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A New Technique for Transceiver Location Data Over LTE Voice Channels

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Abstract : This paper provides a new novel technique for transmitting and receiving the location information over the voice calls of the LTE cellular-networks. This is done by embedding the location data received by the mobile equipment to the voice frames of the LTE standard during the normal call. The new technique inserts all the location information over the padding bits which are set to zero in the audio frames of the AMR-WB codec used in the LTE standard. AS AMR-WB voice codec is a mandatory codec for all LTE implementations according to 3GPP and ITU-T, this technique can be applied for all LTE equipment's without any problems. Also, as this technique of embedding and extraction of location frames is done over padding bits, there is no loss or degradation in the voice quality.
Keywords : LTE, AMR-WB, 3GPP, ITU-T.

I. INTRODUCTION

Due to the highly rising demands of wireless communication systems, the cellular-networks are considered a highly promising demand for its high data rate and mobility. In 2008, the Long-Term Evolution (LTE) was introduced by the 3GPP (3rd Generation Partnership Project) [1] in order to increase the capacity and speed of wireless data networks. LTE is marketed as 4th Generation (4G) wireless services as it is considered an upgrade for 3G UMTS. LTE offers several important benefits for the subscribers as well as to the service provider. Providing peak download rates up to 299.6 Mbit/s and upload rates up to 75.4 Mbit/s with peak delay of 5 ms, it significantly satisfies the users requirement by targeting the broadband mobile applications with enhanced mobility. With the introduction of Smartphone, the application like HDTV, online gaming, video meetings, etc. are certainly become more valuable to the users. Hence, the users under- stand and appreciate the benefits of LTE high data rates and services. A further goal was the redesign and simplification of the network architecture to an IP-based system with significantly reduced transfer latency compared to the 3G architecture. The LTE wireless interface is incompatible with 2G and 3G networks, so that it must be operated on a separate radio spectrum.

For voice communication, which is the main goal of all cellular systems, two main voice solutions are proposed: VoLTE (Voice over LTE) [2] which is supported natively by LTE and CSFB (Circuit- Switched Fall Back) [3]. VoLTE carries on voice calls directly in the 4G system. It utilizes the Voice- over-IP (VoIP) solution, and still offers guaranteed Quality-of-Service (QoS) through LTE networks resource due to the low delay (5 ms). In contrast, CSFB will make use of the deployed predecessor system and works with most current phone models (whereas VoLTE requires new phones). CSFB utilizes the CS technique in the legacy 3G systems to support voice calls for LTE subscribers. For every voice call, CSFB transfers the call procedure from the 4G network to the 3G network. Once the call is terminated, CSFB moves the phone state back to the 4G cellular network. So, LTE offers a readily-accessible and cost-effective solution. The most popular voice solution till now is CSFB which has been widely deployed by most LTE carriers. On the other hand, VoLTE is considered to be the future solution. Due to the higher cost of upgrading mobile networks and phones, its current deployment is not as popular as CSFB. CSFB is the prevalent solution now and continues to be appealing in developing countries. Meanwhile, VoLTE will gain its widespread usage in the long run.

According to 3GPP and ITU-T, the main voice codec used with LTE in VoLTE mode is the Adaptive Multi-Rate Wideband (AMR-WB) [4], [5].

Table 1. Frame Length for each Bit-Rate Configuration Of AMR-WB According To [4]

Frame Length (bits)	Bit rate (kbit/s)
132	6.60
177	8.85
253	12.65
285	14.25
317	15.85

