Adaptivity in Multimedia Flows
a Feedback Control Approach

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Motivation

Two ongoing trends

- **Video is booming**: video applications will account for more than half of the global traffic in 2012
- **Mobile is growing**: mobile data traffic was 1% of total IP traffic in 2010, and will be 8% in 2015

Data Source: CISCO VNI, June 2011
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Issues

- Internet was not designed for delay-sensitive traffic
- Bandwidth is unpredictable in best-effort Internet
- Mobile devices have limited CPU, battery, screen, and network bandwidth resources

Data Source: CISCO VNI, June 2011
The challenge

Content producer PoV

*Provide a seamless multimedia experience maximizing the QoE obtainable given devices and network constraints*
The challenge

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The building block

The key technological enabler is Adaptivity
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The building block

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Adaptive multimedia application requirements

1. It should not harm the network, i.e. no congestion collapse ⇒ Effective congestion control (TCP or application layer)

2. It should provide the maximum audio/video quality possible ⇒ Flows must be made elastic by means of adaptive codecs together with an effective quality adaptation controller
Outline of the talk

1 Motivation

2 Introduction
   - Control Architectures

3 Skype Adaptation Algorithm

4 Adaptive Live Streaming
   - Akamai Adaptive Streaming
   - Feedback control Stream Switching
The stack

Multimedia App
- Video Str.
- Tele Pres.

Adaptive Codecs
- WebM
- H264
- Speex

Adaptation Control
- SS
- FGS
- TC

Congestion Control
- TCP
- DCCP
- UDP

Internet
**The data flow path**

1. The A/V Encoder compresses the raw A/V flow at a bitrate $r_c(t)$
2. The compressed A/V flow is sent via a TCP/UDP over the Internet at a rate $r_T(t)$
3. The receiver decodes the flow

**How to implement adaptivity?**
Introduction to End-to-end Architecture

Adaptation controller regulates the encoding rate $r_c(t)$ based on receiver’s feedback. Controller logic may be implemented at the receiver.

Feedback could be playout buffer, lost frame rate, etc.

**Drawback:** the feedback loop is affected by a time delay due to the RTT of the connection.
**Local Feedback Architecture**

- The feedback is local, control loop is not affected by delays.
- Local feedback can be network available bandwidth (a-la TCP Westwood+), sender buffer size, etc.
- **Drawback**: receiver is not in the loop, CPU overload cannot be taken into account.
Leading applications: are they doing it right?

Video Streaming

- **YouTube**: no adaptivity; each video is encoded into a number of different bitrates and resolutions, the user picks the version making a trade off between quality and his prediction about QoE.

- **NetFlix**: implements adaptivity; each video is encoded into a number of different versions and the client automatically selects the most suitable version.

- **Akamai HD Net**: heuristic-based stream-switching similar to NetFlix approach.

Video-conference

**Skype**: it implements adaptivity by throttling encoding bitrate to match available bandwidth [WWIC07, NOSSDAV08, CDC08, TAC10, COMNET11]

Introduction

- TCP is not suitable for real-time multimedia communication (retransmissions)
- Several multimedia congestion control algorithms have been proposed: TFRC, DCCP, ARC, RAP
- Commercial applications still use proprietary congestion control algorithms (if any) over the UDP
- Skype is the leading VoIP application, generating a very large amount of UDP flows

Goal

To identify Skype VoIP adaptation algorithm mathematical model and to analyze its most relevant properties, i.e. to what extent it is able to adapt to time variable available bandwidth
Skype Adaptation Algorithm

State of the art

- Skype flows adapt its rate to bandwidth variations to some extent (losses occur) [WWIC07, NOSSDAV08]
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The Skype VoIP adaptation exhibits remarkable unfairness \textit{wrt} TCP concurrent flows due to a very slow responsiveness to sudden bandwidth variations (transient lasts $\sim 40$ s)
Three hosts: two Skype, one emulates a wide area network

The emulator can impose available bandwidth, RTT, and loss ratio
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We modified QT libraries to log the Skype tooltips (measured loss ratio $\hat{l}(t)$, $\hat{RTT}(t)$, etc)
Testbed for experimental evaluation

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The emulator can impose available bandwidth, RTT, and loss ratio
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We expect loss ratio \( \hat{l}(t) \) and round trip time \( \hat{RTT}(t) \) measured by Skype are candidate inputs of the controller
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We cannot inspect packets since Skype encrypts flows
Experiments

