

Study of the impact of UMTS Best Effort parameters on QoE of VoIP services

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Abstract— This paper evaluates the performance of VoIP services over mobile accesses, with special focus on currently deployed Best Effort UMTS networks. There are different configuration parameters that have a great impact on end users' perception of quality. This paper tries to help understanding the impact of those parameters, and how they can be tuned for a better VoIP experience.

Quality of Experience, VoIP, Background, UMTS, parameter tuning

I. INTRODUCTION

The Voice over IP (VoIP) service has gained a remarkable relevance over the last few years. Several factors have influenced this growth. The development of new voice codecs allows providing an acceptable listening quality at low bandwidth demands. The broad deployment of Internet access technologies entails that more and more people have the opportunity of using Internet services. And, finally, the development of novel VoIP applications, either free or commercial, has made the technology available to the end users.

However, one of the handicaps that still slow down the deployment of VoIP services is the variability of the delays in the communication. And this effect is even more significant over mobile accesses, due to the characteristics inherent to the radio links.

This paper focuses on the study of the feasibility of provisioning VoIP services over currently deployed Universal Mobile Telecommunications System (UMTS) packet networks. More specifically, the discussion will be centered on the provision of VoIP services over Best Effort UMTS networks, where no QoS differentiation is implemented in the UMTS Terrestrial Radio Access Network (UTRAN).

Section 2 reviews the most relevant technical characteristics of this type of networks, with a special focus on a series of configuration parameters that must be taken into account for enabling the provision of VoIP services. In order to evaluate the feasibility of VoIP services over mobile UMTS accesses, it is convenient to study the problem from the viewpoint of the perceive quality of service (PQoS). This approach allows us to include the end users' perception of services in the study, in order to reflect the global quality of experience (QoE). Thus, Section 3 provides the necessary background for PQoS evaluation on VoIP services and

identifies the main sources of impairments that may result on quality degradations.

The proposed study is carried out through simulations. In Section 4, the simulation scenario and configuration is presented, while Section 5 shows the results obtained for different tests. Finally, Section 6 concludes this paper, and illustrates the convenience of a dynamic service adaptation system that keeps accurate performance levels by setting up an optimum configuration.

II. VOIP SERVICE OVER BEST-EFFORT UMTS

A. Basic VoIP service

The VoIP service encapsulates voice frames into IP packets, which are later sent to the configured layer 2 service. The protocol stack for 3G conversational applications is conformed by the voice frame as the payload of a RTP/UDP/IP header.

This paper focuses on the Adaptive Multi-Rate (AMR) codec as the candidate for VoIP over 3G services. [1] This codec offers a good quality at low bitrate demands and a real-time adaptation capability. Its performance has been proven for UMTS in [2]. The AMR codec may work on eight modes at different bitrates. The voice frame is always 20ms length, with varying frame sizes to fit the target bitrate.

From the voice service perspective the total header overhead is 20 bytes (12 bytes RTP, 8 bytes UDP and 20 bytes IP). Table I shows the different voice frame sizes and the bitrate obtained at IP level due to protocol overhead. The protocol overhead results on a considerable waste of resources, which is critical in mobile Internet accesses.

TABLE I. AMR MODES AND BITRATES

| Codec mode | Voice frame size | Bitrate at IP layer |
|------------|------------------|---------------------|
| AMR 12.20 | 244 bits | 28.2 kbps |
| AMR 10.20 | 204 bits | 26.2 kbps |
| AMR 7.95 | 159 bits | 23.95 kbps |
| AMR 7.40 | 148 bits | 23.4 kbps |
| AMR 6.70 | 134 bits | 22.7 kbps |
| AMR 5.90 | 118 bits | 21.9 kbps |
| AMR 5.15 | 103 bits | 21.15 kbps |
| AMR 4.75 | 95 bits | 20.75 kbps |

The main alternatives to reduce the protocol overhead are header compression or by modifying the packetization, which is the number of voice frames per IP packet. In [3] it is stated that header compression is more efficient than higher packetization. However, header compression cannot be used within the core network, so packetization shall be increased if the bandwidth limitation occurs within the core network.

Figure 1 illustrates the enhancement obtained for different packetization schemes for each AMR mode. The drawback of increasing the packetization factor is that voice frames must be buffered at the sender until all the voice frames sharing an IP packet are generated, and thus introducing additional delays.

