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Self-Clocked Rate Adaptation for Multimedia  
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## Abstract

This memo describes a rate adaptation algorithm for conversational video services. The solution conforms to the packet conservation principle and uses a hybrid loss and delay based congestion control algorithm. The algorithm is evaluated over both simulated Internet bottleneck scenarios as well as in a LTE (Long Term Evolution) system simulator and is shown to achieve both low latency and high video throughput in these scenarios.

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## [1.](#) Introduction

Congestion in the internet is a reality and applications that are deployed in the internet must have congestion control schemes in place not only for the robustness of the service that it provides but

also to ensure the function of the currently deployed internet. As the interactive realtime communication imposes a great deal of requirements on the transport, a robust, efficient rate adaptation for all access types is considered as an important part of interactive realtime communications as the transmission channel

bandwidth may vary over time. Wireless access such as LTE, which is an integral part of the current internet, increases the importance of rate adaptation as the channel bandwidth of a default LTE bearer [[QoS-3GPP](#)] can change considerably in a very short time frame. Thus a rate adaptation solution for interactive realtime media, such as WebRTC, must be both quick and be able to operate over a large span in available channel bandwidth. This memo describes a solution, named SCReAM, that is based on the self-clocking principle of TCP and uses techniques similar to what is used in a new delay based rate adaptation algorithm, LEDBAT [[RFC6817](#)]. Because neither TCP nor LEDBAT was designed for interactive realtime media, a few extra features are needed to make the concept work well within this context. This memo describes these extra features.

### 1.1. Wireless (LTE) access properties

[I-D.[draft-sarker-rmcat-cellular-eval-test-cases](#)] introduces the complications that can be observed in wireless environments. Wireless access such as LTE can typically not guarantee a given bandwidth, this is true especially for default bearers. The network throughput may vary considerably for instance in cases where the wireless terminal is moving around.

Unlike wireline bottlenecks with large statistical multiplexing it is not possible to try to maintain a given bitrate when congestion is detected with the hope that other flows will yield, this because there are generally few other flows competing for the same bottleneck. Each user gets its own variable throughput bottleneck, where the throughput depends on factors like channel quality, network load and historical throughput. The bottom line is, if the throughput drops, the sender has no other option than to reduce the bitrate. In addition, the grace time, i.e. allowed reaction time from the time that the congestion is detected until a reaction in terms of a rate reduction is effected, is generally very short, in the order of one RTT (Round Trip Time).

## [2.](#) Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119](#) [[RFC2119](#)]

## [3.](#) Overview of SCReAM Algorithm

The core SCReAM algorithm has similarities to concepts like self-clocking used in TFWC [[TFWC](#)] and follows packet conservation principles. The packet conservation principle is described as an

important key-factor behind the protection of networks from congestion [[FACK](#)].

The packet conservation principle is realized by including an indication of the highest received sequence number in the feedback, see [Section 5](#), from the receiver back to the sender, the sender keeps a list of transmitted packets and their respective sizes. This information is then used to determine how many bytes can be transmitted. A congestion window puts an upper limit on how many bytes can be in flight, i.e. transmitted but not yet acknowledged. The congestion window is determined in a way similar to LEDBAT [[RFC6817](#)]. This ensures that the e2e latency is kept low. The basic functionality is quite simple, there are however a few steps to take to make the concept work with conversational media. These will be briefly described in sections [Section 3.1](#) to [Section 3.3](#).

The rate adaptation solution constitutes three parts- congestion control, transmission scheduling and media rate adaptation. All these three parts reside at the sender side. The receiver side algorithm is very simple in comparison as it only generates acknowledgements to received RTP packets.

### [3.1.](#) Congestion Control

The congestion control sets an upper limit on how much data can be in the network (bytes in flight); this limit is called CWND (congestion window) and is used in the transmission scheduling.

The SCReAM congestion control method, uses LEDBAT [[RFC6817](#)] to

measure the OWD (one way delay). The SReAM sender calculates the congestion window based on the feedback from SReAM receiver. The congestion window is allowed to increase if the OWD is below a predefined target, otherwise the congestion window decreases. The delay target is typically set to 50-100ms. This ensures that the OWD is kept low on the average. The reaction to loss events is similar to that of loss based TCP, i.e. an instant reduction of CWND.

LEDBAT is designed with file transfers as main use case which means that the algorithm must be modified somewhat to work with rate-limited sources such as video. The modifications are

- o Congestion window validation techniques. These are similar in action as the method described in [[I-D.ietf-tcpm-newcwnd](#)].
- o Fast start for bitrate increase. It makes the video bitrate ramp-up within 5 to 10 seconds. The behavior is similar to TCP slowstart. The fast start is exited when congestion is detected. The fast start state can be resumed if the congestion level is

low, this to enable a reasonably quick rate increase in case link throughput increases.

