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# A Proposed Media Delivery Index (MDI)

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## Abstract

This memo defines a Media Delivery Index (MDI) measurement that can be used as a diagnostic tool or a quality indicator for monitoring a network intended to deliver applications such as streaming media, MPEG video, Voice over IP, or other information sensitive to arrival time and packet loss. It provides an indication of traffic jitter, a measure of deviation from nominal flow rates, and a data loss at-a-glance measure for a particular flow. For instance, the MDI may be used as a reference in characterizing and comparing networks carrying UDP streaming media.

The MDI measurement defined in this memo is intended for Information only.

## 1. Introduction

There has been considerable progress over the last several years in the development of methods to provide for Quality of Service (QoS) over packet-switched networks to improve the delivery of streaming media and other time-sensitive and packet-loss-sensitive applications such as [i1], [i5], [i6], [i7]. QoS is required for many practical networks involving applications such as video transport to assure the availability of network bandwidth by providing upper limits on the number of flows admitted to a network, as well as to bound the packet jitter introduced by the network. These bounds are required to dimension a receiver's buffer to display the video properly in real time without buffer overflow or underflow.

Now that large-scale implementations of such networks based on RSVP and Diffserv are undergoing trials [i3] and being specified by major service providers for the transport of streaming media such as MPEG video [i4], there is a need to diagnose issues easily and to monitor the real-time effectiveness of networks employing these QoS methods or to assess whether they are required. Furthermore, due to the significant installed base of legacy networks without QoS methods, a delivery system's transitional solution may be composed of networks with and without these methods, thus increasing the difficulty in characterizing the dynamic behavior of these networks.

The purpose of this memo is to describe a set of measurements that can be used to derive a Media Delivery Index (MDI) that indicates the instantaneous and longer-term behavior of networks carrying streaming media such as MPEG video.

While this memo addresses monitoring MPEG Transport Stream (TS) packets [i8] over UDP, the general approach is expected to be applicable to other streaming media and protocols. The approach is applicable to both constant and variable bit rate streams though the variable bit rate case may be somewhat more difficult to calculate. This document focuses on the constant bit rate case as the example to describe the measurement, but as long as the dynamic bit rate of the encoded stream can be determined (the "drain rate" as described below in Section 3), then the MDI provides the measurement of network-induced cumulative jitter. Suggestions and direction for calculation of MDI for a variable bit rate encoded stream may be the subject of a future document.

Network packet delivery time variation and various statistics to characterize the same are described in a generic approach in [i10]. The approach is capable of being parameterized for various purposes with the intent of defining a flexible, customizable definition that can be applied to a wide range of applications and further

experimentation. Other approaches to characterizing jitter behavior are also captured such as in [i12]. A wide-ranging report format [i11] has been described to convey information including jitter for use with the RTP Control Protocol (RTCP) [i12]. The MDI is instead intended to specifically address the need for a scalable, economical-to-compute metric that characterizes network impairments that may be imposed on streaming media, independent of control plane or measurement transport protocol or stream encapsulation protocol. It is a targeted metric for use in production networks carrying large numbers of streams for the purpose of monitoring the network quality of the flows or for other applications intended to analyze large numbers of streams susceptible to IP network device impairments. An example application is the burgeoning deployments of Internet Protocol Television (IPTV) by cable and telecommunication service providers. As described below, MDI provides for a readily scalable per-stream measure focused on loss and the cumulative effects of jitter.

## 2. Media Delivery Index Overview

The MDI provides a relative indicator of needed buffer depths at the consumer node due to packet jitter as well as an indication of lost packets. By probing a streaming media service network at various nodes and under varying load conditions, it is possible to quickly identify devices or locales that introduce significant jitter or packet loss to the packet stream. By monitoring a network continuously, deviations from nominal jitter or loss behavior can be used to indicate an impending or ongoing fault condition such as excessive load. It is believed that the MDI provides the necessary information to detect all network-induced impairments for streaming video or voice-over-IP applications. Other parameters may be required to troubleshoot and correct the impairments.

