**3GPP TSG-SA4 Meeting #131 *S4-250133***

**Geneva, , 17th Feb 2025 - 21st Feb 2025**

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| *CR-Form-v12.3* |
| **CHANGE REQUEST** |
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|  | **26.822** | **CR** | **0001** | **rev** | **2** | **Current version:** | **19.0.0** |  |
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| *For* [***HE******LP***](http://www.3gpp.org/3G_Specs/CRs.htm#_blank)*on using this form: comprehensive instructions can be found at* [*http://www.3gpp.org/Change-Requests*](http://www.3gpp.org/Change-Requests)*.* |
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| ***Proposed change affects:*** | UICC apps |  | ME | **x** | Radio Access Network |  | Core Network | **x** |

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| ***Title:***  | Additional Evaluation of Time To Next Burst |
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| ***Source to WG:*** | Huawei, Hisilicon, Nokia |
| ***Source to TSG:*** | SA WG 4 |
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| ***Work item code:*** | FS\_5G\_RTP\_PH2, 5G\_RTP\_PH2 |  | ***Date:*** | 2025-02-05 |
|  |  |  |  |  |
| ***Category:*** | **F** |  | ***Release:*** | Rel-19 |
|  | *Use one of the following categories:****F*** *(correction)****A*** *(mirror corresponding to a change in an earlier release)****B*** *(addition of feature),* ***C*** *(functional modification of feature)****D*** *(editorial modification)*Detailed explanations of the above categories canbe found in 3GPP [TR 21.900](http://www.3gpp.org/ftp/Specs/html-info/21900.htm). | *Use one of the following releases:Rel-8 (Release 8)Rel-9 (Release 9)Rel-10 (Release 10)Rel-11 (Release 11)…Rel-17 (Release 17)Rel-18 (Release 18)Rel-19 (Release 19) Rel-20 (Release 20)* |
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| ***Reason for change:*** | TR 26.822 states that additional evaluation of TTNB is needed |
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| ***Summary of change:*** | Adds additional evaluation of experimental data on TTNB |
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| ***Consequences if not approved:*** | Missing information, incorrect statement in the TR |
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| ***Clauses affected:*** | 6.13.2.5, 6.13.3, 7.13  |
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|  | **Y** | **N** |  |  |
| ***Other specs*** |  | **x** |  Other core specifications  | TS/TR ... CR ...  |
| ***affected:*** |  | **x** |  Test specifications | TS/TR ... CR ...  |
| ***(show related CRs)*** |  | **x** |  O&M Specifications | TS/TR ... CR ...  |
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| ***Other comments:*** |  |
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| ***This CR's revision history:*** | Rev 2:instantaneous (remove the word)ppb (packets per burst) (audio video)paced sender configuration was the deobservation 2: number the notes |

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#### 6.13.2.5 Aggregate Statistics

This clause presents some aggregate statistics for dynamic traffic characteristics are retrieved from the packet traces collected in this solution.

Table 6.13.2.5-1 illustrates the identified metrics.

Deterministic metrics, averages, standard deviations and percentages are considered.

The metrics are coded with a short abbreviation that is used when reporting in subsequent tables.

In Table 6.13.2.5-2 the scenarios described in the previous clauses are summarized and coded with a short abbreviation that is used then reporting the aggregate statistics in subsequent tables.

Table 6.13.2.5-3 illustrates the results of the metrics derived from the packet traces.

For brevity we can leave out the deterministic metrics bt, tbs, #bp, #lp, #bb and #lb that depend heavily on the length of the trace. Instead, the focus is on the ratio between lone packets/bytes and burst packets/bytes and other metrics that are independent of the length of the trace.

Some observations of the statistics in Table 6.13.2.5-3:

- A large percentage of the traffic and packets are part of a burst, highlighting the relevance of burst handling

- Burst durations in the traces are shorter than a millisecond

- PCM audio is tested, transmitted in lone packets, this increases the share of lone bytes (compared to more efficiently compressed audio)

Table 6.13.2.5-1: Metrics and statistics considered for dynamic traffic characteristics

