Evaluation of Network Failure induced IPTV degradation in Metro Networks

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Abstract: In this paper, we evaluate future network services and classify them according to their network requirements. IPTV is used as candidate service to evaluate the performance of Carrier Ethernet OAM update mechanisms and requirements. The latter is done through quality measurements using MDI and subjective evaluations. It is concluded that OAM interval close to 10 ms is a suitable choice for service providers.

Key-Words: Carrier Ethernet, IPTV, MDI, PBT, MPLS-TE

1. Introduction

The next generation broadband services are expected to be the main driver for the future network with capacities well above 1 Gbps to the residential users. The emerging network possibilities merged with the requirements for multimedia contents give rise to a wide range of multimedia related applications like IPTV, Ultra HD IPTV, stereoscopic TV etc. In contrast to some of the driver services today, the multimedia services often require low delay to satisfy the requirements for interactivity.

This places severe requirements to the access and the metropolitan area networks, where the user traffic is aggregated. In the Metro Networks the consequence of network failures is severe due to the large number of affected users. Therefore, in relation to the Carrier Ethernet transport technologies, the restoration time for cable breaks etc. has to be analysed in order to determine the necessary buffer size at the receiver. On the other hand, large buffers increase the delay, which is a problem for delay sensitive applications like videoconferencing.

Hence, in this paper, we first in section 2 analyse the requirements to the network from future services and applications. In section 3 we discuss the Carrier Ethernet in Metro Networks. Then in section 4, we define the measures for quality quantification of IPTV and in section 5, we show the experimental setup and results in section 6. Finally in sections 7 we discuss the results and conclude the work.

2. Future driver services

In the following several representative end user services are described. The bit-rate of the services varies from few kilobits per second (some simpler telemetric services) to 24 Gigabit per second (for immersive Ultra High Definition TV). Furthermore, the tolerable delay varies from tens of seconds (for e-mail) to a few tens of millisecond, the jitter from seconds to less than a millisecond, the packet loss from a few percent (on-line gaming) to less than a tenth of percent (thin client). The applications also have varying requirements on the traffic priority and security and may or may not allow mobility of the end-user. The services today includes video streaming and medium quality video conferencing, while some others, like remote home monitoring, location based services or Ultra High Definition Video are emerging services of the near or middle-term future. Finally, services like Web 3D and robotic assistant are more futuristic. In addition we see services like TV evolve to High Definition TV and further into Ultra High Definition TV. Drawbacks of this development include power consumption in the home and very complex control and management of traffic flows.

It is useful to classify the services according to their network requirements in terms of bandwidth requirements, packet loss, delay and jitter. This is done in Figure 1 for a selected number of the addressed services. The mapping of the services follows three main criteria:

- Delay: Here two classes are defined. Delay requirements below 150 ms and above 150 ms for one way delays. This corresponds to the definition from ITU on elastic and non-elastic services.
- Error-tolerance: This parameter is a combination of the acceptable packet loss and other parameters that
transforms into packet loss and thus creates visible faults or glitches. Any application metric that produces these faults should be considered inside the “error-tolerance” parameter. Initially, the threshold error rate is set to above or below 0.1% error induced packet loss.

- Bandwidth: The applications and services are ordered from low bandwidth to high bandwidth. These numbers range from a few bps up to 24 Gbps for uncompressed Ultra HDTV distribution.

Some of the considered services are illustrated in Figure 1. It is clearly observed that the vast number of available and future applications have quite different requirements to the underlying network architecture.

![Figure 1: Classification of selected services today and tomorrow.](image)

Generally, the video services are among those with heavy requirements on bandwidth depending on display size and quality level. Normal internet video services are hard to categorise, thus it is assumed that they use medium bandwidth level. Unless interactivity is required these services have relaxed delay requirements. The future 3D services are expected to be highly bandwidth consuming. The greatest network requirements stem from the future services in the “entertainment” and the virtual environment area. Here, a number of high bandwidth demanding applications are matched with strict requirements on delay, making these very challenging to successfully implement. These service requirements introduce two main challenges. First, the raw capacity should be available in the network segment, and secondly, in order for a heart monitoring service to operate on the same network as an IPTV, there should be simple ways to prioritise traffic and thus make sure the large data amounts from the IPTV do not exhaust the monitoring flow.

