3GPP TSG-SA WG4 Meeting #132S4-251123

Fukuoka, Japan, 19 – 23 May 2025

**Source: vivo, Qualcomm**

**Title: End to end simulation and channel characteristics**

**Spec: 3GPP TR 26.940 v0.0.1**

**Agenda item: 7.9**

**Document for: Discussion/Agreement**

**1. Introduction**

At the recent SA#107 plenary meeting, the “Study on Ultra Low Bitrate Speech Codec” has been approved. According to the WID description [1], the primary focus of this study is to develop design constraints and performance requirements for a codec supporting use cases like IMS Voice Call over GEO and the resulting transmission parameters. One main objective is to study “GEO channel characteristics and derive service-related dependencies, e.g. bitrates, mouth-to-ear delay or loss/delay/jitter profiles”. This contribution mainly focuses on the methodology to obtain delay-error profiles.

It is evident that the delay-error profile methodology has been previously employed by SA4, particularly during the development of the EVS codec, as detailed in Annex E of TS 26.132 [2]. The resulting profiles from that work are included as attachments in the TS 26.132 [2] ZIP files. This pCR proposes to adopt a similar approach and generate comparable profiles.

However, Annex-E’s purpose is to test JBM presence, not to characterize the channel to design JBM and PLC, the BLER needs to reflect the real characteristics of the new GEO channel.

Therefore, this contribution mainly plans to

- produce error trace of the GEO channel

- utilize Annex-E as a starting point to produce the required delay-error profiles to characterize the GEO channel to design JBM and PLC

- replace BLER parameter in Annex-E with the produced error trace to reflect the real characteristics of GEO channel.

**2. Reason for Change**

The present document provides how to obtain delay-error profiles based on the methodology defined in Annex E in TS 26.132 [2] and how to derive the error trace to reflect the realistic GEO channel.

**3. Proposal**

It is proposed to agree the following changes to 3GPP TR 26.940 [4].

\* \* \* First Change \* \* \* \*

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non‑specific.

- For a specific reference, subsequent revisions do not apply.

- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[22261] 3GPP TS 22.261: "Service requirements for the 5G system".

[22887] 3GPP TR 22.887: "Feasibility Study on satellite access - Phase 4".

[26071] 3GPP TS 26.071: "Mandatory speech CODEC speech processing functions; AMR speech Codec; General description".

[26132] 3GPP TS 26.132: “Speech and video telephony terminal acoustic test specification”.

[38811] 3GPP TR 38.811, “Study on New Radio (NR) to support non-terrestrial networks”.

[38821] 3GPP TR 38.821, “Solutions for NR to Non-Terrestrial Networks (NTN)”

\* \* \* Second Change \* \* \* \*

# 5 Channel characteristics and service-related dependencies

## 5.1 Introduction

This clause introduces the methodology of obtaining channel characteristics and results for developing design constraints and performance requirements for a codec supporting the main scenario as documented in Clause 4.2.1: IMS Voice Call over GEO.

## 5.2 Delay error profiles

The delay-error profile is a model used to describe the network impairments—particularly delay and packet loss—that can impact real-time conversational services such as IMS voice call. Such profile typically reveals the GEO satellite channel characteristics and will be used to evaluate codec robustness, guide jitter buffer design and ensure a fair and comparable testing.

### 5.2.1 End to end simulation model to derive delay error profiles

The intention of this methodology is to reuse the simulation model defined in Annex E of TS 26.132 [26132] to produce the delay error profile.

This Annex E reference LTE access scenario is illustrated in Figure 5.2.1.1-1. Building on the main scenario defined in Clause 4.2.1, the corresponding end-to-end GEO access scenario is shown in Figure 5.2.1.1-2. The primary distinction between the reference LTE scenario and the GEO voice main scenario lies in the introduction of the “new GEO channel”.



Fig.5.2.1.1-1: End-to-end channel of VoLTE using LTE access



Fig.5.2.1.1-2: End-to-end channel of main scenario for IMS voice call using NB-IoT (GEO) satellite access

Based on the functional description in Table E.1 of TS 26.132, the following input parameters are required to implement the simulation model:

**BLER\_tx / BLER\_rx**:

These parameters are required to simulate block error rates in both uplink and downlink.

NOTE: the resulted error trace based on Clause 5.2.2 will be used to serve as the BLER\_tx/BLER\_rx.

**[max\_tx / max\_rx**:

These define the maximum number of HARQ retransmissions for uplink and downlink respectively, which fall under RAN2 scope. In current specifications, NB-IoT supports at most two HARQ processes, which face constraints in high-latency GEO satellite scenarios. For IMS voice over GEO, HARQ feedback is suggested to be disabled per the standard of Release 18 [5].]

**drx\_cycle\_length**:

This parameter represents the duration of the DRX (Discontinuous Reception) cycle in milliseconds. It determines how frequently the device wakes up to monitoring possible scheduling grant. This parameter affects packet scheduling and transmission timing in the simulation context Annex E of TS 26.132. In addition, the values for LTE are 20-40ms, whether these values are suitable for GEO scenarios should be confirmed with RAN2.

**mis\_eNB1\_eNB2**:

This parameter represents the scheduling time mis-align between the two eNBs. In GEO scenarios, it indicates how long packets wait in the buffer before the next transmission opportunity. This should be determined primarily by RAN2 (responsible for dynamic scheduling or Semi-Persistent Scheduling) with possible input from RAN1 about physical layer timing relationship aspects.

**[max\_net\_delay / min\_net\_delay**:

These represent the delay range between eNB1 and eNB2. For GEO voice, they are considered similar to the LTE scenario, and legacy parameter values can be reused.]

Editor’s NOTE: whether the model for the delay between eNB1 and eNB2 for LTE scenarios well reflects the delay in deployment is FFS.