The MDI is updated at the termination of selected time intervals spanning multiple packets that contain the streaming media (such as transport stream packets in the MPEG-2 case). The Maximums and Minimums of the MDI component values are captured over a measurement time. The measurement time may range from just long enough to capture an anticipated network anomaly during a troubleshooting exercise to indefinitely long for a long-term monitoring or logging application. The Maximums and Minimums may be obtained by sampling the measurement with adequate frequency.

## 3. Media Delivery Index Components

The MDI consists of two components: the Delay Factor (DF) and the Media Loss Rate (MLR).

### 3.1. Delay Factor

The Delay Factor is the maximum difference, observed at the end of each media stream packet, between the arrival of media data and the drain of media data. This assumes the drain rate is the nominal constant traffic rate for constant bit rate streams or the piece-wise computed traffic rate of variable rate media stream packet data. The "drain rate" here refers to the payload media rate; e.g., for a typical 3.75 Mb/s MPEG video Transport Stream (TS), the drain rate is 3.75 Mb/s -- the rate at which the payload is consumed (displayed) at a decoding node. If, at the sample time, the number of bytes received equals the number transmitted, the instantaneous flow rate balance will be zero; however, the minimum DF will be a line packet's worth of media data, as that is the minimum amount of data that must be buffered.

The DF is the maximum observed value of the flow rate imbalance over a calculation interval. This buffered media data in bytes is expressed in terms of how long, in milliseconds, it would take to drain (or fill) this data at the nominal traffic rate to obtain the DF. Display of DF with a resolution of tenths of milliseconds is recommended to provide adequate indication of stream variations for monitoring and diagnostic applications for typical stream rates from 1 to 40 Mb/s. The DF value must be updated and displayed at the end of a selected time interval. The selected time interval is chosen to be long enough to sample a number of TS packets and will, therefore, vary based on the nominal traffic rate. For typical stream rates of 1 Mb/s and up, an interval of 1 second provides a long enough sample time and should be included for all implementations. The Delay Factor indicates how long a data stream must be buffered (i.e., delayed) at its nominal bit rate to prevent packet loss. This time may also be seen as a measure of the network latency that must be induced from buffering, which is required to accommodate stream jitter and prevent loss. The DF's max and min over the measurement period (multiple intervals) may also be displayed to show the worst case arrival time deviation, or jitter, relative to the nominal traffic rate in a measurement period. It provides a dynamic flow rate balance indication with its max and min showing the worst excursions from balance.

The Delay Factor gives a hint of the minimum size of the buffer required at the next downstream node. As a stream progresses, the variation of the Delay Factor indicates packet bunching or packet

gaps (jitter). Greater DF values also indicate that more network latency is necessary to deliver a stream due to the need to pre-fill a receive buffer before beginning the drain to guarantee no underflow. The DF comprises a fixed part based on packet size and a variable part based on the buffer utilization of the various network component switch elements that comprise the switched network infrastructure [i2].

To further detail the calculation of DF, consider a virtual buffer VB used to buffer received packets of a stream. When a packet P(i) arrives during a calculation interval, compute two VB values, VB(i,pre) and VB(i,post), defined as:

VB(i,pre) = sum (Sj) - MR * Ti; where j=1..i-1
VB(i,post) = VB(i,pre) + Si

where Sj is the media payload size of the jth packet, Ti is the relative time at which packet i arrives in the interval, and MR is the nominal media rate.

VB(i,pre) is the Virtual Buffer size just before the arrival of P(i). VB(i,post) is the Virtual Buffer size just after the arrival of P(i).

The initial condition of VB(0) = 0 is used at the beginning of each measurement interval. A measurement interval is defined from just after the time of arrival of the last packet during a nominal period (typically 1 second) as mentioned above to the time just after the arrival of the last packet of the next nominal period.

During a measurement interval, if k packets are received, then there are 2*k+1 VB values used in deriving VB(max) and VB(min). After determining VB(max) and VB(min) from the 2k+1 VB samples, DF for the measurement interval is computed and displayed as:

DF = [VB(max) - VB(min)]/ MR

As noted above, a measurement interval of 1 second is typically used. If no packets are received during an interval, the last DF calculated during an interval in which packets did arrive is displayed. The time of arrival of the last previous packet is always retained for use in calculating VB when the next packet arrives (even if the time of the last received packet spans measurement intervals). For the first received measurement interval of a measurement period, no DF is calculated; however, packet arrival times are recorded for use in calculating VB during the following interval.