- o Adaptive delay target. This helps the congestion control to compete with FTP traffic to some degree.

### [3.2.](#) Transmission Scheduling

Transmission scheduling limits the output of data, given by the relation between the number of bytes in flight and the congestion window similar to TCP. Packet pacing is used to mitigate issues with coalescing that may cause increased jitter and/or packet loss in the media traffic.

### [3.3.](#) Media Rate Control

The media rate control serves to adjust the media bitrate to ramp up quickly enough to get a fair share of the system resources when link throughput increases.

The reaction to reduced throughput must be prompt in order to avoid getting too much data queued up in the RTP packet queues. The media



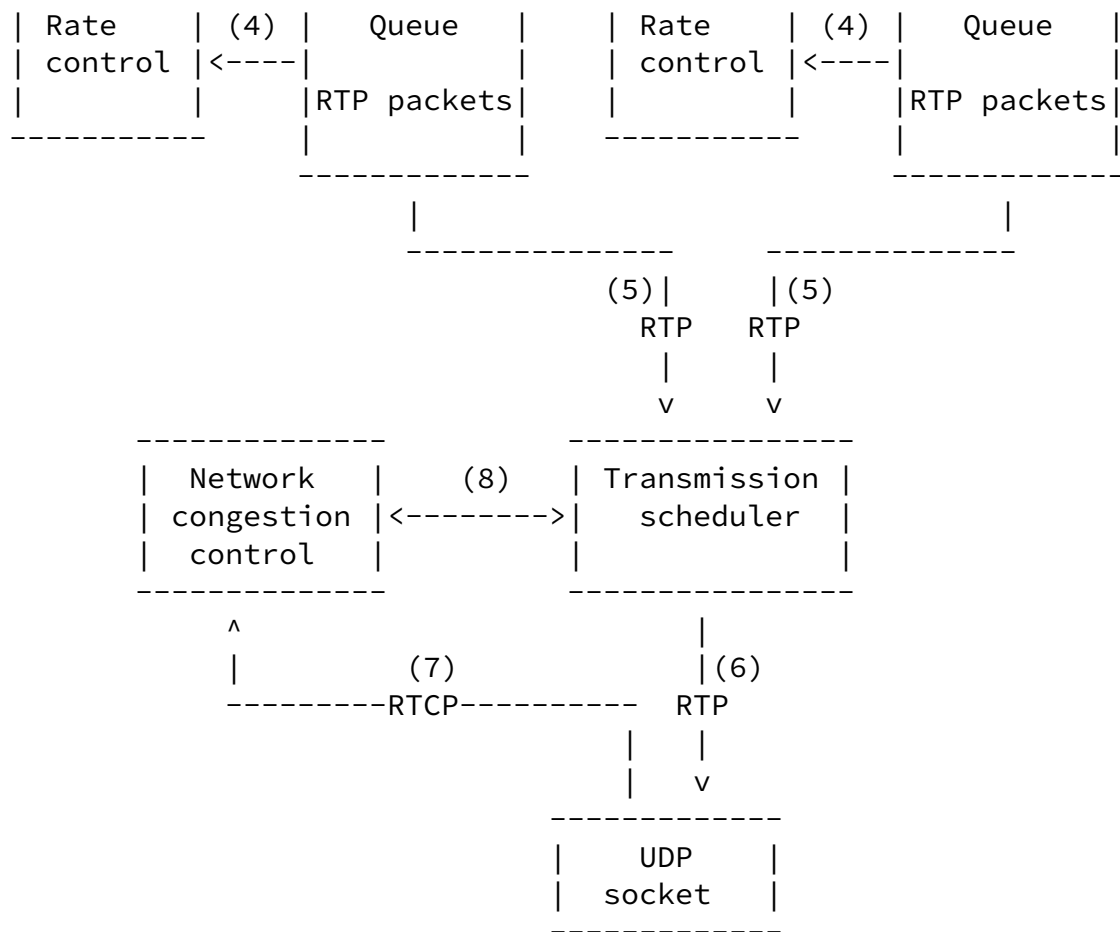


Figure 1: SCReAM sender functional view

Video frames are encoded and forwarded to the queue (2). The media rate adaptation adapts to the size of the RTP queue and controls the video bitrate (1). The RTP packets are picked from each queue based on some defined priority order or simply in a round robin fashion (5). A transmission scheduler takes care of the transmission of RTP

packets, to be written to the UDP socket (6). In the general case all media must go through the transmission scheduler and is allowed to be transmitted if the number of bytes in flight is less than the congestion window. Audio frames can however be allowed to be transmitted immediately as audio is typically low bitrate and thus contributes little to congestion, this is something that is left as an implementation choice. RTCP packets are received (7) and the information about bytes in flight and congestion window is exchanged

between the network congestion control and the transmission scheduler (8).

