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Investigation Of Emergency Call Performance Over Volte

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CHAPTER 1 INTRODUCTION

1.1 Background Information

1.1.1 Current Emergency Call

Starting from March 31st, 2018 [1], all new type M1 (passenger vehicles) and N1 (light-duty vehicles) sold in European Union member countries are all mandatorily equipped with Emergency Call (eCall) system [1]. The eCall system is the European Union's answer to combat vehicular accidents and to provide roadside assistance. In the event of a serious accident or in a situation where the vehicle operator or passengers are in need, the eCall system will be triggered either automatically or manually to dial 112 – a pan European emergency number to seek help.

Figure 1 shows the different stages and components of the eCall system. The Global Navigation Satellite System (GNSS) which includes American GPS, Russian GLONASS, European Galileo, and Chinese Beidou, provides the geo-location of the vehicle. Once the eCall in-vehicle-system (IVS) unit is triggered, a minimum set of data (MSD) is sent to the call center leveraging the cellular communication system. This call is an emergency call using the European 112 systems, which dials the nearest emergency center. The operators at the public-safety answer point (PSAP) call center ideally have both the voice channel open and real-time data uploaded to the call center. The open voice channel will re-assure the seriousness of the need and estimate the conditions of the vehicle operator and passengers. The real-time data may tell the operator what type of vehicle(s) is engaged in an accident and its location. It is also noted that with the current electrification of automobiles, battery packs have become an issue for firefighter and emergency personnel alike. When electric Tesla vehicles are involved in battery fires, such as the one that happened on March 25th, 2018, many times the battery engineers are called onsite to assist with the safety of fire suppression. This is one example to show that if the right type of information is uploaded to the call center correctly at time zero, many unnecessary issues can be avoided which can lead to potentially saving lives.

The fundamental goal of the eCall system is to reduce the time between the accident and the service provided, thus, to reduce the consequences of injury to prevent possible disabilities and deaths. A Swedish study into survivability in fatal road traffic crashes concluded that only 48% of those who died sustained non-survivable injuries. Out of the group who sustained survivable injuries, 5% were not located in time to prevent death, 12% could have survived had they been transported more quickly to the hospital and a further 32% could have survived if they had been transported quickly to an advanced trauma center [3]. In addition, many emergency service providers may receive multiple calls for each incident, for which they may have to respond each time and it is anticipated that eCall may enable them to manage the responses more effectively.

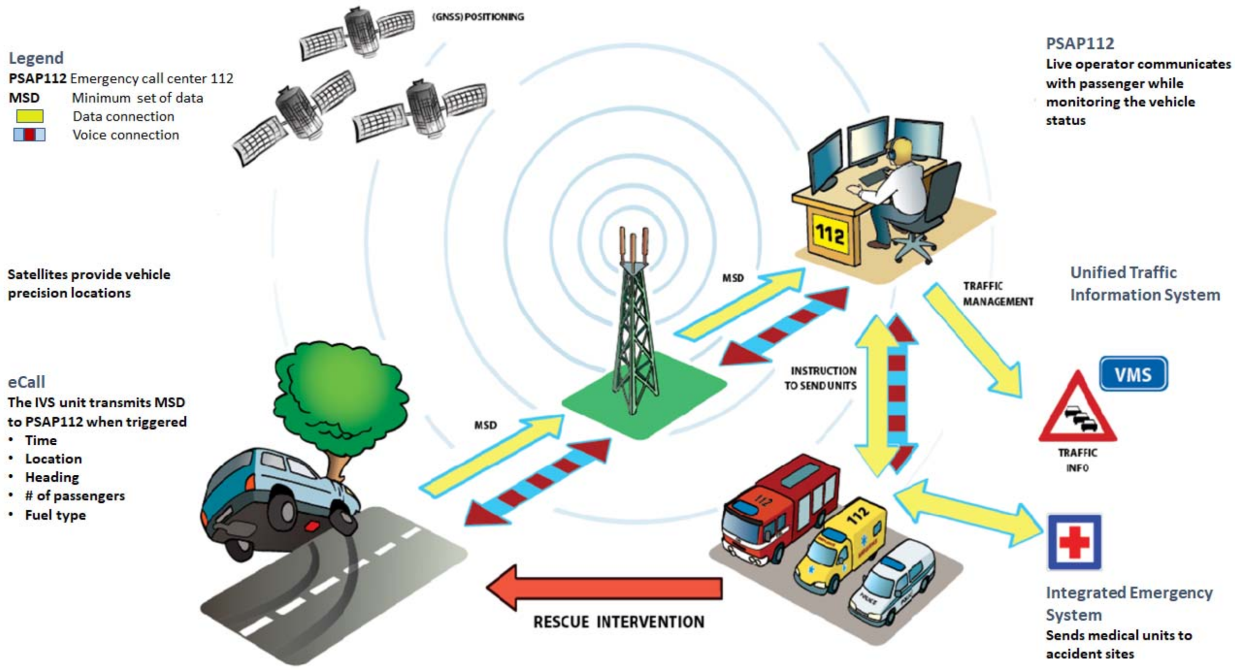

Figure 1. Overall System Diagram

Chart drawing from European HEERO pilot program [2]

This Finnish study also noted that through "the comparison of the 4-8% decrease in traffic accident fatalities arrived at in this study with the figures of other European studies one can see that the results are similar to the German (5%) and Dutch (7%) estimations. The estimations in Sweden (2-4%) and Great Britain (2%) are smaller and the estimate for the whole 25-member state EU area (5-15%) greater than the estimate in this study. The American estimation for the decrease in traffic accident fatalities based on field studies was smaller (2-3%) than in this study. The estimate made by the doctors was, however, greater (9-11%)".

The benefits to cost ratio (BCR) of eCall in Finland have been found to be in the range of 1:2 (minimum estimate) to 2:3 (maximum estimate). A UK benefit to cost analysis concluded that the universal fitment of eCall would result in more costs than benefits [4].

According to the European Transport Safety Council (ETSC) estimates, road deaths are fifty-one (51) per million inhabitants in the European Union in 2016, which amounts to thirty-seven thousand people to forty thousand. Additionally, car accidents leave over one-hundred and fifty thousand people disabled for life. ‘Unfortunately, the number of deaths on European roads is still unacceptably high. With eCall, emergency services’ response time will be reduced by 50% in rural areas and 40% in urban areas, leading to a reduction of fatalities estimated at up to 1,500 saved lives per year,’ said Czech MEP Olga Sehnalová, in charge of the dossier in the Parliament. This is based on the Electronic Medical Records and Genomics (eMERGE) project supported by automotive vehicle manufacturers such as BMW and Volvo in Germany, Italy, the Netherlands, Spain, Sweden, and the UK.

In the year 2014, the death rate is stagnant compared to the previous years; the year 2015 has a 1% increase and a 2% decrease in 2016 [5]. The target death rate reduction is 50% by the year 2020, and the gap is widening between the actual and the desired progress [6]. Figure 1 shows the percentage gap and Figure 2 reflects the percentage change of road deaths from 2015 to 2016.

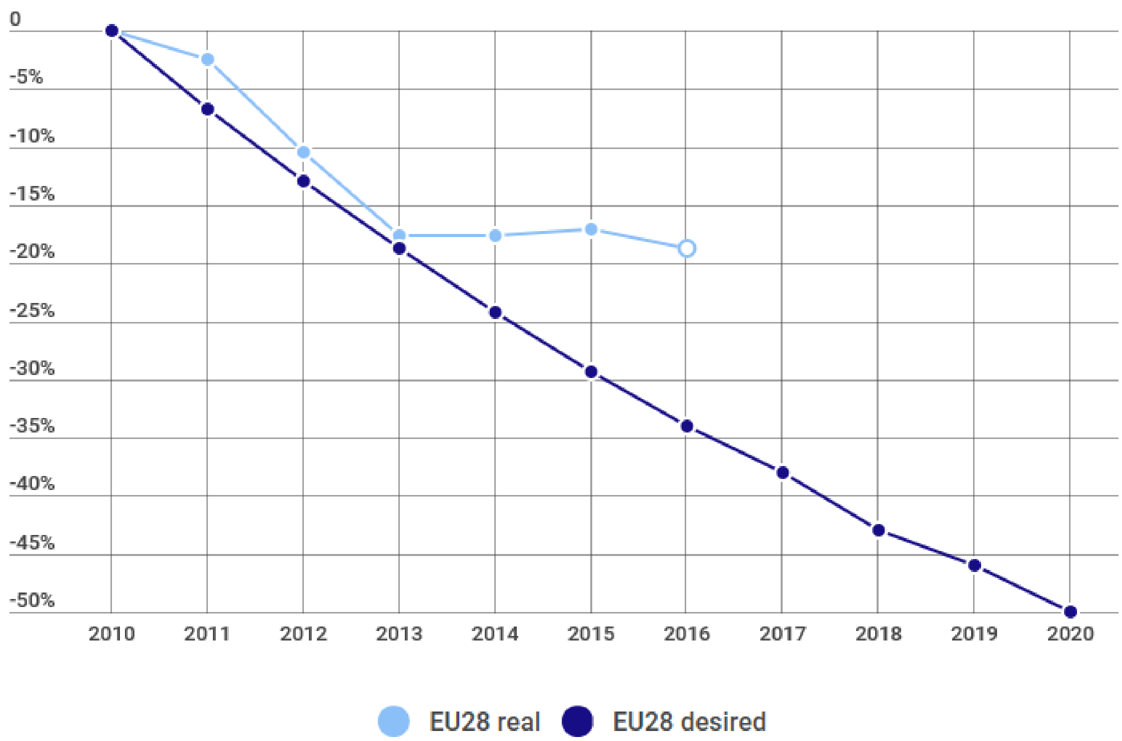

Figure 2a. [7] Gap Between Actual and Desired Progress Towards the EU 2020 Target

Chart from ETSC, Provisional data for 2016

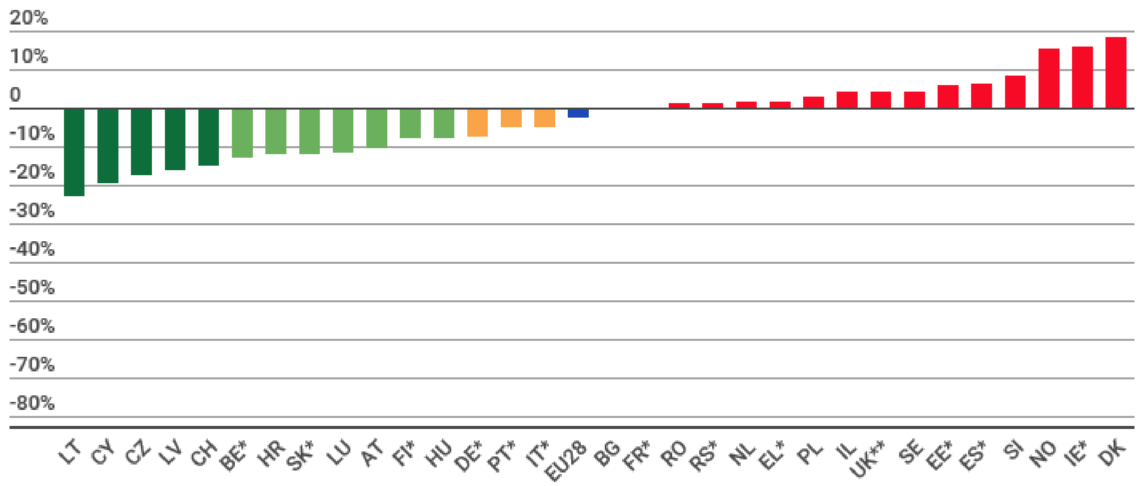

Figure 2b. [8] % Change in Road Deaths 2015-2016

Chart from ETSC, Provisional data for 2016

In a situation where the operator is unconscious or disabled, the automatic triggering mechanism is crucial in saving lives. In a situation where the automatic system is not engaged, yet the eCall service is needed, a button in the vehicle can be pushed to trigger the system. When the eCall system is triggered in a vehicle, the vehicle's precision location and heading information among other data will be transmitted to the local call center to be processed. Depending on the type of assistance needed, the call center will direct different types of services to the vehicle site. The European emergency number operates on a cross-border basis. There is a three-year transitional period allowed such that all the countries within the EU to establish eCall as a standard system, particularly to ease or erase the language barriers across all EU countries including Iceland, Norway, and Switzerland.

As with any new technology, there has been a strong concern regarding the privacy and safety of personal data since the inception of this concept. At the time of the initial vote, the defenders of personal data privacy had expressed strong concerns over misappropriation of this technology, movement tracking, and conversation recording, etc.

‘The eCall system paves the way for private services to use the data sent by the vehicle and may push authorities to use the system to monitor drivers. It would, therefore, have been right to obtain the consent of each vehicle owner to use the system, however, this possibility was not provided in the European regulation,’ said Jan Philipp Albrecht, a spokesperson for the Greens in the EP on Justice and Internal Affairs.

Faced with these arguments, the Parliament provided guarantees: the system “should be designed in such a way that no exchange of personal data between them is possible. Where provided, those services should comply with the applicable safety, security and data protection legislation and should always remain optional for consumers,” stated the European Parliament legislation on eCall.

The current EU eCall standard will be used as a reference to draft future emergency call standards while best to address some of the current performance issues.

1.1.2 Next Generation Emergency Call

A few years prior to the deployment of the current European eCall in 2018, the European Telecommunications Standards Institute (ETSI) had started the draft of the next generation eCall specification ETSI TR 103 140 [9] which would be a packet-based system since the 2G cellular network will be phased out eventually. There will be at least two decades of coexistence of both current eCall system and the future next-generation eCall system based on the typical average vehicle ownerships, and considering that the vehicles last more than fifteen years and a mobile handset will at least last 2-3 years. The ETSI TR 103 140 draft points out eleven (11) potential future issues while offering some possible solutions with uncertainty; however, the media of data transmission on Voice over LTE (Long Term Evolution) is the only recommended option.

LTE is a packet-switched (PS) only network where the current EU eCall is defined in circuit-switched (CS) the only network. These two networks are exclusive and both networks have their advantages and disadvantages. In general, the CS network has better coverage but much less data throughput thus less bandwidth efficiency, whereas the PS network will offer much better data throughput but without guaranteed delivery. In addition, the current eCall system must establish a voice channel first and send the data within this channel. Naturally, VoLTE is considered as the only option to fulfill the legacy requirements set by the current eCall standards.

In order to establish a VoLTE call, an Internet Protocol Multimedia Subsystem (IMS) which is a standalone system that resides outside of the LTE network will need to be connected to the LTE network through Packet Data Network (PDN) Gateway. The IMS can be considered as a service within LTE and the standard set of characteristics will be applied such as QoS (Quality of service) Class Identifier (QCI) and scheduling priority.

There is no platform yet to collect any data on eCall data transmission on VoLTE. The challenge is that there is no speech codec implemented yet to suit eCall applications. It is this paper’s intent to explore some of the fundamentals of speech data transmission on VoLTE by dissecting the original eCall message and code it in IMS format such that transmission can be achieved, and timing analysis can be carried out. Transmission time is the essence of any emergency call system.

1.2 Literature Review and Objectives

In order to evaluate the potential eCall transmission over VoLTE, the data structure of the current eCall will be referenced.

1.2.1 Technical Specification Review – Current eCall

The eCall technical specification can be found at the 3rd Generation Partnership Project (3GPP) group website [10]. The latest release of technical specification 3GPP TS 26.267 is version 14 (March 2017). The C-code reference is provided in 3GPP TS 26.268. The conformance testing specification is provided in 3GPP TS 26.269. The characterization report of the in-band modem is provided in 3GPP TS 26.969. This is a living document that has been updated several times to address some of the function, feature and performance issues. One issue remaining unaddressed is the system's response if either the transmission path or the reception path cannot establish an effective communication link.

An eCall system is a full-duplex communication system that transmits MSD (minimum set of data) from the IVS (in-vehicle system) to the PSAP (public safety answer point) through a cellular voice channel and the PSTN (public switched telephone network) as the uplink, and transmit feedback frames from the PSAP to IVS as the downlink, both by using the in-band modem since the voice and data share the same channel band.

Once the eCall system is triggered, the MSD is sent in between the speech conversations in the uplink. The downlink has no speech conversation but command codes for the IVS receiver. Given that speech communication is real-time, the expectation is that the emergency services will respond much more swiftly and efficiently, which could potentially save lives.

Definitions

Feedback frame – downlink transmitted signal containing feedback data, 140 ms in the time interval or 1120 samples with an 8kHz sampling rate

Frame (speech frame) – 20 ms in the time interval, corresponding to one AMR (adaptive multi-rate audio codec) or FR (full rate) speech frame at 160 samples with an 8kHz sampling rate

MSD – minimum set of data, contains a maximum of 140 bytes

MSD data frame – uplink transmission containing MSD with two variants, the fast modulator has a time interval of 1320 ms or 10560 samples, the robust modulator has 2320 ns and 18560 samples, both with the 8kHz sampled rate

Modulation Frame – two variants, the fast modulator has symbol transmission time of 2 ms or 16 samples with the 8kHz sampling rate, the robust modulator has symbol transmission time of 4 ms or 32 samples with an 8kHz sampling rate

Synchronization Frame – synchronization signal transmission, has a time interval of 260 ms or 2080 samples with an 8kHz sampling rate

Critical Requirements

Both the voice and MSD data must be routed to the same PSAP or emergency center. The transmission of the MSD data must be acknowledged, otherwise, retransmission must be performed. PSAP can request more data during the emergency call if needed. Any corrupted MSD component should not impact speech quality and the MSD should reach the PSAP within four (4) seconds measured with the end-to-end connection between IVS and PSAP. Throughout the emergency call, the PSAP shall send a confirmation to the IVS, request to IVS for MSD retransmission and instruct IVS to terminate the eCall. These three mandatory directives dictate the importance of the IVS downlink receiver.

1.2.1.1 System Architecture

1.2.1.1.1 Speech CODEC

A voice-band within the audio range is used for transmission of speech and it is approximated between 300 Hz to 3400 Hz. The bandwidth of a single voice-frequency transmission channel is thus typically defined to be 4 kHz, including the guard bands, with a Nyquist sampling rate of 8kHz used as the basis of PCM (pulse code modulation) system used for digital PSTN (public switched telephone network).

Speech coding is primarily used in digital telephone and Voice over IP applications [11]. The technique employed in speech coding is similar to those used in audio data compression where the knowledge in psychoacoustics is used to transmit data that is only relevant to the human vocal system. The psychoacoustic model simulates a high-quality lossy data compression that is also highly non-linear that would selectively remove parts of given digital audio to give it better-perceived sound quality. The speech coding is simpler than audio data compression, as a result, the speech codec is based almost entirely on statistical human speech information that emphasizes the preservation of intelligibility and speech pleasantness, at the same time with a very low coding delay [12].