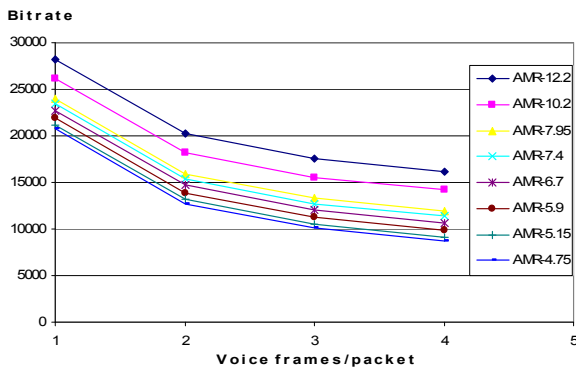


Figure 1. Bitrate vs. packetization for different AMR modes.

B. Basic VoIP service

The UMTS protocol stack for the user plane is shown in Figure 2. [4] Each layer offers a service to the upper layers, including a series of functions. The basic service consists on the transmission of Service Data Units (SDUs) towards the destination node, but each layer additionally includes different functions.

The Packet Data Convergence Protocol (PDCP) service offers the capability of RTP/UDP/IP header compression, which is implementation-specific.

The Radio Link Control (RLC) offers three basic data transmission services: the Acknowledge Mode (AM) makes use of an error recovery method to assure the correct delivery of SDUs; the Unacknowledged Mode (UM) just controls that SDUs are transported in sequence; the Transparent Mode (TM) does not add any RLC protocol information.

In addition to this data transfer functions, the RLC layer is in charge of segmentation and reassembly of upper layer SDUs, as well as concatenation of SDUs to better fitting the MAC capacity.

The MAC layer offers its transfer service as Logical Channels, including control and traffic channels.

Finally, the Physical Channel offers its data transfer service as Transport Channels, which can be common or dedicated channels. The service is available to the upper layers each Transmission Time Interval (TTI), and the upper layers usually send SDUs in a number of Transport Blocks (TB) per TTI. The specific settings to be used in a service are determined by the Transport Format Set (TFS).

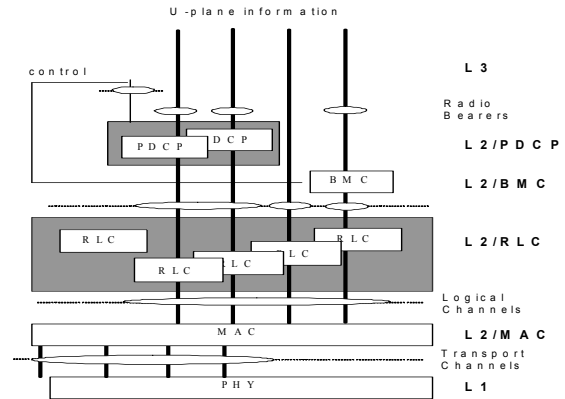


Figure 2. UMTS User plane

C. VoIP over Best-Effort UMTS

Table II illustrates a basic UMTS service configuration based on the Interactive or background / UL:64 DL:384 kbps / PS RAB as defined in [5].

TABLE II. CONSIDERED UMTS UL/DL CONFIGURATION

| UMTS level | Parameter and Value | | |
|------------|---------------------|------------------------|------------------------------|
| | Parameter | UL | DL |
| RLC | Max bitrate | 64 kbps | 384 kbps |
| | SDU size | 320 bits | 320 bits |
| | RLC mode | AM | AM |
| PHY | TFS | 4 TF 0-1-2-4 TB/TBS | 6 TFs 0-1-2-4-8-12 TB/TBS |
| | TB size | 336 bits | 336 bits |
| | TTI | 20 ms | 10 ms |
| | TrCH type | DCH | DCH |

This set of Background UMTS configuration parameters implies different issues that directly influence the performance of VoIP services. First of all, the maximum bitrate parameter determines the capacity of the RLC layer for sending RLC SDUs to the MAC layer. Initially, the values for both the uplink and downlink directions are sufficient for the bitrates shown in Figure 1 for the different AMR configurations.