#### [4.1.1.](#) Constants and Parameter values

A set of constants are defined in Table 1, state variables are defined in Table 2. And finally, local variables are described in Table 3.

An init value [] indicates an empty array.



Constant	Explanation	Value
OWD_TARGET_LO	Min OWD target	0.1s
OWD_TARGET_HI	Max OWD target	0.4s
MAX_BYTES_IN_FLIGHT_HEAD_ROOM	Headroom for limitation of CWND	1.1
GAIN	Gain factor for congestion window adjustment	1.0
BETA	CWND scale factor due to loss event	0.6
BETA_R	Target rate scale factor due to loss event	0.8
BYTES_IN_FLIGHT_SLACK	Additional slack [%] to the congestion window	10%
RATE_ADJUST_INTERVAL	Interval between video bitrate adjustments	0.1s
FRAME_PERIOD	Video coder frame period [s]	
TARGET_BITRATE_MIN	Min target_bitrate [bps]	
TARGET_BITRATE_MAX	Max target_bitrate [bps]	
RAMP_UP_TIME	Timespan [s] from lowest to highest bitrate	10s
PRE_CONGESTION_GUARD	Guard factor against early congestion onset. A higher value gives less jitter possibly at the expense of a lower video bitrate.	0.0..0.2
TX_QUEUE_SIZE_FACTOR	Guard factor against RTP queue buildup	0.0..2.0

Table 1: Constants

Variable	Explanation	Init value
owd_target	OWD target	OWD_TARGET_LO
owd_fraction_avg	EWMA filtered owd_fraction	0.0
owd_fraction_hist	Vector of the last 20 owd_fraction	[]
owd_trend	OWD trend, indicates incipient congestion	0.0
owd_norm_hist	Vector of the last 100 owd_norm	[]
mss	Maximum segment size = Max RTP packet size [byte]	1000
min_cwnd	Minimum congestion window [byte]	2*MSS
in_fast_start	True if in fast start state	true
cwnd	Congestion window [byte]	min_cwnd
cwnd_i	Congestion window inflection point	1
bytes_newly_acked	The number of bytes that was acknowledged with the last received acknowledgement i.e. bytes acknowledged since the last CWND update [byte]. Reset after a CWND update	0
send_wnd	Upper limit of how many bytes that can be transmitted [byte]. Updated when CWND is updated and when RTP packet is transmitted	0
t_pace	Approximate estimate of inter-	0.001

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	interval [s], updated when RTP packet transmitted	
age_vec	A vector of the last 20 RTP packet queue delay samples	[]
frame_skip_intensity	Indicates the intensity of the frame skips	0.0
since_last_frame_skip	Number of video frames since the last skip	0
consecutive_frame_skips	Number of consecutive frame skips	0
target_bitrate	Video target bitrate [bps]	TARGET_BITRATE_MIN
target_bitrate_i	Video target bitrate inflection point i.e. the last known highest target_bitrate during fast start. Used to limit bitrate increase close to the last known congestion point	1
rate_transmit	Measured transmit bitrate [bps]	0.0
rate_acked	Measured throughput based on received acknowledgements [bps]	0.0
s_rtt	Smoothed RTT [s], computed similar to method depicted	0.0

rtp_queue_size	in [RFC6298] Size of RTP packets in queue [bits]	0
rtp_size	Size of the last transmitted RTP packets [byte]	0
frame_skip	Skip encoding of video frame if	false

	true	
--	------	--

Table 2: State variables

Variable	Explanation
owd	OWD = One way delay with base delay subtracted [s]. This is an estimate of the network queueing delay.
owd_fraction	OWD as a fraction of the OWD target
owd_norm	OWD normalized to OWD_TARGET_LO
owd_norm_mean	Average OWD norm over the last 100 samples
owd_norm_mean_sh	Average OWD norm over the last 20 samples
owd_norm_var	OWD norm variance over the last 100 samples
off_target	Relation between OWD and OWD target
scl_i	A general scalefactor that is applied to the CWND or target_bitrate increase
x_cwnd	Additional increase of CWND, used when send_wnd is computed
pace_bitrate	The allowed RTP packet transmission rate, used in the computation of t_pace [bps]
age_avg	Average RTP queue delay [s]
increment	Allowed target_bitrate increase
current_rate	Max of rate_transmit and rate_acked

Table 3: Local temporary variables

#### [4.1.2.](#) Network congestion control

This section explains the network congestion control, it contains two main functions

- o Computation of congestion window at the sender: Gives an upper limit to the number of bytes in flight i.e. how many bytes that have been transmitted but not yet acknowledged.
- o Transmission scheduling at the sender: RTP packets are transmitted if allowed by the relation between the number of bytes in flight and the congestion window. This is controlled by the send window.