The Adaptive Multi-Rate (AMR) audio codec is used in the GSM cellular system and the MSD data is encoded with AMR codec. The AMR frames contain 160 samples and are 20 ms long in the time interval. It is a hybrid speech coder that transmits both speech parameters and a waveform signal. AMR employs various coding techniques such as CELP (code-excited linear prediction), DTX (discontinuous transmission), (VAD) voice activity detection and CNG (comfort noise generation). GSM uses CELP as its standard speech coding, which contains two stages, a linear predictive stage (CELP) that models the spectral envelope and codebook-based model (algebraic code-excited linear prediction - ACELP) on the residual of the linear predictive model. In actual speech coding practices, both speech coding and channel coding methods are chosen as pairs to ensure better transmission, and this is a dynamic adaptive pair.

1.2.1.1.2 In-band Modem Architecture

After the successful establishment of an emergency voice call, the IVS receiver constantly monitors the speech decoder output for any PSAP requests such as MSD transmission. One such request is received, the IVS signal path switches to the speech coder and mutes the speech path during the silent gaps between naturally spoken words. This is also to prevent voice data from interfering with the MSD data. This is considered as the data ‘pull’ mode. The IVS can also trigger the MSD transmission by sending messages to the PSAP to request an MSD transmission which is considered as the data ‘push’ mode.

As shown in Figure 3, the modulated MSD data and speech data are coded by the speech codec for transmission on the IVS side. There is a switch that directs which data path to take. The coded messages will then be transmitted over an RF link. These data will go through the public land mobile network (PLMN) and eventually reach the PSAP, where this network is considered as transparent to the eCall messages. The PSAP will then separate the voice and MSD data. The MSD data will be demodulated and decoded, and feedback data will be sent to the IVS through the PSAP downlink. The PSAP feedback data will go through the same PSTN and reach the IVS, which will be demodulated and decoded. Since there is no voice information, there would be no separation of voice and data information bits.

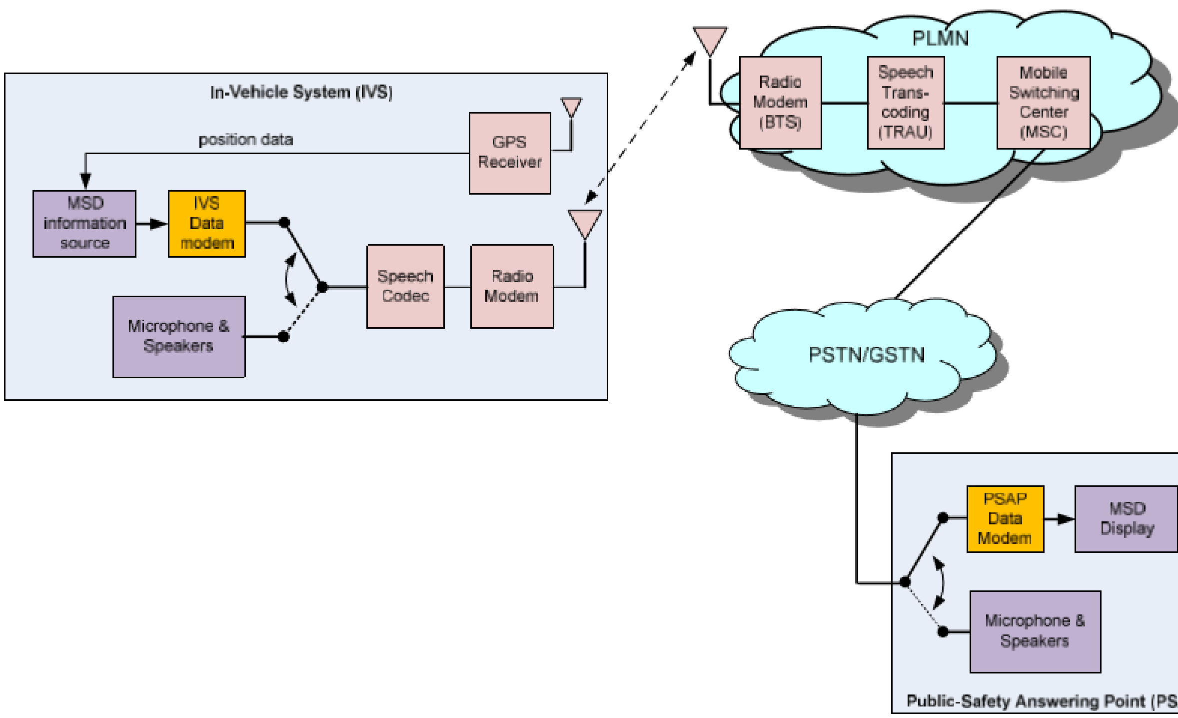

Figure 3. eCall system within the cellular system architecture

Chart from 3GPP TS 26.267

1.2.1.2 IVS Modem Operations and Functions – Transmitter

As illustrated in Figure 4, on the transmission side, the MSD must be CRC coded, then FEC coded before the modulator gets the data ready for speech encoder. On the receiving side, the speech decoded message will go through the demodulation and then decoded.

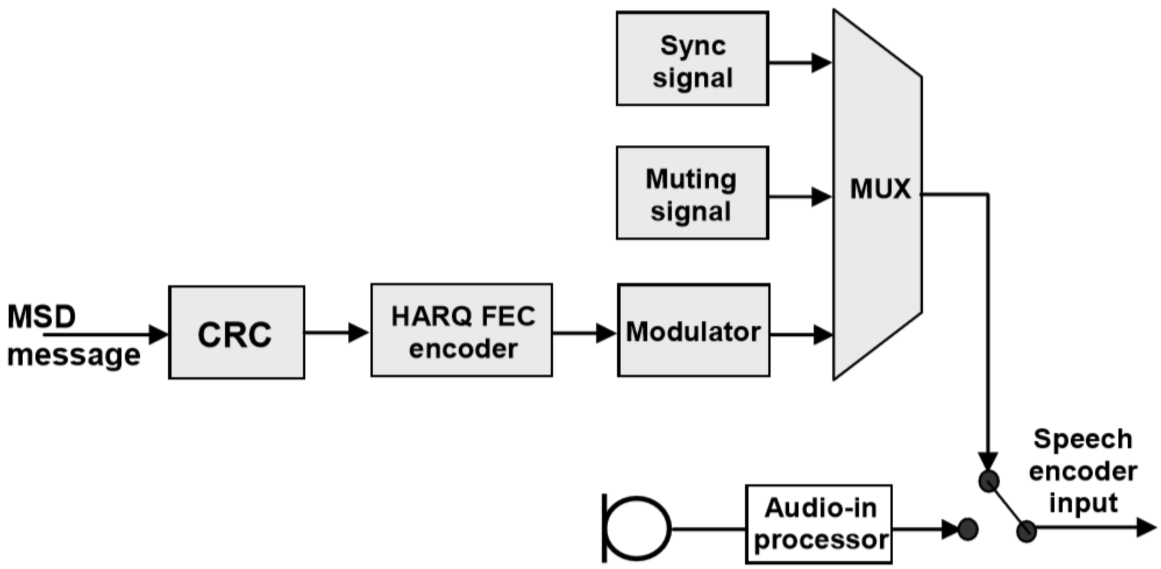

Figure 4. eCall IVS modem overview

Chart from 3GPP TS 26.267

1.2.1.3 MSD Message

The MSD message is 140 bytes long, or equivalently 1120 bits, represented by [13]:

$$a_i, i = 1, \ldots, 1120$$

MSD Data Frame Format

The fundamental time unit in this format is a frame or a time frame, which is 20 ms long. It is designed to match each AMR codec frame which contains 160 samples with a time interval of 20 ms. The modulator has two different modes of operation: fast and robust. When the RF link is good and the error rate is low, the fast mode is deployed. Otherwise, if the RF link is rather poor and the error rate is on the rise, then the robust mode is deployed, as a tradeoff between data rate and bandwidth. The MSD data frame consists of three data fields, our muting gaps, and three synchronization fragments, as shown in Table 1 [14].

Position	Fast Modulator Mode	Robust Modulator Mode
1	M1 (20ms) 1 frame of muting	M1 (20ms) 1 frame of muting
2	D1 (300ms) 15 frames of modulated data	D1 (600ms) 30 frames of modulated data
3	S1(80ms) 4 frames of sync fragment	S1(80ms) 4 frames of sync fragment
4	M2(40ms) 2 frames of muting	M2(80ms) 4 frames of muting
5	D2(300ms) 15 frames of modulated data	D2(600ms) 30 frames of modulated data
6	S2(80ms)4 frames of sync fragment	S2(80ms)4 frames of sync fragment
7	M3(40ms) 2 frames of muting	M3(80ms) 4 frames of muting
8	D3(320ms) 16 frames of modulated data	D3(640ms) 32 frames of modulated data
9	S3(80ms) 4 frames of sync fragment	S3(80ms) 4 frames of sync fragment
10	M4(60ms) 3 frames of muting	M4(60ms) 3 frames of muting
Sum	66 frames (1320ms)	166 frames (2320ms)

Table 1. MSD Data Format

Table from 3GPP TS 26.267

1.2.1.4 Synchronization Signal and Frame Format

The synchronization frame consists of the synchronization tone $s_t(n)$ and the synchronization preamble $s_p(n)$, where both signals are predefined. The synchronization tone is 64 ms long. For the fast modulator mode, the tone is set at 500Hz; and for the robust modulator mode, the tone is set at 800 Hz. The synchronization preamble follows the synchronization tone and the pulse sequence is known at the receiver.

According to the TS specification, this preamble sequence 'has been chosen to optimize the correlation properties in order to allow a very reliable detection and delay estimation at the same time' [15]. The basis of the synchronization preamble is a pseudo-noise (PN) sequence length 15 with values of (1, 1, 1, 1, -1, 1, -1, 1, 1, -1, -1, 1, -1, -1, -1). Each value is a signed 16-bit signal representation, equivalent to a pulse with an amplitude of 20000. The PN sequence is repeated five times with the outer sequence inverted, also the outer sequences have three inner pulses overlapped with sequence number 2 and number 4. The overlapping pulses are transmitted only once, as illustrated in the construction of the pulse sequence of the synchronization preamble in Figure 5 [16]. The synchronization preamble pulse sequence generation can be achieved in Matlab once the pulse sequence construction has been completed as shown in Figure 5. Note that 1 is represented by the '+' symbol and -1 is represented by the '-' symbol.

Five pulse sequences are needed in the synchronization preamble, and twenty-one (21) zero paddings are added in between pulses. In addition, there are seventy-one (71) zero samples placed before the first pulse. The resulting preamble has a duration of: 71+(75-6) +(68x21) = 1568 samples or 196 ms. The overall synchronization frame is the synchronization tone plus the synchronization preamble: 64 ms + 196 ms = 260 ms, which is thirteen (13) time frames. Both the synchronization tone and the synchronization preamble are hard-coded into the ROM table for faster processes.

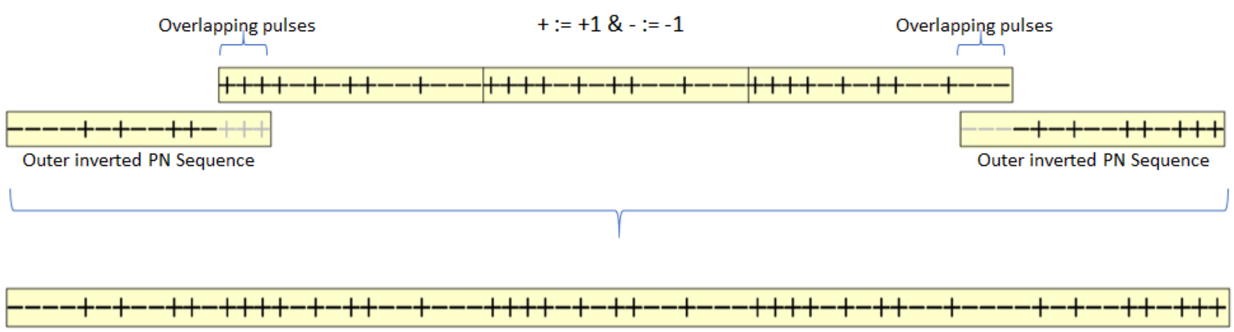

Figure 5. Synchronization Preamble Pulse Sequence Construction

Chart from 3GPP TS 26.267

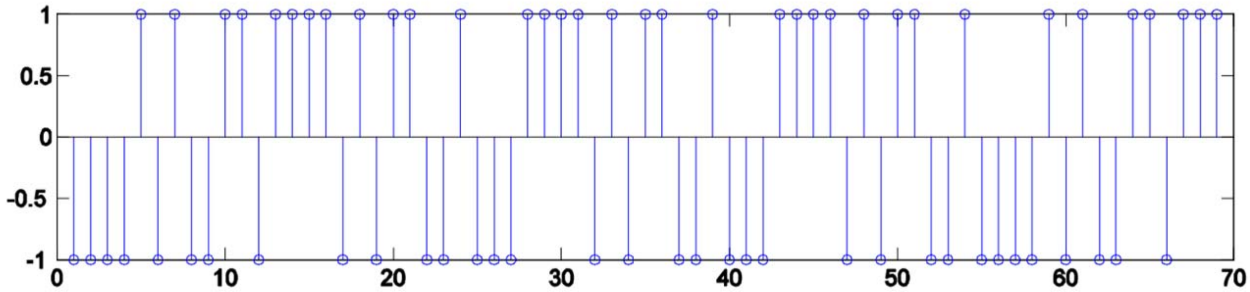

Figure 6. Synchronization Preamble Pulse Sequence Generation

Chart from 3GPP TS 26.267

The last 576 samples of the synchronization preamble, prepended by 64 zero-padding are inserted into the MSD data frame as Position 9 on both the fast modulator mode and the robust modulator mode, as S3 frame which is 640 samples long or 80 ms in the time interval. The uplink data format is shown in Table 2.

MSD

SF	M1	UL Data 1	S1/M2	UL Data 2	S2/M3	UL Data 3	S3/M4

Table 2. Uplink Data Format

Table from 3GPP TS 26.267

1.2.1.5 Modulation Format

The encoded MSD message bits are divided into groups of three bits to form symbols and each modulates on one basic waveform which also corresponds to one modulation frame. As indicated earlier, there are two modulator modes: fast and robust. The default transmission mode is the fast modulator mode and the robust modulator will only serve as a backup method when the fast modulator mode transmission fails. For the fast mode, each frame or speech frame contains ten (10) modulation frames (10 symbols = 30 bits) with a data rate of 1500 bits/sec, not counting the muting gaps and the synchronization frames. For the robust mode, each frame or speech frame contains five (5) modulation frames (5 symbols = 15 bits) with a data rate of 750 bits/sec, also not counting the muting gaps and the synchronization frames.

Choosing the right type of pulse shaping wave is a bit tricky for the eCall application. The GSM Full-rate has the largest bandwidth at 13 kbits/sec and AMR-NB (adaptive multi-rate narrowband) codec used in the GSM network is even less. The signal's bandwidth will increase with the modulation rate. When the signal's bandwidth becomes larger than the channel bandwidth, the channel will introduce inter-symbol interference (ISI). There is no literature yet describing the mathematical fundamentals of the adaption of root raised cosine waveform that is used in eCall.

1.2.1.6 Raised Cosine Filter

The raised cosine filter is an example of a low-pass Nyquist filter that has the property of vestigial symmetry. One of the Nyquist filter's properties is to eliminate ISI. It is shown by Nyquist that the frequency characteristics have odd symmetry at the cutoff frequency and the impulse response has zeroes at uniform spaced intervals [17]. With a raised cosine filter, the effects of jitters may be minimized, and it is much simpler to attain in hardware. The definition of the equations in both the frequency domain and the time domain are as follows:

$$H(f)=\begin{cases} T & 0\leq |f| \leq \dfrac{1-\beta}{2T} \\ \dfrac{T}{2}\left\{1+\cos\left[\dfrac{\pi T}{\beta}\left(|f|-\dfrac{1-\beta}{2T}\right)\right]\right\} & \dfrac{1-\beta}{2T}\leq |f| \leq \dfrac{1+\beta}{2T} \\ 0 & |f| \leq \dfrac{1+\beta}{2T} \end{cases}$$

$$h(t)=\frac{\sin(\pi t/T)}{\pi t/T}\frac{\cos(\pi\beta t/T)}{1-(4\beta^2t^2/T^2)}$$

Where the sampling time is T, bandwidth used for the ideal low pass filter is $B = 1/(2T) = f_c$. Since the impulse response has a sinc term, it ensures that it has zero crossings similar to an ideal low pass filter. The second term decays in time hence reduces the tails which reduces the impact of jitters. The overall channel transfer function must be raised cosine and one to achieve this is to take the square root of the raised cosine filter in the frequency domain and use this new filter on both the transmitting side and the receiving side. By taking the square root of the raised cosine filter response, the equations of frequency domain and time domain become:

$$H(f)=\begin{cases}\sqrt{T} & 0\le |f|\le \frac{1-\beta}{2T}\\ \sqrt{\frac{T}{2}\left\{1+\cos\left[\frac{\pi T}{\beta}\left(|f|-\frac{1-\beta}{2T}\right)\right]\right\}} & \frac{1-\beta}{2T}\le |f|\le \frac{1+\beta}{2T}\\ 0 & |f|\le \frac{1+\beta}{2T}\end{cases}$$

$$h(t)=\frac{2\beta}{\pi\sqrt{T}}\frac{\cos\left[(1+\beta)\pi t/T\right]+\frac{\sin\left[(1-\beta)\pi t/T\right]}{4\beta t/T}}{1-\left(4\beta t/T\right)^{2}}$$

The autocorrelation function of a root raised cosine function is the same as the raised cosine function:

$$R(\tau)=T\left[\frac{\sin(\tau/T)}{\tau/T}\frac{\cos(\beta\pi\tau/T)}{1-(2\beta\tau/T)^{2}}-\frac{\beta}{4}\frac{\sin(\beta\tau/T)}{\beta\tau/T}\frac{\cos(\pi\tau/T)}{1-(\beta\tau/T)^{2}}\right]$$

1.2.1.7 Modulation Format (continued)

The eCall system will use waveforms stored in the ROM to simulate the root raised cosine signal by assigning different values at each sampled point. Hence there is a waveform for each uplink modulator mode.