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| --- | --- | --- | --- | --- |
| Code | Metric | Meaning | Additional notes | Statistics |
| Bt | Burst Time (Bt) | The time of the burst (corresponding to the first packet), measured in seconds  | Bursts are measured as transmission of 2 or more packets in a very short amount of time | Deterministic |
| Ibt | Inter Burst Time (Ibt) | The average duration between 2 consecutive bursts (burst), measured in seconds | This does not consider single ‘lone’ packets transmitted between the bursts | Average and Standard Deviation |
| Bd | Burst Duration (Bd) | The duration between start and end of a burst, measured in seconds | Measures the duration of the burst (in a maximum timeslot of 2 milliseconds ) | Average and Standard Deviation |
| Bs | Burst Size (Bs) | The Size of the Data Burst measured in bytes | Measures the size of the entire burst (in a maximum timeslot of 2 milliseconds) | Average and Standard Deviation |
| Ppb | Packets per burst (Ppb) | Number of packets in each burst | Only bursts of 2 packets or more are considered | Average and Standard deviation |
| #bp | Total Burst packets (#bp) | Number of packets in a burst | This is the number of packets in the trace that is part of a burst | Deterministic |
| #lp | Total Lone packets (#lp) | Number of packets not in a burst | This is the number of packets in the trace that is part not part of a burst | Deterministic |
| #bb | Total Burst Bytes (#bb) | Number of bytes that are part of a burst | This is the number of bytes that is part of a burst | Deterministic |
| #lb | Total Lone Bytes (#lb) | Number of bytes that are not part of a burst | This is the number of bytes in lone packets | Deterministic |
| %lp | Percentage of lone packets (%lp)  | Ratio of lone packets to all packets |  | Percentage |
| %bp | Percentage of burst packets (%bp) | Ratio of burst packets to all packets |  | Percentage |
| %lb | Percentage of lone bytes (%lb) | Ratio of lone packet bytes to all bytes |  | Percentage |
| %bb | Percentage of burst bytes (%bb) | Ratio of burst packet bytes to all bytes |  | Percentage |

Table 6.13.2.5-2: Scenario descriptions and coding in solution #13

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| Scenario code |  |
| w\_h | Windows hangout without network throttling duplex case (Figure 6.13.2.2.2-1) [1] |
| u\_h | Ubuntu Hangout with network throttling simplex case (Figure 6.13.2.2.2-2)[1] |
| g\_v\_a | Gstreamer video only measured on machine A (Figure 6.13.2.3.2-2) [1] |
| g\_v\_b | Gstreamer video only measured on machine B (Figure 6.13.2.3.2-3) [1] |
| g\_va\_a | Gstreamer pcm audio + video at receiver (Figure 6.13.2.3.3-1) [1] |
| g\_va\_b | Gstreamer pcm audio + video at sender (Figure 6.13.2.3.3-2)[1] |
| w\_s | Wowza at server measured at server (Figure 6.13.2.4.2-2) [1] |
| w\_r | Wowza stream measured at player receiver (Figure 6.13.2.4.2-3) [1] |
| m\_rv | MTX server measured vlc player receiver (Figure 6.13.2.4.2-5) [1]  |
| m\_sav | MTX server video + PCM (Figure 6.13.2.4.2-7) [1] monitored at server |
| m\_rav | MTX server video + PCM (Figure 6.13.2.4.2-7) [1] monitored at vlc client |

Table 6.13.2.5-3: Statistical results of experiments in different scenarios in scenario #13

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| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
| Scenario(direction) | ibt(mean) | Ibt(stdev) | Bd(mean) | Bs(mean) | Ppb(mean) | %lp(%) | %bp(%) | %lb(%) | %bb(%) |
| w\_h (a) | 0.0352 | 0.01437 | 0.0002 | 3463 | 3.55 | 23.2 | 76.7 | 6.6 | 93.4 |
| w\_h (b) | 0.01592 | 0.019 | 0.00016 | 3756 | 3.3 | 19.2 | 80.8 | 17.4 | 82.6 |
| u\_h (a) | 0.03132 | 0.00873 | 0.00017 | 13895 | 12,3 | 40.2 | 59.8 | 22.6 | 77.4 |
| u\_h (b) | 0.42321 | 0.37918 | 0.00007 | 213.92 | 2 | 72.2 | 27.8 | 73.8 | 26.2 |
| g\_v\_a | 0.01642 | 0.01175 | 0.00052 | 2864 | 7.73 | 2.4 | 97.6 | 0.7 | 99.3 |
| g\_v\_b | 0.01796 | 0.00967 | 0.00029 | 3109 | 8.6 | 0.2 | 99.8 | 0.2 | 99.8 |
| g\_va\_a | 0.01252 | 0.0114 | 0.00071 | 4144 | 6.6 | 14.2 | 85.8 | 20.1 | 79.9 |
| g\_va\_b | 0.01414 | 0.00955 | 0.00033 | 4472 | 7.5 | 14.8 | 85.2 | 24.7 | 75.3 |
| w\_s (a) | 0.08732 | 0.03483 | 0.00079 | 24822 | 42.3 | 0.2 | 99.8 | 0.0 | 100.0 |
| w\_r (a) | 0.03778 | 0.04627 | 0.00054 | 10671 | 18.24 | 0.7 | 99.3 | 0.1 | 99.9 |
| m\_rv | 0.02749 | 0.0404 | 0.00084 | 6630 | 6.38 | 2.5 | 97.5 | 0.5 | 99.5 |
| m\_sav (a) | 0.04777 | 0.03838 | 0.00056 | 11906 | 12.18 | 12.6 | 87.4 | 3.3 | 96.7 |
| m\_rav (a) | 0.02573 | 0.03412 | 0.00086 | 6692 | 6.8 | 12.2 | 87.8 | 3.2 | 96.5 |

