



Rate Adaptation for the IETF IIAC

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WSI-ICS

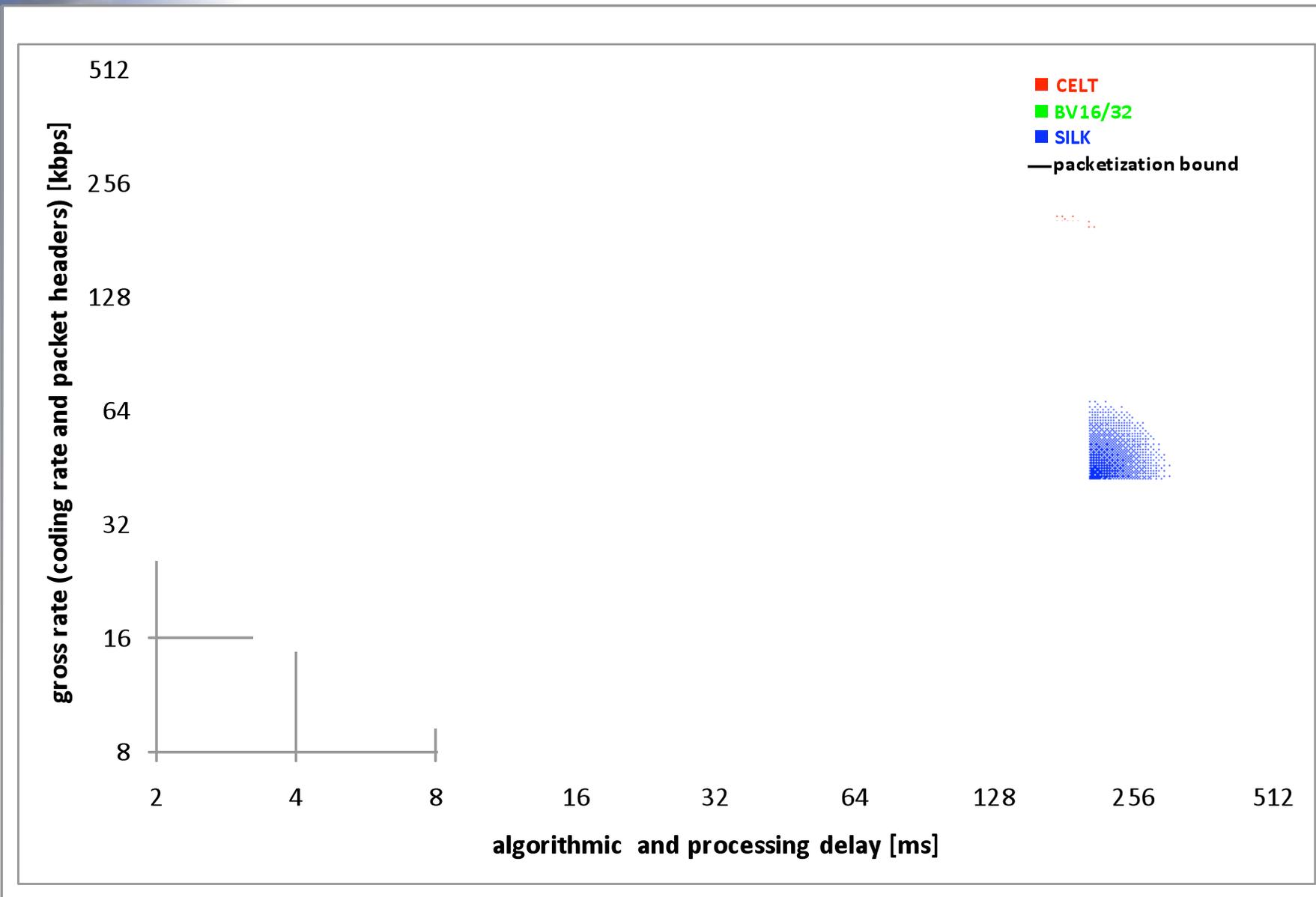
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Problem Statement

