Network Working Group
Internet-Draft
Intended status: BCP
Expires: December 24, 2012

C. Jennings
S. Nandakumar
H. Phan
Cisco

June 22, 2012

Vendors Considered Harmfull
draft-jennings-vendors-considered-harmfull-00

Abstract

There are things vendors will do to make sure that the internet has a robust and stable solution to congestion. However, significantly decreasing the video quality of their currently shipping products is probably not one of the things they will do.

Status of this Memo

Copyright Notice

---

1. Executive Summary

This paper looks at what type of good advice, even when accompanied by many fine lunches and dinners, is likely to be ignored by vendors. For better or worse, many vendors seem to follow the following principals:

> Principal 1 - Tomorrow is Better:
>
> Version N+1 of a product MUST NOT **[1]** have a worse user experience than version N under the same network conditions.
>
> Principal 2 - Yesterday is Past:
>
> If a major vendors products does X, and X turns out to be bad for the overall user population but makes that product look better in some cases to individual users, then it's OK for products from other vendors to also do X.
>
> Principal 3 - Today is not Tomorrow:
>
> Version 1.0 of the product does not need to worry about congestion control, security, management, and so on.

So what does this all mean for a video application on the web? There is some evidence (see **Section 2**) to indicate that existing interactive video products use more bandwidth than a single TCP connection would use in the same network conditions. This is not a unique situation - many applications that run over TCP, such as google maps, iTunes, and Netflix can use more bandwidth than a single TCP connection would use in the same network conditions. One approach is not to abandon the constraints of TCP but instead limit an application to what 4 TCP connections would use. Variants of this approach has been widely deployed with results that don't appear disastrous. It is also the solutions adopted by **[2]** which suggests using up to an "order of magnitude" more bandwidth than TCP.

The IETF needs to find a path to allow interactive video applications to have reasonable congestion control without causing a violation of the above principles. On the other hand, it needs to make it clear that it would be reasonable for standard network operation procedures to block traffic from applications that pushed all the TCP traffic to near zero bandwidth. Working groups that are constrained to do video in a way that is strictly TCP friendly are likely to have their output ignored by the vendors. Just like the vendors have ignored it for interactive audio.

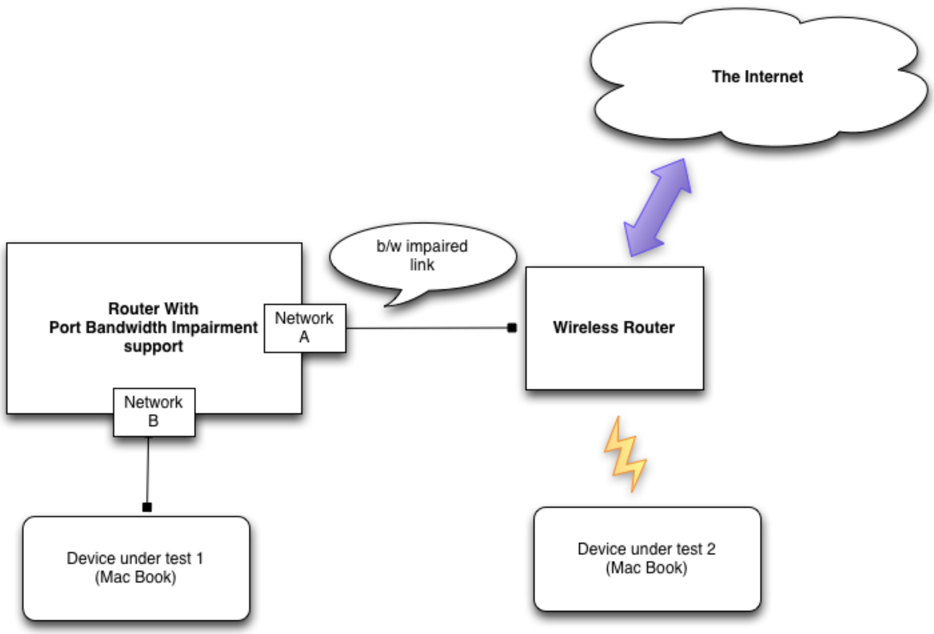
2. Measurements

This section contains measurement results showing interactions between RTP and TCP flows for the following clients - Apple Facetime, Google Hangout, Cisco Movi and Microsoft Skype. We chose these clients since they represent the greatest and the latest VOIP clients used by numerous end-users across the globe.

The idea is to observe and analyze the behavior of RTP bandwidth utilization when in competition with other flows, say TCP.

The measurements represent some early observations and these results have not been carefully verified. We encourage others to duplicate the experiment.

To perform the measurements, the below test-setup was used.



A special version of the Cisco router with pMod tool is used for impairing bandwidth between the networks to which devices under test are individually connected. The device under test (DUT) 1 is physically connected to "Network B", whereas, DUT-2 is wirelessly connected to Internet through Linksys wireless router. The router itself is connected to the wireless router via "Network A" a shown.

Network Link A is bandwidth capped at 512 Kbps and a delay of 250 ms is inserted for the incoming packets, thus resulting in a low download rate and delayed packets at DUT-1.

