Politecnico di Bari
Dipartimento di Ingegneria Elettrica e dell'Informazione

C3LAB
Control of Computing and Communication Systems Lab

Understanding the Dynamic Behaviour of the Google Congestion Control for RTCWeb

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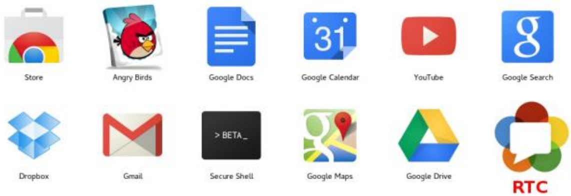
Luca De Cicco, Gaetano Carlucci, Saverio Mascolo

Politecnico di Bari, Dipartimento di Ingegneria Elettrica e dell'Informazione

REAL TIME COMMUNICATION

- Boosted by the large diffusion of mobile devices and high speed connections
- A bunch of real-time communication apps exist
- Each uses a proprietary set of algorithms
- Interoperability is not possible or not convenient

RTC AND THE WEBRTC INITIATIVE



- Lots of apps converging on the web
- Why not leveraging web browsers for RTC?
- WebRTC (W3C) & RTCWeb (IETF) WGs are tackling this challenge

W3C + IETF: THE CHALLENGES

- Javascript API for HTML (W3C)
- Signalling & NAT traversal (IETF RTCWEB)
- Security (IETF RTCWEB)
- Congestion control (IETF RMCAT)

CONGESTION CONTROL REQUIREMENTS FOR RTC FLOWS

- Contain end-to-end delay (queuing delay)
- Contain packet losses (to decrease FEC)
- Reasonable fairness with other flows (both intra-protocol and iter-protocol)
- Prevent starvation when competing with TCP long-lived flows

See R. Jesup, “Congestion Control Requirements For RMCAT”, IETF Draft, Jul. 2013, draft-ietf-rmcat-cc-requirements-00

A WEBRTC STACK

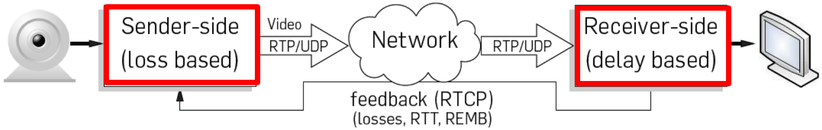
- WebRTC is now available in Chrome, Firefox, and Opera
- The only proposed congestion control algorithm for WebRTC that has been implemented in a browser is Google CC
- Reference implementation in Chrome (Chromium) stable

Experimentally investigate the dynamics and issues of the Google Congestion Control (GCC)

Lundin, Holmer, Alvestrand, "*A Google Congestion Control Algorithm for Real-Time Communication*", IETF RtcWeb -Rmcat, 2013

GOOGLE CONGESTION CONTROL

THE CONTROL ARCHITECTURE



- Audio/video flows sent using RTP over UDP
- Hybrid loss-based delay-based approach
- The <u>sender-side controller</u> probes the available bandwidth
- The <u>receiver-side controller</u> computes the ``Receiver Estimated Maximum Bitrate'' A_r to limit the sending rate A_s and contain the queuing delay

SENDING RATE COMPUTATION IN A NUTSHELL

Sender-side: A_s

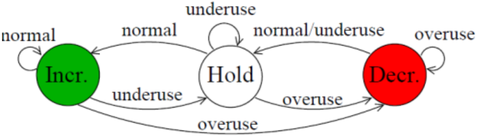
When a RTCP report is received:

- Low losses: multipliative increase
- High losses: multiplicative decrease proportionally to f_l
- Moderate losses: constant

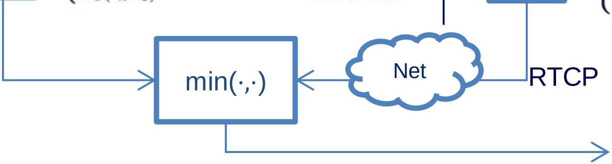
$$A_s(t_k) = \begin{cases} 1.05(A_s(t_{k-1}) + 1\text{kbps}) & f_l(t_k) < 0.02 \\ A_s(t_{k-1})(1 - 0.5 f_l(t_k)) & f_l(t_k) > 0.1 \\ A_s(t_{k-1}) & \text{otherwise} \end{cases}$$