365	18.25
397	19.85
461	23.05
477	23.85

AMR-WB is defined as G.722.2, an ITU-T standard speech codec. When used in LTE cellular-networks, there are three different configurations (combinations of bitrates) that is used for voice channels: Configuration A (Config-WB-Code 0): 6.6, 8.85, and 12.65 kbit/s (Mandatory multi-rate configuration [4]), Configuration B (Config-WB-Code 2): 6.6, 8.85, 12.65, and 15.85 kbit/s and Configuration (Config-WB-Code 4): 6.6, 8.85, 12.65, and 23.85 kbit/s. The frame lengths are shown in table I. This codec leverages frequency bands of (50-6400) Hz for all modes and (6400-7000) Hz (23.85 kbit/s mode only). The frame size is 20 ms and the input sampling rate is 16K-sample/s (each frame is corresponding to 320 audio samples). As we can notice from table I, each frame is padded to have an integer number of bytes with a number of bits according to its length. In this paper, we will consider transceiving the location data within the voice frames of AMR-WB.
The following section briefly reviews some basic information regarding other techniques for transmitting and receiving date over voice frames, which will be used subsequently for comparisons. Section 3 provides a full description of the proposed technique for transceiving location data within AMR- WB voice frames. Finally, conclusion and future work are presented in Section 4.

I. BACK GROUND AND REVIEW OF PREVIOUS WORK

I.1. Description AMR-WB

The Adaptive Multi-Rate Wideband (AMR-WB) speech codec [6], [7] was originally developed and Standardized by the Third-Generation Partnership Project (3GPP) as a codec to be used for GSM 3G cellular-network system.

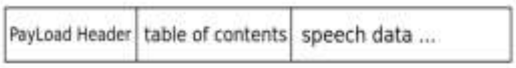

Fig. 1. General layout of the payload frame.

AMR-WB codec is also a multi-mode speech codec. AMR-WB supports 9 wide band voice compression modes with different bit rates ranging from 6.6 to 23.85 kbps as described in table I. The sampling frequency used in AMR- WB is 16K sample/s and the speech coding is achieved every 20 ms per each frame. This means that each AMR-WB encoded frame requires 320 speech samples to be processed.
As we mentioned in Section 1, the future implementation of LTE voice call transmission will be VoLTE. Accordingly, the scope of the new proposed technique will be the use of Real-time Transport Protocol (RTP) frames [8], within UDP segments of VoLTE to transmit and receive location data.

According to [6], the encoded speech payload to be transmitted over the RTP frame consists of a payload header, a payload table of contents, and speech data representing one or more speech frames. Fig.1 shows the general layout of the payload frame. Subsections 2.2. and 2.3 describes the divergence of the payload structure depending on whether the AMR-WB session is configured to use the bandwidth-efficient mode or octet-aligned mode according to [6].

I.2. Bandwidth-efficient mode

In accordance with [5] and [6], the payload header of the bandwidth-efficient mode simply consists of 4-bits only, and represents the Codec Mode Request (CMR) as described in table II. The table of contents (TOC) consists of a list of TOC entries, each representing a speech frame. It consists of 6- bits as described in fig.2.According to fig.2, there are 3 fields in TOC entry. First, F (1 bit): If set to 1, indicates that this frame is followed by another speech frame in this payload. If set to 0, indicates that this frame is the last frame in this payload. Second, FT (4 bits): Frame type index, indicating either the AMR or AMR-WB speech coding mode or comfort noise (SID) mode of the corresponding frame carried in this payload. Last, Q (1 bit): Frame quality indicator. If set to 0, indicates the corresponding frame is severely damaged.

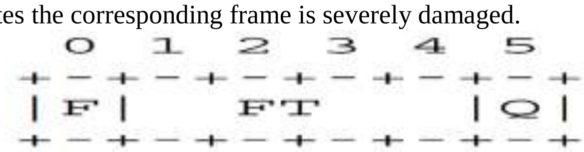

Fig. 2. Layout of TOC for Bandwidth-efficient mode.

Table 2. Frame Length for each Bit-Rate Configuration Of AMR-WB According To [5]

Frame TypeIndex	ModeIndication	ModeRequest	Frame content
0	0	0	6.60 Kbit/s
1	1	1	8.85 Kbit/s
2	2	2	12.65 Kbit/s
3	3	3	14.25 Kbit/s
4	4	4	15.85 Kbit/s
5	5	5	18.25 Kbit/s
6	6	6	19.85 Kbit/s
7	7	7	23.05 Kbit/s
8	8	8	23.85 Kbit/s
9	-	-	AMR-WB SID
10-13	-	-	For future use
14	-	-	speech lost
15	-	-	No Data (No Tx/No

