signal as the transmitter-mobile distance increases. In the presence of strong shadowing, most part of the received packets might be in error and this will happen for a long period of time.

FEC algorithms are designed to detect and correct a maximum number of bits in error in one block. When this threshold is exceeded there is no way to recover the original information. In slow fading situations we assume that the entire packet is corrupted. Some packets will even never reach the mobile terminal as they might be dropped if the base-station experiences congestion. In order to understand the effect of loosing MPEG packets at the mobile terminal we perform the following tests:

Test (5). We divided the encoded data into 53 bytes ATM packets (48 byte data plus 5 byte header), and randomly some of the packets were delivered in error. The main effects appearing in this case are mostly the same as in single random errors. The loss of a packet containing DCT coefficients has almost no effect on the overall quality of the video. However the loss of a packet containing control fields will drastically deteriorate the quality.

Obviously some techniques such antenna diversity can be added to be sure that even in the presence of slow fading the FEC protection scheme will keep the error rate below 10^{-4}, 10^{-5}. However this technique will introduce more complexity (cost) in the system that may not be always affordable. Another way to overcome slow fading situations will require the source to slow down its transmission rate (say reduce the frame rate) so the BER decreases to a point where FEC alone is again enough to kept and acceptable video quality. However, the time involved in telling the source to decrease the bit rate and the time the mobile terminal can start seeing the effects will take to long. For these reasons we consider that an ARQ capability should still remain in the base station to allow retransmission of important packets that might become lost in the air. This leads to a hybrid FEC/ARQ error correction protocol to overcome single and packet level errors.

A packet can be re-transmitted only if the extra time necessary to re-transmit that packet fits into the strict delay boundaries of showing that picture in time. To clarify this point lets suppose that the period of time in which all the packets belonging to one picture should reach the mobile is [(i-1)T, iT], where T could be the 1/picture_rate. A packet received in error at time t, where (i-1)T $\le t \le$ iT, can be retransmitted only if the retransmission time takes place in the interval T_{Ret} = iT - T. If the packet is not received at that time, or if it still contains errors, it should be dropped.

For this reason we propose to snoop the connection at the base station so the retransmission times involved are much shorter allowing ARQ protocols to be more effective (figure 5). This implementation makes it necessary to buffer the MPEG packets at the base station until acknowledgments are received from the mobile terminal. However, as we mentioned in the previous discussion, not all the packets need to be forcefully received correctly. In this case we propose to mark important control packets with a flag that guarantees that those packets will be delivered correctly (packets that may be buffered and re-transmitted if necessary). Other packets (mainly packets with DCT coefficients only) will not be tagged with the flag. The application layer at the encoder should be modified to send priority information to the transport layer in order to set the flag.

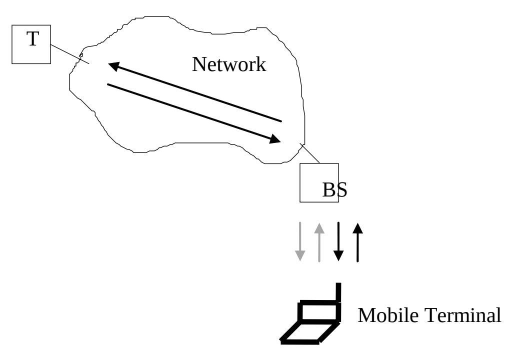


Figure 5. Error recovery termination. (a) → At the Transmitter, (b) → At the base-station.

Obviously, depending of the fade duration, the number of packets corrupted by errors is variable. Because MPEG generates different type of pictures is important to understand how the lost of a picture affects the quality of the following pictures. Since B pictures need one of the previous pictures and one of the following pictures to be decoded they are not causal because they require out of order transmission. The use of B pictures at the decoder makes it necessary to have bigger buffers to save at least two pictures in order to decode the corresponding B picture. B pictures will also imply a larger reconstruction delay that may not be tolerated. For these reasons B pictures are not suitable for real time applications. Once B pictures have been taken away of the group of pictures a sequence of MPEG pictures may look as follows:

I I I I I I I I I I . . . or I P P P P P P P P I . . .