The TTI parameter gets different values in each direction. In the uplink, the TTI is set to 20ms, which matches up with the AMR frame size of 20ms. Thus, in an ideal case each voice frame will be transmitted within a TTI, and no additional delays are introduced due to the UMTS slotted transmission method. In the downlink, the TTI parameter is configured to 10ms, and each TTI has enough capacity to transmit an entire voice frame, so additional delays are avoided as well.

The RLC layer is configured in Acknowledge Mode, which implies that SDUs are confirmed and recovered if lost in the radio link.

The size of the RLC header introduced with AM is 2 bytes. As a result, although the physical Transport Block size is 336 bits, the SDU size is limited to 320 bits. In the uplink direction, the RLC layer is able to access the physical medium every 20ms, thus having 50 time slots per second. At each TTI slot, the RLC can send a maximum number of 4 SDUs of 320 bits.

As previously stated, a higher VoIP packetization scheme results on decreasing the required IP bitrate but higher delays. Initially, the packetization delay can be calculated as the voice frame size (20 ms for AMR) multiplied by number of voice frames per packet. However, in UMTS access networks, additional delays may arise due to the slotted access to the physical medium. As shown in Table II for the uplink direction, a packetization scheme of 4 voice frames per packet generates IP packet sizes high enough to require 5 RLC SDUs to be sent to the MAC layer. In consequence, two consecutive TTIs are needed to transport one IP packet, introducing additional 20ms delays to the communication.

TABLE III. PACKETIZATION DELAYS WITH RLC AM

| Frames per packet | Parameter and Value | | |
|-------------------|-------------------------|---------------|--------|
| | AMR-12.2 IP packet size | Required SDUs | Delay |
| 1 | 564 bits | 2 | 20 ms |
| 2 | 808 bits | 3 | 40 ms |
| 3 | 1052 bits | 4 | 60 ms |
| 4 | 1296 bits | 5 | 100 ms |

Another effect to be considered is inherently introduced by the use of RLC AM. This operating mode is designed for optimizing the performance of applications with high tolerance to delays. Since the radio medium introduces frame loss ratios higher to wired links, the TCP recovery mechanism may result on slow data transfers. In order to avoid this effect, the RLC local recovery allows a faster recovery and a better performance of TCP applications.

However, this mechanism may not be beneficial for real-time UDP communications as VoIP, since recovered packets could not arrive on time to the destination endpoint. The impact of this effect on VoIP performance depends on the dejittering buffer size. This buffer is introduced in the destination between the receiving and playing functions in order to store a determined number of voice frames. Thus, voice frames are delayed before being played out, but the application results more resilient to delay variations.

Thus, for VoIP services over Background UMTS networks, the effect of packet losses in the radio link is reflected in the performance of the dejittering buffer. If a VoIP packet recovered by the RLC arrives to the destination endpoint while the subsequent voice frame is still in the dejittering buffer, it could be provided to the playing function on time. Otherwise, the packet will be computed as lost in the dejittering buffer.

Finally, another detected effect that must be taken into account when evaluating the performance of VoIP services over Best/Effort UMTS networks is the switching between transmission channels. Table II shows Dedicated Channels (DCH) as transmission channels. However, assigning a DCH to every user limits the capacity of a cell, reducing the number of users who can access data services at the same time. If data rates are low or intermittent, the alternative configuration of using common FACH/RACH channels results more efficient. To solve this trade-off, a commonly adopted approach consists on initially assigning common channels, switching to dedicated channels when a threshold is exceeded. This threshold values is configuration-dependant, and may result on disturbing voice quality degradations at the initial periods of voice communications. [6]

III. PERCEIVED QUALITY OF VOIP SERVICES

The predominant method for assessing the PQoS of VoIP services is the E-model, defined by the ITU-T in [7]. This model allows computing the value of a rating factor R, which provides an evaluation of the communication impairment as shown in (1).

$$R = R_o - I_s - I_d - I_{e-eff} + A \quad (1)$$

The R score computed with default values proposed in [7] is 93.2, which corresponds to an estimated Mean Opinion Score (MOS) score of MOSCQE=4.41.