Receiver-side: A_r

Driven by a three sates FSM



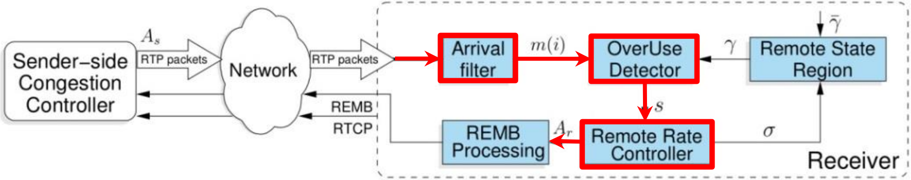
$$A_r(t_i) = \begin{cases} \eta A_r(t_{i-1}) & \text{Increase} \\ \alpha R(t_i) & \text{Decrease} \\ A_r(t_{i-1}) & \text{Hold} \end{cases}$$



Sending rate A_s

THE RECEIVER

<u>IDEA</u>: to compute the rate A_r based on the estimated one-way queuing-delay <u>variation</u> *m*

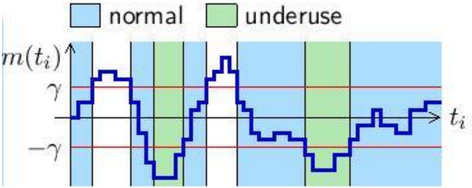


- <u>Arrival filter:</u> estimates the QD variation $m(t_i)$ using a Kalman filter
- <u>Overuse detector (OUD):</u> based on $m(t_i)$ generates a signal
- <u>Remote rate controller:</u> the signal produced by the OUD drives the state of a FSM that sets the rate A_r

THE OVERUSE DETECTOR

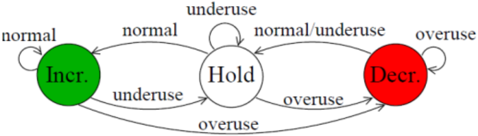
Arrival Filter

$m(t_i)$



The signal is compared to two thresholds $-\gamma$, γ

- $|m| \leq \gamma$: the network is considered uncongested ⇒ ``normal"
- $m < -\gamma$: the network is considered underused ⇒ ``underuse"
- $m > \gamma$: for more then t_{OU} (100 ms) the network is considered congested ⇒ ``overuse"



A *overuse* always triggers a switch to Decrease state

TESTBED

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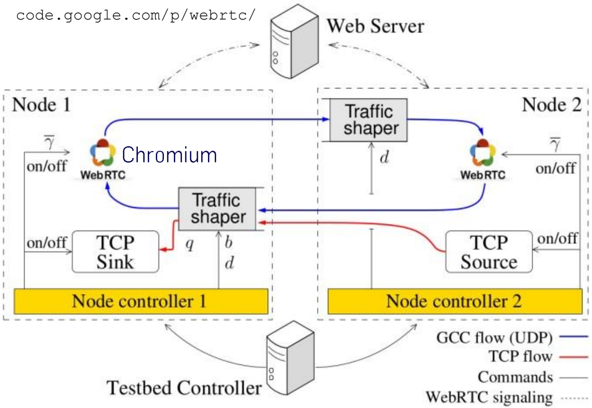

- Two hosts, one web server, one testbed controller to orchestrate experiments
- TC+NetEm to set bandwidth, buffer sizes, and base RTT
- TCP long-lived traffic using iperf
- Modified Chromium to change γ and log internal variables

METRICS

- Channel Utilization U=R/b: where *b* is the known available bandwidth and *R* is the average received rate
- Loss Ratio *l* =(packet lost)/(packet received)
- Queuing delay T_q: measured averaging the value $RTT(t)-RTT_m$ over all the RTT samples reported in the RTCP feedbacks
- Number of delay based decreased event n_{dd}: the number of times the remote controller makes the sending rate to be decreased.