In the fist case only I pictures are encoded. However, as we mentioned before since, I pictures do not take advantage of the high amount of temporal redundancy in video sequences they require too many bytes to be encoded. Wireless bandwidth will always be limited so the scenario of mixed I and P pictures seems to be the most suitable option.

Since different type of pictures will be delivered to the mobile terminal, it is important to understand how the video degrades when errors corrupt different type of pictures. For picture level tests we only introduce errors in a single picture to see the propagation effect of these errors in following pictures.

Test (6). In this test errors are introduced but just in the first picture of the group of pictures (I picture). These errors will corrupt some macroblocks in that picture that remains during the following P pictures (refer to figures 7 and 9). The reason is that P pictures are decoded with respect to the previous I or P pictures. This issue makes it imperative to protect I pictures with a stronger FEC algorithm or by using retransmissions. If the I picture is completely corrupted it is no sense to transmit the rest of the group of pictures (P pictures), since all the information in the remaining pictures will be useless at the mobile terminal.

Test (7). In this test, errors do not corrupt packets in I pictures, but errors will corrupt P pictures. Common errors we got are mostly DCT coefficients out of range that introduce dark spots on the screen that in some sense may be tolerable. However, protecting I pictures against link errors improve the quality of the video noticeably (refer to figures 10, 11 and 12). Some times dark macroblocks appear and remain on the following pictures if errors hit P pictures macroblocks that are used as reference by the followings P pictures (as in sequence 2). This suggests that important control fields of P pictures should be also guaranteed an error-free delivery.

Sequence 3 (No errors in I pictures, BER 10^{-4})

Figure 10, Picture 1 (I picture)

Figure 11, Picture 2 (B Picture)

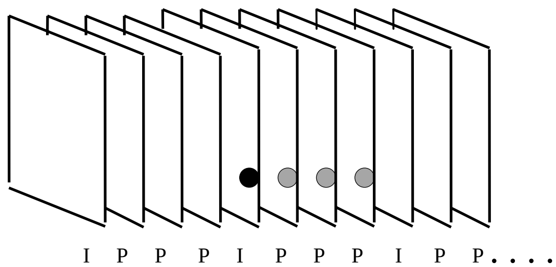
Figure 12, Picture 4 (P Frame)

In a mixed Intraframe/interframe scenario except for the first picture, the remaining pictures (P pictures) need the previous picture to be decoded. This means that if errors hit one picture macroblock in the i th. picture, that error will also appear in picture i+1, i+2, i+3 ... N-i, where N is the number of pictures in the group of pictures. As a result, one macroblock in error is going to remain on the screen until a new I picture arrives to a decoder. Knowing this, the mobile terminal can request the encoder to reduce the ratio of intraframe to interframe pictures so corrupted blocks will just propagate a maximum of N-1 pictures. Reducing the number of P pictures among I pictures will noticeably increase the bit rate but this can be compensated with a bigger step size when quantization is taken place at the encoder. An example of this can be seen in Figures X and Y, where the video bit rate has been reduced from 500 to 100 Kbps.

Figure 13, Bit rate = 500 Kbps

Figure 14, Bit rate = 100 Kbps

In the case that bandwidth is still a limitation to support more I pictures, the encoder might even reduce the picture rate to compensate for the additional I pictures bit rate. The basic design criteria will be that it is better to show few good pictures per second in the mobile terminal rather than show many bad pictures.



● Macroblock corrupted by link errors.
● P picture's Macroblock decoded erroneously.

Figure 15. Propagation of errors in a MPEG's GOP.

For picture level errors we propose to mark all the packets belonging to I pictures so that these packets will reach the mobile correctly with an acceptable high probability. The picture spreading algorithm mentioned in section 3 will be used further to increase the interval of time in which retransmissions of I packets can take place. It also has to be guaranteed that packets containing important fields of P pictures reach the mobile correctly. Finally, depending of the error rate behavior at any particular time, we may dynamically increase or reduce the number of P pictures between I pictures.