- The IIAC is likely to have many parameters:
 1. Coding rate (kbit/s)
 2. Sampling rate (kHz)
 3. Packet length (ms)
 4. DTX/VAD/music/speech mode
 5. Complexity
 6. Look-ahead (ms)
 7. Channels (x)
- The IIAC will have a broad range of operation
 - 8 till $192 \cdot x$ kbit/s
 - 8 till 48 kHz
 - 2 till 160ms delay
- Many different devices
- Many different link qualities on the Internet

Problem: *When to set which codec parameter how?*

Operational Range of Contributed Codecs



Many different platforms (and interests)

		CPU	DSP	RAM	ROM	Example device	OS	Call Capacity
End device	PC	>2 GHz (i386 or x64)	-	> 2 GB	HD	-	Windows, Linux	ca. 100 ?
	Smart-Phone	ARM11, 500 MHz	-	192 MB	256 MB	HTC Dream, MSM7201A	iPhone, Android...	ca. 10?
	VoIP Phone	275-MHz MIPS32 CPU	125-MHz ZSP DSP	>1 MB external	>1 MB external	BCM1103	Linux	2 to 3'?
Gateway	PC based	two Xeon dual core, 2.33 GHz	-	4 GB	HD	Asterisk v1.4.11	Linux	400 calls with G.711 to G.729
	Intel server based	two 4/6 core Xeon	-	12 GB	HD	IVR and conference server	Linux	400 to 10,000
	High density	-	six TI C64x +™ DSP	5,5 MB +external RAM	?	TNETV3020	Telogy Software	AMR 6*216, G.711 6*504
	Spatial Audio	>2 GHz (i386 or x64)	-	> 2 GB	HD	research prototypes	Linux	hardly 1

Goal: Optimize Quality of Experience

- ITU-T P.10/G.100 defines “Quality of Experience”

The overall acceptability of an application or service, as perceived subjectively by the end-user.

- Extension at ITU-T G.RQAM

Quality of experience includes the complete end-to-end system effects (client, terminal, network, services infrastructure, etc.).

Overall acceptability may be influenced by user expectations and context.