To identify the mathematical model of the rate $r_s(t)$ generated by Skype VoIP flow we conducted several experiments using the testbed:

1. Variable RTT $RTT(t)$: we found it has no effect on $r_s(t)$
2. Variable loss ratio $l(t)$ (lossy link): Skype uses FEC to counteract losses when losses are not due to congestion
3. Variable bandwidth $b(t)$: we found that Skype uses a low-passed version of $l(t)$ as the main driver for $r_s(t)$
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\[ L_i(0) = L_{init} \]
\[ \hat{l}(0) = 0 \]
Skype Audio over a variable loss rate link

Loss ratio $l(t)$: square wave, period 200 s, max value 0.4, min value 0

Results

1. When $l(t)$ increases, the sending rate increases $\Rightarrow$ not a loss based algorithm

2. The measured loss ratio $\hat{l}(t)$ (red) is a low pass filtered version of $l(t)$ (blue)

3. Packet size increases when $l(t)$ increases $\Rightarrow$ FEC to counteract persistent packet losses (state $S_3$ - Losses)
FEC action $f(t)$ and $\hat{l}(t)$

FEC action model $f(t)$

- Skype adds at most one redundant frame for each new sent frame
- when FEC action is maximum rate doubles
- $f(t) \in [0, 1]$ fraction of redundant frames: $f(t) = 0$ FEC is off, $f(t) = 1$ FEC is max
FEC action $f(t)$ and $\hat{l}(t)$

**FEC action model $f(t)$**
- Skype adds at most one redundant frame for each new sent frame when FEC action is maximum rate doubles.
- $f(t) \in [0, 1]$ fraction of redundant frames: $f(t) = 0$ FEC is off, $f(t) = 1$ FEC is max.

**Measured loss ratio $\hat{l}(t)$**
- Loss ratio: $l(t) = o(t)/r(t)$, where $o(t)$ is the overflow rate, $r(t)$ is the sending rate.
- Skype measured loss rate: $l(t)$ filtered by a first order filter with time constant $\tau$ ($\tau \approx 11$ s)

$$\dot{\hat{l}}(t) = -\frac{1}{\tau}\hat{l}(t) + \frac{1}{\tau}l(t)$$
Proposed Model

**Hypothesis:** the encoder is multi-rate, it can choose among $N$ levels of encoding $\mathcal{L} = \{L_1, \ldots, L_N\}$

Proposed model: controller + encoder

$$r_s(t) = (1 - \hat{l}(t)) \left(1 + f(t)\right) L_i(t)$$

$L_i(t)$ is the encoding level selected at time $t$
Validation of the proposed model

- The sending rate predicted by the model follows the rate produced by Skype
- $f(t) = 0$ during congestion (after a bandwidth drop occurs)
- FEC is activated after the available bandwidth increases to its maximum value (packet size doubles) to counteract packet losses in the “probing phase”

The long transient time exhibited by the CC algorithm is due to the filtering of the loss ratio
A Skype flow accessing a bottleneck

Setting: available bandwidth $b(t)$, drop-tail queue length $q_M$, fw path delay $T_1$, bw path delay $T_2$. State $[q(t), \hat{l}(t)]^T$

Queue length model $q(t)$

$$
\dot{q} = \begin{cases} 
0 & q = 0, r \leq b \text{ or } q = q_M, r \geq b \\
r - b & \text{otherwise}
\end{cases}
$$

(1)

where $r$ is the queue input rate. The overflow rate is given by:

$$
o = \begin{cases} 
r - b & q = q_M, r > b \\
0 & \text{otherwise}
\end{cases}
$$

(2)

Measured loss ratio $\hat{l}(t)$ model

Recalling the model for the measured loss ratio is

$$\dot{\hat{l}} = -\frac{1}{\tau} \hat{l} + \frac{1}{\tau} \frac{o r T_2}{r T_2}$$

and plugging (2) it turns out:

$$
\dot{\hat{l}} = \begin{cases} 
f_1 = \frac{1}{\tau} - \frac{\hat{l}}{\tau} - \frac{b T_2}{\tau(1-\hat{l} T_2)(1+f T_2) L T_2} & q = q_M, r > b \\
f_2 = -\frac{1}{\tau} \hat{l} & \text{otherwise}
\end{cases}
$$

