CESNET Technical Report 5/2002

Influence of Network QoS Characteristics on MPEG Video Transmission

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June 21, 2002

1 Experiment objective

One of the prospective applications expected to be used frequently in National Research and Educational Networks (NRENs) is high-quality video transmission. Applications of this type use mostly MPEG encoding for video and audio data (MPEG1, MPEG2 or MPEG4). Required bandwidth, ranging approximately from 2 Mb/s to 10 Mb/s, is relatively small when compared to the backbone capacity of current NRENs. However, these video transmissions are supposed to be available also in points farther away from the backbone. A typical example is broadcasting a conference session from a lecture hall connected using a wireless link with the capacity in the order of 10 Mb/s to 100 Mb/s having less than optimal network QoS characteristics. The objective of this experiment was to verify influence of primary QoS characteristics (loss rate, delay and jitter) on the quality of video and audio signal transmitted using MPEG encoding.

2 Test configuration

The test configuration is shown in Fig. 1. The sending PC was equipped with Optibase MPEG MovieMaker 200 encoding card and the receiving PC was equipped with Optibase Videoplex Xpress decoding card. Both cards support MPEG1 and MPEG2 encodings in SIF, QSIF, Full-D1 and Half-D1 formats. The Moviemaker 200 card can transmit MPEG1 data encoded online

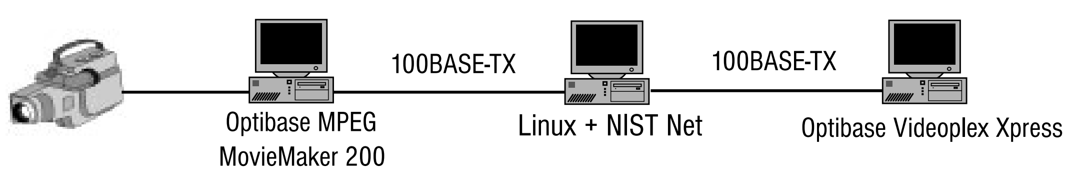


Figure 1: Test configuration

from the S-video input or it can read MPEG1 or MPEG2 data stored in a file. We used this particular hardware because it was just what we managed to get for testing purposes.

The sending and the receiving computer communicated over a router based on a Linux machine running NIST Net [1] emulation package. NIST Net allows to configure network QoS characteristics experienced by traffic passing the router. These characteristics include adjustable throughput, loss rate, duplications and delay (including standard deviation and linear correlation). The user can supply his own distribution function for delay emulation, if required.

3 Using NIST Net emulation package

Using the NIST Net emulation package involves three steps: loading the `nistnet` kernel module, turning the NIST Net on and configuring the QoS characteristic to be emulated.

To load the `nistnet` kernel module, run the `Load.Nistnet` script. In most cases you can also just type `insmod nistnet`.

To turn NIST Net on, use `cnistnet -u` (up) command.

To configure the QoS characteristics to be emulated, use `cnistnet` command with `-a` (add) option, specifying source and destination IP addresses of the flow (optionally including protocol and port numbers) and required QoS characteristics. For example:

```
cnistnet -a 195.113.147.1 195.113.147.2 --drop 0.1 --delay 100/10 --dup 0.1
```

This command will insert packet loss rate of 0.1%, mean delay of 100 ms with standard deviation of 10 ms and duplications of 0.1% into the flow of packets sent from IP address 195.113.147.1 to IP address 195.113.147.2.

4 Observations

We found the following observations to apply almost equally to different stream types (MPEG1 SIF, MPEG1 QSIF and MPEG2 Half D1) and rates ranging from 4 Mb/s to 10 Mb/s.

As expected, packet loss rate was a critical parameter. MPEG without FEC proved to be completely intolerable to packet losses as indicated in Table 1. With 10 Mb/s stream in 1500-byte packets, loss rate of 0.02% represents one lost packets in 6 seconds. These single packet losses were observable as image pixelization when looking carefully at the video sequence. We used a rather dynamic demonstrational clip from Optibase. It is likely that less dynamic sequences, such as broadcasting from a conference would suffer less from lost data. The effect of loss rate of 0.02% and 0.1% is illustrated in Fig. 2. The figure shows some of the worst pixelizations that have occurred, these faults were interspersed with period of acceptable quality.

On the other hand, the video transmission proved to be highly resilient to delay and jitter. We tried various combinations of mean delay and standard deviation resulting in observations summarized in Table 2. The system was able to adapt to the mean delay of up to 10 seconds which is well beyond conditions in real-world networks.

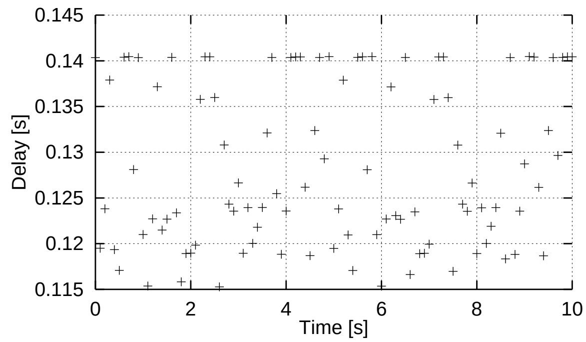
Loss rate	Effect
0.01%	No observable effect
0.02%	Little pixelization in fast sequences
0.1%	Lot of pixelization and interruptions
0.2%	Still image