TESTBED PARAMETERS

Single flow scenario

Bandwidth (kbps)	Buffer size (kB)	RTT (ms)	Threshold γ (ms)
500, 1000, 1500, 2000	15, 30, 45, 60, 75	50	10/60, 15/60,…,65/60

GCC vs TCP

Bandwidth (kbps)	Buffer size (kB)	RTT (ms)	Threshold γ (ms)
1000, 2000, 3000	15, 30, 45, 60, 75	50	10/60, 15/60,…,65/60

For each tuple of parameters we run three experiments
Dataset : 120 hours of active measurements, around 1300 calls

EXPERIMENTAL RESULTS

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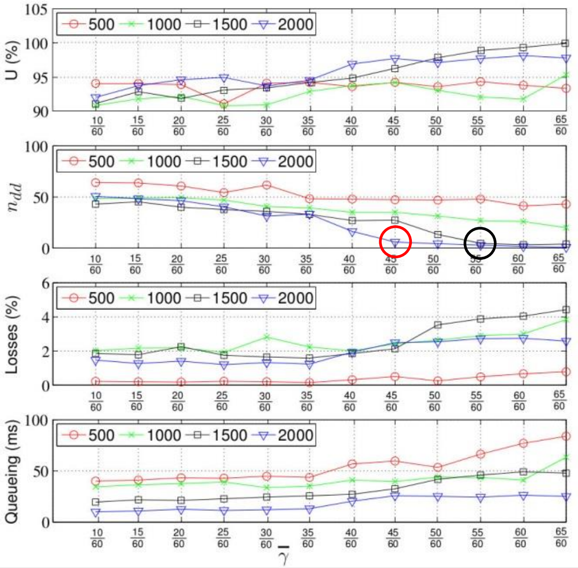
ONE GCC FLOW OVER A BOTTLECK (buffer size 30 kB)



γ=25/60 default value

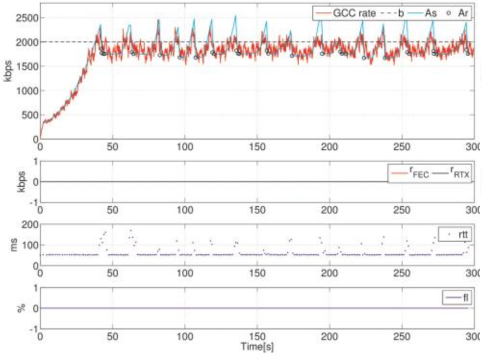
Utilization increases with γ

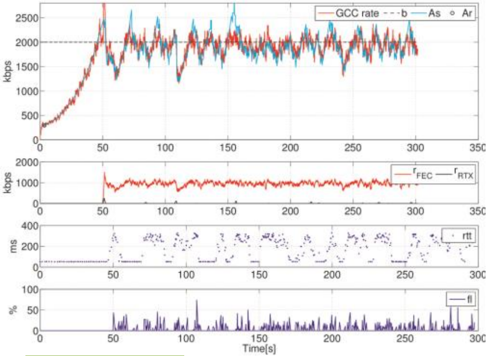
The number of delay-based decrease events decreases with γ

n_{dd} decreases faster at higher bandwidths

When n_{dd} decreases losses and queuing delay increase (more loss-based)

EFFECT OF γ ON QUEUING AND LOSSES (BW=2Mbps, QS=75kB)





γ=10/60

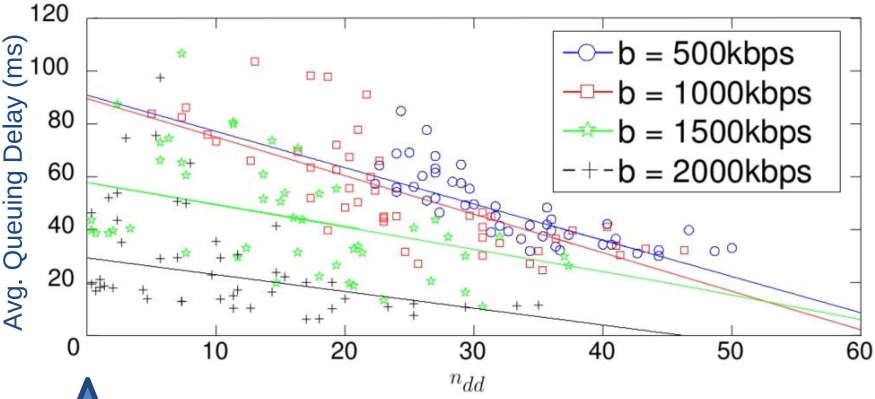
γ=65/60

No losses, queuing delay less than 150 ms, several DB decrease events

Losses (4%), queuing delay doubled, zero DB decrease events. No influence of the DB algo

