Title: Evaluation of Codec Behavior in IP and ATM Networks

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Abstract

As multimedia applications are becoming more and more widespread there is a need for video data to be transmitted over IP as well as ATM networks. While ATM with its Quality of Service features is able to provide adequate transmission guarantees for the strong time and bandwidth constraints of video data, the transfer of video over IP networks is an issue wherever ATM is not available and applications are used that can tolerate longer delays and occasional impairments.

The study evaluates a codec for teleconferencing and teleteaching applications over an IP network. A direct comparison is made to a MJPEG codec for similar applications over ATM with empirical measurements in a simple testbed. The main focus lies on the Quality of Service parameters delay, jitter and subjective picture quality. Network overload situations and realistic network behavior are simulated and their impact on picture quality is evaluated.

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1 Preface

With an increased use of video and audio transmissions over data networks the term Quality of Service (QoS) has become a primary focus of interest. Multimedia applications with their strong time constraints cannot rely on best effort service, but are in need for a guaranteed transmission quality if video and audio sequences are to be displayed at the receiver without interruptions.

Before the video signal of a camera can be sent over an IP or ATM network coders compress and transform the signals into packets suitable for transmission. Once the packets have reached their destination it is the responsibility of decoders to regain the video signal and make it available for display.

In the first chapter of the paper QoS parameters for video transmissions are explained. This paragraph is followed by a discussion of QoS in IP and ATM networks. In chapter three an IP codec for teleconferencing and teleteaching applications is evaluated with the main focus on delay, jitter and picture quality over an IP network. A direct comparison is made to a MJPEG codec for similar applications over ATM. The influence of network impairment or overload situations on the picture quality is evaluated as well.

2 Quality of Service Parameters for Video Transmissions

Quality of Service (QoS) can be described as the collective effort of service performance and as such determines the overall degree of satisfaction of a user with a service (ITU-94). The ISO (ISO-85) further extends this definition of QoS and includes all characteristics which can be measured or recognized by a user. Any evaluation of the quality of a service should therefore not only consider objective measurements, but a user's subjective impressions as well. Objective and subjective quality can be described by several parameters (ISO-96, ITU-92). Examples for objective measurements are throughput, delay, delay variation and errors or losses. The overall transmission quality can be considered as an example for a subjective parameter.

One of the main Quality of Service requirements of video applications is to have enough bandwidth available during the transmission to make room for a continuous flow of data which allows the receiving side to display a video sequence with its original data rate. If data packets are delayed the receiver can no longer display 25 frames per second (as in PAL) to deliver a flowing and uninterrupted motion picture. Movements become jerky as frames are lost and the picture may even freeze temporarily. The delay variation must be kept within very small limits and the maximum distance between arriving packets should not be exceeded. Every packet received after the given time frame is considered late and must be discarded. In addition to timing errors transmissions may also be affected by losses due to hardware or software errors. Packet loss can be caused by congestion in network nodes with buffer overflow.

To keep bandwidth requirements at a minimum a digital video signal of 270 Mbit/s is compressed to much smaller bandwidths before transmission. Data streams typically range from 1.5 Mbit/s to 40 Mbit/s streams after compression. This process of compression - and subsequent decompression on the receiving end - adds a significant amount of delay to the overall transmission time and is accomplished by so-called codecs (encoders/decoders). Encoding and decoding delay times are very much determined by the equipment that is used and as such have - next to the underlying quality of the network - a strong impact on the overall quality of a transmission.

3 ATM vs. IP Technology for Video Transmissions

In today's networks most of the high quality video is transmitted over ATM networks (GT-00). ATM technology was developed especially for multimedia traffic and therefore provides excellent mechanisms for bandwidth reservations and guaranteed Quality of Service. But with the increasing integration of all kinds of services into one infrastructure, the IETF proposed the addition of QoS functionality to the IP protocol as well.