The relationship between R and MOSCQE is determined by (2):

$$MOS_{CQE} = \begin{cases} 1 & R < 0 \\ 1 + 0.035R + R(R - 60)(100 - R) \cdot 10^{-6} & 0 < R < 100 \\ 4.5 & R > 100 \end{cases} \quad (2)$$

Regarding service and network performance, the main impairment factors due to VoIP over 3G specific characteristics are I_d and I_{e-eff} . I_d is defined as the impairment due to delay effects. If perfect echo cancellation is considered, I_d is determined by the total one-way delay at application layer (T_a) as in (3):

$$I_d = \begin{cases} 0 & T_a \leq 100\text{ms} \\ 25 \left\{ \left(1 + X^6 \right)^{\frac{1}{6}} - 3 \left(1 + \left[\frac{X}{3} \right]^6 \right)^{\frac{1}{6}} \right\} + 2, X = \frac{\log\left(\frac{T_a}{100}\right)}{\log 2} & T_a > 100\text{ms} \end{cases} \quad (3)$$

I_{e-eff} is defined as the impairment factor due to the combined effect of the codec and packet losses. In (4), I_e is the equipment impairment factor, and reflects the degradation due to due to codification. Thus, it is a codec-dependant parameter. When there are packet losses, the consequent degradation is determined by the codec-dependant Packet-loss Robustness Factor (Bpl), which

reflects the resiliency of a codec to losses, and by the Burst Ratio (BurstR), which is determined by the bursts intensity of experienced losses. The Ie and Bpl values for AMR-12.2 mode (compatible with GSM EFR) can be set to 5 and 10 respectively. [8]

$$Ie_{eff} = Ie + (95 - Ie) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl} \quad (4)$$

The end-to-end delay can be calculated as (5): [3]

$$Ta = d_{cod} + d_{pack} + d_{UTRAN} + d_{netw} + d_{jit} + d_{decod} \quad (5)$$

d_{cod} and d_{decod} are the codification and decodification delays, which for the AMR codec can be considered as fixed values of 20ms each.

d_{pack} is the delay introduced by the packetization scheme, which can be calculated as n times 20ms, being n the number of voice frames per packet.

d_{UTRAN} is defined as the delay introduced in the UTRAN, which includes the transmission delay and the additional delay due to RLC functions. This RLC delay includes the delay due to the slotted access to the physical medium and the delays introduced by the SDU recovery mechanism. For the presented case study, the former can be considered negligible, as discussed in the previous section, except for a packetization of 4 (see Table III). The latter highly depends on the UTRAN loss ratio, which determines the number of SDUs to be recovered, and on the burstiness of the losses, which may influence the time needed for recovering a lost SDU.

d_{netw} is the delay introduced by the transport of packets through the core network, which may include transmission delays and queuing delays.

Finally, d_{jit} is the delay introduced by the dejittering buffer, which is determined by the buffer size.

Similarly, the total packet loss probability (Ppl) and the BurstR parameter are determined by the individual contributions of each packet loss ratio due to the different sources of losses.

$$Ppl = f \{ \rho_{UTRAN}, \rho_{netw}, \rho_{jit} \} \quad (6)$$

ρ_{netw} is the packet loss ratio through the core network, mainly due to congestion effects.

ρ_{UTRAN} is the packet loss ratio caused by packet lost in the UTRAN. In RLC AM, SDUs are considered as not recoverable when the SDU Discard Timer is exceeded.

ρ_{jit} is the packet loss ratio experienced in the dejittering buffer, due to voice frames arriving later than its play out time.

While ρ_{netw} depends only on core network dimensioning, both ρ_{UTRAN} and ρ_{jit} are dependant on the values of the SDU Discard Timer and dejittering buffer size, as well as on the end-to-end delay (Ta), which at the same time is determined

by the statistical characteristics of the SDU losses in the UTRAN.

IV. SIMULATION ENVIRONMENT

In order to study the feasibility of provisioning VoIP services over Background UMTS networks, a general UMTS scenario is implemented using OPNET© 14.0. Figure 3 depicts network topology which is compounded of a mobile end user terminal which establishes a VoIP session to a fixed VoIP workstation through an UMTS network. This UMTS network includes the UTRAN (Node B and RNC nodes) and the UMTS core network (SGSN and GGSN nodes).

The figure represents a general network topology adopted for the performance of different simulations, which allow the study of the source of impairments. For this purpose, it is necessary to set different network configurations using the facilities given by OPNET©.