**nFrames**:

This refers to the number of frames for the simulation. In the reference LTE scenario, one IP packet corresponds to 20 ms of speech. In contrast, the GEO voice scenario introduces additional considerations shown as follows due to the propagation delay from GEO satellite altitude.

* **Speech sequence (frame length)**: For GEO, a longer frame length may be used. The maximum frame length of **80 ms**, as defined by 3GPP, is assumed in this simulation. Final confirmation is expected from SA4.
* **Voice packet size**: This depends on the protocol overhead as illustrated in Figure 5.2.1.1-3 for the reference LTE access scenario and Figure 5.2.1.1-4 for the GEO voice main scenario. The exact overhead depends on the transport path of the RTP packet—user plane or control plane, via IP or via Non-IP (NIDD)—and must be confirmed by RAN2 and SA2. For simulation purpose, the assumed range is as follows:

- **Lower bound**: ~**7 bytes**. Assuming user plane with ROHC (3 bytes) and NB-IoT protocol layers: PDCP (1 byte), RLC (1 byte), MAC (2 bytes);

- **Upper bound**: ~**52 bytes**. Assuming control plane with IPv4/UDP/RTP (20/8/12 = 40 bytes), NAS reduction (7 bytes, optimized by CT1), and NB-IoT layers: RRC (2 bytes), RLC (2 bytes), MAC (1 byte);



Fig. 5.2.1.1-3: VoIP RTP packet in reference LTE access scenario





Fig.5.2.1.1-4: Example of RTP packet in GEO voice main scenario

- **RTP Payload Size**: This is computed as the product of frame length and codec bit rate.

Editor’s Note: whether the size of RTP payload affects the delay-error profile is FFS.

Once the parameters regarding GEO channel are confirmed, the simulation methodology as described in Table E.1 will be updated with these new parameters and used to produce the required delay-error profiles.

### 5.2.2 Simulation Model to derive error traces

The NTN link consists of a service link (between the UE and the satellite) and a feeder link (between the satellite and the ground station). The bottleneck is the link due to the limited TX power and small antenna at the UE. The feeder link is typically characterized by large capacity and high reliability and can be abstracted as an ideal link in the end-to-end simulation. The RAN simulation addresses the service link only.

The objective is to generate multiple loss traces for a combination of frame loss rate (target BLER), raw bitrate (TBS), voice bundling period and Doppler spread, while maintaining channel consistency among different combinations.

The multiple loss traces are the result of using multiple random seeds, and the number is 10. For each combination, all 10 seeds are used in generating the error traces.

Each trace represents a duration of 400 seconds (or 6.67 minutes). Therefore, for 80ms bundling, there are 5000 TBs, and for 160ms bundling there are 2500 TBs.

The following parameters are for the uplink of the service link.

**Channel model**: NTN-TDL-C [38811]

**Modulation:** QPSK

**SCS**: 3.75kHz, 15kHz

**Number of tones:** 1

**Voice bundling period**: 80ms, 160ms, 320ms

NOTE: the 40ms bundling is not considered because for SCS 3.75kHz the minimum time-domain allocation is 32ms and it leaves insufficient time for downlink data (NPDSCH) and control (NPDCCH) transmissions in the same 40ms time interval.

**Doppler spread**: 1Hz, 5 Hz

**Target BLER**: 1%, 2%, 6%, 10%

**Target SNR**: Company report, 3GPP SET-1 SNR +[X] dB, X is TBD, where 3GPP SET-1 parameters are defined in [38821] and the [X] dB accounts for better performance of commercial satellites.

**TBS** **values and PHY bitrates**: The TBS values are selected from table 16.5.1.2-2 for NB-IoT for NPUSCH in TS36.213 and the corresponding PHY bitrates and codec bitrate (assuming 8 bytes of packet header with RoHC) are calculated for each bundling period.

[

Table 1 TBS and PHY bitrate for 80ms bundling

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| TBS (bits) | 144 | 256 | 328 | 424 |
| PHY bitrate (kbps) | 1.8 | 3.2 | 4.1 | 5.3 |
| Codec bitrate (kbps) | 1.0 | 2.4 | 3.3 | 4.5 |

Table 2 TBS and PHY bitrate for 160ms bundling

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| TBS (bits) | [208] | 424 | 600 | 808 |
| PHY bitrate (kbps) | [1.30] | 2.65 | 3.85 | 5.05 |
| Codec bitrate (kbps) | [0.9] | 2.25 | 3.45 | 4.65 |

Table 3 TBS and PHY bitrate for 320ms bundling

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| TBS (bits) | [328] | 776 | 1000 | 1544 |
| PHY bitrate (kbps) | [1.025] | 2.425 | 3.125 | 4.825 |
| Codec bitrate (kbps) | [0.8025] | 2.225 | 2.925 | 4.625 |

**]**

]

Editor’s NOTE: Company can report candidate values of TBS.

NOTE: A PHY bitrate is considered feasible if the required SNR to meet a target BLER is lower than the target SNR.

**Channel consistency**: The same set of channel realizations are used across all combinations.

### 5.2.3 Results

Editor’s Note: the results are FFS

\* \* \* End of Changes \* \* \* \*

[1] SP-250378, "Study on Ultra Low Bitrate Speech Codec"

[2] TR 26.132, "Speech and video telephony terminal acoustic test "

[3] TS 26.114, "Media handling and interaction"

[4] TR 26.940, "Study on Ultra Low Bitrate Speech Codecs"

[5] TR 36.321, "Evolved Universal Terrestrial Radio Access (E-UTRA); Medium Access Control (MAC) protocol specification"

[6] S4-141392, "Extra profiles used for EVS characterization testing".