The uplink wave for the fast modulator mode is:

P_{UL} *(n)* = (0 0 0 40 -200 560 -991 -1400 7636 15000 7636 -1400 -991 560 -200 40)

The uplink wave for the robust modulator mode is:

P_{UL} *(n)* = (0 0 0 0 0 40 -200 560 -991 -1400 7636 15000 7636 -1400 -991 560 -200 40 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0)

The mapping relation between the symbols and the waveform is given by a cyclic right-shift of *k* samples as $p \rightarrow k$ with a sign of the basic waveform *q*.

For the fast modulator mode, the waveform equation is: $W_{UL} = q(p(n) \rightarrow k, n = 1,\ldots,15)$

For the robust modulator mode, the waveform equation is:

$$W_{UL} = q(p(n) \rightarrow k, n = 1,\ldots,31)$$

A mapping table between the symbol and waveform is presented as the following, as shown in Table 3.

Symbol		Uplink waveform fast modulator mode		Uplink waveform robust modulator mode	
symbol #	bits	sign q	cyclic shift	sign q	cyclic shift
0	000	1	0	1	0
1	001	1	4	1	8
2	010	1	8	1	16
3	011	1	12	1	24
4	100	-1	12	-1	24
5	101	-1	8	-1	16
6	110	-1	4	-1	8
7	111	-1	0	-1	0

Table 3. Uplink Symbol Modulation Table

Table from 3GPP TS 26.267

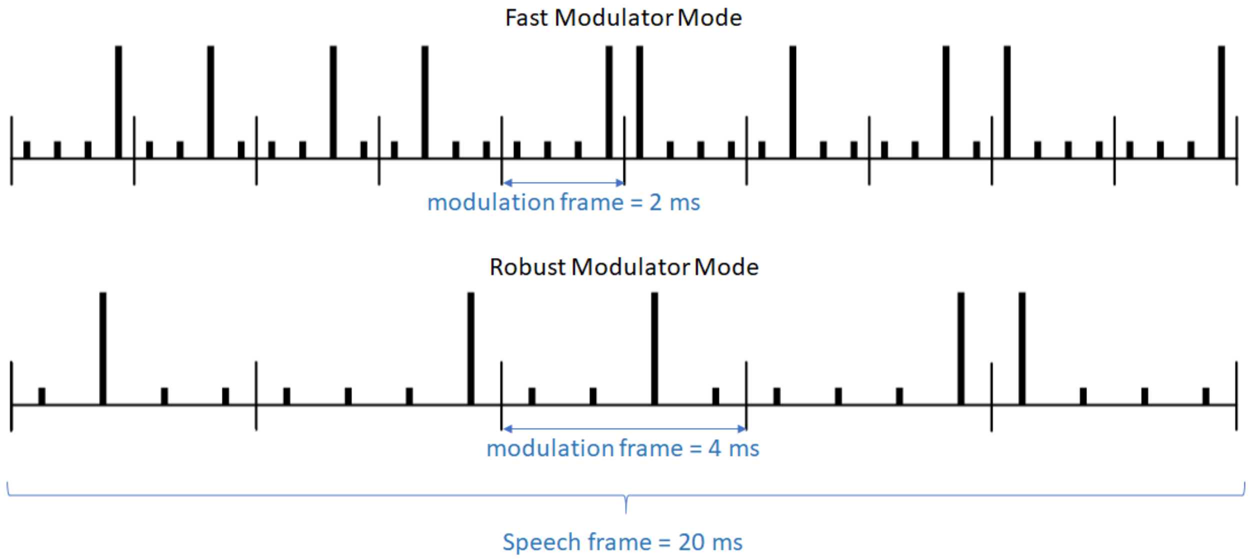

Figure 7. Time Slot Structure of Uplink Modulators

Chart from 3GPP TS 26.267

1.2.1.8 CRC

The eCall message is embedded in the data transport layer between the IVS and the PSAP. Messages shorter than 140 bytes must be padded, typically with zeroes before being sent to the IVS transmitter. The CRC (cyclic redundancy code) block uses the entire MSD message which is 1120 bit long to calculate the parity bits. The length of the CRC-encoded word is 1148 bits, thus the degree of the generator polynomial is 28 (=1148-1120). The parity bits are generated with the following cyclic generator polynomial [13]:

$$\begin{aligned} g_{CRC28}(D) = {} & D^{28} + D^{26} + D^{24} + D^{23} + D^{18} + D^{17} + D^{16} \\ & + D^{15} + D^{14} + D^{11} + D^{8} + D^{4} + D^{3} + 1 \end{aligned}$$

To follow the same standard notations used in the spec, bits in the MSD are denoted by $a_1,\ldots,a_k,$ and the parity bits are denoted by $p_1,\ldots,p_{28}$. The encoding polynomial formula becomes:

$$a_1 D^{k+27} + a_2 D^{k+26} + \ldots + a_k D^{28} + p_1 D^{k+27} + p_2 D^{k+26} + \ldots + p_{27} D^1 + p_{28}$$

1.2.1.9 H-ARQ FEC Encoder

H-ARQ (hybrid automatic repeat request) encoder block performs the function of bit scrambling, turbo coding, and H-ARQ scheme. The H-ARQ is a combination of high-rate forward error correction (FEC) coding and ARQ error control algorithm. With the standard ARQ algorithm, redundant bits are first generated by error-detecting codes such as CRC, then these bits are added into the data stream. With H-ARQ, the original message is first coded with FEC coding, then the parity bits are generated. However, the parity bits may or may not be sent depending on the request received. When the coded data block is received, the receiver first decodes the error-correction code. If the channel quality is good enough, all transmission errors should be correctable, and the receiver can obtain the correct data block. If the channel quality is bad, and not all transmission errors can be corrected, the receiver will detect this situation using the error-detection code, then the received coded data block is rejected and a re-transmission is requested by the receiver. In eCall, the H-ARQ can generate eight different redundancy versions (RV), *rv0, …, rv7*, from the channel coded bit buffer. Each RV consists of a subset of 1380 bits in the bit buffer. Messages *rv0, rv2, rv4* and *rv6* consist of MSD+CRC total bits. The redundancy version will increase with each transmission regardless of whether a success transmission was achieved.

The error-detection codes are typically a few bytes long, where the FEC can easily double or triple the message length. In the eCall application where FEC turbo codes are used, the rate is 1/3 which triples the original messages. In a more sophisticated pattern, the encoded message alternates between data bit along with error detecting bit and FEC parity bits only. When the first transmission is received error-free, then the FEC parity bit is never sent. Two subsequent transmissions can be combined for error correction if neither is error-free.

In digital communications, a scrambler transposes, inverts or encodes the original message at the transmitter side to make it unintelligible to any receiver without the descrambling algorithm. The scrambler can be placed before or after FEC and it needs to be distinguished from encryption. In the eCall application, bit scrambling is applied to MSD prior to turbo encoding, with the following function:

$$a_s(i) = a_{crc} XOR b_{scm}, i = 0, \ldots, 1147$$

Where a_{crc} is the MSD and CRC bitstream, and b_{scm} is the scrambling sequence.

In information theory, turbo codes are considered a class of high-performance FEC codes closest approaching the channel capacity, known as the Shannon limit – a theoretical maximum of code rate that given a specific noise level, reliable communication is still achievable. The name ‘turbo code’ comes from the feedback loop used during the decoding process within turbo codes, similar to the exhaust-driven turbocharged engines. Turbo codes are used in 3G/4G (UMTS/LTE) and satellite communication systems to best achieve reliable information transfer over either bandwidth-constrained or latency-constrained communication channels with data-corrupting noises.

There are many different instances of the turbo code, such as Parallel Concatenated Convolutional Codes (PCCC), Serial Concatenated Convolutional Code (SCCC) and Repeat-accumulate codes, using different component encoders, interleavers, puncture patterns, and input/output ratios. For a PCCC code, there are two identical encoders and two identical decoders. The encoder outputs will only differ the parity bits for each payload sent, due to the process that one encoder has n/2 parity bits for the payload and the other has n/2 parity bits for a known permutation of the payload, and this permutation is carried out by an interleaver. The decoder is constructed in a similar way with two decoders interconnected to each other, in series from decoder 1 to decoder 2, not in parallel.

An interleaver sits in between the two decoders to scatter error bursts coming from the first decoder. The output of decoder 2 will be fed into decoder 1 again to complete one iteration. The key innovation of turbo codes is how they use the likelihood data to reconcile differences between the two decoders. Each of the two convolutional decoders generates a hypothesis (with derived likelihoods) for the pattern of *m* bits in the payload sub-block. The hypothesis bit-patterns are compared, and if they differ, the decoders exchange the derived likelihoods they have for each bit in the hypotheses. Each decoder incorporates the derived likelihood estimates from the other decoder to generate a new hypothesis for the bits in the payload. Then they compare these new hypotheses. This iterative process continues until the two decoders come up with the same hypothesis for the *m-bit* pattern of the payload, typically in 15 to 18 cycles [18].

The eCall system FEC turbo code used is a Parallel Concatenated Convolutional Code (PCCC), also the first turbo code that was discovered, with one turbo code interleaver and two eight-state constituent encoders with the following polynomial [19].

$$g_0(D) = g_1(D) = 1 + D^2 + D^3$$

The Turbo coder's coding rate results at 1/3. The shift register of the first eight-state encoder will be initialized with zeros prior to encoding the MSD, while the output bits from the internal interleaver will be sent to the second eight-state encoder. The Turbo coder is constructed in Figure 5.

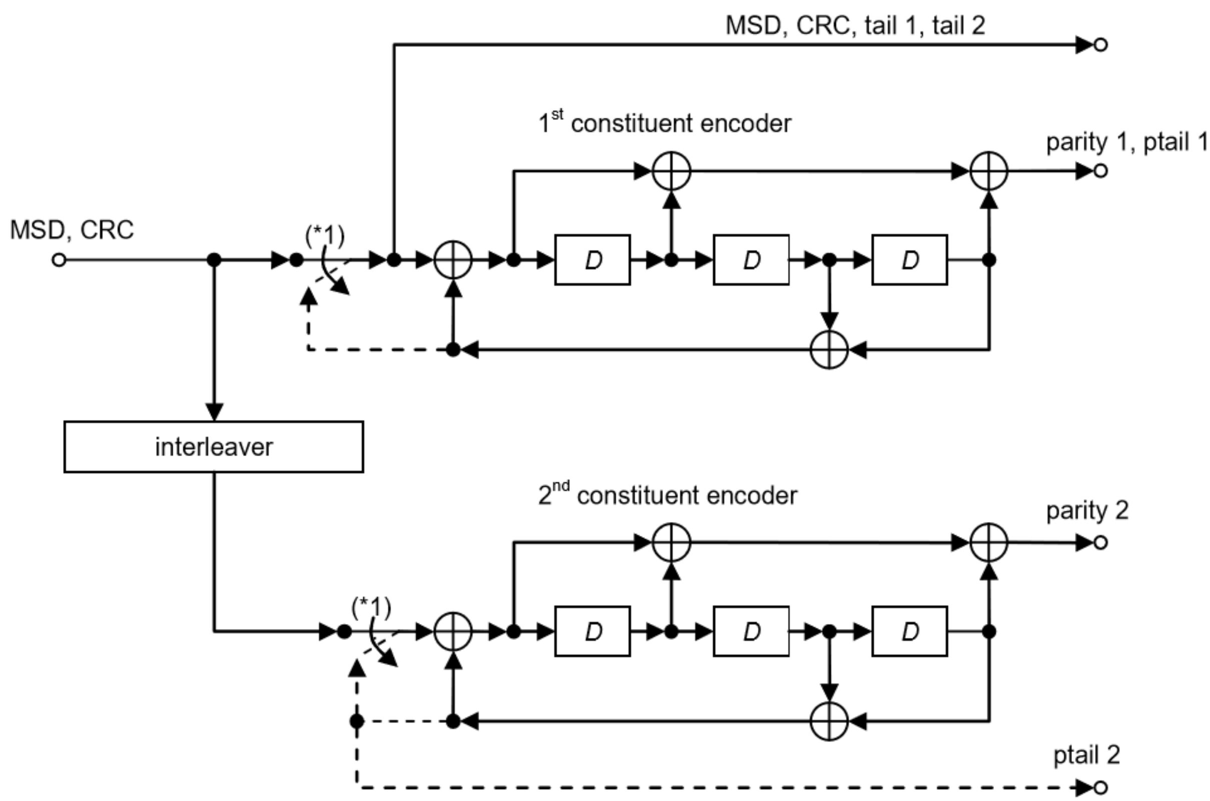

Figure 8. Structure of rate 1/3 Turbo encoder

Chart from 3GPP TS 26.267

Trellis termination of turbo codes is critical for maintaining its good performance, especially for short information blocks when the deterministic interleaver is sought to reduce the complexity of signaling the interleaver permutation [20]. Dotted lines imply trellis termination, which is performed by taking the padded tail bits *ptail1* and *ptail2* from the shift register feedback after the MSD is encoded. The first three tail bits terminates the first eight-state constituent encoder while the second eight-state constituent decoder is temporarily disabled. The last three tail bit terminates the second eight-state constituent encoder while the first eight-state constituent encoder is disabled. The three tail bits are generated from the FEC encoder states and the parity blocks are generated with the same convolutional encoder. The interleaver consists of bit-input to a rectangular matrix with padding, intra-row and inter-row permutations of the rectangular matrix, and bit-output from the rectangular matrix with pruning [21]. The encoder output is fed to the channel coded bit buffer in the following format [22], shown in Table 4.

MSD+CRC	tail1	tail2	Parity 1	Ptail1	Parity 2	Ptail2

Table 4. Channel Coded Bit Buffer Format

Table from 3GPP TS 26.267

Data collected in this format now is ready to be appended with voice data, then modulated and transmitted over the RF link.

1.2.1.10 Uplink Signal and Retransmission

The uplink signal starts with a synchronization frame (SF), then the MSD messages or MSD redundant message will follow, with increasing revision counter, all using the fast modulator mode as the default. The signal format is shown in Table 5. When the channel condition is ideal, the MSD data should be decoded by the PSAP receiver and the PSAP is able to send an ACK message down to the IVS.

However, if the ACK message is not received or decoded, the IVS will keep sending MSD messages, each time with a different version of the incremental redundancy (IR), up to eight different versions. On the PSAP side, if the MSD message is not decoded after one full transmission cycle (all eight transmissions), then the PSAP will instruct the IVS to restart the transmission using the robust modulator mode and internal to the PSAP, the IR buffer will be reset to recombine all the MSD messages. If the PSAP loses its signal synchronization, it can interrupt IVS uplink transmission by sending a START command, which will also start the IVS transmission in the default fast modulator mode. Overall, the IVS will transmit MSD messages when START or NACK messages are received, and the transmission will terminate when the ACK message is received. Since this is a two-way full-duplex asymmetrical communication, the questions arise when no valid PSAP message can be decoded.

SF	MSD rv0	MSD rv1	MSD rv2	MSD rv3	MSD rv4	MSD rv5	MSD rv6	MSD rv7

Table 5. Uplink Signal Format

Table from 3GPP TS 26.267

1.2.1.11 IVS Modem Operations and Functions - Receiver

PSAP messages START, NACK, link-layer ACK and higher-layer ACK (reserved) are sent to the IVS to demodulate and decode. After decoding with the speech codec, the MSD data is separate from the voice data. The MSD data will go through a synchronization detector/tracker for further process. In the demux section, the original message frames, synchronization frames, and muting frames are all extracted.

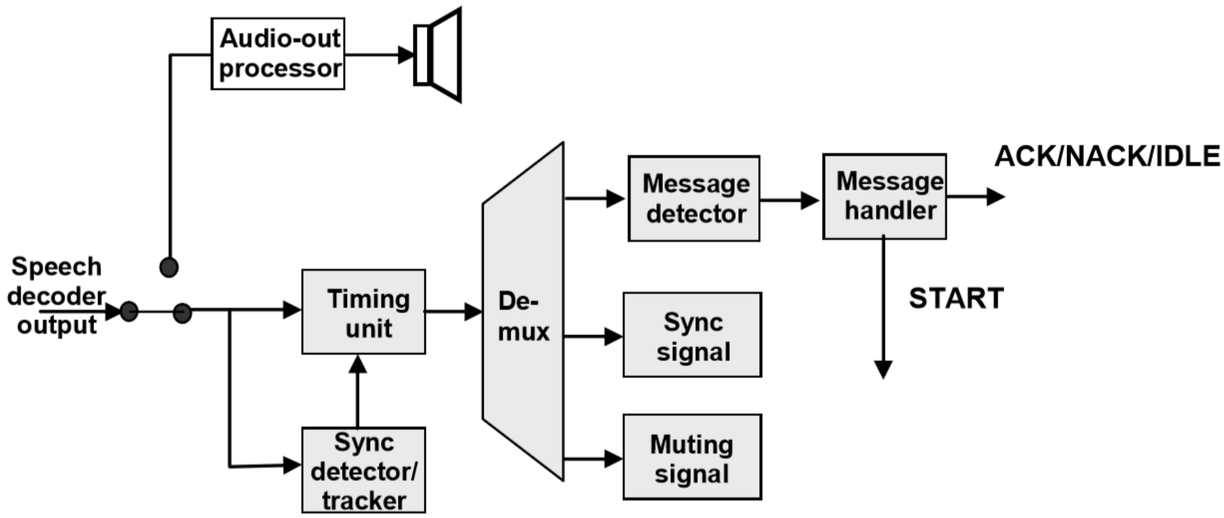

Figure 9. IVS Receiver Block Diagram

Chart from 3GPP TS 26.267

1.2.1.12 Synchronization Detector/Tracker

The eCall system is a full-duplex but asymmetrical communication system. The uplink and downlink data formats are different, even though the downlink synchronization frame is identical to the uplink synchronization frame where only the 500 Hz frequency tone is used for the synchronization tone.

The synchronization detector/tracker performs three different types of functions:

1) Synchronization: detection of the start of an eCall data transmission
2) Synchronization: identify the data frame timing. This timing will determine the sampling start timing which is crucial in determining the incoming bit
3) Tracking: frame timing tends to shift and with each incoming data frame, there is a synchronization frame. The tracker will track this timing shift and identify a new timing for each frame

The IVS synchronization detector/tracker will process three (3) synchronization frames in sequence to avoid false detection. A detection flag is set to positive if synchronization preambles are detected successfully in all three frames. This is a feature required to prevent false detection of START message and keep synchronization as precise as possible. From Figure 6, given the pulse sequence of the synchronization preamble, the autocorrelation properties of the synchronization preamble is generated, as shown in Figure 10. The synchronization algorithm for the preamble will be checking the distances between peak pairs of peaks (2, 4) and peaks (1, 5). The synchronization preamble can only be considered as detected if

1) Correct distances are detected between peaks
2) Amplitude difference of no more than a factor of 3

3) Average more than half of the global maximum synchronization filter output
4) Or, detect an additional peak

To distinguish the link-layer ACK message and the higher-layer ACK message (reserved), the peak signs are used. If an inverted sign is detected at the beginning of the reception, then the synchronization detector will assume that all the bits have a reverse sign and all samples received will multiply by -1 for the rest of the MSD data transmission.

