Taxonomy of Issues in Internet Media
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Abstract

   This document provides a taxonomy of networking issues that pertain
   to quality of experience in delivery of video or other high-bitrate
   media over the internet.

Status of This Memo

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1.  Introduction

   As the internet has grown, an increasingly large share of the traffic
   delivered to end users has become video.  Estimates put the total
   share of internet video traffic at 75% in 2019, expected to grow to
   82% by 2022.  What's more, this estimate projects the gross volume of
   video traffic will more than double during this time, based on a
   compound annual growth rate continuing at 34% (from Appendix D of
   [CVNI]).

   In many contexts, video traffic can be handled transparently as
   generic application-level traffic.  However, as the volume of video
   traffic continues to grow, it's becoming increasingly important to
   consider the effects of network design decisions on application-level
   performance, with considerations for the impact on video delivery.

   This document aims to provide a taxonomy of networking issues as they
   relate to quality of experience in internet video delivery.  The
   focus is on capturing characteristics of video delivery that have
   surprised network designers or transport experts without specific
   video expertise, since these highlight key differences between common
   assumptions in existing networking documents and observations of
   video delivery issues in practice.

   Making specific recommendations for mitigating these issues is out of
   scope (though some existing mitigations might be mentioned in
   passing).  The intent is to provide a point of reference for future
   solution proposals to use in describing how new technologies address
   or avoid these existing observed problems.

2.  Bandwidth Provisioning

2.1.  Scaling Requirements for Media Delivery

2.1.1.  Video Bitrates

   Video bit-rate selection depends on many variables.  Different
   providers give different guidelines, but an equation that
   approximately matches the bandwidth requirement estimates from
   several video providers is given in [MSOD]:

   Kbps = (HEIGHT * WIDTH * FRAME_RATE) / (7 * 1024)

   Height and width are in pixels, and frame rate in frames per second.
   The actual bit-rate required for a specific video will also depend on
   the codec used and some other characteristics of the video itself,
   such as the frequency of high-detail motion, which may influence the
   compressability of the content, but this equation provides a rough
   estimate.

   Here are a few common resolutions used for video content, with their
   minimum per-user bandwidth requirements according to this formula:

   +-----------+---------------+------------------------------+
   | Name      | Width x Height | Approximate Bit-rate for 60fps |
   +-----------+---------------+------------------------------+
   | DVD       | 720 x 480     | 3 Mbps                       |
   |           |               |                              |
   | 720p      | 1280 x 720    | 8 Mbps                       |
   |           |               |                              |
   | 1080p     | 1920 x 1080   | 18 Mbps                      |
   |           |               |                              |
   | 2160p (4k) | 3840 x 2160  | 70 Mbps                      |
   +-----------+---------------+------------------------------+

2.1.2.  Virtual Reality Bitrates

   TBD: Reference and/or adapt content from expired work-in-progress
   [I-D.han-iccrg-arvr-transport-problem].

   The punchline is that it starts at a bare minimum of 22 Mbps mean
   with a 130 Mbps peak rate, up to 3.3 Gbps mean with 38 Gbps peak for
   high-end technology.

2.2.  Path Requirements

   The bit-rate requirements in Section 2.1 are per end-user actively
   consuming a media feed, so in the worst case, the bit-rate demands
   can be multiplied by the number of simultaneous users to find the
   bandwidth requirements for a router on the delivery path with that
   number of users downstream.  For example, at a node with 10,000
   downstream users simultaneously consuming video streams,
   approximately up to 180 Gbps would be necessary in order for all of
   them to get 1080p resolution at 60 fps.

   However, when there is some overlap in the feeds being consumed by
   end users, it is sometimes possible to reduce the bandwidth
   provisioning requirements for the network by performing some kind of
   replication within the network.  This can be achieved via object
   caching with delivery of replicated objects over individual
   connections, and/or by packet-level replication using multicast.

   To the extent that replication of popular content can be performed,
   bandwidth requirements at peering or ingest points can sometimes be
   reduced to a per-feed requirement instead of a per-user requirement.