### 3.2. Media Loss Rate

The Media Loss Rate is the count of lost or out-of-order flow packets over a selected time interval, where the flow packets are packets carrying streaming application information. There may be zero or more streaming packets in a single IP packet. For example, it is common to carry seven 188 Byte MPEG Transport Stream packets in an IP packet. In such a case, a single IP packet loss would result in 7 lost packets counted (if those 7 lost packets did not include null packets). Including out-of-order packets is important, as many stream consumer-type devices do not attempt to reorder packets that are received out of order.

### 3.3. Media Delivery Index

Combining the Delay Factor and Media Loss Rate quantities for presentation results in the following MDI:

DF:MLR

Where:

DF is the Delay Factor
MLR is the Media Loss Rate

At a receiving node, knowing its nominal drain bit rate, the DF's max indicates the size of buffer required to accommodate packet jitter. Or, in terms of Leaky Bucket [i9] parameters, DF indicates bucket size b, expressed in time to transmit bucket traffic b, at the given nominal traffic rate, r.

### 3.4. MDI Application Examples

If a known, well-characterized receive node is separated from the data source by unknown or less well-characterized nodes such as intermediate switch nodes, the MDI measured at intermediate data links provides a relative indication of the behavior of upstream traffic flows. DF difference indications between one node and another in a data stream for a given constant interval of calculation can indicate local areas of traffic congestion or possibly misconfigured QoS flow specification(s) leading to greater filling of measurement point local device buffers, resultant flow rate deviations, and possible data loss.

For a given MDI, if DF is high and/or the DF Max-Min captured over a significant measurement period of multiple intervals is high, jitter has been detected but the longer-term, average flow rate may be nominal. This could be the result of a transient flow upset due to a coincident traffic stream unrelated to the flow of interest causing packet bunching. A high DF may cause downstream buffer overflow or underflow or unacceptable latency even in the absence of lost data.

Due to transient network failures or DF excursions, packets may be lost within the network. The MLR component of the MDI shows this condition.

Through automated or manual flow detection and identification and subsequent MDI calculations for real-time statistics on a flow, the DF can indicate the dynamic deterioration or increasing burstiness of a flow, which can be used to anticipate a developing network operation problem such as transient oversubscription. Such statistics can be obtained for flows within network switches using available switch cpu resources due to the minimal computational requirements needed for small numbers of flows. Statistics for all flows present on, say, a gigabit Ethernet network, will likely require dedicated hardware facilities, though these can be modest, as buffer requirements and the required calculations per flow are minimal. By equipping network switches with MDI measurements, flow impairment issues can quickly be identified, localized, and corrected. Until switches are so equipped with appropriate hardware resources, dedicated hardware tools can provide supplemental switch statistics by gaining access to switch flows via mirror ports, link taps, or the like as a transition strategy.

The MDI figure can also be used to characterize a flow decoder's acceptable performance. For example, an MPEG decoder could be characterized as tolerating a flow with a given maximum DF and MLR for acceptable display performance (acceptable on-screen artifacts). Network conditions such as Interior Gateway Protocol (IGP) reconvergence time then might also be included in the flow tolerance DF resulting in a higher-quality user experience.

## 4. Summary

The MDI combines the Delay Factor, which indicates potential for impending data loss, and Media Loss Rate as the indicator of lost data. By monitoring the DF and MLR and their min and max excursions over a measurement period and at multiple strategic locations in a network, traffic congestion or device impairments may be detected and isolated for a network carrying streaming media content.

## 5. Security Considerations

The measurements identified in this document do not directly affect the security of a network or user. Actions taken in response to these measurements that may affect the available bandwidth of the network or the availability of a service is out of scope for this document.

Performing the measurements described in this document only requires examination of payload header information (such as MPEG transport stream headers or RTP headers) to determine nominal stream bit rate and sequence number information. Content may be encrypted without affecting these measurements. Therefore, content privacy is not expected to be a concern.

## 6. Informative References

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