The downlink feedback messages are shorter compared to the uplink MSD messages, hence the synchronization is only required once for every transmission, instead of three for an uplink MSD message. However, the synchronization is checked constantly by evaluating the received peak positions with the expected peak positions. This synchronization tracking feature will re-use the original synchronization algorithm and monitor the cross-correlation of the incoming signals and the known synchronization sequence within a data interval, and this data interval is ±480 samples. The modulated feedback message is 480 samples long and equivalent to 60ms.

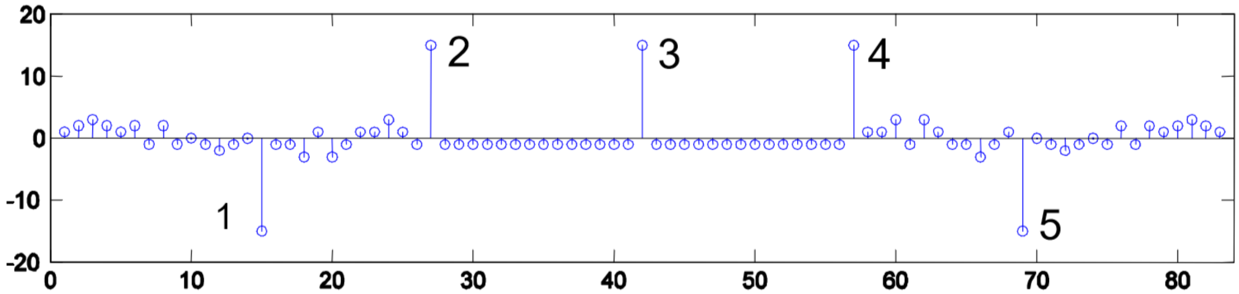

Figure 10. Autocorrelation Properties

Chart from 3GPP TS 26.267

1.2.1.13 Timing unit

The incoming signal bits need to adjust its timing based on the detector/tracker input where the synchronization information is extracted.

1.2.1.14 De-multiplexing

The original MSD message will be extracted by removing the muting and synchronization bits.

1.2.1.15 Demodulation and FEC Decoding

The demodulation and decoding are done with a single correlator detector that matches to the downlink modulated waveforms stored on the ROM. After the correlation process, a maximum likelihood decision will be derived on the feedback messages received, only one out the four messages will be the correct message, the equations are as follows [23];

$$metric(k) = \sum_{0}^{479} pcm\_data(i) * dlPCMMatch(k)(i), k = 0,1,2,3 \quad (1)$$

$$message = \arg\max_{k}(metric(k)) \tag{2}$$

Equation (1) is the correlator. *pcm_data(i)* is the incoming message and *dlPCMMatch(k)* is the stored basis downlink waveform in the IVS. The correlator will multiply the waveforms and integrate the result over 480 samples which is the length of the modulated feedback message. Function (1) will generate four different correlation values and function (2) will select the highest value to be the decoded messages. There might be situations where the synchronization is detected, but the demodulator is unable to reach valid messages. For the ACK and the NACK messages, the unreachable messages are all marked as reliable and ignored. For the START message, the first six unreliable messages are ignored and the subsequent ones are not compared. The reserved higher-layer ACK message is considered reliable if two consecutive messages are detected with identical data or three successful consecutive detections.

1.2.1.16 Message Handler

This block manages how the messages are received and processed. The MSD is sent upon the synchronization lock and three reliable START messages decoded. The IVS will keep sending the MSD when the NACK message is received. The IVS will stop transmission of the MSD only when ACK messages are received or the preamble is not detected.

1.2.1.17 PSAP Modem Operations and Functions - Transmitter

On the incoming path, after being triggered by the IVS to send MSD request, the PSAP receiver continuously listens for incoming signals from this IVS. Data need to be detected and synchronized before being demodulated and then decoded. The H-ARQ decoder will first process the initial MSD transmission data with any subsequent transmission data, combine them if necessary, then decode the FEC to extract the information bits, and finally estimates the CRC encoded MSD message bits. In a successful transmission, the PSAP operator will be able to correctly extract the MSD information and no re-transmission request will be sent to the IVS. However, if the CRC detects a non-recoverable error in the MSD bits, then the PSAP receiver will send NACK to the IVS for retransmission.

1.2.1.18 PSAP Transmitter

The feedback messages will be generated by the PSAP transmitter for downlink transmission. The functional blocks identical to the IVS transmitter. The only difference lies in the message and message coding.

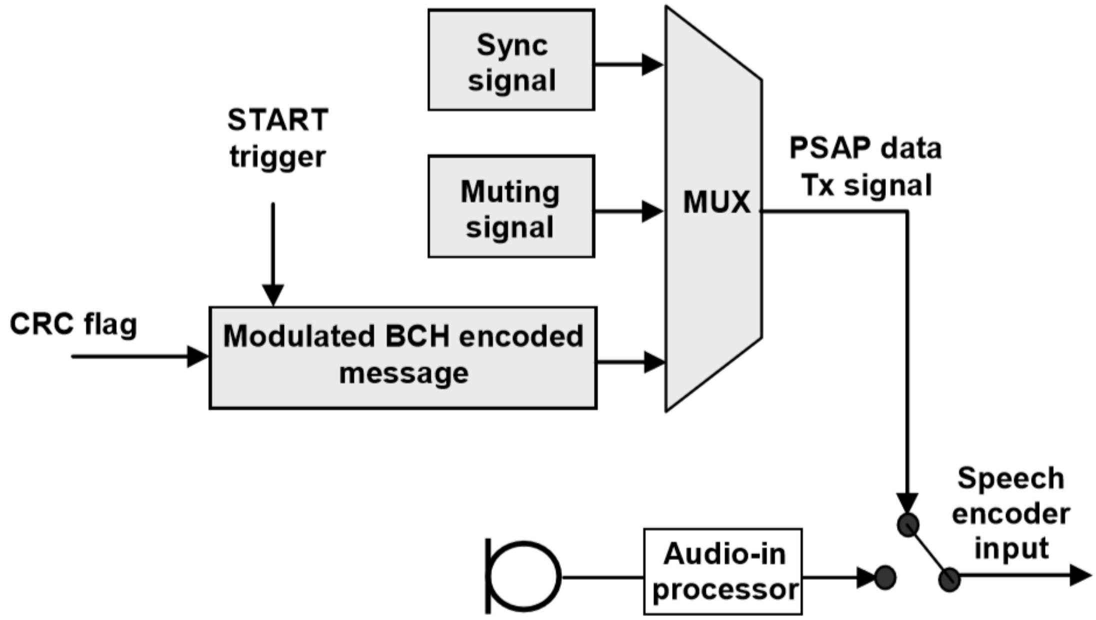

Figure 11. eCall PSAP Transmitter

Chart from3GPP TS 26.267

1.2.1.19 Message Encoding

There is a total of four different messages that can be sent to the IVS: START, NACK, link-layer ACK and higher-layer reserved for later definition. They are defined as [25]:

1) START signal – triggers start of the IVS MSD transmission
2) NACK – negative acknowledgment upon CRC check failure
3) Link-layer ACK – positive acknowledgment upon CDC check success
4) Higher-layer ACK – reserved

1.2.1.20 BCH Coding

The four feedback messages generated consist of three link-layer messages and one higher-layer message. These messages are encoded with a shortened (60, 4) BCH (Bose-Chaudhuri-Hocquenghem) code, which is a class of cyclic error-correcting codes, as illustrated in Table 6.

Message	Layer	CRC Flag	Binary	BCH Coded
START (DL1)	Link	NA	0000	A72 F298 41FA B376
NACK (DL2)	Link	0	0001	4C4 1FD6 6ED2 7179
ACK (DL3)	Link	1	0010	97A 8C41 FAB3 7693
Reserved (ACK)	Higher Link	NA	0011	DBE 9397 9461 07EA
Not used			0100-1111	

Table 6. Downlink Message

For the link-layer messages, the downlink message data format consists of the synchronization frame (13 frames) plus eight frames and each frame is 20 ms long in a time interval, as shown in Table 7 and Table 8:

SF	M1	M2	M3	DL	DL	DL	M2

Table 7. Downlink Message Format (link-layer)

Position	Frame
1	SF- Synchronization
2	M1 - Muting
3	M2 - Muting
4	M3 - Muting
5	DL – link-layer data
6	DL – link-layer data
7	DL – link-layer data
8	M3 - Muting

Table 8. Downlink Message Frame (link-layer)

When the PSAP transmitter is ready to send a higher-layer ACK message, the synchronization frame is inverted, and all the messages are multiplied by -1. The message data format is shown in Table 9 and Table 10.

SF	M1	DL	DL	DL	DL	DL	DL

Table 9. Downlink Message Format (higher layer)

Position	Frame
1	SF - Synchronization
2	M1 - Muting
3	DL – link-layer or higher layer data
4	DL – link-layer or higher layer data
5	DL – link-layer or higher layer data
6	DL – link-layer or higher layer data
7	DL – link-layer or higher layer data
8	DL – link-layer or higher layer data

Table 10. Downlink Message Frame (higher layer)

The PSAP handles the received signals in three steps. First, the PSAP receiver will try to detect the synchronization frame. If it does not detect the synchronization frame, then it will retransmit the START message multiple times until the receiver detects the synchronization frame. Secondly, after the synchronization is detected, the receiver will send a series of NACK messages, until a successful CRC check of the MSD has been cleared. After the MSD is successfully decoded, the PSAP will send an ACK message to the IVS receiver.

1.2.1.21 Modulation

For the IVS transmitter, in fast mode, each speech frame is equivalent to 10 modulation frames; in the robust mode, each speech frame is equivalent to 5 modulation frames and each symbol carries 3 bits of binary information. For the PSAP transmitter, each speech frame is equivalent to 5 modulation frames and each symbol carries 4 bits of binary information. The waveform is 4 ms long and the modulation data rate is 1000 bps. The basic downlink waveform is:

$P_{DL}(n)$ = (40 -200 560 -991 -1400 7636 15000 7636 -1400 -991 560 -200 40 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0), for n = 0, … , 31

This downlink waveform is the robust modulator mode basic wave function but shifted by k samples, as denoted in the mapping table, shown in Table 11:

Symbol		Downlink waveform	
Symbol #	bits	Sign q	Cyclic shift k
0	0000	1	0
1	0001	1	4
2	0010	1	8
3	0011	1	12
4	0100	1	16
5	0101	1	20
6	0110	1	24
7	0111	1	28
8	1000	-1	28
9	1001	-1	24
10	1010	-1	20
11	1011	-1	16
12	1100	-1	12

13	1101	-1	8
14	1110	-1	4
15	1111	-1	0

Table 11. Symbol Modulation Mapping

Chart from3GPP TS 26.267

1.2.1.22 PSAP Modem Operations and Functions - Receiver

The PSAP receiver continuously monitors the data traffic. The eCall sync burst from the IVS transmission will wake up the PSAP receiver if it is in any standby mode. The PSAP receiver will output the CRC flag and the MSD messages.

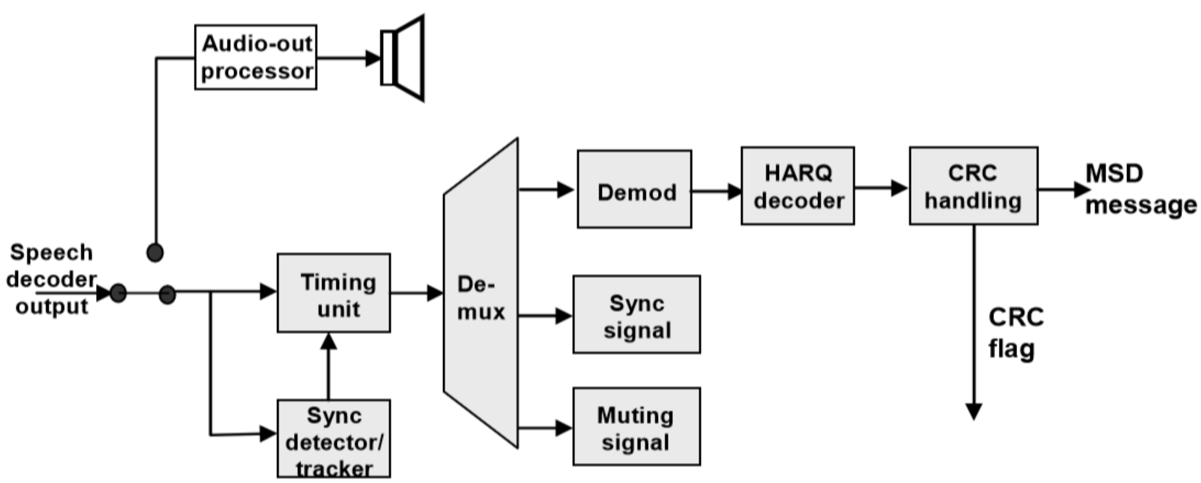

Figure 12. eCall PSAP Receiver

Chart from3GPP TS 26.267

1.2.1.23 Synchronization Detector/Tracker

The PSAP Synchronization detector/tracker has the same hardware as the IVS detector/tracker, with some minor functional differences. PSAP receiver can be trigger with one synchronization preamble detection, and it checks for another ten speech frames to 1) make sure the detection is accurate 2) to restart the MSD message if a better preamble is found. Based on the uplink synchronization frames, the Sync Check function constantly checks the delay value. The Sync Tracker function will assign a new delay value if the value found by the Check function is invalid. This delay is set at ±240 samples. The receiver will also detect the two tones either at 500 Hz or 800 Hz to distinguish between if the fast modulator or the robust modulator is used.

1.2.1.24 Demodulator

The data modulator is also a correlator detector, where the incoming modulation waves are matched to the known modulated waveform stored in the ROM.

The correlator detector equation is:

$$r(i) = \sum_{j=0}^{n} ulPulse(j) * ulPulseMatch(i)(j)$$

$$r(i+4) = -r(3-i), i = 0,1,2,3$$

For a fast modulator, n=15; for a robust modulator, n=31.

The final value fed to the FEC decoder is the correlation value *r(i)* normalized by its variance, then subtracted by its mean.

1.2.1.25 H-ARQ FEC Decoder

The FEC decoder is a Parallel Concatenated Convolutional Codes (PCCC) turbo decoder. The turbo decoder starts the decoding process with the three data messages of the first redundant MSD message, rv1. The descrambling will follow the turbo decoder.

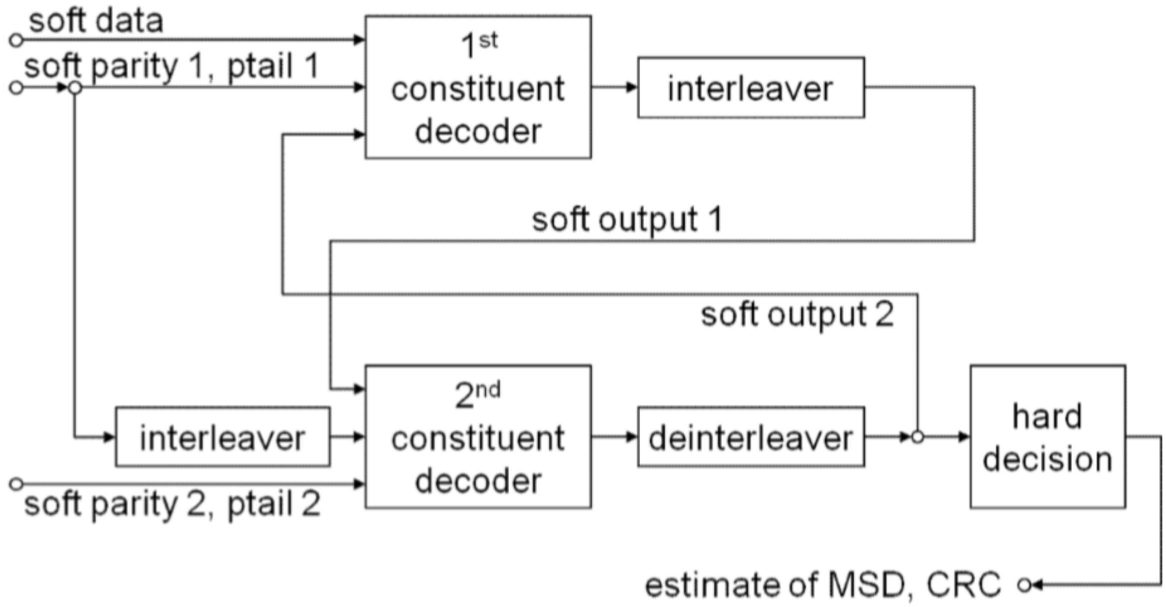

Figure 13. FEC Turbo Decoder

Chart from3GPP TS 26.267

1.2.1.26 Objectives of Research

The eCall system is designed to be a potentially lifesaving product for mass consumers. Hence, accurate and timely data delivery is a key performance indicator (KPI). QUALCOMM Inc claims that ‘Under normal channel conditions, the MSD information is received at the PSAP on an average of 1.36 seconds, well below the

4-second requirement for the eCall in-band data modem. Low rate speech channels and bad radio conditions increase the overall MSD transmission time due to required retransmissions. However, even when including these less favorable conditions, the overall average transmission time of the Qualcomm modem across all required conditions is 2.03 seconds, still well below 4 seconds [36].' However, this is all simulated data for uplink only and there is no definite data on downlink yet. In the NXP (acquired by Telit in 2014) European tryout in 2013, the shortest uplink time was well over five seconds with average over 30 seconds and there is no guarantee of reception neither uplink nor downlink. The overall call drop rate was close to 30% [37]. This number is much higher than average voice channel calls. The fundamental cause is digital waveforms passing through the speech codec. The speech codec is specially designed for the human voice and anything else would be treated as noise, thus highly distorted or filtered out. More importantly, there is yet any real data collected on any 4G/LTE network as of this report.