Table 1: Influence of packet loss rate on image quality

Mean delay	Standard deviation	Effect
100 ms	5 ms	No observable effect
100 ms	7.5 ms	No observable effect
100 ms	10 ms	A short interruption of video every about 5 s
100 ms	20 ms	Frequent interruptions of video and audio

Table 2: Influence of delay and jitter on image quality

Figure 2: Effect of loss rate of 0.02% (left) and 0.1% (right)



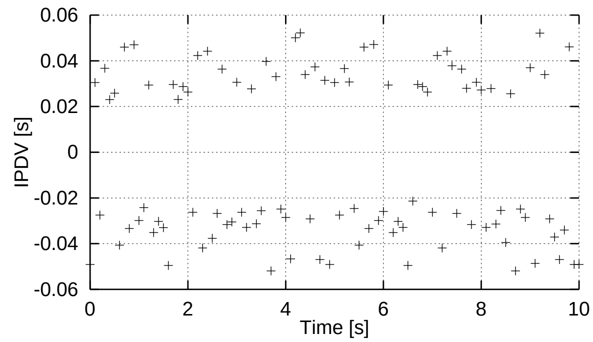


Figure 3: Measured maximum delay (left) and jitter (right) over intervals of 100 ms produced by Nist Net configured for mean delay of 100 ms and standard deviation of 10 ms

A particularly interesting point is the value of standard deviation which was still acceptable. For example, the mean delay of 100 ms with standard deviation of 10 ms means approximately 47% (!) of reordered packets. This was made possible probably because of layered nature of MPEG data which is designed to work with certain reordering of data packets [4]. Delay and jitter experienced by the video streams in this particular example is illustrated in Fig. 3. These diagrams were obtained using our system for precise measurement of network QoS characteristics [2, 3].

The video transmission was also tolerable to high packet duplications (tested up to 10%), which could be expected and is well beyond conditions in real-world networks.

In addition to emulation, we tried transmission over a real network between two buildings in the university campus. There were three routers interconnected with Fast and Gigabit Ethernet links along the path. The image was free of interruptions or visible pixelization, but with slight constant fluttering.

We did not have an opportunity to try transmission over a wirelless link, but loss rate measurement that we performed on one of our wirelless links suggest that it may be difficult to transmit MPEG video over a wirelless link. The packet loss rate measured on the Microcom wirelless 10 Mb/s link from Prague to Podebrady over a period of 5 days is indicated in Fig. 4. There was an almost constant loss rate of about 1.7% although the link was only lightly loaded. Such loss rate would probably render MPEG video transmission impossible. We currently do not know whether this is a standard operation condition of the wireless technology used or if there is some technical problem.

5 Conclusion

We found that MPEG video transmission without FEC is highly susceptible to packet losses. Maximum acceptable packet loss rate for flawless image was approximately 0.01%. Such packet loss rate is probably difficult to achieve on wireless links which means that MPEG video transmission without FEC is probably not possible over wireless links. Tests on real wireless links would have to be conducted to find out the actual behaviour of this class of application on

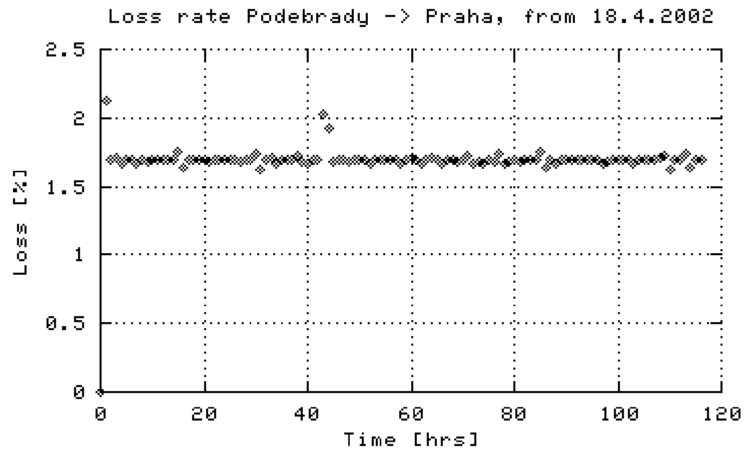


Figure 4: Measured packet loss rate on the wireless link from Prague to Podebrady

wirelless links.

We also suppose that the observable image quality can depend on the decoder implementation. If the decoder could render duplicate frames instead of damaged frames, it would be probably considered much less disturbing by users.

References

[1] "NIST Net", Internetworking Technology Group (ITG), National Institute of Standards and Technology (NIST), `http://snad.ncsl.nist.gov/itg/nistnet`.

[2] Sven Ubik, Vladimir Smotlacha (Cesnet), Sampo Saaristo (Tampere University of Technology), Juha Laine (Soon Communications). "Low-Cost Precise QoS Measurement Tool", Cesnet Technical Report 7/2001, `http://www.cesnet.cz/doc/techzpravy/2001/07`.

[3] Sven Ubik. "Přesné a jednoduché měření kvalitativních parametrů sítě" (in Czech), Sdělovací technika 2002/5.

[4] Komura Takaaki, Fujikawa Kenji, Ikeda Katsuo. "Layered Transmission and Control of Packet Transmission Order for Multimedia Broadcasting", Proceedings of INET 2000, Tokyo, 18-21.7.2000.