(3)
Skype accessing a bottleneck: hybrid model

It is simple to show that the following hybrid automaton models the considered system (equations (1) and (3)):

Hybrid Automaton $\mathcal{H}$

Characterizing the equilibria of $\mathcal{H}$

- Characterization of the equilibrium states of $\mathcal{H}$
- Steady state characterization of a Skype Audio flow accessing a bottleneck with capacity $b$
- The time delay is neglected since $\tau \gg RTT$
Qualitative Phase Portrait of $H$ [TAC10]

Two different phase portraits depending on equilibrium inputs $b^*, f^*, L^*$

Unique equilibrium:
$\hat{l}^* = 1 - \sqrt{\frac{b^*}{L^*(1+f^*)}}$; $q^* = q_M$

$\forall(\hat{l}(t_0), q(t_0))$ only reachable state: $\Sigma_3$ (full queue) ⇒
Persistent congestion

Unique equilibrium:
$\hat{l}^* = 0$; $q^* = 0$

$\forall(\hat{l}(t_0), q(t_0))$ only reachable state $\Sigma_1$ (empty queue) ⇒
No congestion
**Phase Portrait** $b^* < (1 + f^*)L^*$

Example $b^* = 30$ kb/s, $f^* = 0$, $L^* = 56$ kb/s $\Rightarrow l^* = 0.2546$, $q^* = q_M$
Characterization of the steady state

**Proposition**

Let us consider the equilibrium inputs $b^*, L^*$ and $f^*$ the automaton $\mathcal{H}$ is characterized by the following equilibria that are globally asymptotically stable:

$$\hat{l}^* = 1 - \sqrt{\frac{b^*}{L^*(1 + f^*)}}; \quad q^* = q_M \text{ iff } b^* < (1 + f^*)L^* \quad (4)$$

$$\hat{l}^* = 0; \quad q^* = 0 \text{ iff } b^* \geq (1 + f^*)L^* \quad (5)$$

**Corollary**

The controller employed by Skype is not able to avoid congestion unless $b^* > L^*(1 + f^*)$.

**Proof** We are under the hypothesis of Prop 1 $\Rightarrow$ (4) is asymptotically stable. At steady state it results $o^* = (1 - \hat{l}^*)L^*(1 + f^*) - b^*$, thus substituting (4) in the equation just written we have:

$$o^* = \sqrt{b^*L^*(1 + f^*)} - b^* > 0$$
Proposition validation

Available bandwidth drop at $t = 50$, $b^* = 30 \text{ kb/s}$, $f^* = 0$, $L^* = 56\text{kb/s} \Rightarrow r^* = 41\text{kb/s}$, $o^* = 11 \text{ kb/s}$

Conclusions

The steady state values for $r(t)$ and $o(t)$ recover the ones predicted by the model. Persistent overflow rate is present: Skype VoIP is not able to avoid congestion.
Akamai Adaptive Streaming Algorithm

- **Akamai Client**: Decoder, Player, Measurement, Adaptation Controller

- **Akamai Server**: TCP buffer, Video Levels, Actuator

- **Internet**: HTTP traffic, TCP rx buffer

- **Adaptation Controller** selects $l(t) \in L$

- **Parameters**:
  - $q(t)$
  - $r(t)$
  - $B(t)$
  - $f(t)$
  - $F(t)$
  - $c(t)$
  - $T(t)$
  - $\tau_b$
  - $\tau_f$

- **Events**:
  - HTTP POST
  - TCP buffer
  - Selects $l(t) \in L$
Five video levels $\mathcal{L} = \{l_0, \ldots, l_4\}$ from 300 kbps (320x180) up to 3500 kbps (1280x720)

- Adaptation logic is client side
- Control loop is distributed: time-delay (RTT) affects the loop
- Adaptation logic is made of two coupled modules: 1) a buffer level controller and 2) a stream-switching heuristics
**The client-server protocol**

1. Client clicks on video, a HTTP GET request is sent to download a SMIL file.

2. The SMIL file is sent to the client. The client parses the file which contains information on video levels with encoding bitrates.

3. At time $t_0$, the adaptation algorithm starts. HTTP POST requests are sent specifying a command $c(t_0)$, the video level $l(t_0)$ and several feedback variables $F(t_0)$.