3. **Metro Networks**

The Metro Network is located between the access network, which might be based on GPON, EPON, xDSL or similar, and the core network. It is interconnecting the service providers access points and usually contains IPTV servers, if such services are offered. It is thus relevant to consider this network part and its resiliency mechanisms.

One recent emerged technology in this network segment is Carrier Ethernet, which is basically covering technologies like MPLS-TP [1] and PBB-TE [2]. The motivation for this is to make packet network behave more like transport network and ease the migration from the existing networks. Basically, this means that the network follows the principle outlined in the ITU-T recommendation G.805, and the operation is focused on transport rather than services. The advantages are among others higher scalability, lower cost per bit, higher availability and better management.

One important aspect of this suite of technologies is their OAM information, i.e., the information used for constantly monitor the transmission, and in case of degradation or loss of signal initiate protection switching.

For the Provider Backbone Transport (PBT), the OAM information is specified through the ITU-T recommendation Y.1731. This includes “heartbeat” continuity check signals, which operates with 7 possible rates, i.e., different OAM update intervals to indicate whether the link is operating. If a receiver does not receive three consecutive OAM messages it determines that the link has failed and it will attempt protection switching depending on the setup. The failure types comprise hard failures like cables cuts and node outages or soft failures like errors in the routing protocols or corrupted OAM messages. If multicast is implemented other error types are problems with the multicast tree.

The choice of the OAM update interval is a trade-off between how fast the systems reacts to failures and the overhead imposed by the messages. The experimental studies reported in this paper evaluates the required OAM update time acceptable for an IPTV user. We have chosen IPTV as the main driver for Carrier Ethernet solutions, because most of the considered services in section 2 use IPTV in different variants. In the experimental evaluation we only focus on the one-way properties of the restoration process, however, the two-way interactions can easily be derived from this.
4. Measuring video quality

In order to evaluate the quality of a video stream we need to define some definitions for acceptable video quality. This can be grouped into two major categories as shown in Figure 2. The subjective evaluation is based on human perceived quality and mathematical or network evaluations based on measurable parameters.

Among the subjective evaluations, the Mean Opinion Score (MOS) is the most well known. Here a panel of participants are formed and they give a subjective quality score for the shown video and/or audio. MOS is inherited from the audio world and an expert panel scores the perceived stream from 1 to 5 which is from bad to excellent, respectively. This has mainly been used to compare the quality from different codecs.

More specifically the Perceptual Evaluation of Video Quality (PEVQ) uses a MOS evaluation for video perception. In addition to the overall MOS, other measurable parameters like PSNR are taken into account.

The second group includes the measurable parameters. First the MDI provides clear values of the delay and loss of a media stream.

In addition we have the Peak Signal to Noise Ratio (PSNR). Here the “signal” is the original uncompressed signal and the noise is the differences in the shown picture caused by the codec in the encoding phase.

The Czenakowski Distance (CZD) is close to the PSNR and operates per pixel. Furthermore the Structural Similarity (SSIM) measures the differences between two pictures or two frames.

However, while these are very relevant tools for measuring compression quality they do not necessarily give a clear picture on the artifacts caused by the network layer. Only the MDI gives this information and even here the perceived quality can be very different depending on where or when the failure occurs and which type of video is shown.

Therefore we focus on a subjective evaluation backed by MDI measures.

Thus, the Media Delivery Index (MDI) is used for monitoring the video stream and it is probably the most useful measure on the network layer and below. The results from the MDI can be used to indicate for a service provider the required buffer length in the network layer. Off course this is also a trade off with the interactivity and the time it takes to provide control functions to the setup.

Basically, MDI comprises measures of the delay, jitter and the loss, which are called the Delay Factor and the Media Loss Rate. Hence an MDI value is described as:

$$MDI = DF : MLR$$

The Delay Factor (DF) can be used for the service provider to indicate how much buffer he or she should include in the solution, however he should usually stay within acceptable values of less than 50 ms. Otherwise the user will find zapping and other controls annoying.