Unlike TCP, SReAM is not a byte oriented protocol, rather it is an RTP packet oriented protocol. Thus it keeps a list of transmitted RTP packets and their respective sending times (wall-clock time). The feedback indicates the highest received RTP sequence number and a

timestamp (wall-clock time) when it was received. In addition, an ACK list is included to make it possible to determine lost packets.

#### 4.1.2.1. Congestion window update

The congestion window is computed from the one way (extra) delay estimates (OWD) that are obtained from the send and received timestamp of the RTP packets. LEDBAT [[RFC6817](#)] explains the details of the computation of the OWD. An OWD sample is obtained for each received acknowledgement. No smoothing of the OWD samples occur, however some smoothing occurs anyway as the computation of the CWND is in itself a low pass filter function.

SReAM uses the terminology "Bytes in flight (bytes\_in\_flight)" which is computed as the sum of the sizes of the RTP packets ranging from the RTP packet most recently transmitted down to but not including the acknowledged packet with the highest sequence number. As an example: If RTP packet was sequence number SN with transmitted and the last ACK indicated SN-5 as the highest received sequence number then bytes in flight is computed as the sum of the size of RTP packets with sequence number SN-4, SN-3, SN-2, SN-1 and SN.

CWND is updated differently depending on whether the congestion control is in fast start or not and if a loss event is detected. A Boolean variable in\_fast\_start indicates if the congestion is in fast

start state.

A loss event indicates one or more lost RTP packets within an RTT. This is detected by means of inspection for holes in the sequence number space in the acknowledgements with some margin for possible packet reordering in the network. As an alternative, a timer for loss detection similar to TCP RACK may be used.

Below is described the actions when an acknowledgement from the receiver is received.

bytes\_newly\_acked is updated.

The OWD fraction and an average of it are computed as

$$\text{owd\_fraction} = \text{owd}/\text{owd\_target}$$
$$\text{owd\_fraction\_avg} = 0.9 * \text{owd\_fraction\_avg} + 0.1 * \text{owd\_fraction}$$

The OWD fraction is sampled every 50ms and the last 20 samples are stored in a vector (owd\_fraction\_hist). This vector is used in the computation of an OWD trend that gives a value between 0.0 and 1.0

depending on how close to congestion it is. The OWD trend is calculated as follows

Let  $R(\text{owd\_fraction\_hist}, K)$  be the autocorrelation function of `owd_fraction_hist` at lag  $K$ . The 1st order prediction coefficient is formulated as

$$a = R(\text{owd\_fraction\_hist}, 1) / R(\text{owd\_fraction\_hist}, 0)$$

The prediction coefficient  $a$  has positive values if OWD shows an increasing trend, thus an indication of congestion is obtained before the OWD target is reached. The prediction coefficient is further multiplied with `owd_fraction_avg` to reduce sensitivity to increasing OWD when OWD is very small. The OWD trend is thus computed as

$$\text{owd\_trend} = \max(0.0, \min(1.0, a * \text{owd\_fraction\_avg}))$$

The `owd_trend` is utilized in the media rate control and to determine

when to exit slow start.

An off target value is computed as

$$\text{off\_target} = (\text{owd\_target} - \text{owd}) / \text{owd\_target}$$

A temporal variable is  $scl\_i$  is computed as

$$scl\_i = \max(0.2, \min(1.0, (\text{abs}(\text{cwnd} - \text{cwnd}_i) / \text{cwnd}_i * 4)^2))$$

$scl\_i$  is used to limit the CWND increase when close to the last known max value, before congestion was last detected.

The congestion window update depends on whether a loss event has occurred, and if the congestion control is if fast start or not.

-----

On loss event:

If a loss event is detected then  $\text{in\_fast\_start}$  is set to false and CWND is updated according to

$$\text{cwnd}_i = \text{cwnd}$$
$$\text{cwnd} = \max(\text{min\_cwnd}, \text{cwnd} * \text{BETA})$$

otherwise the CWND update continues

-----

$\text{in\_fast\_start} = \text{true}$ :

$\text{in\_fast\_start}$  is set to false and  $\text{cwnd}_i = \text{cwnd}$  if  $\text{owd\_trend} \geq 0.2$  and otherwise CWND is updated according to

$$\text{cwnd} = \text{cwnd} + \text{bytes\_newly\_acked} * scl\_i$$

-----

in\_fast\_start = false:

Values of `off_target > 0.0` indicates that the congestion window can be increased. This is done according to the equations below.

```
gain = GAIN*(1.0 + max(0.0, 1.0 - owd_trend/ 0.2))
```

The equation above limits the gain when near congestion is detected

```
gain *= scl_i
```

This equation limits the gain when CWND is close to its last known max value

```
cwnd += gain * off_target * bytes_newly_acked * mss / cwnd
```

Values of `off_target <= 0.0` indicates congestion, CWND is then updated according to the equation

```
cwnd += GAIN*off_target*bytes_newly_acked*mss/cwnd
```

The equations above are very similar to what is specified in [\[RFC6817\]](#). There are however a few differences.