4. The server sends to the client the video level requested at time $t_0$.

5. The algorithm repeats at time $t_i$. 

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Client (Flash player) | Akamai HD Server
--- | ---
1 | GET('videoname.smil')
2 | videoname.smil
3 | POST($c(t_0)$, $l(t_0)$, $F(t_0)$)
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**Diagram:**

- **Client (Flash player):**
  - GET('videoname.smil')
  - POST($c(t_0)$, $l(t_0)$, $F(t_0)$)

- **Akamai HD Server:**
  - videoname.smil
  - GET('videoname.smil')
  - POST($c(t_0)$, $l(t_0)$, $F(t_0)$)
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Client (Flash player) | Akamai HD Server
---|---
1. GET('videoname.smil') |
2. | videoname.smil
3. POST($c(t_0), l(t_0), F(t_0)$) |
4. | ...
5. POST($c(t_i), l(t_i), F(t_i)$) |
   ...
The commands (cmd)

POST /control/Fname?cmd=c_i,a_1,...,a_n&v=1.0&r=ABCDE&g=token&lvl1=F_1,...,F_{12}

**POST arguments**

1. **cmd**: command $c_i$ to be issued on the server
2. **lvl1**: client feedback variables $F_1,...,F_{12}$ sent to the server

**Commands $c_i$**

1. **throttle**: issued periodically on average each 2s to adjusts the receiver buffer using a feedback control loop
2. **rtt-test**: issued periodically, on average each 11s, triggers greedy send mode (lasts 5 seconds) to estimate available bandwidth and RTT under congestion
3. **SWITCH_UP**: asks the server to switch the video level up
4. **BUFFER_FAIL**: asks the server to switch the video level down
The feedbacks (lvl1)

1. Receiver buffer size $q(t)$ [s]
2. Receiver buffer set point $q_T(t)$ [s]
3. Decoded frame rate $f(t)$ [fps]
4. Estimated bandwidth $B(t)$ [kbps]
5. Received goodput $r(t)$ [kbps]
6. Current video level $l(t)$ [kbps]
7. Round trip time $R(t)$ [s]
The buffer level controller (server)

Goal of the controller

To steer the client buffer length $q(t)$ to the set point $q_T(t)$

The throttle percentage

The throttle command specifies as the only argument the **throttle percentage** $T(t)$. We have identified the following control law:

$$T(t) = \max \left( 100 \left( 1 + \frac{q_T(t) - q(t)}{q_T(t)} \right), 10 \right)$$

Akamai server throttles the TCP send buffer filling rate $\bar{r}(t)$ by using $T(t)$:

$$\bar{r}(t) = l(t) \frac{T(t)}{100}$$

When the error $e(t) = q_T(t) - q(t) > 0$ (buffer below the threshold) $T(t) > 100$ so that $\bar{r}(t) > l(t)$. This allows to send the video at a higher rate than $l(t)$ letting the receiver to fill the buffer.
The stream-switching heuristic (client)

Identified stream-switching heuristics

\[
\begin{align*}
&k < i < j \\
&\text{BUFFER\_FAIL} \rightarrow & l_k & \quad q(t) < q_L \land B(t) > 1.2l_k \\
&\text{otherwise} & l_i & \quad q(t) \geq q_L \land B(t) > l_j(1 + S(t)) \\
&\quad \quad \quad & l_j & \quad \text{SWITCH\_UP}
\end{align*}
\]
**The stream-switching heuristic (client)**

**Identified stream-switching heuristics**

- \( k < i < j \)
- \( q(t) < q_L \wedge B(t) > 1.2l_k \)
- \( q(t) \geq q_L \wedge B(t) > l_j(1 + S(t)) \)

**Safety factor \( S(t) \)**

When \texttt{rtt-test} is issued, \( T(t) = 500 \) allowing the server to send in greedy mode and to probe for the available bandwidth and measure the RTT \( R(t) \) under congestion.

\[ S(R) = 2.5R + 0.15 \]
Step Response (1/2)

- Step-like available bandwidth from 0.5Mb/s to 4 Mb/s
Step Response (1/2)

- Step-like available bandwidth from 0.5Mb/s to 4 Mb/s
- Each time $B(t) > l_i(1 + S(t))$ a SWITCH_UP is sent.