The proposed study and experiment are to implement the current eCall structure to the current 4G/LTE network and measure its performance. The 2G/3G European Emergency Call system has been in operations since April 1st, 2018 with limited success. There is a strong discussion in the mobile communications community that 3G will be phased out before 2G where 2G will still be present due to its more stable coverage. Thus, the study of emergency call system performance in the 4G/LTE environment becomes of interest since 2G and 4G/LTE will co-exist for a very long time. Unlike the United States, the European Union and China intend to keep the current CS network for a much longer time, and several standard bodies have already started the discussion of PS-network based emergency call proposals. The IETF (Internet Engineering Task Force) group is currently working on the next generation PS-based emergency call system specification within Europe and ACN (Automated Crash Notification) globally. China is working on its own emergency call system and is scheduled to release as a mandatory standard by the year 2020.The modeling will be based on Matlab and some eCall published C codes will be used to generate needed data.

The reference used will include 3GPP TS specifications 26.267 and 26.268 and ETSI TR 103 140.

1.2.2 Future PS-Based eCall

Cellular Long Term Evolution (LTE) is a large topic and only the relevant pieces will be discussed for this experiment. LTE is a PS only network and inherently there is no guarantee of timely delivery of data. It is based on GSM/EDGE (Global System for Mobile/Enhanced Data rates for GSM Evolution) and UMTS/HSPA (Universal Mobile Telecommunication System/High-Speed Packet Access) technologies. The standard is developed by the 3GPP group and it is also known as 3.95G, not as 4G [26] since LTE does not meet the technical criteria of 4G wireless services. However, due to marketing pressure and significant advancements in WiMax, Evolved High-Speed Packet Access and LTE compared to 3G, the International Telecommunications Union (ITU) decided to call these 4G technologies [27]. The LTE Advanced (LTE-A) technology satisfies the ITU-R (Radiocommunication Sector) requirements and is considered to be IMT Advanced (International Mobile Communication Advanced). To distinguish from LTE, the LTE Advanced is called the 'True 4G' technology [28] [29].

CHAPTER 2 SYSTEM ARCHITECTURE

At the time this dissertation is written, there is yet any report on non-speech data transmission on the VoLTE network. A new and unique method needs to be designed to evaluate non-speech data transmission, most importantly this method needs to be implementable on the actual carrier network. Most of the carriers have implemented VoLTE services on individual networks, however, cross-carrier network VoLTE services are still in the early trial phase. The challenge becomes that most of the modern mobile handsets are capable of speech VoLTE services but cannot offer data transmission on VoLTE while none of the LTE IoT (Internet of Things) 4G/LTE modules can offer the VoLTE feature.

The current VoLTE speech codec uses both the standard G.722.2 AMR-WB (wideband) codec, which corresponds to 3GPP TS 26.190 and TS 26.194 and the G.711 (A-law) AMR-NB [11] codec. Only the AMR-WB is used under the VoLTE environment without CSFB and AMR-NB is used with CSFB and VoLTE that's in-between MNOs (Mobile Network Operators) in theory. The AMR-WB modes are sampled at 16 kHz and processed at 12.8 kHz. Frequency band 6.4 kHz to 7 kHz is transmitted at the highest bitrate 23.85 kbps only while the decoder would generate sound by using the lower frequency bands of 75Hz to 6.4 kHz mixed with random noise to simulate higher frequency band. In order to be compatible with the legacy systems, AMR-NB must be used when coding experimental data.

The fundamental challenge of this experiment is to develop a potential solution for such transmission. The initial experiment is to conduct transmission of non-speech data on the VoLTE network of a single cellular carrier since cross-carrier communication is dependent on carrier solutions only. Multiple carriers were tested out for this experiment and China Mobile 4G services were selected at the end since it has a broader regional coverage along the test routes and the availability of VoLTE service on multiple hardware platforms.

2.1 LTE-TDD and LTE-FDD

LTE Time-Division Duplex (LTE-TDD) is a 4G telecommunications technology and standard co-developed by an international coalition of companies, including China Mobile, Datang Telecom, Huawei, ZTE, Nokia Solutions and Networks, Qualcomm, Samsung, and ST-Ericsson. LTE-TDD is one of the two mobile data transmission technologies of the Long-Term Evolution (LTE) technology standard, the other being LTE Frequency-Division Duplex (LTE-FDD). While some companies refer to LTE-TDD as "TD-LTE" for familiarity with TD-SCDMA, there is no reference to that acronym anywhere in the 3GPP specifications [30] [31] [32].

2.2 Building Blocks

2.2.1 LTE Structure

LTE is formally referred to as Evolved UMTS Terrestrial Radio Access (E-UTRA) and Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) [33]. It is based on standards developed by the 3GPP group with increased uplink and downlink peak data rates over the 3G services. It provides improved spectral efficiency with scalable bandwidths. Two key characteristics of the network are that it is all IP network and the interfaces may support a multitude of user types. The LTE network is intended to bridge the data exchange gap between fixed wireless Local Area Networks (LAN) and mobile-cellular networks. The key driving factors for LTE standards are primarily making the spectrum flexible to increase its utilization efficiency to improve the overall system capacity and coverage with higher data rates and reduced latency, thus financial benefits to the operators.

One of the biggest differences between LTE networks and legacy 3G mobile communication systems is the base station. In 3G systems, there is an intelligent and centralizing node like the RNC (Radio Network Controller), and it needs to control all the radio resources and mobility over multiple NodeBs (3G base stations) underneath it in a hieratical radio access network (Figure 14). All NodeBs need to do is behave exactly according to commands from the RNC sent over the Iub interface. In LTE, on the other hand, eNodeBs as base stations must manage radio resource and mobility in the cell and sector to optimize all the UE's communication in a flat radio network structure (Figure 15). Therefore, the performance of an LTE eNodeB depends on its radio resource management algorithm and its implementation.

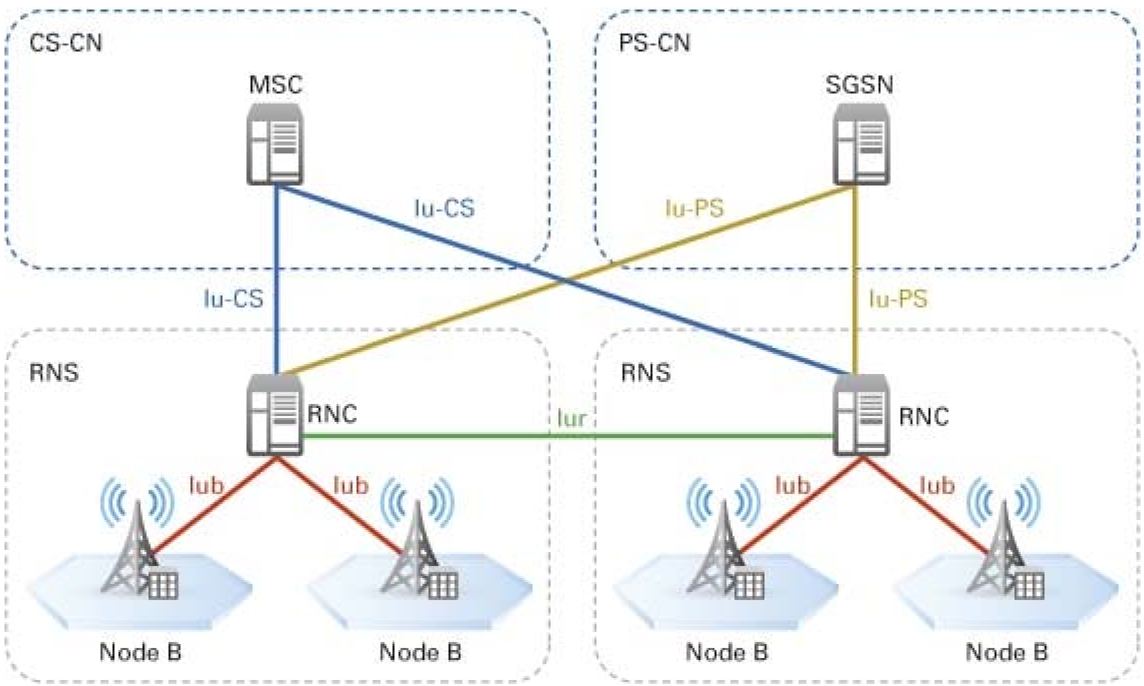

Figure 14. 3G Radio Access Network (UTRAN) Architecture

Figure from Artizanetworks.net

Definitions

VoLTE UE – User Equipment
eNodeB – Evolved Node B (or Base station, eNB)
E-UTRA – Evolved-UMTS Terrestrial Radio Access
E-UTRAN – Evolved-UMTS Terrestrial Radio Access Network
RRC – Radio Resource Control
EPC – Evolved Packet Core
MME – Mobility Management Entity

SGW – Serving Gateway
X2 GW – X2 Gateway
X2-AP – X2 Application Protocol
SCTP – Stream Control Transmission Protocol
PDCD – Packet Data Convergence Protocol
RLC – Radio Link Control
MAC – Media Access Control

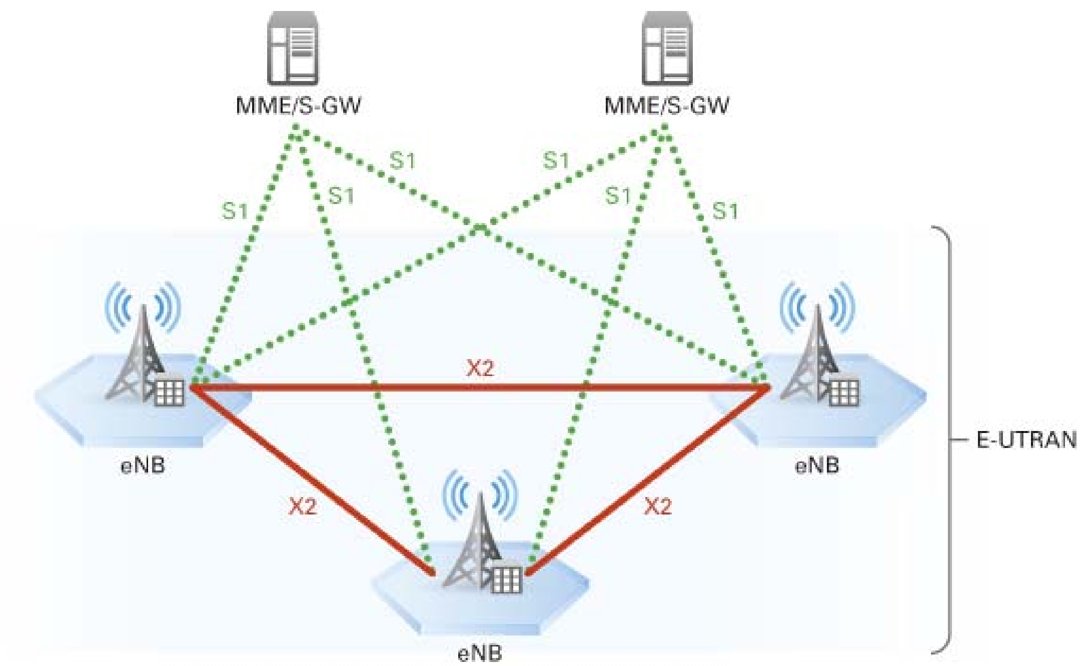

Figure 15. High-level Architecture

Figure from Artizanetworks.net

As shown in Figure 15, taken from [33], the E-UTRAN is consists of eNodeBs that provide E-UTRA user planes such as PDCP/RLC/MAC/PHY and control plane RRC protocol terminations towards the User Equipment (IoT devices, handsets, and etc.). The X2 interfaces provide the connections between eNodeBs when switching of broadcast nodes are needed. The UE can have information of up to sixteen (16) eNodeBs at any given time to plan the best packet routing strategy and to maximize the overall system bandwidth efficiency. The eNodeBs are also connected to the Evolved Packet Core (EPC) through S1-MME (Mobility Management Entity) or S-GW (Serving Gateway) interfaces which is under the control plane that provides guaranteed data delivery. The S1 (Serving) interface supports a many-to-many protocol relation between MMEs/S-GWs and eNodeBs. The X2-AP that resides within the SCTP (Streaming Control Transmission Protocol) is used mainly for handover sequences between eNodeBs.

The eNodeB has the following functions:

- Manage radio resource functions: Radio Bearer Control, Radio Admission Control, Connection Mobility Control, and the scheduling of uplink, downlink, and sidelink.
- IP head encryption and compression used in the user data stream.
- Routing of User Plane data towards the Serving Gateway.
- Scheduling and transmission of paging messages that are originated from the MME.
- Scheduling and transmission of broadcast information originated from MME.
- Measurement and measurement reporting configuration for mobility and scheduling.

When an emergency call or a voice service is triggered on 3G/4G/LTE network, Circuit Switched Fall Back (CSFB) is the method to provide such services until VoLTE becomes widely available. In addition to the fallback time, another drawback is the call setup latency of more than 4 seconds and times of over 10 seconds have been observed in actual practices [34], which is much longer than the sub-second call setup time of VoLTE. This would not meet the current emergency call conformance testing standard, specified in 3GPP TS 26.269.

Packets in the PS networks arrive out of order, packet delays are high and variable while some packets can be repeated (echoing), lost or discarded due to signal distortion. Consequently, the conversation quality of a VoIP (Voice over IP) session could vary dramatically. The primary issue with real-time media provided on the PS-based network is the lack of QoS (quality of service) measurement because the PS network does not guarantee bandwidth allocation for any connection nor for any application. LTE uses a QoS Class Identifier (QCI) as a mechanism to handle the QoS of its bearers (tunnels or channels) to ensure appropriate resource allocations. Table 16 shows the LTE standard QCI classes.

QCI	Resource Type	Packet loss rate	Delay budget(ms)	QCI Priority	Service Examples
1	GBR	10^{-2}	100	2	Conversational voice
2		10^{-3}	150	4	Conversational video
3			50	3	Real-time gaming
4		10^{-6}	300	5	Buffered video
5	Non-GBR		100	1	IMS signaling
6			300	6	Buffered video, email
7		10^{-3}	100	7	Interactive gaming
8		10^{-6}	300	8	TCP-based services

Table 12. LTE Standard QCI Class

Table from 3GPP TS 23.203

2.2.2 IMS Architecture

The IMS framework integrates SIP (Session Initiation Protocol) to manage real-time communication services and applications between two or more endpoints within a network. The entire IMS architecture is divided into three layers: Transport Layer, Application & Services Layer and Session & Control Layer. SIP is defined in the Session & Control Layer and can be directly accessed by the software stack on both the client and the server. Figure 16 is the overall definition of how

IMS handles a call initiated from any UE (User Equipment) to any other terminal including CS-based or PS-based UEs. Since the experiment is to test exclusively in the VoLTE environment, a simplified system with only what's needed will be explained. The goal is to initiate a call from a UE (User Equipment) client, going through CSCF (Call Session Control Function) and route the messages back to another UE server. Many data interfaces and exchanges are only allowed by MNOs thus appear transparent to the terminal users or UEs in the form of AT (Attention) commands.

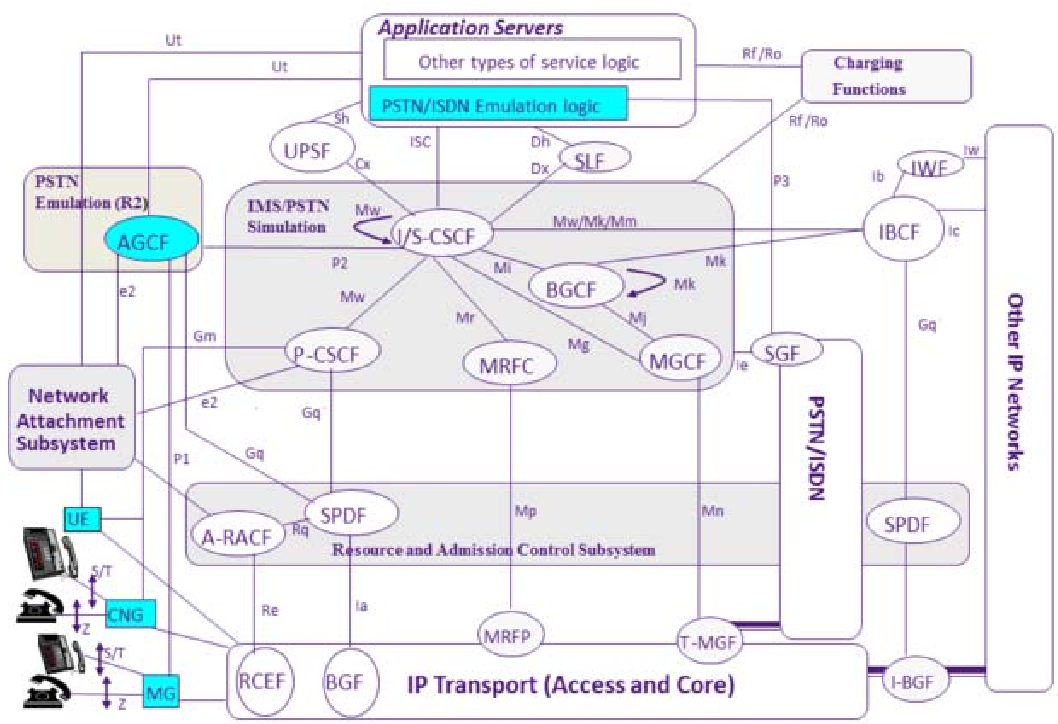

Figure 16. IMS Architecture

Figure from Artizanetworks.net

As shown in Table 12, the SIP signaling has the LTE QCI set to 5 which indicates that SIP signaling messages are not guaranteed to be delivered; however, once the call is set up, the real-time voice data has a guaranteed packet loss rate of 10^{-2} and a delay budget of 100 ms, which indicates that this loss may not present much an issue for speech data but there will potentially be somewhat significant non-speech data loss.

2.2.3 VoLTE Structure

LTE leverages IMS to provide VoLTE services where the IMS is a standalone system that resides outside LTE and connected to the PDN (Packet Data Network) gateway. Figure 17 shows the top-level VoLTE IMS architecture where block 1 is typically a VoLTE enabled mobile phone or module, block 2 is the LTE network, block 3 is the IMS SIP (Session Initiate Protocol) servers and block 4 is the traditional PSTN (Public Switched Telephone Network) where CS-based call would occur. Blocks 1, 2 and 3 will be the primary interest of this experiment.