ONE GCC FLOW OVER A BOTTLECK

Bandwidth (kbps)	Base rtt (ms)	Runs	Buffer size kB	γ (ms)
500, 1000, 1500, 2000	50	3	15, 30, 45, 60, 75	[10/60-65/60]

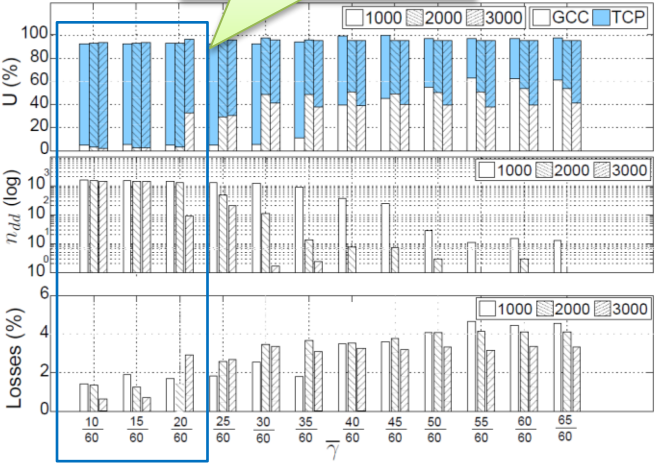


Loss-based only

Increasing influence of delay-based algo

ONE GCC FLOW vs ONE TCP FLOW (QS=30 kB)

Why starvation occurs?



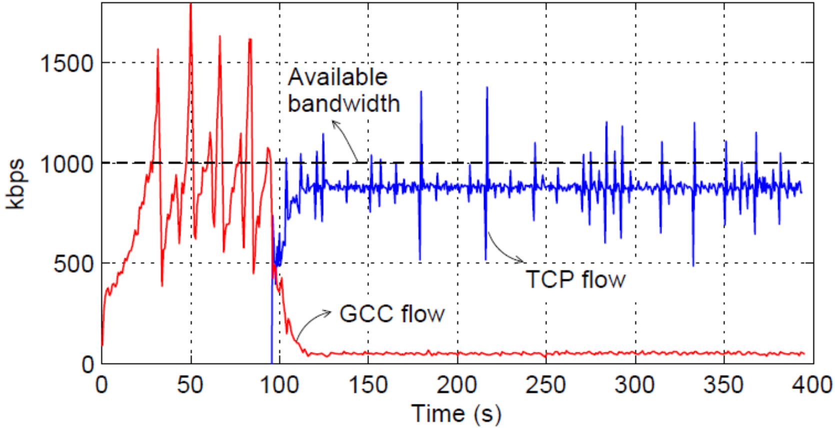
Large number of delay-based decrease events

With increased queue sizes a larger and larger γ is required to avoid starvation of GCC

For low γ GCC is starved

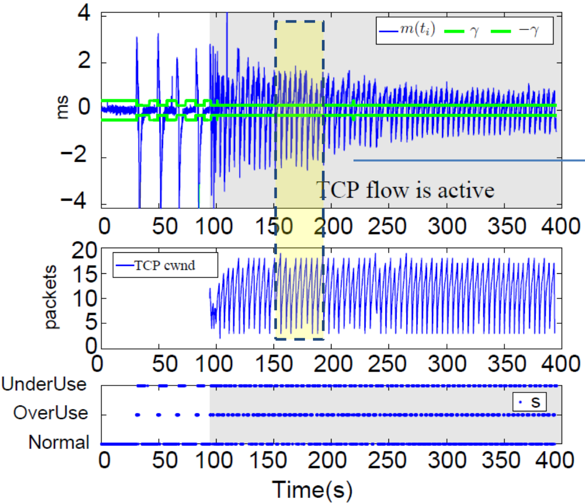
GCC STARVED BY TCP (1/4)

b=1000kbps, q=30kB, γ=25/60



GCC STARVED BY TCP (2/4)