The algorithm for forming the RTP payload in bandwidth-efficient mode is simple. Simply, packing bits from the payload header, table of contents, and speech frames in order. Starting with the payload header, and then all TOC entries, then all speech frames in the same order as TOC entries and encoded according to CMR value of the payload header (table II). Then, padding the payload to complete an integer number of bytes. Padding bits MUST be set to zero and MUST be ignored at the decoder. If any speech frame is lost due to any reason, a ToC entry with FT set to NO DATA (table II) SHALL be included in the ToC for each of the missing frames, but no data bits are included in the payload for the missing frame.

```
 0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15
+-+-+-+-+-+-+-+-+-------------------
|  CMR  |R|R|R|R|  ILL  |  ILP  |
+-+-+-+-+-+-+-+-+-------------------
```

Fig. 3. Layout of payload header for Octet-aligned mode.

```
+----------------------+
| list of ToC entries  |
+----------------------+
| list of frame CRCs   | (optional)
```

Fig. 4. Layout of TOC for Octet-aligned mode.

I.3. Octet-aligned mode

According to [6] and [5], fig.3 describes the payload headers for octet-aligned mode. It consists of a 4-bit (CMR), 4 reserved bits, and optionally, an 8-bit interleaving header. As shown in fig.3, the payload header consists of CMR (4 bits) which is the same as defined in subsection II-C and table II. R, is a reserved bit that MUST be set to zero and MUST be ignored by the receiver. ILL (4 bits, unsigned integer) and ILP (4 bits, unsigned integer) are OPTIONAL fields.

The table of contents (TOC) in octet-aligned mode is the same as Bandwidth-efficient mode with extra two padding bits to complete one byte. These two padding bits MUST be set to zero and MUST be ignored by the receiver. Same as Bandwidth- efficient mode, TOC is consisting of a list of ToC entries where each entry corresponds to a speech frame contained in the same order in the payload and an additional optional list of speech frame CRCs as described in fig.4. The CRC is calculated as shown in equation 1.

$$C(x) = 1 + x2 + x3 + x4 + x8 \quad (1)$$

The Speech data in octet-aligned mode is com- posed in a similar way as in the bandwidth- efficient mode as discussed in subsection II-B with only one different that each speech frame MUST be padded with zero bits separately at the end if all bits in the octet are not used. The padding bits MUST be ignored on reception.

The algorithm for forming the RTP payload in octet-aligned mode is the same as bandwidth efficient mode except one different that two different packetization methods for packetizing the speech data named Normal order and Robust sorting order. In normal order, all of the bytes forming a speech frame are appended to the payload as one unit. The speech frames are packed in the same or- der as their corresponding TOC entries are arranged in the TOC list. For robust sorting order, the bytes of all speech frames are interleaved together at the byte level. So, the data part of the payload begins with the first byte of the first frame, followed by the first byte of the second frame, then the first byte of the third frame, and so on. After the first byte of the last frame has been packed, the cycle is repeated with the second byte of each frame. The operation continues until all bytes are finished. If the frames do not have an equal length, a shorter frame is skipped once all bytes in it have been packed.

I.4. Previous work

H. Miao et al. [9] introduced a technique called "Adaptive Suboptimal Pulse Combination Con- strained" (ASOPCC). They embed data on com- pressed speech signal of AMR-WB codec. By taking the advantage of the" redundancy", they employed features of the Adaptive Multi-Rate Wide- band (AMR-WB) encoder adopted in 3G communications, their approach modifies the search phase of words belonging to the algebraic codebook used during the encoding process, i.e., some bits of the resulting codebook are altered to carry the hidden data to represent secret information. So, a very destructive modification of codec's codewords highly reduces the SNR (Signal to Noise Ratio), accordingly resulting into noticeable artifacts. Eventually, the stenographic method must also take into consideration channel loss model, since the AMR- WB codec is a lousy codecs and eventually alters the features of its original voice stream according to the measured SNR. For these reasons, authors proposed a time-varying embedding factor η used to control the trade-off between the SNR and the amount of hidden data. To achieve the best tradeoff, $\eta = 4$ leading to a stenographic bandwidth of 1, 400 bps, which can be used for short periods and under high quality communication channels to avoid the degradation of the voice intelligibility.