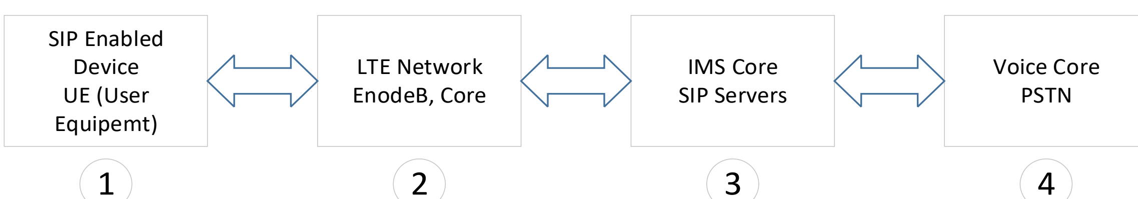

Figure 17. VoLTE IMS Architecture

Most of GSM MNOs (mobile network operators) in Europe are anticipating a switchover of GSM networks and moving all communications to VoLTE technology.

Considering that typical mobile phones last 2-3 years and vehicles last 15 years, the current European eCall system with in-band technology must be supported beyond the year 2040 and the migration to IMS (IP Multimedia Subsystem) LTE needs much immediate attention.

Section 6.1.3 of [9] outlines 11 key issues in defining IMS emergency call solutions relating to the IVS and PSAP (Public Safety Answering Point) structure, MSD data format that may involve data security and the overall system management. While there is no definition of what information the next generation MSD should contain yet, the common understanding is that it should at least maintain the current data format defined in 3GPP TS 26.267. Although, suggestions for removal of muting requirement and 140-byte limit, and provision of bi-directionality have been proposed. These suggestions will improve the user experience in the LTE environment, but it may create issues when switching to a CS-based network (or fallback) is needed. Overall, ETSI TR 103 140 recommends minor modifications of the current emergency call solution as the future long-term underlying communication mechanism.

Figure 18 shows the proposed eCall migration scenarios from ETSI 103 140 Sec. 7.7.2 wherein the hope that there will be very few crossed calls between the CS network and the LTE network (shown in red). We especially need to take into consideration millions of older vehicles by future time with CS-only equipped IVS units still in service which tend to have more issues that may trigger emergency call services. Currently, there are already hundreds of thousands of such vehicles in service.

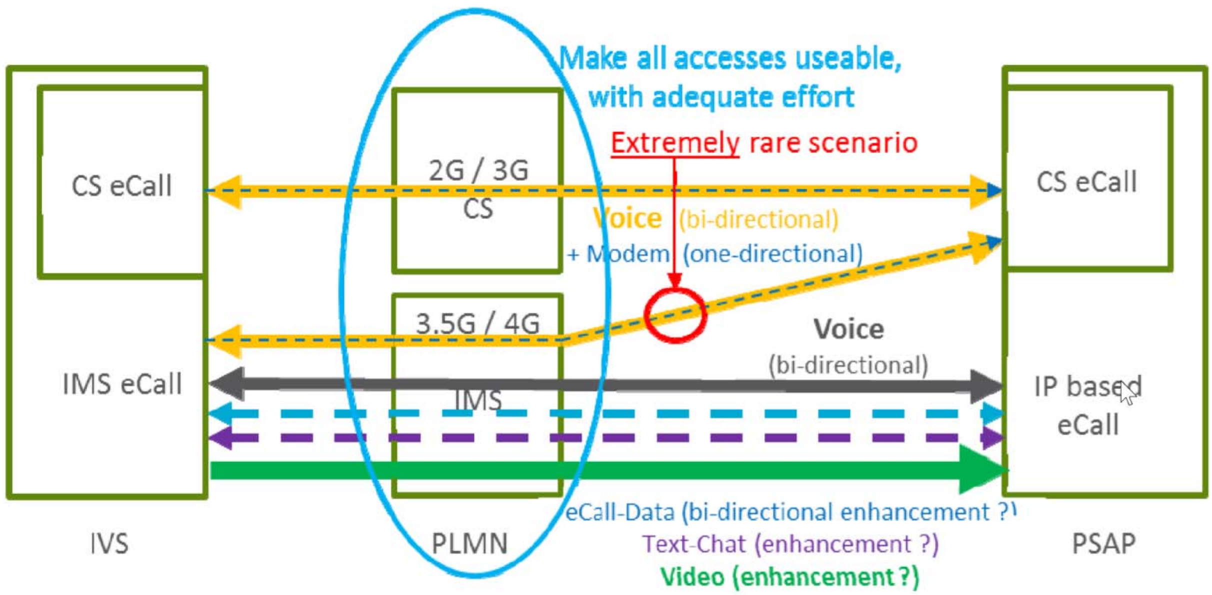

Figure 18. Migration Scenarios

Chart from ETSI 103 140

2.2.4 SIP (Session Initiate Protocol)

The SIP protocol was standardized as IETF RFC 2543 [35] (now obsoleted by IETF RFC 3261) and was accepted as a 3GPP signaling protocol and permanent element of IMS architecture for IP-based streaming multimedia services in the cellular networks. Specification IETF RFC 3261 [36] is the revised standard and offers various extensions and clarifications. SIP was designed to provide a signaling and call setup for IP-based communications supporting the call processing functions and features presented in the PSTN with support to multimedia applications.

The routings of data packets vary depending on the network conditions. Even with SIP implemented in IMS to provide VoLTE calls, there is still plenty of echoing and loss of data. SIP is a text-based protocol with syntax similar to that of HTTP. There are two different types of SIP messages: requests and responses. The first line of a request has a method, defining the nature of the request, and a Request-URI (Uniform Resource Identifier) indicating where the request should be sent to. Figure 19 shows the basic operation of a SIP call between two users. As indicated, the SIP protocol is only responsible or setting up or terminating the call, the actual voice exchange uses other protocols which will be explained in the later chapters.

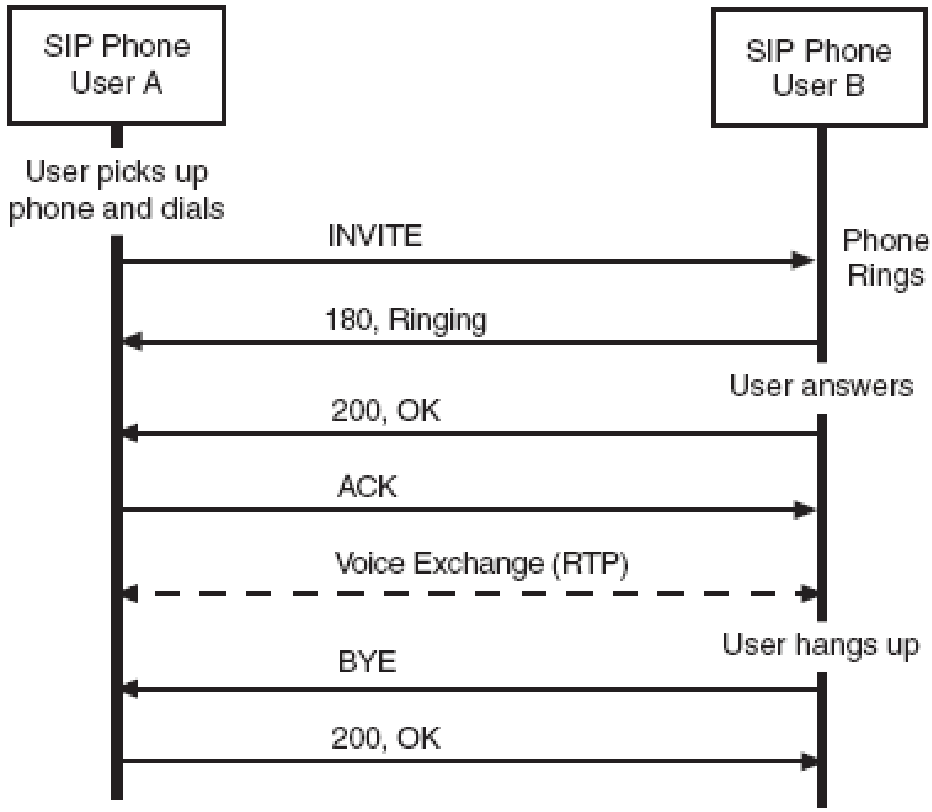

Figure 19. SIP Basic Operation

Chart from IETF RFC 2543

2.2.5 TCP (Transmission Control Protocol)

The Transmission Control Protocol (TCP) is one of the main protocols of the Internet protocol suite. It originated in the initial network implementation in which it complemented the Internet Protocol (IP). Therefore, the entire suite is commonly referred to as TCP/IP. TCP provides reliable, ordered, and error-checked delivery of a stream of octets (bytes) between non-realtime applications running on hosts communicating via an IP network. Major internet applications such as the World Wide Web, email, remote administration, and file transfer rely on TCP, which is part of the Transport Layer of the TCP/IP suite. TCP is connection-oriented, and a connection between client and server is established (passive open) before data can be sent. Three-way handshake (active open), retransmission, and error-detection adds to reliability but lengthens latency. Applications that do not require reliable data stream service may use the User Datagram Protocol (UDP), which provides a connectionless datagram service that prioritizes time over reliability. TCP employs network congestion avoidance. However, there are vulnerabilities to TCP including denial of service, connection hijacking, TCP veto, and reset attack. For network security, monitoring, and debugging, TCP traffic can be intercepted and logged with a packet sniffer. This paper references the standard IETF RFC 1180: A TCP/IP Tutorial [37].

TCP was originally designed for wired communication networks. Packet loss is considered to be the result of network congestion and the congestion window size is reduced dramatically as a precaution. However, wireless links are known to experience sporadic and usually temporary losses due to fading, shadowing, handoff, interference, and other radio effects, that are not strictly congestion. After the erroneous back-off of the congestion window size, due to wireless packet loss, there may be a congestion avoidance phase with a conservative decrease in window size. This causes the radio link to be underutilized. There has been extensive research on combating these undesirable effects. Suggested solutions can be categorized as end-to-end solutions, which require modifications at the client or server, Link Layer solutions such as Radio Link Protocol (RLP) in cellular networks, or proxy-based solutions which require some changes in the network without modifying end nodes.

For many applications, TCP is not appropriate. One problem (at least with normal implementations) is that the application cannot access the packets coming after a lost packet until the retransmitted copy of the lost packet is received. This causes problems for real-time applications such as streaming media, real-time multiplayer games, and voice over IP (VoIP) where it is generally more useful to get most of the data in a timely fashion than it is to get all of the data in order. In this experiment, the Resource ReSerVation Protocol (RSVP) and Real-Time Streaming Protocol (RTSP) are used with SIP to provide QoS (Quality of Service) during a voice change, such as the Real-time Transport Protocol (RTP) shown in the Figure 19 example.

2.2.6 RTP (Real-time Transport Protocol)

RTP is designed for end-to-end, real-time transfer of streaming media. The protocol provides facilities for jitter compensation and detection of packet loss and out-of-order delivery, which are common especially during UDP transmissions on an IP network. RTP allows data transfer to multiple destinations through IP multicast. RTP is regarded as the primary standard for audio/video transport in IP networks and is used with an associated profile and payload format. The design of RTP is based on the architectural principle known as application-layer framing where protocol functions are implemented in the application as opposed to in the operating system's protocol stack.

Real-time multimedia streaming applications require timely delivery of information and often can tolerate some packet loss to achieve this goal. For example, loss of a packet in an audio application may result in loss of a fraction of a second of audio data, which can be made unnoticeable with suitable error concealment algorithms but is a much bigger issue for non-speech data contents. The Transmission Control Protocol (TCP), although standardized for RTP use, is not normally used in RTP applications because TCP favors reliability over timeliness.

RTP was developed by the Audio/Video Transport working group of the IETF standards organization and is used in conjunction with other protocols such as H.323 and RTSP. The RTP specification describes two protocols: RTP and RTCP. RTP is used for the transfer of multimedia data, and the RTCP is used to periodically send control information and QoS parameters.

RTP carries real-time data. Information provided by this protocol includes timestamps (used for synchronization), sequence numbers (used for packet loss and reordering detection) and the payload format which indicates the encoded format of the data. The control protocol, RTCP, is used for quality of service (QoS) feedback and synchronization between the media streams. The bandwidth of RTCP traffic compared to RTP is small, typically around 5% [38] [39].

RTP sessions are typically initiated between communicating peers using a signaling protocol, such as H.323, the Session Initiation Protocol (SIP), RTSP, or Jingle (XMPP). These protocols may use the Session Description Protocol to specify the parameters for the sessions. An RTP session is established for each multimedia stream. Audio and video streams may use separate RTP sessions, enabling a receiver to selectively receive components of a particular stream. The specification recommends that RTP port numbers are chosen to be even and that each associated RTCP port be the next higher odd number. However, a single port is chosen for RTP and RTCP in applications that multiplex the protocols. RTP and RTCP can typically use any port as the protocol design is transport independent. This paper references the standard IETF RFC 3550: RTP: A Transport Protocol for Real-Time Applications [40].

2.2.7 RTCP (RTP Control Protocol)

The RTP Control Protocol (RTCP) is a sister protocol of the Real-time Transport Protocol (RTP). Its basic functionality and packet structure is defined in IETF RFC 3550 [40]. RTCP provides out-of-band statistics and control information for an RTP session. It partners with RTP in the delivery and packaging of multimedia data but does not transport any media data itself.

The primary function of RTCP is to provide feedback on the quality of service (QoS) in media distribution by periodically sending statistics information such as transmitted octet (byte) and packet counts, packet loss, packet delay variation, and round-trip delay time to participants in a streaming multimedia session while not providing any methods on data authentication or encryption. RTCP protocol gathers statistics on quality aspects of the media distribution during a session and transmits this data to the session media source and other session participants. Such information may be used by the source for adaptive media encoding (codec) and detection of transmission faults. If the session is carried over a multicast network, this permits non-intrusive session quality monitoring. It also provides the session control function whereas the RTP protocol cannot. An application may use this information to control quality of service (QoS) parameters by limiting packet size, transmission/queue timing, or using different codecs.

RTCP reports are expected to be sent by all participants, even in a multicast session which may involve thousands of recipients. Such traffic will increase proportionally with the number of participants. Thus, to avoid network congestion, the protocol must include session bandwidth management. This is achieved by dynamically controlling the frequency of report transmissions. RTCP bandwidth usage should generally not exceed 5% of the total session bandwidth. Furthermore, 25% of the RTCP bandwidth should be reserved for media sources at all times, so that in large conferences new participants can receive the Canonical End-Point Identifiers (CNAME) of the senders without excessive delay. It is important to understand that the RTCP reporting interval is randomized to prevent unintended synchronization of reporting. The recommended minimum RTCP report interval per station is 5 seconds. Stations should not transmit RTCP reports more often than once every 5 seconds. This is helpful if streaming content is long, however, this feature might not be as useful if the streaming is short and bursty, but nonetheless, this feature should play an important role in eCall applications.

2.2.8 IP (Internet Protocol)

An IP address serves two principal functions: 1) identifies the host, or more specifically its network interface, and 2) provides the location of the host in the network, and thus the capability of establishing a path to that host. The header of each IP packet contains the IP address of the sending host, and that of the destination host. There are two versions of the IP used today, namely IPv4 and IPv6. This experiment uses IPv4 which is also IPv6 compatible. In IPv6, the address size was increased from 32 bits in IPv4 to 128 bits, thus providing up to 2^{128} addresses. A large number of IPv6 addresses allow large blocks to be assigned for specific purposes and aggregated for efficient routing where appropriate. IP addresses are assigned to a host either dynamically as they join the network, or persistently by configuration of the host hardware or software. Persistent configuration is also known as using a static IP address. In contrast, when a computer's IP address is assigned each time it restarts, this is known as using a dynamic IP address.

Dynamic IP addresses are assigned by network using Dynamic Host Configuration Protocol (DHCP). DHCP is the most frequently used technology for assigning addresses. It avoids the administrative burden of assigning specific static addresses to each device on a network. It also allows devices to share the limited address space on a network if only some of them are online at a particular time. Typically, dynamic IP configuration is enabled by default in modern desktop operating systems.

The address assigned with DHCP is associated with a lease and usually has an expiration period. If the lease is not renewed by the host before expiry, the address may be assigned to another device. Some DHCP implementations attempt to reassign the same IP address to a host (based on its MAC address) each time it joins the network. A network administrator may configure DHCP by allocating specific IP addresses based on the MAC (Media Access Control) address which is used for this experiment. This experiment references the original IETF RFC 791 with updates [41].

2.2.9 PPP (Point-to-Point Protocol)

Point-to-Point Protocol (PPP) is used over many types of physical networks including serial cable, phone line, trunk line, cellular telephone, specialized radio links, and fiber optic links such as SONET. Internet service providers (ISPs) have used PPP for customer dial-up access to the Internet since IP packets cannot be transmitted over a modem line on their own without some data link protocol that can identify where the transmitted frame starts and where it ends.

A PPP is a layered protocol that has three components: 1) an encapsulation component that is used to transmit datagrams over the specified physical layer, 2) a Link Control Protocol (LCP) to establish, configure, and test the link as well as negotiate settings, options and the use of features, and 3) one or more Network Control Protocols (NCP) used to negotiate optional configuration parameters and facilities for the network layer. There is one NCP for each higher-layer protocol supported by PPP. LCP initiates and terminates connections gracefully, allowing hosts to negotiate connection options. It is an integral part of PPP and is defined in the same standard specification. LCP provides an automatic configuration of the interfaces at each end (such as setting datagram size, special characters, etc) and for selecting optional authentication. The LCP protocol runs on top of PPP and therefore a basic PPP connection must be established before LCP is able to configure it. This experiment references IETF RFC 1661 [42].

CHAPTER 3 EXPERIMENT PLATFORM

There are three goals of this experiment: 1) establish a testbed for an emergency call on Voice over LTE (VoLTE) network, 2) establish measurement of Key Performance Indicator (KPI) on successful transmissions and the transmission timing, and 3) compare the collected data with the simulated data from ETSI 103 140.