Effects of mobility:

In wired networks, the users are connected to terminals that are always attached to the same place. This makes the task of routing packets much easier and, since users are static, the network can be designed to provide a minimum amount of resources (bandwidth) guaranteeing a minimum QoS. Wireless capabilities allow users to move to places covered possibly for different base-stations. The new issues that mobility creates affect both the network and transport layers. In a wireless connection, when the quality of the link degrades at certain point, the mobile terminal needs to find a new link with better quality to continue the communication leading to a hand-off to other base-station. However it can not be guaranteed that the resources available in the new link satisfy the minimum requirements for MPEG. In general, the new link will always provide less, equal or more bandwidth. Depending on the topology of the network, the mobile terminal may find in the new location a completely different link performance in terms of delay and bit error rate. This leads again to the requirements for an algorithm that can dynamically adapt to the conditions in the network along the connection's lifetime.

Processing capacity limitations at the decoder.

The network might provide resources (bandwidth) to transmit the packets properly but, if the resources in the mobile/wireless terminal are not fast enough to decode the incoming data in time, may result in poor video quality. It is important to remember that still the processing capacity of a powerful portable computer will not be comparable as the capacity of a fixed host. In the previous section, we pointed out that the first target bit rate may be given by the network capacity (considering both, the wired and wireless paths). However this bit rate has to agree with the maximum bit rate that can be processed at the mobile. The portable computer should specify at the beginning of the connection what is the maximum transmission speed that can be decoded and the buffer space that will be available. When this information is received in the encoder, it can be interpreted and translated to set parameters such as frame rate, picture size, color format, etc. in order to fit the decoder resources.

Conclusions and Future Work:

As we showed in this work, the periodic peaks in the bit rate generated for MPEG encoders plus the picture encoding type (compression scheme involved) complicate the task of designing an efficient protocol to transmit these sources over broadband access wireless networks. We extensively show the effects of high error rates on MPEG streams pointing out an algorithm to compensate for these errors. We show that for the most part it is the type of field in error which determines the level of damage to the quality of the video rather than the raw bit error rate only. We show that an ARQ capability at the base-station should be preserved in combination with common FEC techniques as a way to compensate for both, the high error rate characteristics of the links and the impact in image quality of different type of pictures. We also discuss the advantages of periodically feedback information to the source to readapt to changes in the network.

The analysis discussed in this work is based on MPEG-1, in future we will look at how the scalability issues available in MPEG-2 behaves and can be used to overcome the aggressive characteristics of mobile/wireless applications.