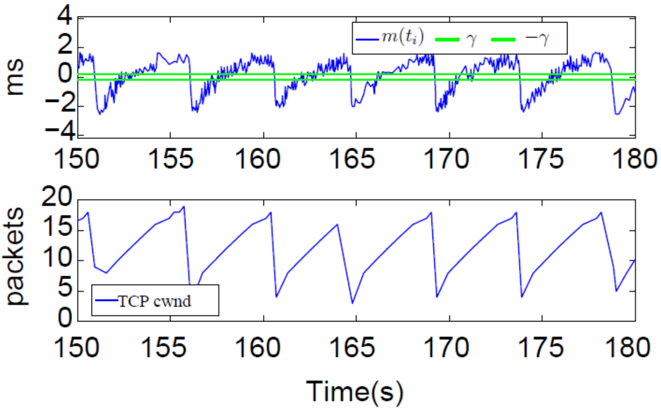
The receiver dynamics



Zoom

Large number of overuse signals produced after the TCP flow joins

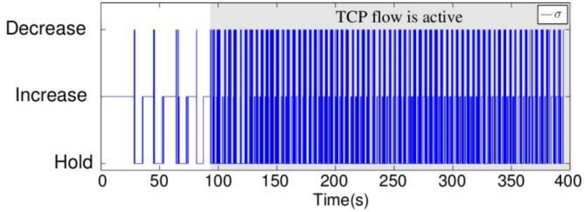
GCC STARVED BY TCP (3/4)



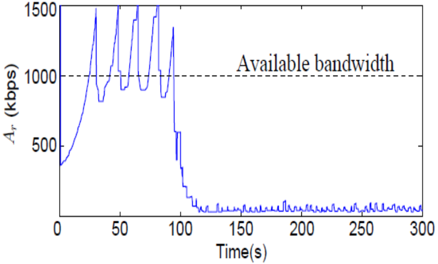
large QD variations due to TCP cong. control dynamics and not to self-inflicted delay

GCC STARVED BY TCP (4/4)

Remote rate controller state



When TCP is on the Remote Rate controller FSM switches to Decrease mode frequently



A_r is quickly decreased, leading to starvation

CONCLUSIONS

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<u>Single GCC flow</u>

- the threshold γ has a remarkable impact on the performance
- for lower value of γ queuing and losses are contained

<u>GCC vs TCP</u>

- the threshold γ has a remarkable impact on the friendliness
- for sufficiently high value of γ reasonable fairness is reached

The threshold should be made adaptive to provide optimal performance and prevent starvation in the case of concurrent TCP traffic

QUESTIONS

QUESTIONS?

BACKUP SLIDES

ARRIVAL FILTER: THE QUEUING DELAY VARIATION COMPUTATION

One way delay variation model

OWD var. = Transm. Time var. + Queuing delay var. + net jitter

$d(t_i)$ $\frac{L(t_i) - L(t_{i-1})}{C(t_i)}$ $m(t_i)$ $n(t_i)$

i-th frame size Bottleneck bw considered gaussian

One way delay variation measurement

OWD var. = Inter-arrival time - Inter-departure time

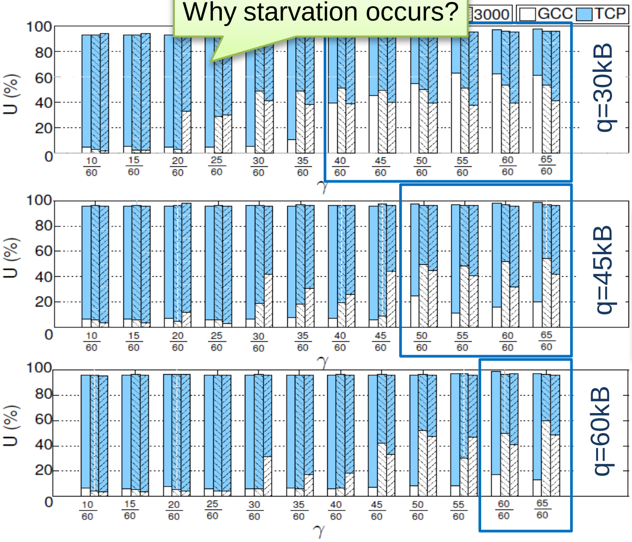
$d_m(t_i)$ $t_i - t_{i-1}$ $T_i - T_{i-1}$

QD variation $m(t_i)$ computation

A Kalman filter computes $m(t_i)$ and $1/C(t_i)$ to steer the residual measurement error $d(t_i) - d_m(t_i)$ to zero

CHANNEL UTILIZATION vs QUEUE SIZE

Why starvation occurs?



With increased queue sizes a larger and larger γ is required to avoid the starvation of GCC flows