- o [\[RFC6817\]](#) specifies a constant GAIN, this specification however limits the gain when CWND is increased dependent on near congestion state and the relation to the last known max CWND value.
- o [\[RFC6817\]](#) specifies that the CWND increased is limited by an additional function controlled by a constant ALLOWED\_INCREASE. This additional limitation is removed in this specification.

---

A number of final steps in the congestion window update procedure are outlined below

---

Resume fast start:



Fast start can be resumed in order to speed up the bitrate increase in case congestion abates. The condition to resume fast start (`in_fast_start = true`) is that `owd_trend` is less than 0.2 for 1.0 seconds or more.

---

Competing flows compensation, adjustment of `owd_target`:

Competing flows compensation is needed to avoid that flows congestion controlled by SCReAM are starved out by flows that are more aggressive in their nature. The `owd_target` is adjusted according to the `owd_norm_mean_sh` whenever `owd_norm_var` is below a given value. The condition to update `owd_target` is fulfilled if `owd_norm_var < 0.16` (indicating that the standard deviation is less than 0.4). `owd_target` is then update as:

```
owd_target = min(OWD_TARGET_HI,max(OWD_TARGET_LO, owd_norm_mean_sh*
OWD_TARGET_LO*1.1))
```

---

Final CWND adjustment step:

The congestion window is limited by the maximum number of bytes in flight over the last 1.0 seconds according to

```
cwnd = min(cwnd, max_bytes_in_flight*MAX_BYTES_IN_FLIGHT_HEAD_ROOM)
```

This avoids possible over-estimation of the throughput after for example, idle periods.

Finally `cwnd` is set to ensure that it is at least `min_cwnd`

```
cwnd = max(cwnd, MIN_CWND)
```

#### [4.1.2.2.](#) Transmission scheduling

The principle is to allow packet transmission of an RTP packet only if the number of bytes in flight is less than the congestion window. There are however two reasons why this strict rule will not work optimally:

- o Bitrate variations: The video frame size is always varying to a larger or smaller extent, a strict rule as the one given above will have the effect that the video bitrate have difficulties to increase as the congestion window puts a too hard restriction on the video frame size variation, this further can lead to occasional queuing of RTP packets in the RTP packet queue that will prevent bitrate increase because of the increased RTP queue size.
- o Reverse (feedback) path congestion: Especially in transport over buffer-bloated networks, the one way delay in the reverse direction may jump due to congestion. The effect of this is that the acknowledgements are delayed with the result that the self-clocking is temporarily halted, even though the forward path is not congested.

Packets are transmitted at a pace given by the send window, computed below

The send window is computed differently depending on OWD and its relation to the OWD target.

- o If  $owd > owd\_target$ :  
The send window is computed as  
 $send\_wnd = cwnd - bytes\_in\_flight$   
This enforces a strict rule that helps to prevent further queue buildup.
- o If  $owd \leq owd\_target$ :  
A helper variable  
 $x\_cwnd = 1.0 + BYTES\_IN\_FLIGHT\_SLACK * \max(0.0, \min(1.0, 1.0 - owd\_trend / 0.5)) / 100.0$   
is computed. The send window is computed as  
 $send\_wnd = \max(cwnd * x\_cwnd, cwnd + mss) - bytes\_in\_flight$   
This gives a slack that reduces as congestion increases, `BYTES_IN_FLIGHT_SLACK` is a maximum allowed slack in percent. A large value increases the robustness to bitrate variations in the source and congested feedback channel issues. The possible drawback is increased delay or packet loss when forward path congestion occur.

#### [4.1.3.](#) Video rate control

The video rate control is operated based on the size of the RTP packet send queue and observed loss events. In addition, `owd_trend` is also considered in the rate control, this to reduce the amount of induced network jitter.

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A variable `target_bitrate` is adjusted depending on the congestion state. The target bitrate can vary between a minimum value (`target_bitrate_min`) and a maximum value (`target_bitrate_max`).

For the overall bitrate adjustment, two network throughput estimates are computed :

- o `rate_transmit`: The measured transmit bitrate
- o `rate_acked`: The ACKed bitrate, i.e. the volume of ACKed bits per time unit.

Both estimates are updated every 200ms.

The current throughput `current_rate` is computed as the maximum value of `rate_transmit` and `rate_acked`. The rationale behind the use of `rate_acked` in addition to `rate_transmit` is that `rate_transmit` is affected also by the amount of data that is available to transmit, thus a lack of data to transmit can be seen as reduced throughput that may itself cause an unnecessary rate reduction. To overcome this shortcoming; `rate_acked` is used as well. This gives a more stable throughput estimate.