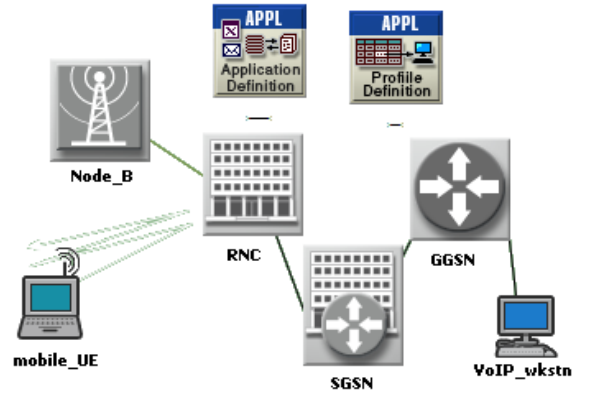


Figure 3. Considered OPNET UMTS scenario

There are some scenario attributes which remain the same through all the network simulations, such as the application and the profile definitions. Concerning to the service, a new type of application is defined: VoIP, whose attributes are shown in Figure 4 and in which the Type of Service is set to Best-Effort.

| Attribute | Value |
|-------------------------------|-----------------|
| Encoder Scheme | [...] |
| Voice Frames per Packet | 1 |
| Type of Service | Best Effort (0) |
| RSVP Parameters | None |
| Traffic Mix (%) | All Discrete |
| Signaling | None |
| Compression Delay (seconds) | 0.02 |
| Decompression Delay (seconds) | 0.02 |
| Conversation Environment | [...] |

Figure 4. OPNET VoIP service configuration

Since this paper focuses on AMR encoding schemes, a new Voice attribute (Figure 5) is created in order to take into account the eight AMR modes, which should be previously defined in the Voice Encoder Schemes Attribute.

| | Encoder | Type |
|---|----------|---------|
| 0 | AMR_12.2 | sendrcv |
| 1 | AMR_10.2 | sendrcv |
| 2 | AMR_7.95 | sendrcv |
| 3 | AMR_7.4 | sendrcv |
| 4 | AMR_6.7 | sendrcv |
| 5 | AMR_5.9 | sendrcv |
| 6 | AMR_5.15 | sendrcv |
| 7 | AMR_4.75 | sendrcv |

Figure 5. Implementation of AMR modes considered

To end with the service configuration, it is defined a Voice User profile configured with the VoIP application previously created. This profile is set into both end user terminal attributes. With regard to the UMTS network, the mobile terminal is configured to send traffic through the background class.

Node B attributes allows the configuration of the cell pathloss parameters, such as pathloss model or shadow fading standard deviation, which allows studying the UMTS metrics in different environments.

Since the background class is being used, it is necessary to configure the related attributes in the RNC node, such as the radio link control mode, which is set to Acknowledged or the RLC MAC concatenation of SDUs, set to enabled. Additionally, the TrCH TTI is set to 10ms and 20ms respectively for the downlink and uplink bearers.

V. RESULTS

With this configuration, we analyze the impact of mobility on QoE. As Figure 6 illustrates, different mobility patterns result on different loss patterns at the radio transmission channel. The figure shows the CDF of the gap periods, i.e. the duration of loss-free time periods, for the two selected trajectories: the first one simulates a mobile device in an indoor environment and the second one represents a vehicular trajectory within the limits of the Node-B coverage.

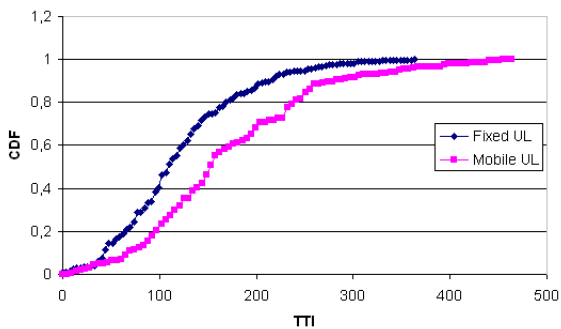


Figure 6. CDF for gap periods

Since the RLC AM operating mode recovers these radio errors locally, the distribution of delays is directly affected. Figure 7 shows the values of end-to-end delays in both cases for the uplink direction.

