                    A Proposed Media Delivery Index (MDI)

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   information.

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.

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