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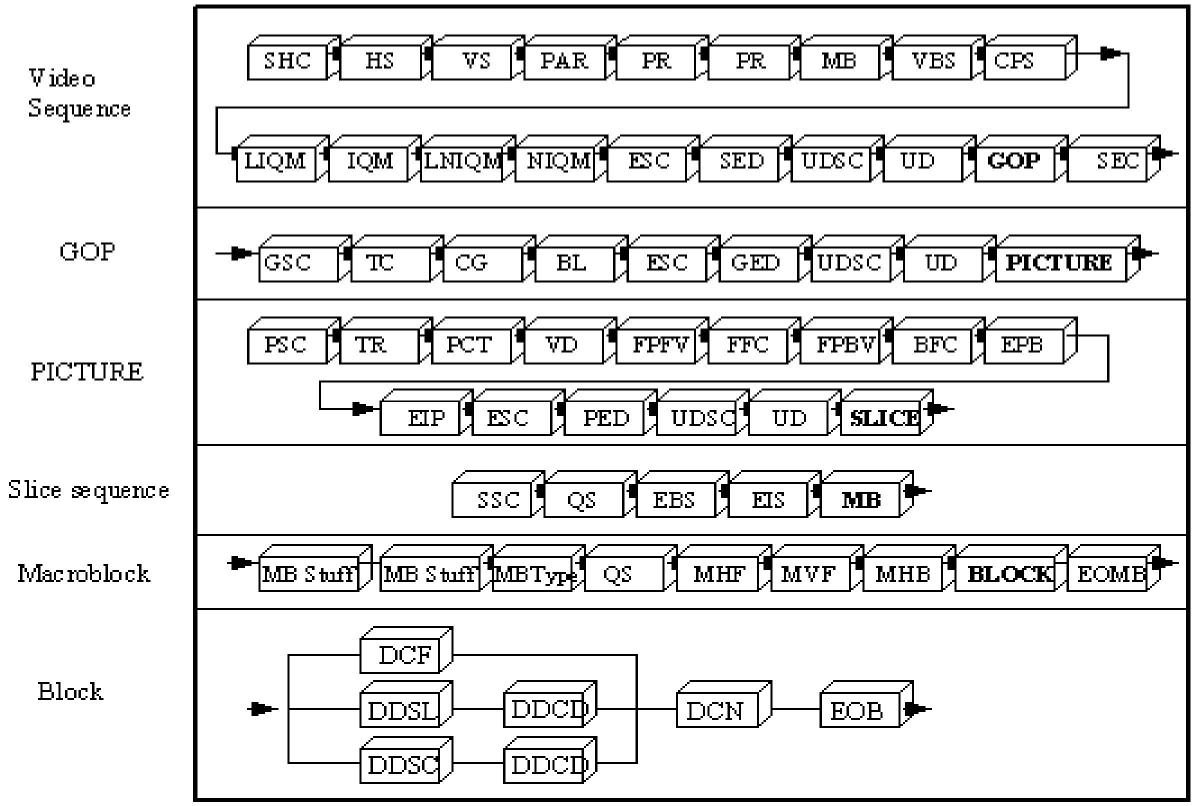
MPEG-1 FORMAT



Sequence Header

SHC: Sequence Header Code
HS: Horizontal Size
VS: Vertical Size
PAR: Pel Aspect Ratio
PR: Picture Rate

BR: Bit Rate
MB: Market Bit
VBS Video Buffering Ver.
CPF: Const. Par. Flag

LIQM: Load Intra Quant Matrix
IQM: Intra Quantize Matrix
LNIQM: Load Non Intra Quant Matrix
NIQM: Non Intra Quantize Matrix
ESC: Ext. Start Code
SED: Seq. Extension Data
UDSC: Use Data Start Code
UD: User Data
SEC: Sequence Start Code

GOP

GSC: Group Start Code
TC: Time Code
CG: Closed GOP
BL: Broken Link
ESC: Extension Start Code

GED: Group Ext. Data
UDSC: User Data Start Code
UD: User Data

Picture

PSC: Picture Start Code
TR: Temporal Reference
PCT: Picture Coding Type
VD: VBV Delay
FPFV: Full Pel Forward Vector
FFC: Forward f Code
FPBV: Full Pel Backwards Vector
BFC: Backwards f Code

EBP: Extra Bit Picture
EIP: Extra Information Picture
PED: Picture Extension Data
UDSC: Use Data Start Code
UD: User Data

Slice

SSC: Slice Start Code
QS: Quantizer Scale
EBS: Extra Bit Slice
EIS: Extra Info. Slice

Macroblock

MB Stuff: Macroblock Stuffing
MB ESC: Macroblock Escape

MBAI: MB Address Increment
MB TYPE: Macroblock Type
QS: Quntizer Scale
MHF: Motion Hor. Forward Code
MVF: Motion Ver. Forward Code
MHB: Motion Hor. Backward Code
MVB: Motion Ver. Backward Code
CBP: Coded Block Pattern
EOM: End of Macroblock

Block

DDSL: DCT DC Size Luminance
DDSC: DCT DC Size Chrominance
DDCD: DCT DC Differential
DCF: DCT Coefficient First
DCN: DCT Coefficient Next
EOB: End of Block