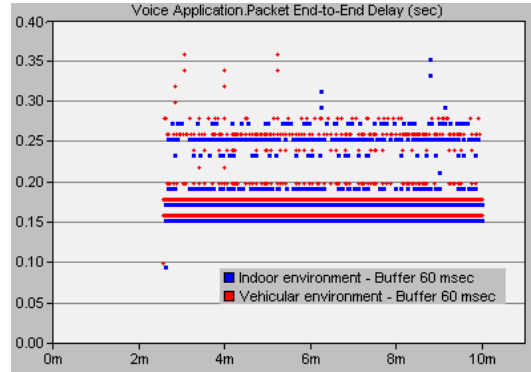


Figure 7. End-to-end delay distribution

As stated on Section 3, the MOS scores resulting from these end-to-end delay distributions depend on the dejittering buffer size. A higher buffer size introduces additional delays, but decreases packet loss ratio in the buffer. Therefore, it is a trade-off to select the most accurate buffer size to maximize the PQoS, being different in function of the composite network effect.

Figure 8 shows the MOS scores obtained for different dejittering buffer sizes, analyzed for the case of a mobile user in a vehicular environment. As can be observed, the optimal dejittering buffer size for the considered scenario is 60ms.

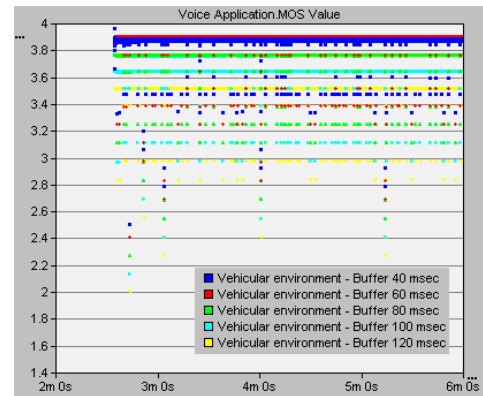


Figure 8. MOS score vs. buffer size

Figure 9 shows the DeJitter Loss Rate Values obtained for different simulation runs configured with the same SDU Discard Timer and varying the dejittering buffer size. This value provides a measure of how many packets arrive at the sender with too high delay values, so they are dropped in the application buffer since they can not be played. As can be observed, for the end-to-end delay distributions found in this study case, there is a high difference between the case of 40ms and those of 60ms or higher values.

Thus, we can state that for the gap between 40ms and 60ms of buffer size, the enhancement on total packet loss ratio is more important than the impairment due to the additional delay introduced.

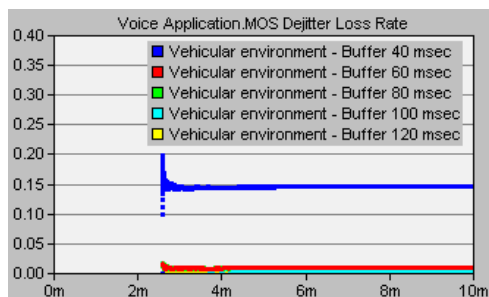


Figure 9. Dejitter loss rate

VI. CONCLUSIONS

This paper focuses on the evaluation of VoIP services over Best Effort UMTS networks. We have analyzed the most relevant configuration parameters in order to evaluate the performance of VoIP communications in different conditions. From a QoE perspective, the impact of the UMTS RLC AM has a great influence on users' perception of quality, since losses in the radio access are mitigated by introducing additional delay variation. Results show the combined impact of dejittering buffer and UTRAN loss pattern.

We have shown that the performance of VoIP services over Best Effort UMTS network depends on a heterogeneous set of configuration parameters, which should be carefully tuned in order to maximize the PQoS experienced by VoIP users, while keeping the performance of the cell in suitable levels for other data applications. Several studies have analyzed this fine tuning previously. For example, in [9] the dynamic selection of AMR modes was studied taking into account the characteristics of 3G networks. Likewise, the optimum dejittering buffer size can be analyzed [10] in combination with the impact of VoIP packet sizes and different UTRAN parameters. [3]

All these results provide a step forward towards a real-time adaptation method, which could allow a dynamic tuning of the set of parameters based on the status of the network. However, in order to find a generalized method, this study must be extended in order to take into account the network variability, and especially the burstiness of losses.

ACKNOWLEDGMENT

The work here presented has been performed within the research framework of the EU-funded FP7 ICT-214751 ADAMANTIUM Project.

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