3.1 Quality of Service in the ATM Protocol

In the ATM protocol, extensive QoS and traffic management functionalities are implemented. To achieve connections with different QoS characteristics the ATM Forum has defined several service categories (ATM-96). A distinction is made between real-time and non-real-time service classes: Constant Bit Rate (CBR) and real-time Variable Bit Rate (rt-VBR) provide real-time characteristics. Two other classes, the non real-time Variable Bit Rate (nrt-VBR) and Unspecified Bit Rate (UBR) are non-real-time services with no QoS guarantees.

To manage the network capacities and control the flow of data traffic management functions (ATM-96) such as the Call Admission Control (CAC) and the Usage Parameter Control (UPC) are specified. During the connection setup the CAC ensures that enough resources are available in the network and rejects a request if the required capacities cannot be provided.

If a call is accepted, the necessary performance and QoS parameters are guaranteed during the lifetime of the connection. To avoid congestion in network components the UPC monitors and controls the traffic and the validity of a connection. The UPC discards cells if the sending rate is higher than the previously negotiated rate.

3.2 Quality of Service in the IP Protocol

The Differentiated Services (BLA-98) architecture is a scalable approach to add QoS to the IP protocol in order to provide different service classes to applications. IP packets are classified according to the value of the Type of Service (TOS) field in the IP header, with each service class providing different Quality of Service features. The quality of the service class determines the queueing and scheduling algorithms and has an impact on resource allocation in the network nodes. Traffic profiles are checked to ensure that they conform with the characteristics of the class and packets are dropped if the Service Level Agreement (SLA) is violated. In the worst case scenario of a congested network, however, this approach can only provide best effort service.

Another approach of the IETF is the Integrated Services proposal (BRA-94), which is based on an admission control unit, a packet forwarding mechanism and a protocol to reserve resources in network components and end devices. One such protocol presented by the IETF is called Resource Reservation Protocol (RSVP) (BRA-97).This protocol does not offer scalability in WANs because of its complex signaling and the storage of flow states in each network node and end device. But the concept is able to offer absolute guarantees concerning the quality of a service.

4 Measurements and Tests

The study evaluates two pairs of codecs with the main focus on the parameters delay, delay variation and picture quality: A pair of Cellstack Classic (KNET) (CEL-97, CEL-98) codecs for ATM networks using MJPEG as compression algorithm, and a pair of CAMVision-2 7615 (LIT-00) Litton codecs compressing in MPEG-2 (4:2:0) MP@ML format. The Litton codec was equipped with both an ATM interface and a 10/100 Ethernet interface. For the tests only the Fast Ethernet card was used.

4.1 Delay and Jitter

To measure the delay the codecs were setup in loopback (Fig. 1) mode without any network components involved and all parameters mentioned above were evaluated:

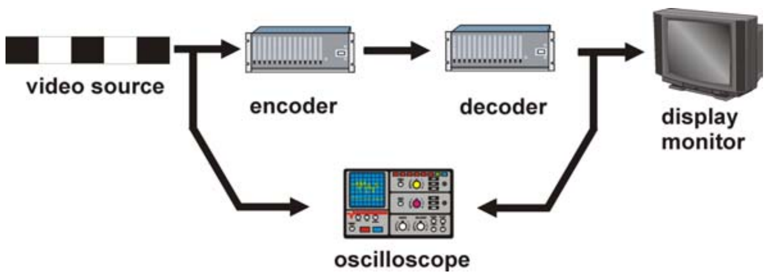

Fig.1: Test Setup for Delay Measurements

As a video source a PAL sequence was used where 12 black and 12 white frames alternated in a nonstop loop. The video signal was connected to an oscilloscope (Tektronix 2220) (TEK-86) and at the same time fed into the encoder video input slot. After the encoding process the compressed signal arrived at the decoder and was subsequently transmitted to a second input channel on the oscilloscope. For observation and control the signal was also displayed on a monitor. With the original signal on one input channel and the delayed signal on the other channel the oscilloscope showed the time spent on the encoding and decoding process. The jitter was derived using 100 samples of measured delay values and was calculated as the difference between maximum and minimum delay within the obtained value range (Table 1). The indicated values include both encoding and decoding times.