The bitrate is updated at regular intervals, given by `RATE_ADJUST_INTERVAL` and differently depending the fast start state

The rate change behavior depends on whether a loss event has occurred, and if the congestion control is if fast start or not.

---

On loss event:

First of all the `target_bitrate` is updated if a new loss event was indicated and the rate change procedure is exited.

```
target_bitrate_i = target_bitrate
```

```
target_bitrate = max(BETA_R* target_bitrate, TARGET_BITRATE_MIN)
```

If no loss event was indicated then the rate change procedure continues.

-----  
in\_fast\_start = true:

An allowed increment is computed based on the congestion level and the relation to target\_bitrate\_i

$scl\_i = (target\_bitrate - target\_bitrate\_i) / target\_bitrate\_i$

$increment = TARGET\_BITRATE\_MAX * RATE\_ADJUST\_INTERVAL / RAMP\_UP\_TIME * (1.0 - \min(1.0, owd\_trend / 0.1))$

$increment *= \max(0.2, \min(1.0, (scl\_i * 4)^2))$

target\_bitrate += increment

target\_bitrate is reduced further if congestion is detected.

$target\_bitrate *= (1.0 - PRE\_CONGESTION\_GUARD * owd\_trend)$

target\_bitrate =  
 $\min(TARGET\_BITRATE\_MAX, \max(TARGET\_BITRATE\_MIN, target\_bitrate))$

-----  
in\_fast\_start = false:

target\_bitrate\_i is updated to the current value of target\_bitrate if in\_fast\_start was true the last time the bitrate was updated.

A pre-congestion indicator is computed as

$pre\_congestion = \min(1.0, \max(0.0, owd\_fraction\_avg - 0.3) / 0.7)$

```
pre_congestion += owd_trend
```

The target bitrate is computed as

```
target_bitrate=current_rate*(1.0-  
PRE_CONGESTION_GUARD*pre_congestion)-TX_QUEUE_SIZE_FACTOR  
*rtp_queue_size
```

```
target_bitrate =  
min(TARGET_BITRATE_MAX,max(TARGET_BITRATE_MIN,target_bitrate))
```

#### [4.2.](#) SCReAM Receiver

The SCReAM receiver is very simple in its implementation. The task is to feedback acknowledgements of received packets. For that purpose a set of state variables are needed, these are explained in Table 4.

One set of state variables are maintained per stream.

Variable	Explanation	Init value
rx_timestamp	The wall clock timestamp when the latest RTP packet was received	0
highest_rtp_sequence_number	The highest received sequence number	0
ack_vector	A 16 bit vector that indicates received RTP packets with a sequence number lower than highest_rtp_sequence_number	0
n_loss	An 8 bit counter for the number of lost RTP packets, separate counters are	0

n_ECN	maintained for each SSRC An 8 bit counter for the number of ECN-CE marked RTP packets, separate counters are maintained for each SSRC	0
pending_feedback	Indicates that an RTP packet was received and that an RTCP packet can be generated when RTCP timing rules permit	false
last_transmit_t	Last time an RTCP packet was transmitted, this is used to ensure that RTCP feedback is generated fairly for all streams.	-1.0

Table 4: State variables

Upon reception of an RTP packet, the state variables in Table 4 should be updated and the RTCP processing function should be

notified. An RTCP packet is later generated based on the state variables, how often this is done depends on the RTCP bandwidth.

## 5. Feedback Message

The feedback is over RTCP [[RFC3550](#)] and is based on [[RFC4585](#)]. It is implemented as a transport layer feedback message (RTPFB), see proposed example in Figure 2. The feedback control information part (FCI) consists of the following elements.

- o Highest received RTP sequence number : The highest received RTP sequence number for the given SSRC
- o n\_lost : Accumulated number of lost RTP packets for the given SSRC
- o Timestamp: A timestamp value indicating when the last packet was received which makes it possible to compute the one way (extra) delay (OWD).

- o n\_ECN : Accumulated number of ECN-CE marked RTP packets for the given SSRC
- o Source quench bit (Q): Makes it possible to request the sender to reduce its congestion window. This is useful if WebRTC media is received from many hosts and it becomes necessary to balance the bitrates between the streams.

