Application Note 24

Media Delivery Index (MDI)

Introduction

The transmission of real-time media content over IP-based networks is not always possible without problems. If audio / video streams are transported over packet switched networks these can be influenced in the following ways:

1. network jitter
2. packet loss (either single packet or burst loss)
3. packets arriving in wrong order

These parameters are the main indicators for the quality of streamed audio / video content. A first step is the monitoring of these values and an evaluation of the measurement results. The next step is the troubleshooting in the network. The first requirement can be accomplished by the Media Delivery Index (MDI). It can be used to monitor and alarm in case of degradation of the network.

Normally all receivers (decoders) are capable of handling a certain amount of network jitter. This is usually done with the help of specified input buffers on the decoding side. These buffers prefetch data in order to compensate time delays or data bursts of the incoming stream(s). One could claim a sufficient buffer in every decoder in order to minimize this problem. However this would increase the end-to-end latency of the transmission which is not acceptable in all cases. If the latency is not a main issue, the decoder should always be set to a major de-jitter buffer size. In case of “latency-sensitive” applications there is always a trade-off between acceptable delay and picture / audio quality. A small decoder buffer is not always capable to smooth network delays and / or packet loss.

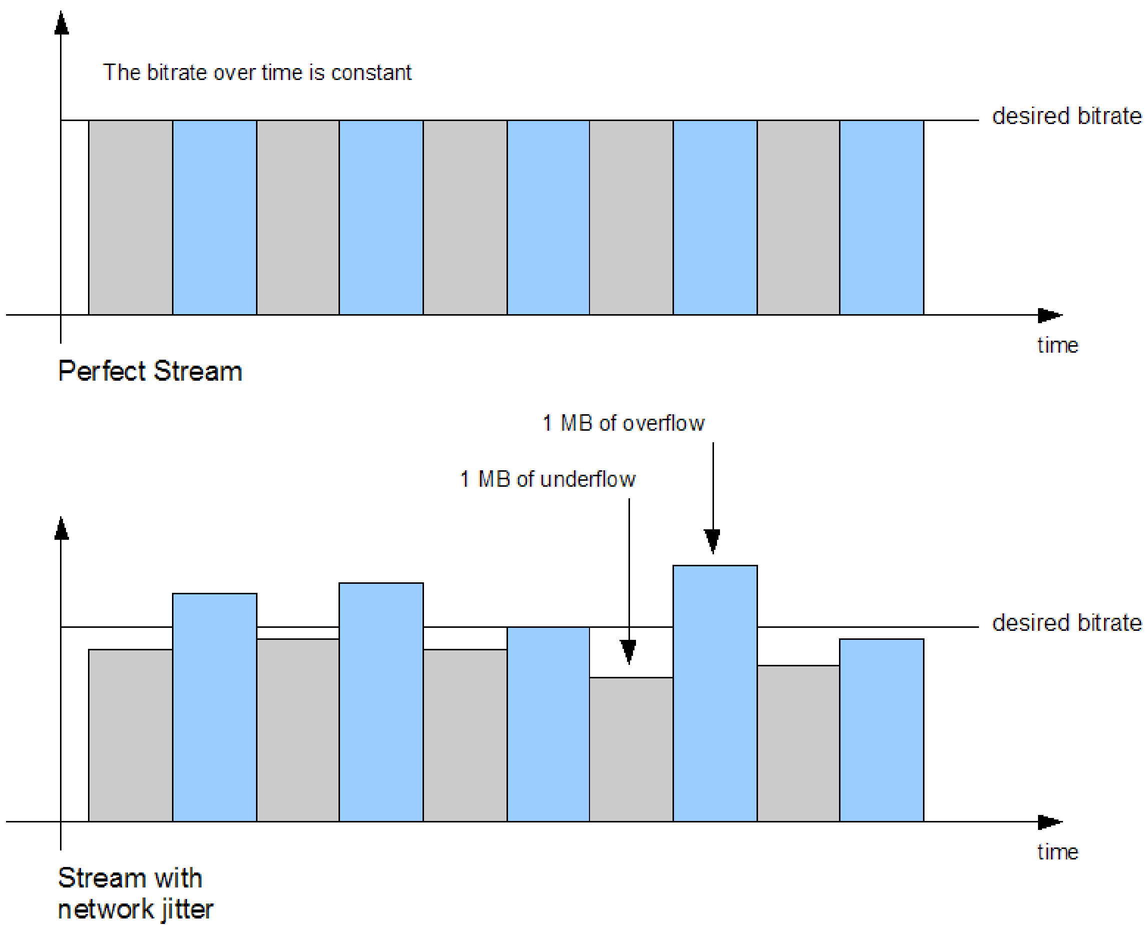

Figure 1: Influence of network jitter

Figure 1 illustrates the influence of network jitter on audio / video streams. The upper graphic

shows a perfect stream. The bit rate over time is constant, i.e. the decoder is fed with a steady-going input bit rate. There are no time variations or data bursts.

The lower graphic demonstrates the impact of network jitter on the data rate. It can easily be seen that the bit rate is not constant – to put it bluntly it is fluctuating to a maximum deviation of 1 MB. To ensure acceptable audio / video quality an input buffer of 1 MB is required.

How does MDI fit into this? MDI can be used to evaluate network performance that might have changed due to altered traffic loads. Therefore MDI has two components that measure known, critical network parameters:

1. Delay factor (DF)
2. Media loss ratio (MLR)

Generally it is down-to-earth to detect lost packets in the network infrastructure. This can be done via monitoring of the different internal links. Even if there is no packet loss the infrastructure might affect the round trip delay of the stream which could be caused by congestion of packets on routers / switches. This may result in over- or underflow of the decoder buffer.