Video levels are switched up after a delay of around 15s. Sluggish response due to conservativeness. Large oscillations due to the greedy/non-greedy phases triggered by rtt-test. Poor channel utilization.
Step Response (1/2)

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![Graph showing step response](image-url)
Step Response (1/2)

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- Each time $B(t) > l_i(1 + S(t))$ a SWITCH_UP is sent
- Video levels are switched up after a delay of around 15s
- Sluggish response due to conservativeness

- Large oscillations due to the greedy/non-greedy phases triggered by rtt-test
- Poor channel utilization

Remarkable oscillations due to intermittent greedy mode
Channel Utilization: 67%

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Step response (2/2)

- Buffer tracks the target with a steady state error
- Each time a rtt-test command is sent throttle is set to 500
- The identified law for the throttle $T(t)$ fits well the measured throttle signal
Feedback-based Adaptive Streaming Algorithm

Player

HTTP traffic

r(t), l(t)

Internet

Controller

selects

l_i \in L

Server

sender buffer

q(t)

Video Levels

l(t)

Decoder

buffer

Controller
The proposed quality adaptation controller (QAC)

Design choices

- Controller is **server-side**, no feedback from the client required
The proposed quality adaptation controller (QAC)

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The proposed quality adaptation controller (QAC)

**Design choices**

- Controller is **server-side**, no feedback from the client required
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The proposed quality adaptation controller (QAC)

Design choices
- Controller is **server-side**, no feedback from the client required
- Control architecture: local feedback using the sender buffer
- The control loop is not distributed thus the system is **delay-free**
- The controller is designed using classic **feedback control**
- Any player can be employed
QAC - The control loop

Goal of the controller: steer the sender buffer at a desired target $q_T(t) > 0$
QAC - The control loop

- **Goal of the controller**: steer the sender buffer at a desired target \( q_T(t) > 0 \)
- The sender buffer is filled at rate \( l(t) \) and drained by the available bandwidth \( b(t) \)
**QAC - The control loop**

- **Goal of the controller:** steer the sender buffer at a desired target $q_T(t) > 0$
- The sender buffer is filled at rate $l(t)$ and drained by the available bandwidth $b(t)$
- $l(t)$ belongs to a discrete set $\mathcal{L} = \{l_0, \ldots, l_4\}$, thus control signal $u(t)$ is quantized

![Control Loop Diagram](image-url)
QAC - The control loop

- **Goal of the controller**: steer the sender buffer at a desired target \( q_T(t) > 0 \)
- The sender buffer is filled at rate \( l(t) \) and drained by the available bandwidth \( b(t) \)
- \( l(t) \) belongs to a discrete set \( \mathcal{L} = \{l_0, \ldots, l_4\} \), thus control signal \( u(t) \) is quantized
- To make the control loop linear, we introduce the equivalent (bounded) disturbance \( d_{eq}(t) = d_q(t) + b(t) \) where \( d_q(t) \) is the mismatch between \( u(t) \) and \( l(t) \)
Goal of the controller: steer the sender buffer at a desired target $q_T(t) > 0$

- The sender buffer is filled at rate $l(t)$ and drained by the available bandwidth $b(t)$
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- To make the control loop linear, we introduce the equivalent (bounded) disturbance $d_{eq}(t) = dq(t) + b(t)$ where $dq(t)$ is the mismatch between $u(t)$ and $l(t)$
- To get zero steady state error and reject step disturbances we employ a PI controller: $G_c(s) = K_p + K_i/s$
Controller tuning and implementation

Closed loop transfer function

$$G_0(s) = \frac{K_p s + K_i}{s^2 + K_p s + K_i}$$

Controller tuning

- **Settling time** $T_s = 30$ s (system bandwidth 0.06Hz in order to reduce video level switches)
- **damping factor** $\delta = 0.707$
- It turns out $K_p = 0.2667$, $K_i = 0.0356$

Implementation

**Discretized control law** with sampling time $\Delta T = 0.5$ s

$$u(t_k) = K_p e(t_k) + K_i \sum_{j=0}^{k} \Delta T e(t_j)$$
From theory to practice: QuavStreams

Implementation

1. Raw video
From theory to practice: QuavStreams

Implementation

① Raw video → ② Multiple Encoding
From theory to practice: QuavStreams

Implementation

1. Raw video → 2. Multiple Encoding → 3. QAC selects video level $l_i$
From theory to practice: QuavStreams

Implementation

1. Raw video → 2. Multiple Encoding → 3. QAC selects video level \( l_i \) → 4. \( i \)-th file version is stored in the producer’s sender queue
From theory to practice: QuavStreams