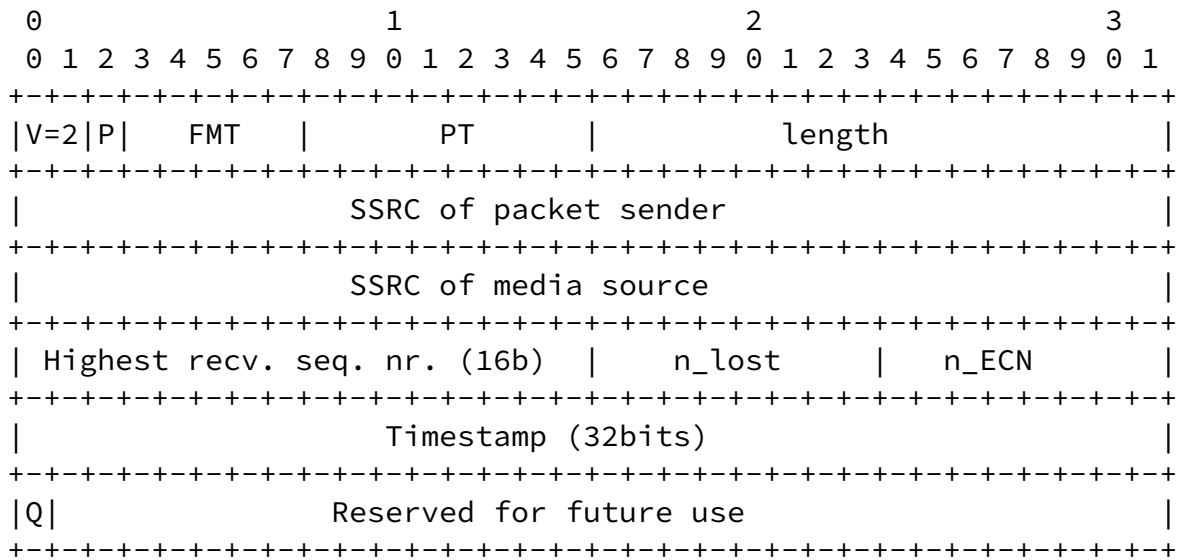


Figure 2: Transport layer feedback message

To make the feedback as frequent as possible, the feedback packets are transmitted as reduced size RTCP according to [\[RFC5506\]](#).

The timestamp clock time is recommended to be set to a fixed value such as 1000Hz, defined in this specification. The n\_lost and n\_ECN makes it possible to take necessary actions on the detection of lost and ECN marked packets.

[Section 4](#) describes the main algorithm details and how the feedback is used.

## 6. Additional features

This section describes additional features. They are not required for the basic functionality of SCReAM but can improve performance in

certain scenarios and topologies.

### [6.1.](#) Packet pacing

Packet pacing is used in order to mitigate coalescing i.e. that packets are transmitted in bursts.

Packet pacing is enforced when `owd_fraction_avg` is greater than 0.1. The time interval between consecutive packet transmissions is then enforced to equal or higher than `t_pace` where `t_pace` is given by the equations below.

$$\text{pace\_bitrate} = \max(50000, \text{cwnd} * 8 / \text{s\_rtt})$$
$$\text{t\_pace} = \text{rtp\_size} * 8 / \text{pace\_bitrate}$$

`rtp_size` is the size of the last transmitted RTP packet

### [6.2.](#) Frame skipping

Frame skipping is a feature that makes it possible to reduce the size of the RTP queue in the cases that e.g. the channel throughput drops dramatically or even goes below the lowest possible video coder rate. Frame skipping is optional to implement as it can sometimes be difficult to realize e.g. due to lack of API function to support this.

Frame skipping is controlled by a flag `frame_skip` which, if set to 1 dictates that the video coder should skip the next video frame. The frame skipping intensity at the current time instant is computed according to the steps below

The queuing delay is sampled every frame period and the last 20 samples are stored in a vector `age_vec`

An average queuing delay is computed as a weighted sum over the samples in `age_vec`. `age_avg` at the current time instant is computed as

$$\text{age\_avg}(n) = \text{SUM}_{k = [0..20[} \text{age\_vec}(n-k) * w(k)$$



$w(n)$  are weight factors arranged to give the most recent samples a higher weight.

The change in `age_avg` is computed as

$$\text{age\_d} = \text{age\_avg}(n) - \text{age\_avg}(n-1)$$

The frame skipping intensity at the current time instant  $n$  is computed as

- o If `age_d > 0` and `age_avg > 2*FRAME_PERIOD`:  
`frame_skip_intensity = min(1.0, (age_vec(n)-2*FRAME_PERIOD)/(4*FRAME_PERIOD)`
- o Otherwise frame skip intensity is set to zero

The `skip_frame` flag is set depending on three variables

- o `frame_skip_intensity`
- o `since_last_frame_skip`, i.e the number of consecutive frames without frame skipping
- o `consecutive_frame_skips`, i.e the number of consecutive frame skips

The flag `skip_frame` is set to 1 if any of the conditions below is met, otherwise it is set to 0.

- o `age_vec(n) > 0.2 && consecutive_frame_skips < 5`
- o `frame_skip_intensity < 0.5 && since_last_frame_skip >= 1.0/frame_skip_intensity`
- o `frame_skip_intensity >= 0.5 && consecutive_frame_skips < (frame_skip_intensity -0.5)*10`

The arrangement makes sure that no more than 4 frames are skipped in sequence, the rationale is to ensure that the input to the video encoder does not change too much, something that may give poor prediction gain.