A 2 way audio and video RTP call with a competing FTP flow was instantiated to perform the measurements.

In the order of Skpye, Hangout, Facetime and Movi, the below graphs capture the interactions between RTP and TCP (FTP-Data) as observed from DUT-1.

Graphs shown below are read as follows:

Red Line --> TCP Flow

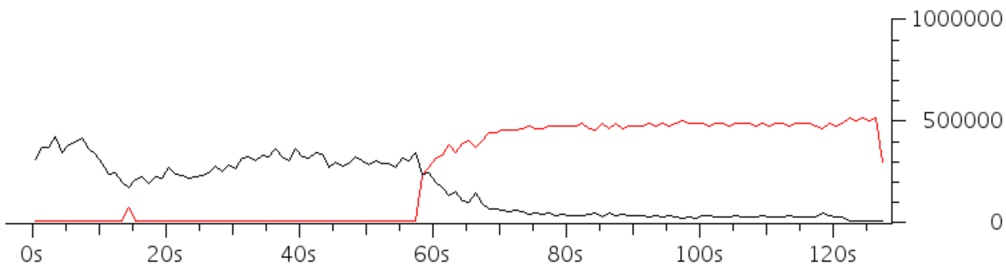
Black Line --> RTP Flow

X-axis --> Time in seconds

Y-axis --> Bits per second

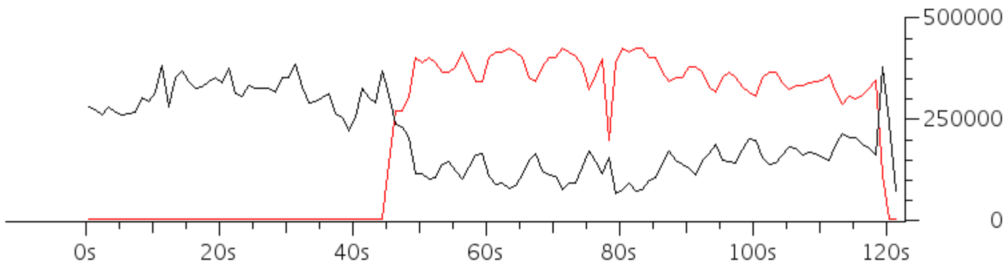
2.1. Skype

Mircosoft Skype Graph - 512 Kbps Link with 250 ms latency



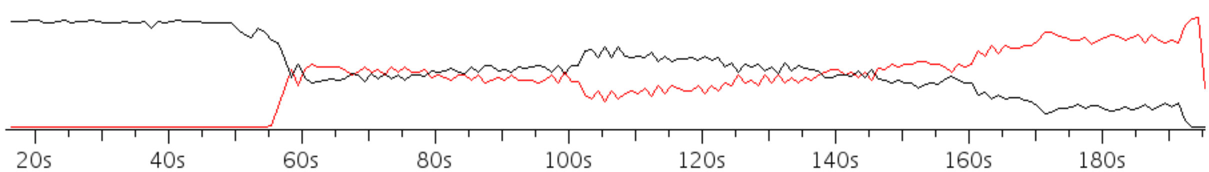
2.2. Hangouts

Google Hangout Graph - 512 Kbps Link with 250 ms latency



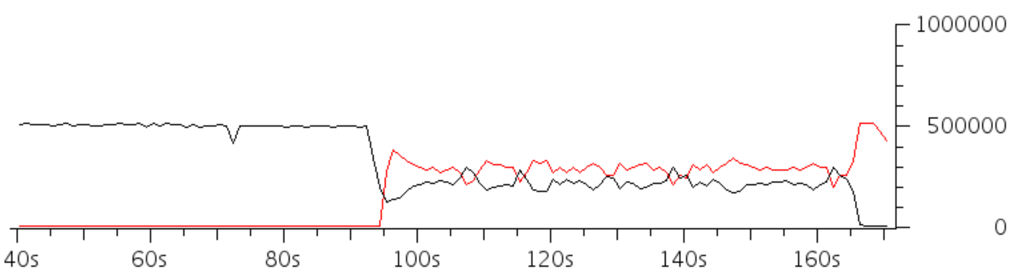
2.3. Facetime

Apple Facetime Graph - 512 Kbps Link with 250 ms latency



2.4. Movi

Cisco Movi Graph - 512 Kbps Link with 250 ms latency



3. Normative References

[1] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels," BCP 14, RFC 2119, March 1997 (TXT, HTML, XML).
[2] Perkins, C. and V. Singh, "RTP Congestion Control: Circuit Breakers for Unicast Sessions," draft-perkins-avtcore-rtp-circuit-breakers-00 (work in progress),

March 2012 (**TXT**).

Authors' Addresses

Cullen Jennings
Cisco
170 West Tasman Drive
San Jose, CA 95134
USA
Email: **fluffy@cisco.com**

Suhas Nandakumar
Cisco
170 West Tasman Drive
San Jose, CA 95134
USA
Email: **snandaku@cisco.com**

Hein Phan
Cisco
170 West Tasman Drive
San Jose, CA 95134
USA
Email: **hiphan@cisco.com**