Codecs	Compression Format	Bandwidth	Delay	Mean Delay	Jitter
Cellstack Classic (ATM)	MJPEG	11.5 Mbit/s	90ms - 102ms	97.66ms	12ms
Litton (IP)	MPEG-2 (4:2:0) GOP=1 (I frames only)	7.2 Mbit/s	182ms -222ms	202.78ms	40ms

Table 1: Delay and Jitter Measurements

The Cellstack Classic codecs scored considerably better in both jitter and delay measurements compared to the Litton CAMVision CV2. According to the recommendation of the ITU (ITU-96a) the delay has a disturbing impact if the video transmission is bidirectional and 150 ms are exceeded. Therefore the Cellstack Classic codecs can be considered more suitable for bidirectional videoconferencing.

4.2 Picture Quality

The quality of a video sequence can be measured objectively or subjectively (WOL-97, FIB-97). So far subjective tests have been used predominantly, since in the end it depends on how a human spectator evaluates the perceived quality of a service (FEN-98). Subjective testing, however, has the disadvantage that a large number of people are required to evaluate the picture quality in a laboratory under conditions that remain the same for every test and can be reproduced at all times, if statistically relevant information is to be obtained. For this reason objective methods of evaluation have been developed which focus on high conformity with the subjective perception of a human spectator and use this degree of correlation to validate their tests (SCH-97).

The objective of this study is not to reach evaluations in a standardized laboratory that are statistically representative; instead the intention of the authors is to assess picture quality subjectively and provide some interesting results and insights into codec behavior using different compression formats over different types of networks. WAN behavior is simulated in tests with an impairment tool and by creating overload situations with synthetically generated traffic.

4.2.1 Evaluation of Picture Quality with Impairment Tool

A camera zoomed in onto a moving metronome was used as a video source (Fig. 2). The video sequences were rated according to the Mean Opinion Score (MOS) with its categories "excellent", "good", "fair", "poor" and "bad". (ITU-96). The picture qualities were varied by introducing errors with an Impairment Tool (Interwatch 95000) to simulate WAN behavior.

Fig. 2: Video Input Signal for Subjective Evaluation of Picture Quality

The video signal traveled from the encoder over two network components to the decoder and was then displayed on a control monitor for subjective evaluation (Fig. 3).

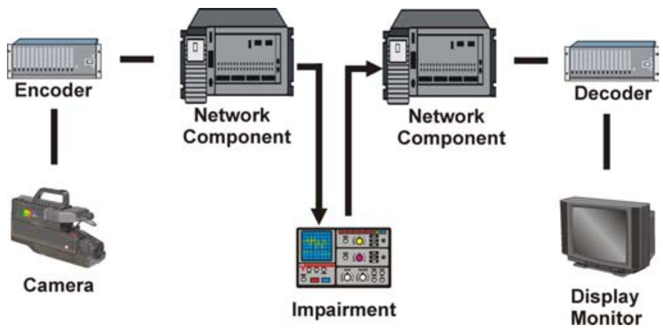

Fig. 3: Test Setup with Impairment Tool

4.2.1.1 Cellstack Classic Codecs

The Cellstack Classic codecs for ATM technology were connected to two FORE LE-155 ATM switches as network components. One ATM PVC (Permanent Virtual Circuit) was configured to carry the video stream. The PVC was classified as UBR (Unspecified Bit Rate) traffic, since both switches involved in the test did not carry any other traffic and the full bandwidth of 149.76 Mbit/s was available across the interfaces. In a first reference test no errors were introduced into the network connection and the picture quality was rated as excellent (Table 2).