Implementation

1. Raw video → 2. Multiple Encoding → 3. QAC selects video level $l_i$ →
4. $i$-th file version is stored in the producer’s sender queue →
5. Producer sends the video through HTTP to the client
Experimental comparison: Akamai vs QAC

**Controlled testbed scenario**
- Investigate systems dynamics
- Evaluate network utilization

**High Speed Data Packet Access scenario**
- How the algorithms work in a mobile scenario in terms of video continuity and video quality achieved
- Investigate fairness with concurrent TCP traffic
Adaptive Live Streaming

Feedback control Stream Switching

**Step response**

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**Akamai**
- Transient: 150s
- Ch. utilization: 67%
- Complex dynamics

**QAC**
- Transient: 30s (settling time)
- Ch. utilization: 93%
- Predictable dynamics
Adaptive Live Streaming
Feedback control Stream Switching

Concurrent TCP flow: Akamai vs QAC

Akamai
- Remarkable TCP oscillations
- Ch. utilization: 76%
- Takes 62 s to switch down

QAC
- Typical TCP burstiness
- Ch. utilization: 99%
- Switches between $l_2$ and $l_3$

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Adaptivity in Multimedia Flows
LAAS-Toulouse-16/02/2012
Experimental campaign over HSDPA

Comparing Akamai and QAC over a HSDPA network

- For fair comparison QAC use the same video with the same video levels set of Akamai
- Two considered scenarios: 1) One video flow, 2) one video flow vs one TCP flow
- More than 200 experiments in a two months time-span

Metrics considered for each experiment

- Average goodput $\bar{g}_i$
- Paused time: total time the client is paused due to rebuffering
One video flow

Goodput
Akamai goodput is around 40% less wrt QAC due to its conservativeness (switch up/down safety bands max(\(S\)) = 0.4)

Paused time
QAC 90-th percentile is < 15s, Akamai 90-th percentile is > 100s. Akamai performance is hard to be predicted (high standard deviation)
One video flow vs one TCP flow

Goodput
Akamai does not get a fair share due to on-off greedy phases (TCP is greedy) QAC obtains the fair share since it is always greedy ($q_T > 0$)
One video flow vs one TCP flow

Relative distance from fair share:

\[
\chi(g_{vid}, g_{TCP}) = \begin{cases} 
\frac{g_{vid}}{g_{TCP}} - 1 & g_{vid} \geq g_{TCP} \\
-\frac{g_{TCP}}{g_{vid}} + 1 & g_{vid} < g_{TCP}
\end{cases}
\]

In yellow the “fair region” video or TCP gets at most twice the goodput of the concurrent flow (\(\chi = \pm 1\))

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Adaptive Live Streaming

Feedback control Stream Switching

One video flow vs one TCP flow

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Paused time
QAC median is $< 10$ s, Akamai median is $> 100$ s

Akamai: 19% QAC: 73%

Akamai: 110 s QAC: 9 s

σ_{QAC} = 16 s σ_{Akamai} = 63 s

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L.De Cicco (Poliba-DEE-Bari, Italy) Adaptivity in Multimedia Flows
Conclusions

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- Skype Audio adaptation law is not able to avoid congestion and Akamai Adaptive Video Streaming is based on heuristics that are not able to efficiently adapt to bandwidth variations.
Conclusions

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- Skype Audio adaptation law is not able to avoid congestion and Akamai Adaptive Video Streaming is based on heuristics that are not able to efficiently adapt to bandwidth variations.
- We have proposed a simple adaptation controller (QAC) based on classic control theory which outperforms Akamai Video Streams both in terms of dynamics and QoE.
Ongoing and future research

**CPU overload control for mobile devices**
- design a cascade control architecture to control both bandwidth variations and CPU overload
- propose standardization in the recent WebRTC (Web real time communication) IETF standardization effort

**Cloud-based adaptive live multimedia distribution network**
- develop a model of the overall system (considering all nodes of the distribution tree)
- design of optimal cloud resource management algorithm minimizing costs and maximizing QoE using optimal control framework
Bibliography

Skype


De Cicco, Mascolo, Palmisano, “Skype Video Responsiveness to Bandwidth Variations”, in Proc. of *ACM NOSSDAV ’08*, Germany, May, 2008


Adaptive Video Streaming