Some additional observation/evaluation of these results considering the inter-burst time or time to next burst:

**Observation 1**: burst durations (Bd) are under 1 millisecond.

**Observation 2**: inter burst times (TTNB) (ibt) are dynamically changing (standard deviation of the inter burst time is in same order of magnitude as the mean inter burst time)

**Observation 3**: Average inter burst time observed in experiments is between 3 ms and 500 ms (423 ms rounded up)

NOTE 1: Inter burst time could be signaled (TTNB) in milliseconds, or 10th of milliseconds to also accommodate future applications with higher frame rates

NOTE 2: Inter burst time signaling (TTNB) with granularity lower than 1 ms may not be useful in some cases due to impact of N6 jitter for DL cases, for UL cases it could be useful in some cases.

**Observation 4**: In typical RTP cases for real time communication up to 99 percent of bytes is carried in bursts, enabling optimization for would optimize traffic up to 99 percent.

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### 6.13.3 Discussion and conclusion

**Observation 1:**

The data from the GStreamer RTP and from WebRTC browser implementation show bursts of data being transmitted in short time intervals (for webRTC default settings in browsers for the paced sender were used). Average inter-burst time observed in experiments is between 3 ms and 500 ms (423 ms rounded up). Burst durations (Bd) are under 1 millisecond.

There is no evidence that the encoder/packager gradually produces and transmits packets, as in the experiments only short bursts were observed. It could happen on a per frame(s) basis in a short time interval. It may also be that some frames are combined in a single burst, but this cannot be observed from the current results.

Similar observations were made when an intermediate server was used for redistributing the streams.

It is therefore expected to be possible for these applications to calculate the burst size a priori before sending it out. The added latency is expected to be limited to around a few milliseconds even in the worst case.

In this scenarios, a large percentage of the total traffic (packets/bytes) is part of a burst. In typical RTP cases for real time communication up to 99% of bytes is carried in bursts, enabling optimization for bursts would optimize up to 99% of the traffic.

In this study only burst of size more than 1 packet were considered.

The results show that in cases a large percentage of the bytes/packets (generally over 90%) are transmitted in bursts.

This highlights that improvement of transmission of bursty traffic can be beneficial as this is such a large percentage of traffic.

NOTE 1: The latency is related to sending the packets on the network, not to the encoder or RTP packager generating the packets. Detailed study of the cause of the delay is FFS. The preliminary conclusion is that a few milliseconds of delay may be introduced is only a worst-case estimate.

**Observation 2:**

The inter-burst time interval seems regular, but not constant. Inter-burst times (TTNB) (ibt) are dynamically changing (stddev of inter burst time is greater than zero). Therefore, signaling of the time to next burst, if known by the application, may be suitable for signaling as a dynamic traffic characteristic.

Aggregate metrics for inter burst times have been derived (average/standard deviation), but these do not relate to the potential to predict or measure these in an application.

NOTE 2: This requires more study of different patterns and situations.

**Observation 3:**

It can be derived that when the network is not overloaded the WebRTC and GStreamer implementations do not differ too much and the influence of the paced sender module is limited.

In addition, when the network bandwidth changes all of a sudden, the rate control may adapt so well that the effect of the paced sender is not significant.

**Observation 4 (extra on P2P versus Server based):**

When comparing the end-end latency of the peer-to-peer setups to the server based setups, for the peer to peer setups latencies < 1 seconds are achieved while in the server routing setup latencies of around 3 seconds are observed.

**Conclusion:**

Short periodic traffic bursts in short intervals occur in typical real time conversational applications using real-time video + audio.

Given the observed traffic behaviour and the observed application behaviour it seems achievable to include information about a burst size before it is being sent out as the durations of the burst are in the order of less than1-2 milliseconds (mostly less than 1 millisecond), but it may require some changes in the sender implementation to achieve this.

A large percentage of the transmitted bytes/packets are part of a data burst.

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## 7.13 Conclusions for Key Issue #12

**Enhancements of Data Burst Marking**

The following aspects are concluded as principles for normative work:

- Do normative work for signaling burst size, when deterministically known, from RTP senders in a HE.

- Do normative work for signaling time to next burst, when deterministically known, and dynamically changing, from RTP senders in a HE

- Revisit definition of a data burst in TS 26.522 [2] to indicate what is meant by idle time and if it is required.

NOTE 1: Communication with SA2 and RAN2 might be necessary about the SA4 work on the definition of data burst.

- Enable the application to provide data boosting indication to the 5G network for downlink using RTP/RTCP signaling.

- Define TTNB in coordination with SA2 and RAN2.

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| \*\*END OF CHANGES\*\* |