Error Rates	Metronome, 60 beats per minute
none	**excellent**
10^{-8}	**excellent**
10^{-5}	**Good** (a few blocks)
10^{-4}	**Fair** (many block errors)
10^{-3}	**Bad** (picture freezes)

Table 2: Picture Quality of the Cellstack Classic Under the Influence of Error Rates

As errors were introduced with the impairment tool the quality of the picture started to deteriorate. The video sequence turned into a frozen picture with an error rate of 10^{-3}.

4.2.1.2 Litton Codecs

The Litton codecs were connected to two Cisco 7500 routers over Fast Ethernet interfaces. The bandwidth was set to 7.2 Mbit/s. The codecs were first evaluated without the introduction of errors with the impairment tool. As soon as strong motion was added to the video input, the picture froze and the codecs locked up. For this reason the picture quality was only rated as good at best (Table 3).

Fig.4: Severe Blocking in Areas with Strong Movements

Error Rates	Metronome, 60 beats per minute, GOP size = 1	Metronome, 120 beats per minute, GOP size = 15
none	**Good**	**Good**
10^{-8}	**Fair** (single errored blocks)	**Fair** (blocking, picture starts trembling in areas with a lot of motion)
10^{-7}	**Poor** (complete lines are errored, severe blocking in areas with strong movements (Fig. 4))	**Poor** (severe trembling in picture areas with strong movements)
10^{-6}	**Bad** (picture starts trembling and freezes)	**Bad** (picture is trembling, extreme blocking, picture freezes)
10^{-5}	**Bad** (picture freezes)	**Bad** (picture freezes)

Table 3: Picture Quality of the Litton Codecs Under the Influence of Error Rates

4.2.2 Evaluation of Picture Quality with Background Traffic

A camera focused on a moving metronome was used again as video input. The video signal traveled from the encoder over two network components to the decoder and was then displayed on a control monitor for subjective evaluation (Fig. 3). Background traffic was generated with a HP 4200B Analyzer and caused an overload situation at the outgoing interface of the network component nearest to the encoder (Fig.5).

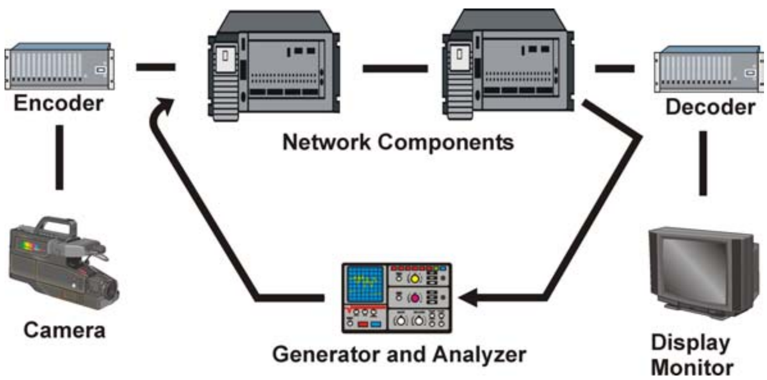

Fig. 5: Test Setup with Background Traffic

4.2.2.1 Cellstack Classic Codecs

Fore LE-155 ATM switches were also used as network components for testing the Cellstack Classics with background traffic. The FORE LE-155 STM-1 interfaces are capable of handling a total load of 149.76 Mbit/s. The coder was sending 6800 Protocol Data Units (PDU) per second.