The delay factor is calculated at a specified time interval (normally 1 second). The largest span and the smallest span between received bytes and drained bytes within the time interval is measured. This virtual buffer size is used for the calculation of the delay factor. The set phrase for the calculation of the DF is as follows:

$$X = \left| bytes\,received - bytes\,drained \right| \quad \left(virtual\,buffer\,size \right)$$

$$DF = \frac{\left[\max(X) - \min(X) \right]}{media\ rate}$$

The DF is the equivalent to the time required to empty the virtual buffer at the used audio and / or video bit rate. It may also be regarded as the relationship between the network latency and the latency created by delivery from source to destination.

If the DF value steadily grows this is an indication for unassimilated buffer size as the drain rate is different from the fill rate. Usually this is caused by a problem of the audio / video source as network upsets are typically transient in nature.

To avoid network problems from the start it is inevitable to characterize the network in advance. Therefore it might be useful to check all discrete parts of the network in order to find out how much headroom is required to minimize or rather avoid packet loss. Completely checking the network infrastructure helps to avoid bottle necks in the network.

The MLR is the amount of lost media packets per second. It is computed by subtracting the number of expected packets from the number of media packets expected during one time interval (usually 1 s):

$$MLR = \frac{(packets\ expected - packets\ received)}{time\ interval\ (in\ s)}$$

1. *many intervals with a positive but similar magnitude:*

   This is mostly traced back to the fact of buffer overflows which are the direct consequence of network overload or misconfiguration of particular network devices.

2. *transient effects:*

   Transient effects may be caused by electrical noise interference. This is normally only of temporary nature. A constant MLR with varying but low traffic loads indicates server or other media issues at the source.

Realization

The IO [io] 8001 offers the monitoring of several networking parameters. The front panel offers this option via the “status” menu item in the root folder of the front panel menu.

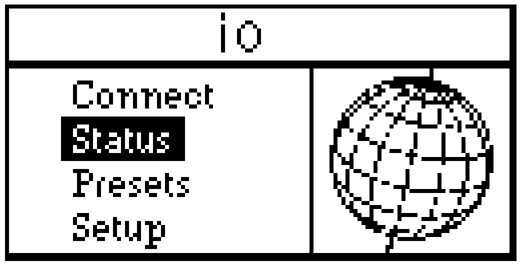

Figure 2: Menu item „status“

As soon as a connection between the encoder and the decoder is established it is possible to retrieve the following values from the encoder / decoder:

1. UDP (User Datagram Protocol):

   IPA (incoming packets): restricted to input

   The amount of incoming UDP - packets is counted no matter if they are destined for this interface or not.

   OPA (outgoing packets): restricted to output

   The amount of outgoing UDP - packets is counted. This value is not restricted to the audio and video stream(s).

   IBR (incoming bit rate): restricted to input

   not yet supported

OBR (outgoing bit rate): restricted to output

The *calculated* bit rate of all encoders (audio and video) including the header sizes of the encapsulated payload (IEEE 802.3 + IP + UDP).

IER (IN errors): restricted to input

Amount of received UDP – packets that could not be forwarded to the decoder.

NoP (no port): restricted to input

The amount of UDP – packets that arrived on ports that were not listening. Thus the packets have been discarded.

2. RTP (Real-Time Transport Protocol):

   IBR (input bit rate): restricted to decoder

   The measured input bit rate on the specified decoder including headers (IEEE 802.3 + IP + UDP + RTP).

   OBR (outgoing bit rate): restricted to encoder

   not yet supported

   LIS (Listener): restricted to encoder

   The amount of active listeners to the stream created by the encoder. The registration of a listener is only possible for bi-directional connections.

   RCJ (receive jitter): restricted to decoder

   The receive jitter is calculated according to RFC 3550 (comparison of RTCP timestamps on the decoder).

LOP (Lost packets): restricted to decoder

The amount of lost packets. Lost packets always lead to loss of audio / video. The first value represents the complete amount of sequence errors since start of the connection, the second is restricted to the last minute.

SEQ (Sequence errors): restricted to decoder

The amount of sequence errors (swapped packets). Up to a certain amount of packets the correct reconstruction of the sequence is possible. If this is not the case a sequence error is counted as a LOP. The first value represents the complete amount of sequence errors since start of the connection, the second is restricted to the last minute.

MDI (Media delivery index): restricted to decoder

The MDI comprises of 2 values. The presentation of the MDI is as [DF : MLR].

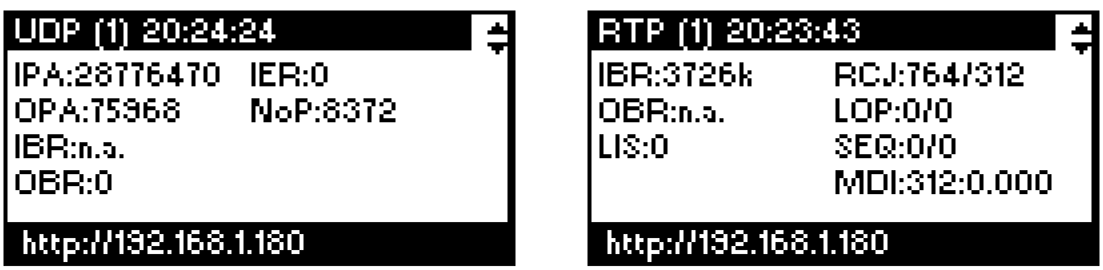

Figure 3: UDP and RTP networking statistics

Besides these networking parameters the status menu offers additional information on the stream.

- Duration of the connection (e.g. 20:24:24 for UDP)
- Connection type (http://)
- Source address of the encoder (192.168.1.180) or destination address of encoder
- Number of sub codec running (e.g. 1)