2.3.  Caching Systems

   TBD: pros and cons of caching decisions at different locations within
   the network?

   Peak vs. average provisioning, and effects on peering point
   congestion under peak load?

   Provisioning issues for caching systems?

2.4.  Predictable Usage Profiles

   TBD: insert charts showing historical relative data usage patterns
   with error bars by time of day in consumer networks?

   Cross-ref vs. video quality by time of day in practice for some case
   study?  Not sure if there's a good way to capture a generalized
   insight here, but it seems worth making the point that demand
   projections can be used to help with e.g. power consumption with
   routing architectures that provide for modular scalability.

3.  Adaptive Bit Rate

3.1.  Overview

   Adaptive Bit-Rate (ABR) is a sort of application-level congestion
   response strategy in which the receiving media player attempts to
   detect the available bandwidth of the network path by experiment or
   by observing the successful application-layer download speed, then
   chooses a video bitrate that fits within that bandwidth, typically
   adjusting as changes in available bandwidth occur in the network.

   The choice of bit-rate occurs within the context of optimizing for
   some metric monitored by the video player, such as highest achievable
   video quality, or lowest rate of expected rebuffering events.

3.2.  Segmented Delivery

   ABR strategies are commonly implemented by video players using HLS
   [RFC8216] or DASH [DASH] to perform a reliable segment delivery of
   video data over HTTP.  Different player implementations and receiving
   devices use different strategies, often proprietary algorithms, to
   perform the bit-rate selection and available bandwidth estimation.

   This kind of bandwidth-detection system can experience trouble in
   several ways that can be affected by networking design choices.

3.2.1.  Idle Time Between Segments

   When the bit-rate selection is successfully chosen below the
   available capacity of the network path, the response to a segment
   request will complete in less absolute time than the video bit-rate
   speed.

   The resulting idle time within the connection carrying the segments
   has a few surprising consequences:

   o  Mobile flow-bandwidth spectrum and timing mapping.

   o  TCP Slow-start when restarting after idle requires multiple RTTs
      to re-establish a throughput at the network's available capacity.
      On high-RTT paths or with small enough segments, this can produce
      a falsely low application-visible measurement of the available
      network capacity.

3.2.2.  Head of Line Blocking

   In the event of a lost packet on a TCP connection with SACK support
   (a common case for segmented delivery in practice), loss of a packet
   can provide a confusing bandwidth signal to the receiving
   application.  Because of the sliding window in TCP, many packets may
   be accepted by the receiver without being available to the
   application until the missing packet arrives.  Upon arrival of the
   one missing packet after retransmit, the receiver will suddenly get
   access to a lot of data at the same time.

   To a receiver measuring bytes received per unit time at the
   application layer, and interpreting it as an estimate of the
   available network bandwidth, this appears as a high jitter in the
   goodput measurement.

   Active Queue Management (AQM) systems such as PIE [RFC8033] or
   variants of RED [RFC2309]} that induce early random loss under
   congestion can mitigate this by using ECN [RFC3168] where available.
   ECN provides a congestion signal and induce a similar backoff in
   flows that use Explicit Congestion Notification-capable transport,
   but by avoiding loss avoids inducing head-of-line blocking effects in
   TCP connections.

3.3.  Unreliable Transport

   In contrast to segmented delivery, several applications use UDP or
   unreliable SCTP to deliver RTP or raw TS-formatted video.

   Under congestion and loss, this approach generally experiences more
   video artifacts with fewer delay or head of line blocking effects.
   Often one of the key goals is to reduce latency, to better support
   applications like video conferencing, or for other live-action video
   with interactive components, such as some sporting events.

   Congestion avoidance strategies for this kind of deployment vary
   widely in practice, ranging from some streams that are entirely
   unresponsive to using feedback signaling to change encoder settings
   (as in [RFC5762]), or to use fewer enhancement layers (as in
   [RFC6190]), to proprietary methods for detecting quality of
   experience issues and cutting off video.

4.  IANA Considerations

   This document requires no actions from IANA.

5.  Security Considerations

   This document introduces no new security issues.

6.  References

6.1.  Normative References

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