### [6.3.](#) Q-bit semantics (source quench)

The Q bit in the feedback is set by a receiver to signal that the sender should reduce the bitrate. The sender will in response to this reduce the congestion window with the consequence that the video bitrate decreases. A typical use case for source quench is when a receiver receives streams from sources located at different hosts and they all share a common bottleneck, typically it is difficult to apply any rate distribution signaling between the sending hosts. The solution is then that the receiver sets the Q bit in the feedback to the sender that should reduce its rate, if the streams share a common bottleneck then the released bandwidth due to the reduction of the congestion window for the flow that had the Q bit set in the feedback will be grabbed by the other flows that did not have the Q bit set. This is ensured by the opportunistic behavior of SCReAM's congestion control. The source quench will have no or little effect if the flows do not share the same bottleneck.

The reduction in congestion window is proportional to the amount of SCReAM RTCP feedback with the Q bit set, the below steps outline how the sender should react to RTCP feedback with the Q bit set. The reduction is done once per RTT. Let :

- o  $n$  = Number of received RTCP feedback messages in one RTT
- o  $n_q$  = Number of received RTCP feedback messages in one RTT, with Q bit set.

The new congestion window is then expressed as:

$$cwnd = \max(\text{MIN\_CWND}, cwnd * (1.0 - 0.5 * n_q / n))$$

Note that CWND is adjusted at most once per RTT. Furthermore The CWND increase should be inhibited for one RTT if CWND has been decreased as a result of Q bits set in the feedback.

The required intensity of the Q-bit set in the feedback in order to achieve a given rate distribution depends on many factors such as RTT, video source material etc. The receiver thus need to monitor the change in the received video bitrate on the different streams and adjust the intensity of the Q-bit accordingly.

## [7.](#) Discussion

This section covers a few open discussion points

- o RTCP feedback overhead: SCReAM benefits from a relatively frequent feedback. Experiments have shown that a feedback rate roughly

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equal to the frame rate gives a stable self-clocking and robustness against loss of feedback. With a maximum bitrate of 1500kbps the RTCP feedback overhead is in the range 10-15kbps with reduced size RTCP, including IP and UDP framing, in other words the RTCP overhead is quite modest and should not pose a problem in the general case. Other solutions may be required in highly asymmetrical link capacity cases. Worth notice is that SCReAM can work with as low feedback rates as once every 200ms, this however comes with a higher sensitivity to loss of feedback and also a potential reduction in throughput.

- o AVPF mode: The RTCP feedback is based on AVPF regular mode. The SCReAM feedback is transmitted as reduced size RTCP so save overhead, it is however required to transmit full compound RTCP at regular intervals, this interval can be controlled by trr-int depicted in [[RFC4585](#)].
- o BETA, CWND scale factor due to loss: The BETA value is recommended to be higher than 0.5. The reason behind this is that congestion control for multimedia has to deal with a source that is rate limited. A file transfer has "unlimited" source bitrate in comparison. The outcome is that SCReAM must be a little more aggressive than a file transfer in order to not be out competed.

## [8.](#) Conclusion

This memo describes a congestion control algorithm for RMCAT that it is particularly good at handling the quickly changing condition in wireless network such as LTE. The solution conforms to the packet conservation principle and leverages on novel congestion control algorithms and recent TCP research, together with media bitrate determined by sender queuing delay and given delay thresholds. The solution has shown potential to meet the goals of high link utilization and prompt reaction to congestion. The solution is realized with a new [RFC4585](#) transport layer feedback message.

## [9.](#) Open issues

A list of open issues.

- o Describe how clock drift compensation is done

- o Describe how FEC overhead is accounted for in `target_bitrate` computation
- o Investigate the impact of more sparse RTCP feedback, for instance once per RTT

## 10. Source code

Source code for SCReAM is available in two formats :

- o C++ code, that is apt for experimentation. The code maintained as Visual Studio project. This code can possibly be included in simulators such as NS3. Available at <https://github.com/EricssonResearch/scream>
- o OpenWebRTC implementation : Work in progress, see <http://www.openwebrtc.io/> for information about the OpenWebRTC project

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## 12. IANA Considerations

A new [RFC4585](#) transport layer feedback message needs to be standardized.

## 13. Security Considerations

The feedback can be vulnerable to attacks similar to those that can affect TCP. It is therefore recommended that the RTCP feedback is at least integrity protected.

## 14. Change history

A list of changes:

- o -04 to -05 : ACK vector is replaced by a loss counter, PT is removed from feedback, references to source code added
- o -03 to -04 : Extensive changes due to review comments, code somewhat modified, frame skipping made optional
- o -02 to -03 : Added algorithm description with equations, removed pseudo code and simulation results
- o -01 to -02 : Updated GCC simulation results

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- o -00 to -01 : Fixed a few bugs in example code

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