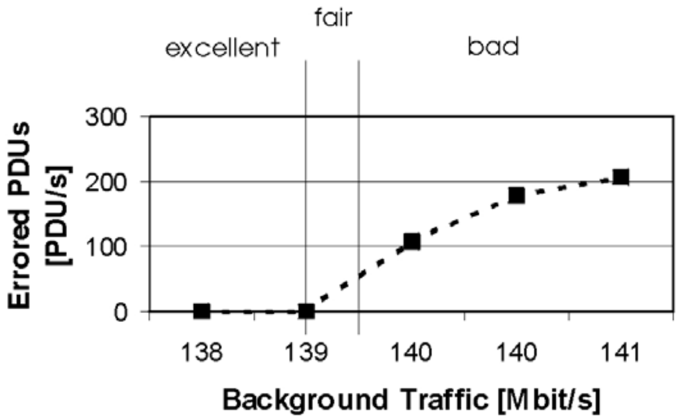

Fig. 6: Influence of Background Traffic on PDUs

Figure 6 shows the influence of background traffic on the PDUs of the video stream.

4.2.2.2 Litton Codecs

As in the impairment test Cisco 7500 routers were used as network components. The codecs were configured to use 7.2 Mbit/s for data packets. With the overhead of the protocol layer a total of 8.32 Mbit/s was required. The connection between the routers was implemented as an ATM connection with ATM Interface Processors (AIPs) with a maximum capacity of 138.56 Mbit/s. The GOP size was set to 1 (I frames only).

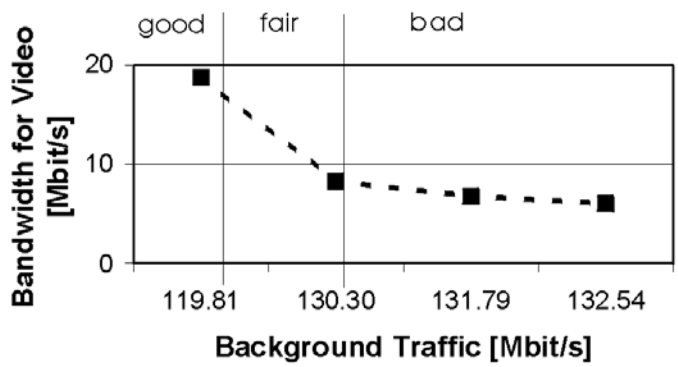

Fig. 7: Influence of Background Traffic on the Litton Codecs

As soon as the background traffic takes more bandwidth than 8.32 Mbit/s the video signal lacks bandwidth and the picture quality declines (Fig. 7). With only 6.02 Mbit/s bandwidth available the picture freezes.

It is obvious that for both codecs even a few losses already reduce the picture quality to the point where the spectator is no longer satisfied.

4.3 QoS Features

As was shown in the previous chapters small impairments of the transmission quality already have a severe impact on the picture quality and thus on the satisfaction of the user.

The Cellstack Classic Codecs compress the audio and video signals in two separate streams which can then be transported over ATM using two PVCs. Since ATM technology provides the possibility to configure each traffic stream as Continuous Bit Rate (CBR) traffic which reserves the requested bandwidth as a flat rate, the video signal can be shielded from any network overload situations.

The only mechanism to offer Quality of Service over IP implemented in the Litton codecs is the setting of the Type of Service (TOS) bit via the registry in the Windows NT platform (LIT-00). Unfortunately this feature could not be tested, since the available codecs were issued with an older software version where the TOS bit capability was not enabled.

To receive guaranteed QoS in networks video codecs must be able to reserve resources in the network, for example with RSVP. At the moment the tested Litton codec is not able to compensate for the lack of Quality of Service (QoS) features of the transport protocol in IP traffic.

5 Conclusion

The tests show, that QoS guarantees are required for the transport of video over networks. If parameters such as bandwidth, delay and jitter are not guaranteed or kept within certain bounds, packets are lost and the picture quality deteriorates, even if loss rates are small. To ensure high quality transmissions, the two-way delay, the delay variation and the loss rates should be kept to a minimum. Since the stability of the Litton codecs was limited further tests are recommended.

High quality applications require a networking infrastructure which is able to make Quality of Service guarantees. The Internet with its best effort IP service is not yet able to meet these demands and is therefore still not suitable for the transmission of high quality video data.

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