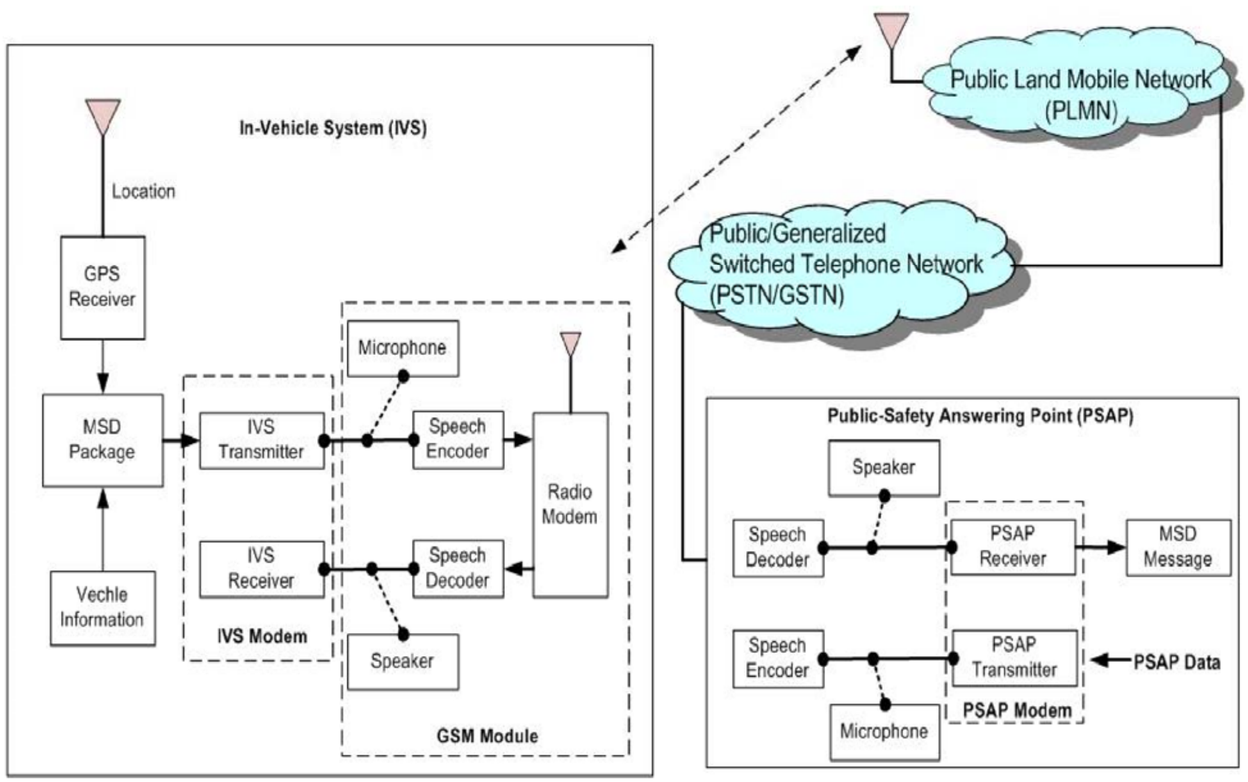

Figure 20. 2G/3G eCall System Architecture

Figure from 3GPP TS 26.267

Figure 20 shows the current overall 2G/3G - based eCall architecture of the including IVS, PLMN, PSTN, and PSAP. The IVS is used to initiate emergency calls and transmit MSD and comprised of a Global Positioning System (GPS) receiver, an IVS modem, and a GSM module. The GPS receiver provides the vehicle coordinates, time, speed, and direction. The GPS information and vehicle information are then packaged into the 140-byte MSD packet and sent to the transmitter of the IVS modem. The modulated MSD information or voice information is transmitted through the voice codec, radio modem, and GSM antenna to the PSAP. The PSAP demodulates and decodes the received uplink signal and then obtains the MSD data. Once the emergency call is established, the IVS transmits an ”INITIATION” signal which consists of a synchronization tone of 500Hz and data bits modulated by the in-band modem. The ”INITIATION” signal followed by the modulated MSD signals are sent to the PSAP through the digital cellular voice channel. The microphone in the IVS is muted during the transmission of the MSD to minimize noise and interference. The PSAP receiver detects the signal, performs demodulation and decoding for the MSD. If the decoding is successful, the PSAP sends an acknowledgment (ACK) signal to the IVS. Upon receiving the ACK signal, the IVS switches to talking mode for people in the vehicle to talk with the person at the PSAP for further assistance.

The eCall transmission timing is the KPI of interests. According to [43], the eCall transmission timing should be 4 seconds or less under the ideal conditions, meaning using Narrow Band Adaptive Multirate (AMR) Codec of 12.2kbps to code the 140-byte long MSD message under the error-free RF environment and it is also used for eCall system conformance testing. For this experiment, only the MSD data without GPS information is used for data transmission without the actual human voice data since none of them do not play an important role at this point. The MSD data is taken from 3GPP TS 26.268 which is a standard ‘campaign.txt’ file.

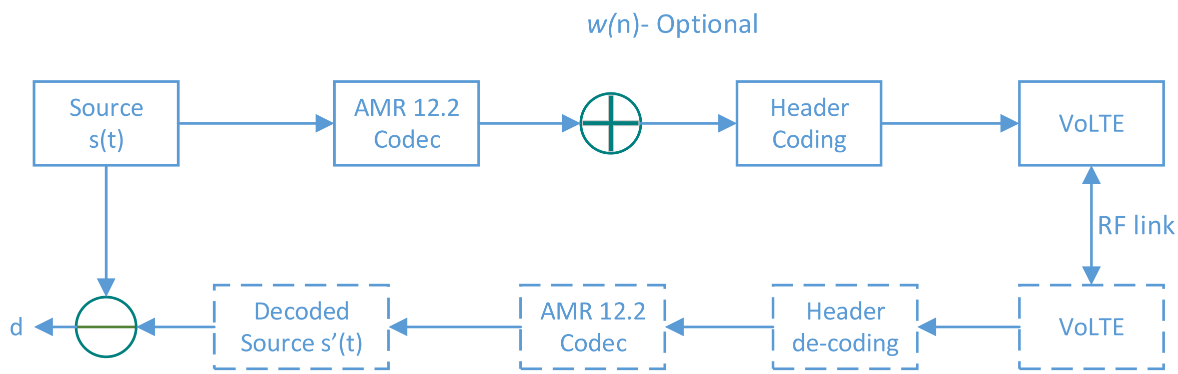

Figure 21. VoLTE eCall System Architecture

Figure 21 shows the VoLTE eCall system architecture for this experiment. Source file s(t) is the ‘campaign.txt’ file. Optional noise w(n) can be introduced to evaluate the effects of channel noise when needed. Result d is the difference between the original source file and the received and decoded file. Only the AMR 12.2 kbps is used for this experiment. Header coding and decoding are carried out by AT Commands. Decoded files s’(t) are stored in a cloud server with timing and successful transmission rate data, not the bit error rate. These files are used to compare with the original file sent.

3.1 Hardware Platform

For simultaneous data transmission and collection, a pair of systems are used which simulates IVS and PSAP separately. The IVS units are located in the vehicle and the PSAP units are located at a server. The wireless service is provided by China Mobile and the cloud service is provided by Baidu Cloud in China. Figure 22 shows the VoLTE in-vehicle unit.

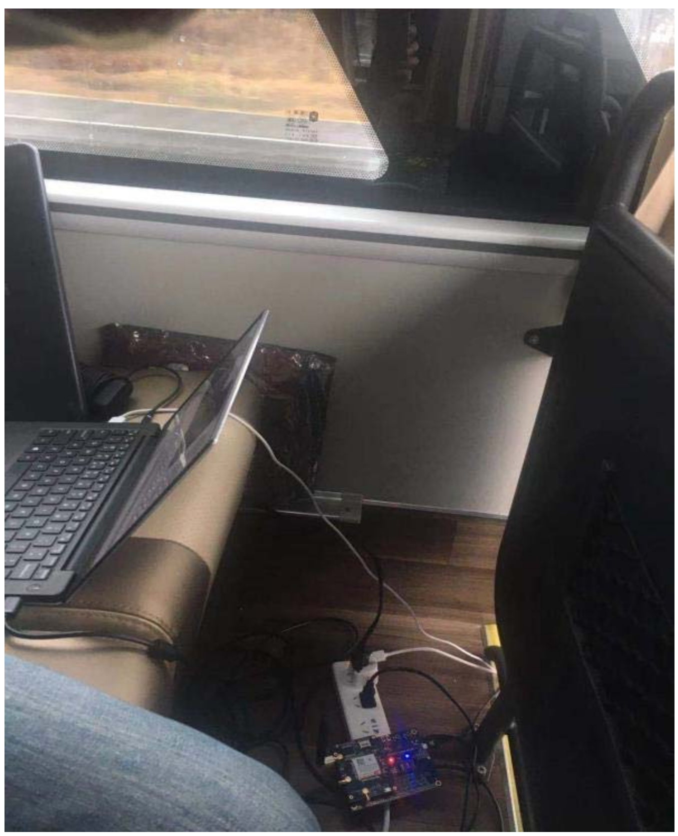

Figure 22. VoLTE In-Vehicle Unit

This test is part of a long-distance data collection based on an OEM customer's request. Over five thousand data points were collected over a period of two weeks. Figure 23 shows the vehicle routes. Highway driving consists of 80% in the data collection plan, actual high-speed driving over 70kmph is less than 50% due to traffic congestions. There were a few incidents when the vehicle speeds were over 120 km/h.

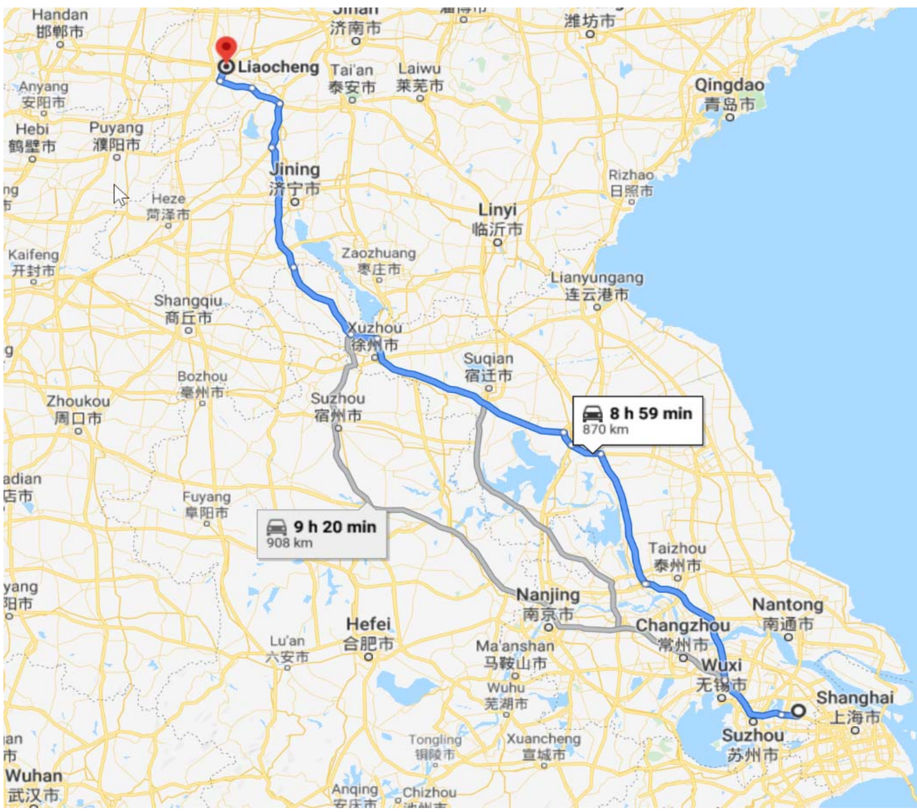

Figure 23. Test Routes

The main processor board is the Freescale i.MX6Q-based from TopEET in Beijing, China running on Ubuntu 16.04 build Linux. The VoLTE system uses the Simcom CE7600 series modules for wireless transmission. The Freescale i.MX6 series processor is chosen due to its extensive usage in automotive applications such as many safety-critical products and it has a history of being stable. Figure 24 shows the evaluation board from ToPEET used in this experiment which can be expanded to carry out future tests with the extra data ports which are not available from Freescale factory evaluation boards.

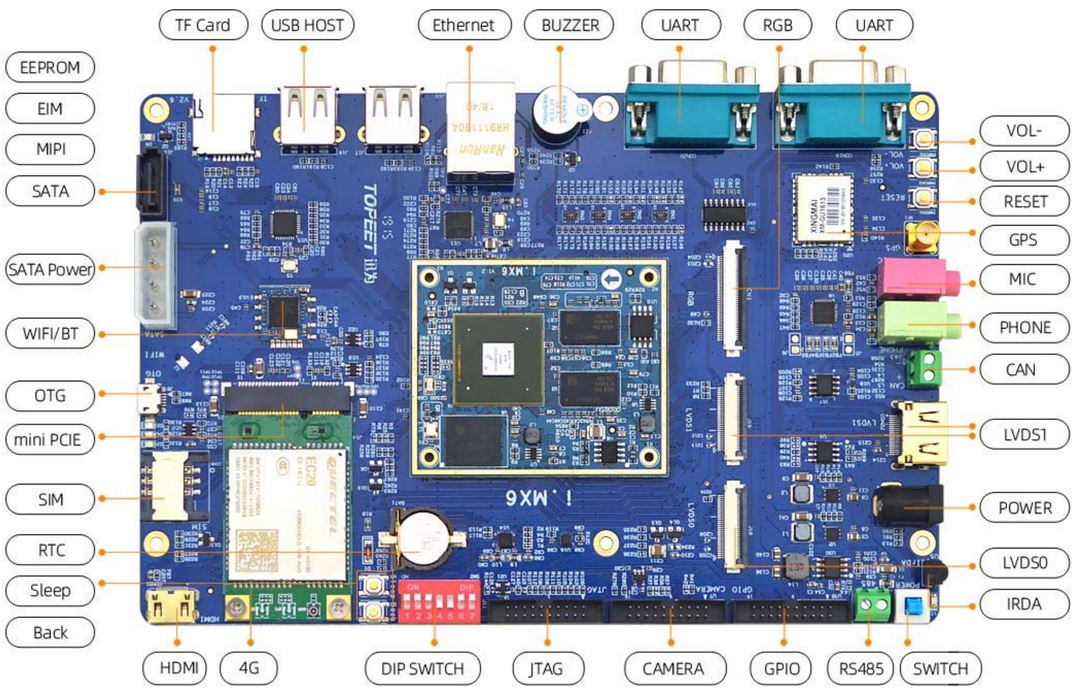

Figure 24. Evaluation Board from TopEET

3.2 Software Platform

The i.MX6 runs on Ubuntu 16.04 build with the stable USB driver. A software interface is provided by TopEET to execute AT commands for both wireless modules. Most of the AT commands provided are standard commands, a few crucial ones were written exclusively for this experiment. These AT commands will generate and concatenate the header information and execute data transmission and reception. Each wireless module maker also provides its own software interface for AT command execution.

The sender (IVS equivalent) and the receiver (PSAP equivalent) have different application software that encodes and decodes this proprietary data. The receiver is also the server unit for all the data collection. Once the cloud service has been obtained, the server ensures the correct portal is always open to receive data transmission while decoding and storing the data for later analysis.

CHAPTER 4 METHODOLOGY

During a typical VoIP (Voice over IP) call session which is a purely packet-based call without IMS, the Presentation Layer uses the File Transfer Protocol (FTP) headers. In order to have a faster transmission timing, a socket communication system running Transmission Control Protocol (TCP) is implemented instead of FTP for this experiment. On the Session Layer, the Session Initiate Protocol (SIP) is the standard protocol used for packet-based media applications regardless of VoLTE call or VoIP call. On the Transport Layer, the Real-time Transport Protocol (RTP) is used in combination with TCP to ensure the packets are sorted with the correct timing and guaranteed delivery. The Network Layer would typically use the Internet Protocol (IP) to route the data packets to specific instance addresses and port numbers from both hosts. The Data Link Layer supports many types of connection protocols and for this experiment, we have chosen the Point-to-Point Protocol (PPP) with padded Cyclic Redundant Code (CRC) option. The Physical Layer would be the standard LTE implementation.

4.1 Data Structure

Table 13 represents the OSI model interpretation of the data structure used for this experiment. From the top layer down, the VoLTE Call Manager (executed with AT commands) in the Application Layer with AH (Application Header) initiates a call and specifies to send the raw MSD data which is already pre-processed with AMR-NB (campaign. txt file processed with AMR 12.2 kbps) in PCM format with little-endian byte order.

Layer	Header	Application/Protocol
Application	AH	VoLTE Call Manager
Presentation	PH	Socket/TCP
Session	SH	SIP
Transport	TH	RTP/TCP
Network	NH	IP
Data Link	DLH	Point-to-Point Protocol (PPP)
Physical		Raw Data

Table 13. MSD Data Structure – OSI Model

As part of the data structure, socket communication must be established between the client and the server and TCP protocol is used at the Presentation Layer, as shown in Table 14. Each TCP packet is designed to contain 16~40 bytes with 16 bytes of header information and a maximum of 24 bytes of payload.

<table>
<tr><td colspan="7">Source port (1024)
2 bytes</td><td>destination port number (1024)
2 bytes</td></tr>
<tr><td colspan="8">sequence number (1000)
4 bytes</td></tr>
<tr><td rowspan="2">Header Length
4 bits</td><td rowspan="2">reserved
3 bits</td><td>URG</td><td>ACK</td><td>PSH</td><td>RST</td><td>SYN</td><td rowspan="2">window size
2 bytes</td></tr>
<tr><td colspan="5">9 bits</td></tr>
<tr><td colspan="7">checksum
2 bytes</td><td>urgent pointer
2 bytes</td></tr>
</table>

Table 14. TCP Header

The TCP packet is next concatenated by the SIP header. The SIP header structure is shown in Table 15, with a total length of 16 bytes. The flow label (also defined as IMS signaling) contains eight (8) messages as follows: INVITE, 100 Trying, 180 Ringing, 200 OK, ACK, Media, BYE and 200 OK. The hop limit is set to the maximum value of 31. The payload type is one of the challenges experienced in this experiment.

<table>
<tr><td>version 0
4 bits</td><td colspan="3">flow label
28 bits</td></tr>
<tr><td colspan="2">payload length
2 bytes</td><td>payload type
1 byte</td><td>hop limit (31)
1 byte</td></tr>
<tr><td colspan="4">source address
4 bytes</td></tr>
<tr><td colspan="4">destination address
4 bytes</td></tr>
</table>

Table 15. SIP Header

Table 16 shows the latest standard SIP payload types supported. Since Frame Type 10~13 are reserved for future usage and not used in the current standard, Frame Type 10 is allocated temporarily and defined to transmit PCM (Pulse Code Modulated) files. This required SIP protocol modification at the code level.