A similar approach is introduced in [10], where they deal with the 3rd Generation Partnership Project (3GPP) single-channel narrowband Adaptive Multi- Rate (AMR) codec. The same as the previous work, A stenographic scheme utilizes some bits of the code-book to be modified resulting into a manipulation of both the frequency and time envelopes of the voice signals. The resulting data bandwidth is of 2 kbps. Finally, for a low-rate standard, the work is done by Tingting and Zhen utilizes the G.723.1 low- rate codec [11], offering a data capacity of 133.3 bps (i.e., altering the 5 least significant bits of a compressed frame).

II. THE PROPOSED TECHNIQUE FOR TRANSMITTING AND RECEIVING LOACTION DATA OVER AMR-WB

II.1. Design idea

Our proposed technique is transmitting the location data (which is a small amount of data) over the reserved bits and the padding bits, so that the quality of the voice call will never be affected or degraded. In the following subsections, we will discuss in details how to implement this new technique. According to [12], the location data is simply a small message of type (Geographic position, latitude / longitude) which may not exceed 80 bytes. An example for this representation is:" $GPGLL, 3751.65, S, 14507.36, E ∗ 77" or" $GPGLL, 4916.45, N, 12311.12, W, 225444, A".

II.2. Implementation for bandwidth-efficient mode

For bandwidth-efficient mode of transmission, as discussed in subsection II-B, the payload of RTP frame consist of 4-bits for payload header, 6-bits for TOC for each speech frame and number of bits for each speech data depends on the codec rate as discussed in table I. The following equation calculate how many bits can be inserted per each payload frame:

$$B = 8 - [(4 + 6 \times N + M \times N) \text{ Mod8}] \qquad (2)$$

where B is the maximum number of bits to be inserted in this RTP frame, N is the number of speech frames included in this RTP frame and M is the number of bits per speech frame obtained from table I according to the encoding rate. From equation 2, it is clear that the number of bits to be inserted in bandwidth-efficient mode depends on the number of speech frames per RTP frame and the chosen encoding rate for AMR-WB codec. For example, if the encoding bit-rate was 6.6 Kbit/s and the RTP frame contains 6 speech frames, then according to equation 2 there is no space left to insert any bits within this RTP frame.

Another example, if the encoding bit-rate was 8.85 Kbit/s and the RTP frame contains 3 speech frames, there exist a location for inserting 7-bits within the RTP frames. The maximum number of bits to be inserted in all cases of bandwidth-efficient mode will be 7- bits all over the single RTP frame.

II.3. Implementation for octet-aligned Mode

In the case of the octet-aligned mode, the payload header, as mentioned in subsection II-C, is con- structed only by one byte with 4-bits are reserved. The TOC contains two reserved bits. Each speech frame is padded to be byte aligned. So, the following equation shows how many bits can be inserted per each payload frame in octet-aligned mode:

$$B = 4 + (2 \times N) + (M \times N) \quad (3)$$

where B is the maximum number of bits to be inserted in this RTP frame, N is the number of speech frames included in this RTP frame and M is the number of mandatory padding bits per speech frame obtained from table III according to the encoding rate. Equation 3 proves that there is at least 4-bits per speech frame for inserting data (like location messages) within each RTP frame. Moreover, the AMR-WB coding rate of 8.85 Kbit/s has the largest capacity for carrying the additional data according to table III. For the coding rate of 8.85 Kbit/s, according to equation 3 and table III, at least 9 bits per each speech frame and 450 bits/s can be transmitted.

III. CONCLUSIONS AND FUTURE WORK

We have discussed in details a new novel technique for transmitting small messages and data over the AMR-WB speech coding frames for LTE standard. Also, we have proved that using the AMR- WB with rate of 8.85 Kbit/s in the octet-aligned mode is the most efficient setup for transmitting date without loss. Moreover, using the bandwidth- efficient mode is the least efficient setup for this pur- pose. According to [6], there exist another method for transmitting, receiving and storing the AMR- WB codec using special file format, which is not in the scope of this paper as it is outside the LTE standard. The new proposed technique may be used within this format with some adaptation as a future work.

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