Promoting the use of DCCP instead of UDP

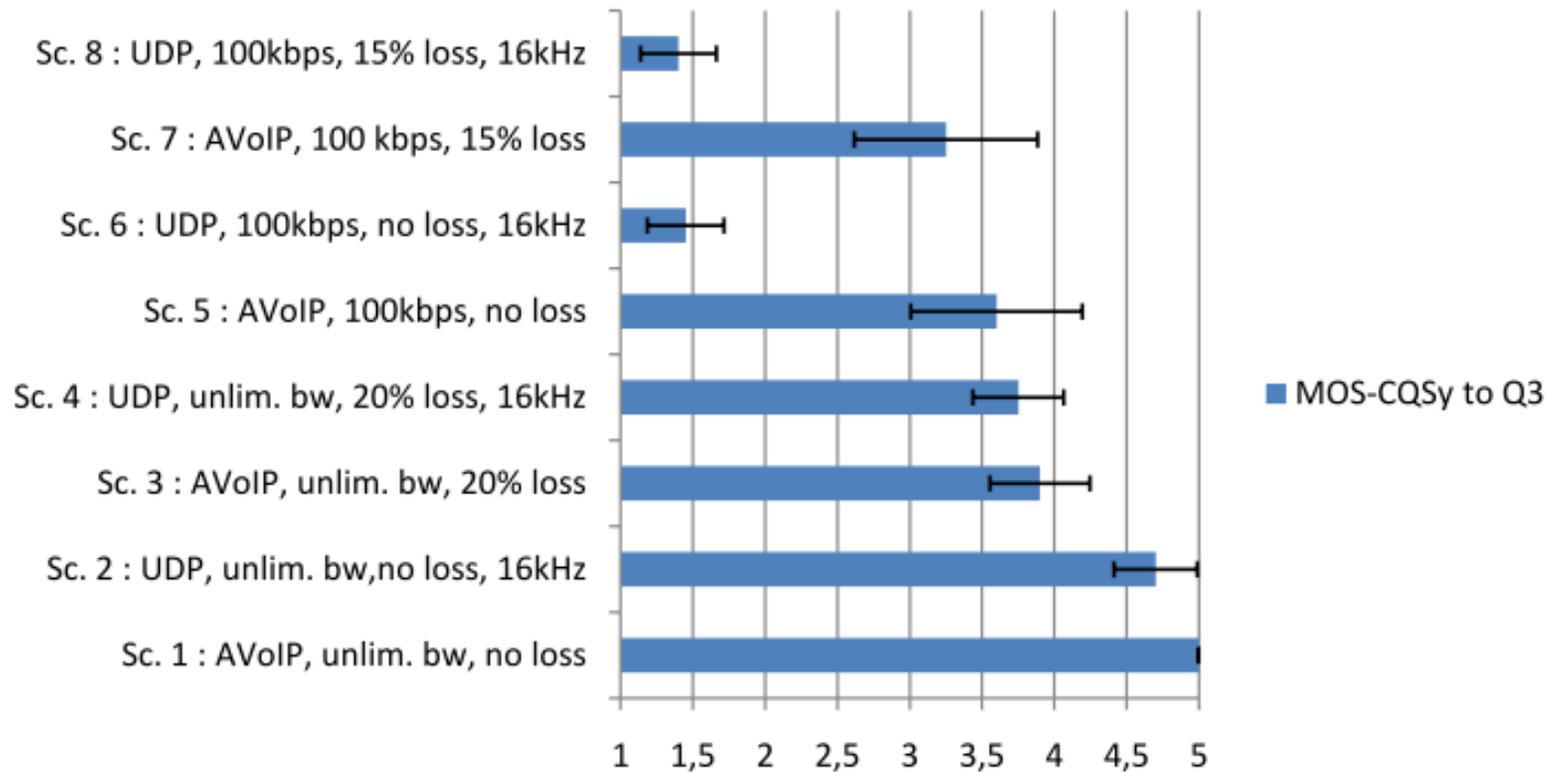
- Offering congestion control and fairness like TCP
 - but fast delivery (no retransmissions)
- Easy application interface
 - API gives you currently available TX rate and RTT
- Implementations available
 - user-space and Linux kernel
- Supports variable packet sizes
 - important for VoIP
- Does DCCP solve all problems?
 - Highly variable bw feedback
 - No feedback on month-to-ear delay
 - which is important for QoE
 - No feedback on computational latency
 - Which important for predicting MtE delay and for low cost devices

Running Code(c)

- Master Thesis of Patrick Schneider
- Implemented DCCP+SBC+PLC
 - (SBC as replacement for a ~~not yet existing~~ IIAC)
- Supporting
 - Rate control without difficulties
(Optimal parameter selection is not yet achievable)
- Switches to Push-To-Talk mode
 - if link speed falls below gross coding rate
- We conducted conversational-tests comparing
 - UDP+packet loss
 - DCCP+Push To Talk

Research Results

Q3: "What level of effort did you need to understand what the other person was telling you? "



- Using SBC mono with 16 to 48 kHz
- Using a network simulator for bw limits and addit. losses
- AVoIP refers to DCCP plus Push to Talk mode

Summary

- IIAC+RTP+DCCP is useful combination
 - make things easier
- but need protocol support for QoE control loop
 - mouth-to-ear delay (when frame have been play out)
 - feedback on complexity (computational delay)
 - in RTP payload, RTCP-XR, or RTP header extensions?
- Vendor specific optimizations on parameter trade-off shall be possible
 - to adapt to different user needs
 - to find an „optimal“ solution in respect to QoE
 - to cope with DCCP's highly variable rate feedback
- Push-To-Talk mode helps
 - for low bandwidth lines
 - also for short handover interruptions