The Media Loss Rate (MLR) measures lost packets during a given time period, e.g., 1 sec. For HDTV the MLR should be no more than 0.05 percent.

5. Experimental setup

The objectives of the experimental studies are to evaluate the effect of the Carrier Ethernet signalling and monitoring. In our setup we use three Carrier Ethernet enable nodes and observe and measure the quality degradation when a failure is induced. We use the MDI value in combination with a subjective evaluation of a video stream.

The experimental setup is shown in Figure 3.
A PBT channel with a working and a protecting path is defined and implemented in each of the CE switches. The blue solid path is our working path the dashed blue is the protecting path. The orange path is a reference path so that we can have two displays operating.

The Carrier Ethernet switches is three logical switches based on a TPACK Longmorn switch, which has the very powerful feature that it can instantiate several virtual switches as long as there are enough available ports. This makes it possible for us to use one physical switch to provide three logical switches.

The network is established in a few basic steps. First, PBT tunnels are defined and for each node it is indicated whether this is a termination or a tunnelling node. Then working and protection paths are established for each flow. Finally, the virtual switches are instantiated and the device operates completely as three individual Carrier Ethernet PBT switches.

To evaluate the different OAM interval times this is adjusted as shown in Figure 4, as derived from the Carrier Ethernet Switch interface. We specifically evaluate the values 3 ms, 10 ms, 100 ms and 1 s.

To measure the MDI values we use an Agilent N2X Solution. It is a very powerful measurement equipment that is extremely flexible due to its FPGA platform and it can be used to emulate the behaviour and flow characteristics of several thousand of triple play users.

6. Experimental results

The experimental results are based on the subjective evaluation a video streaming over the network and the corresponding MDI values. Unfortunately, the video clips cannot be shown here on paper, however, the following results can be summarised as in Table 1

<table>
<thead>
<tr>
<th>Update Interval</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 sec</td>
<td>Long breaks and it seems that the stream loses synchronisation. Definitely not acceptable.</td>
</tr>
<tr>
<td>100 msec</td>
<td>Often this induces breaks in the stream, and severe artifacts are present. The result depends on the failure time with respect to the MPEG stream. The quality is not acceptable</td>
</tr>
<tr>
<td>10 msec</td>
<td>Minor artifacts observed. Rarely loss of sync or frames. Acceptable</td>
</tr>
<tr>
<td>3 msec</td>
<td>Very few artifacts. Definitely acceptable</td>
</tr>
</tbody>
</table>

It is seen from the subjective evaluation that the OAM update intervals of 10 msec or less is acceptable for a representative user.

The MDI values for the four different scenarios are shown in Figure 5, Figure 6, Figure 7 and Figure 8.

![Figure 5: MDI for 1 sec update interval](image5.png)

![Figure 6: MDI for 100 msec update interval](image6.png)
It is noted that the delay axis for the three last figures is altered compared to the first. In each graph two peaks are shown, the first peak is the situation when the failure occurs and the restoration process is initiated. Then the second failure is because the original working path is again available and the system switches back to this. As previously described, restoration is initiated if three consecutive OAM frames are lost. This is in accordance with the results. It is noted that the results are obtained in intervals of 1 sec. The results and the subjective evaluation indicates that a service provider, who plans to use a Carrier Ethernet for providing IPTV services should use an OAM update interval of no more than 10 ms to ensure his customers an acceptable quality even in case of failures. Should he also provide interactive services, it might even be relevant to lower the interval to 3 ms, as this has a dramatic impact on the MDI quality measure.

7. Conclusion

In this paper we have discussed the future services in the internet with special focus on the Metro Network. We evaluated the expected future services, which we classified according to their network requirements. Then we chose IPTV as a representative candidate, and we showed how a Carrier Ethernet can be operated to support such application. Finally, we provided a subjective evaluation backed by MDI measurements to define the most optimal OAM update frequency for such application in a specific PBT enable Carrier Ethernet setup. It was indicated that an OAM update interval no longer than 10 ms provides acceptable user perceived quality.

Acknowledgement

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References:

[1] IEEE 802.1Qay