Frame Type Index	Mode Indication	Mode Request	Frame Content
0	0	0	AMR-WB 6.60kbps
1	1	1	AMR-WB 8.85kbps
2	2	2	AMR-WB 12.65kbps
3	3	3	AMR-WB 14.25kbps
4	4	4	AMR-WB 15.85kbps
5	5	5	AMR-WB 18.25kbps
6	6	6	AMR-WB 19.85kbps
7	7	7	AMR-WB 23.05kbps

8	8	8	AMR-WB 23.85kbps
9	-	-	AMR-WB SID (Comfort Noise)
10~13	-	-	Future use
14	-	-	Speech lost
15	-	-	No data

Table 16. SIP Payload Type

Table 17 shows the RTP header structure. The RTP protocol version used is 2 which is the latest. The payload type is set to 20 which is unassigned that supports audio. SSRC and CSRC are all set to 0 since this experiment is a point-to-point only communication. Timestamp used is generated by the local system clock.

V(2) 2 bit	P 1 bit	X 1 bit	CC 4 bits	M 1 bit	PT (20) 7 bit	Sequence number 16 bits
timestamp 32 bits						
synchronization source (SSRC) identifier 32 bits						
contributing source (CSRC) identifier(s) 32 bits						

Table 17. RTP Header

Table 18 shows the network layer IP protocol. Newer technologies require real-time streaming and therefore make use of the DSCP (Differentiated Services Code Point) and in this experiment, it is set to VoIP services which will support QoS measurement during data transmission. The flag is set to 2 which will allow fragmentation if needed.

V (4) 4 bits	Len (6) 4 bits	DSCP (VoIP) 8 bits	ECN (not used) 16 bits		
Identification (not used) 16 bits			Flags (2) 3 bits	Fragment Offset 13 bits	
Time 8 bits		Protocol (89) 8 bits	Header checksum 16 bits		
Source IP address 32 bits					
Destination IP address 32 bits					
Options 24 bits				Padding 8 bits	

Table 18. IP Header

Data of variable sizes will be packaged into the PPP frame for physical layer (LTE network) transmission. Figure 19 shows the PPP frame structure. The Flag indicates the beginning of a PPP frame and the Protocol is the PPP ID (set to 0x0021 which stands for IP data type) of the embedded data. The frame check sequence (FCS) field is used for determining whether an individual frame has an error. It contains a checksum computed over the frame to provide basic protection against errors in transmission. This is a CRC code. According to RFC 1662, it can be either 16 bits (2 bytes) or 32 bits (4 bytes) in size (default is a 16 bits Polynomial $x^{16} + x^{12} + x^5 + 1$).

Flag 1 byte	Address 1 byte	Control 1 byte	Protocol 1 byte	Payload variable	FCS 2 bytes	Flag 1 byte

Table 19. PPP Frame

The final data packet structure D becomes as shown in Figure 25 with a total maximum length of 119 bytes and it will be sent to LTE transmission. There will be a minimum of six (6) frame transmissions for each MSD dataset.

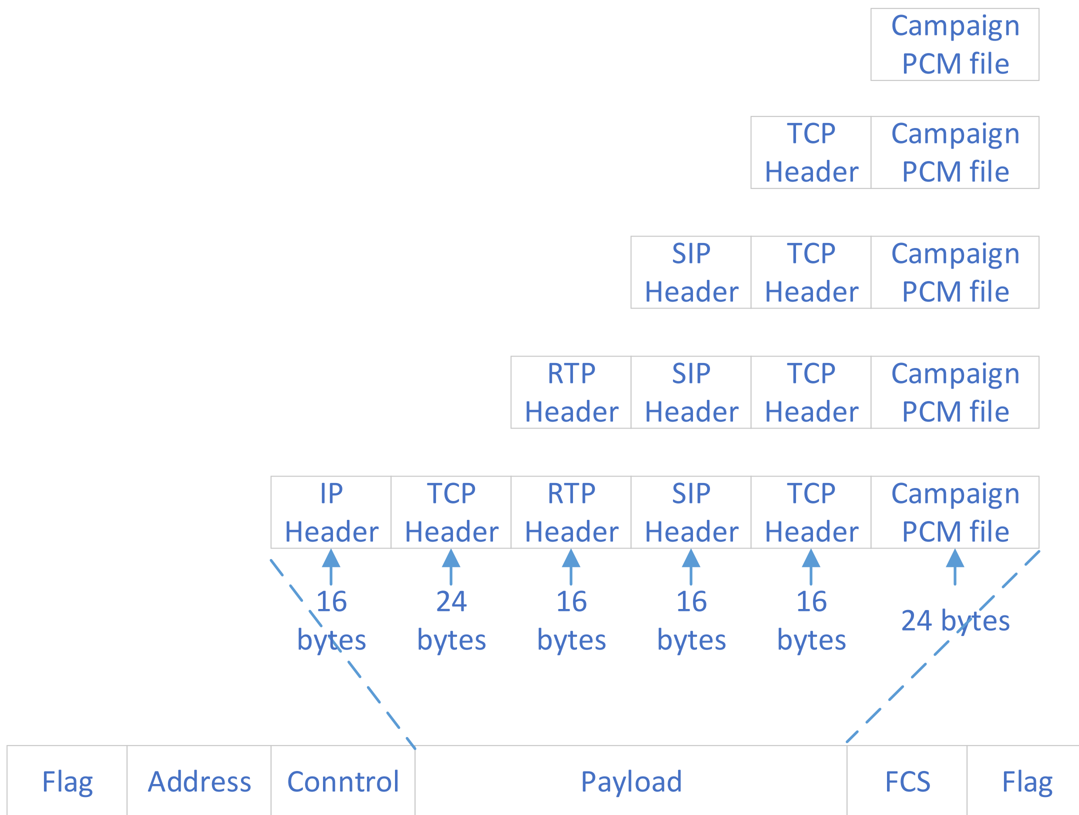

Figure 25. Data Packet Structure

4.2 LTE – Physical Layer

The uplink and downlink are different on the LTE physical layer. The signal bandwidth and frame structure are determined by multiple access methods where the uplink uses Single Carrier Frequency Division Multiple Access (SC-FDMA) and the downlink uses Orthogonal Frequency Division Multiple Access (OFDMA). The uplink modulation methods include QPSK (Quadrature Phase Shift Keying), 16QAM (Quadrature Amplitude Modulation), 64QAM or 256QAM. The downlink modulation is OFDM (Orthogonal Frequency Division Multiplexing). The OFDM signals have a higher peak-to-average power ratio than SC-FDMA. The reason being that in the time domain, a multicarrier signal is the sum of many narrow bands. This sum could be large or small at different time instances which means the peak value is much higher than the average. Thus it reduces the efficiency of the RF power amplifier. This power issue is much more of a challenge on a UE than on an eNobeB. The tradeoff is that the uplink data rate is about half of the downlink data rate. The LTE structure is symmetrical for both the uplink and the downlink, including PDCP (Packet Data Convergence Protocol), RLC (Radio Link Control) and MAC (Medium Access Control), as shown in Figure 26. PDCP will compress all the header information again while not affecting the payload data. In order to guarantee the delivery of a full packet of data, LTE uses HARQ at the MAC Layer and Segmentation ARQ at the RLC Layer. The MAC Layer would employ CRC and FEC (Forward Error Correction) codes where the RLC Layer would use CRC only. The bitstream transmission is segmented into frames which are 10 msec in duration. Each frame consists of 20 slot periods of 0.5 msec. A sub-frame contains two (2) slot periods and is 10 msec in duration. The Transport Channel will scramble all the data bits generated and get it ready for 1/3 rate turbo FEC code and modulation selection on the PHY Layer and the RF transmission level.

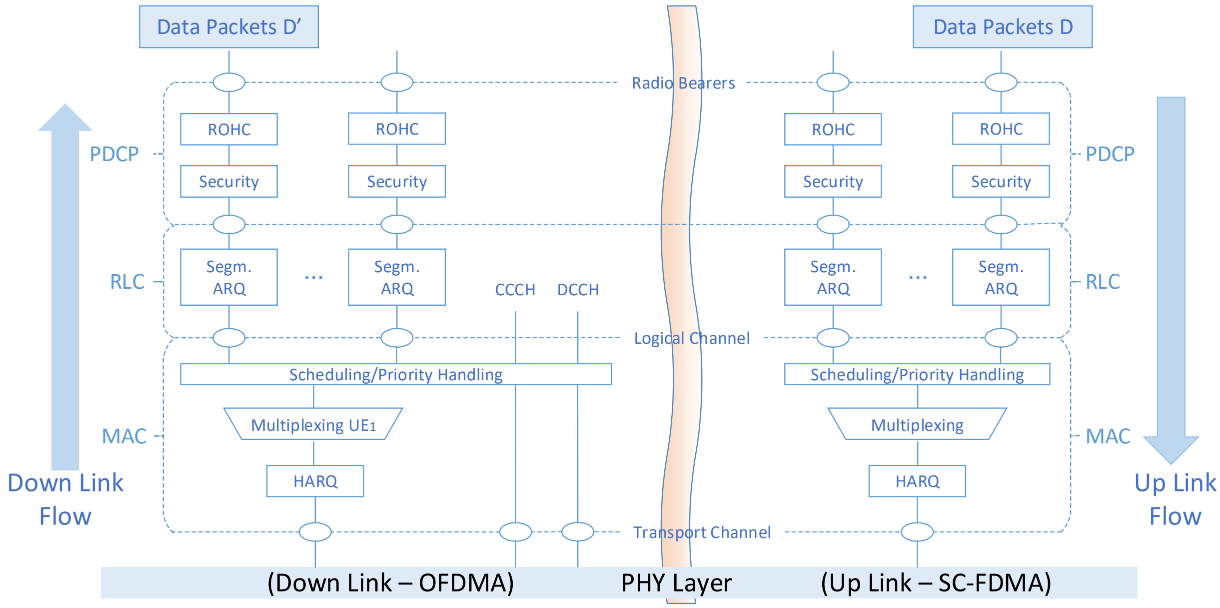

Figure 26. LTE Uplink and Downlink Process

4.3 Transmission Timing Measurement

Definitions

T – Transmission time

t1 – Start of data transmission

t2 – End of data transmission

As defined by 3GPP TS 26.267, the MSD transmission starts with the IVS module with synchronization frame and follows with one or more redundant version of the MSD data, e.g. MSD *rv*0, MSD *rv*1,…, MSD *rv*7, each with a different version of incremental redundancy (IR) that is processed with different CRC coding scheme. The IVS will continue sending all eight (8) fast-mode modulated data unless interrupted by the PSAP or receives 'ACK' message. If no 'ACK' message is received, ten (10) or more ‘NACK’ messages are received, or a ‘Start’ message is received, then the IVS will continue with robust-mode modulated data. If both the fast-mode and the robust-mode modulated data transmission fail, then the whole system fails and each IVS module manufacture is responsible for its own system definition on dealing with failed transmissions.

The transmission timing starts with MSD preamble data transmission, triggered with a timestamp t1. A timestamp t2 is triggered on the PSAP once the ‘ACK’ message is sent by the PSAP. The overall transmission time is T = t2 - t1. However, if the synchronization can’t be achieved or is lost during transmission, then T becomes meaningless. It is also possible that the last ‘ACK’ message sent by the PSAP not be able to reach the IVS which may put the IVS in different modulation modes.

The transmission timing measurement is similar in the VoLTE channel application which is carried out by the SIP protocol. At the SIP application layer, timestamp t1 starts with a handshake of sending the ’Register’ messages, and time stamp t2 is triggered by ‘Bye’ message and confirmation of ‘200’ message. The transaction layer can return with timeout error if ‘Register’ message yields no response, and the UE does not immediately re-attempt another registration to the same registrar since it will most likely return with another timeout, and instead waiting a reasonable amount of time would cause the timeout to be corrected and it reduces the unnecessary load on the network. No specific time interval is mandated since it is network dependent.

The transmission timing is of great interests as it is stated in Appendix 2.11 of [44] that the MSD data should typically reach the PSAP within four (4) seconds with the end-to-end connection under optimal conditions which is defined as an error-free radio channel, and GSM Full Rate Codec with AMR 12.2kbps mode. The test MSD data of 140 bytes long will be coded with AMR 12.2kbps and used for this experiment. This will represent the best effort of system performance.

CHAPTER 5 TIMING KPI PERFORMANCE EVALUATION

All the data are collected in 4G VoLTE exclusive settings to ensure data integrity. The data transmission timeout period for VoLTE is set to 25 seconds after some trials. When there is LTE coverage, there is no bit error observed between file transmitted and file received. However, LTE service can drop out rather quickly and sometimes come back very quickly. This results in sporadic extended transmission timing. Scenarios could be that the transmissions could have stayed within one LTE eNode coverage area or the transmission has switched to a new eNode. It is impossible to monitor the signal level, data throughput and serving eNode handover without commercial carrier-grade software tools. The best way to measure the performance would be to have control over which eNode to stay on and when to switch to a different eNode.

The 4G/LTE coverage without switching to 3G seems to be a very binary operation without much transitional time. The data throughput is very good when coverage is available. The only noticeable indication of 4G/LTE coverage dropout is when transmission hangs without any response. This hints a potential issue with network switch over for data over voice, which is whether to switch to a 2G, an SMS or VoIP over a 3G network and how to structure the MSD data format.

Figure 27 shows the Probability Density Function (PDF) and the Cumulative Distribution Function (CDF) of the transmission timing under stationary VoLTE condition. Based on 5000 all successful trials, the data mean is 4.579 seconds and the median is 4.579 seconds. 99.92 percent of the transmission can be completed within 4 seconds and 99.86 percent of the transmission can be achieved within 2 seconds. The longest transmission timing took 8.883 seconds and the shortest took 0.2749 seconds. Table 20 shows the statistic table based on Figure 27.

Stationary data were taken on Jan 11, 2020 in Kunshan office and the mobile data were taken from Jan 20, 2020 to Jan 22, 2020 following the planned route outlined in Figure 23. The server is located about 5km away from the Kunshan office. The data collection was not collected in an automated fashion, but rather each data point was collected manually. Received files were stored and processed in the server.

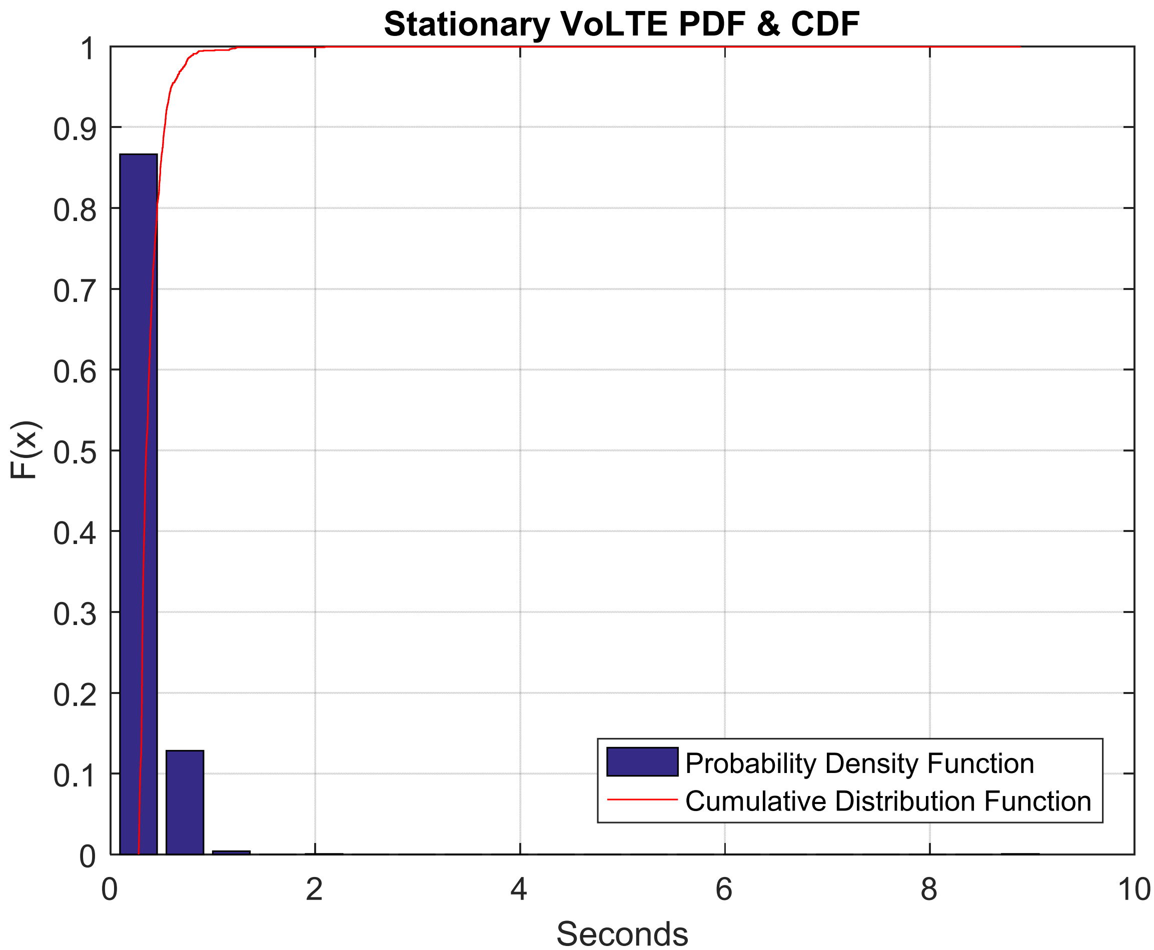

Figure 27. PDF & CDF of Stationary VoLTE Transmission Timing

	PDF X	PDF Y	CDF X	CDF Y
min	2.2749	0	-Inf	0
max	8.883	0.8665	Inf	1
mean	4.579	0.05	NaN	0.5805
median	4.579	0	0.3851	0.6241
mode	0.2749	0	0.2749	0
std	2.68	0.1943	NaN	0.2871
range	8.608	0.8665	Inf	1

Table 20. Statistics based on Figure 27

According to [45], LTE performance should be stable for speeds less than 120 km/hour and this is observed. However, for the few moments when speeds were around and over the 120 km/hour limit, longer transmission timing was noticed. In this experiment, most of the longer timing was recorded at higher speeds.

The PDF and CDF of the transmission timing under mobile VoLTE are shown in Figure 28. Also based on 5000 all successful trials, the data mean is 5.155 seconds and the median is 5.155 seconds. 99.52 percent of the transmission can be completed within 4 seconds and 98.8 percent of the transmission can be achieved within 2 seconds. The longest transmission timing took 10.04 seconds and the shortest took 0.2739 seconds.

It has been observed that most of the times the system would drop the packets when the transmission is over 25 seconds long under the VoLTE environment while most of the transmissions are completed within 1 seconds, suggesting that the number of re-transmissions and the overall timing has exceeded the usefulness of the data. Table 21 shows the statistic table based on Figure 28.

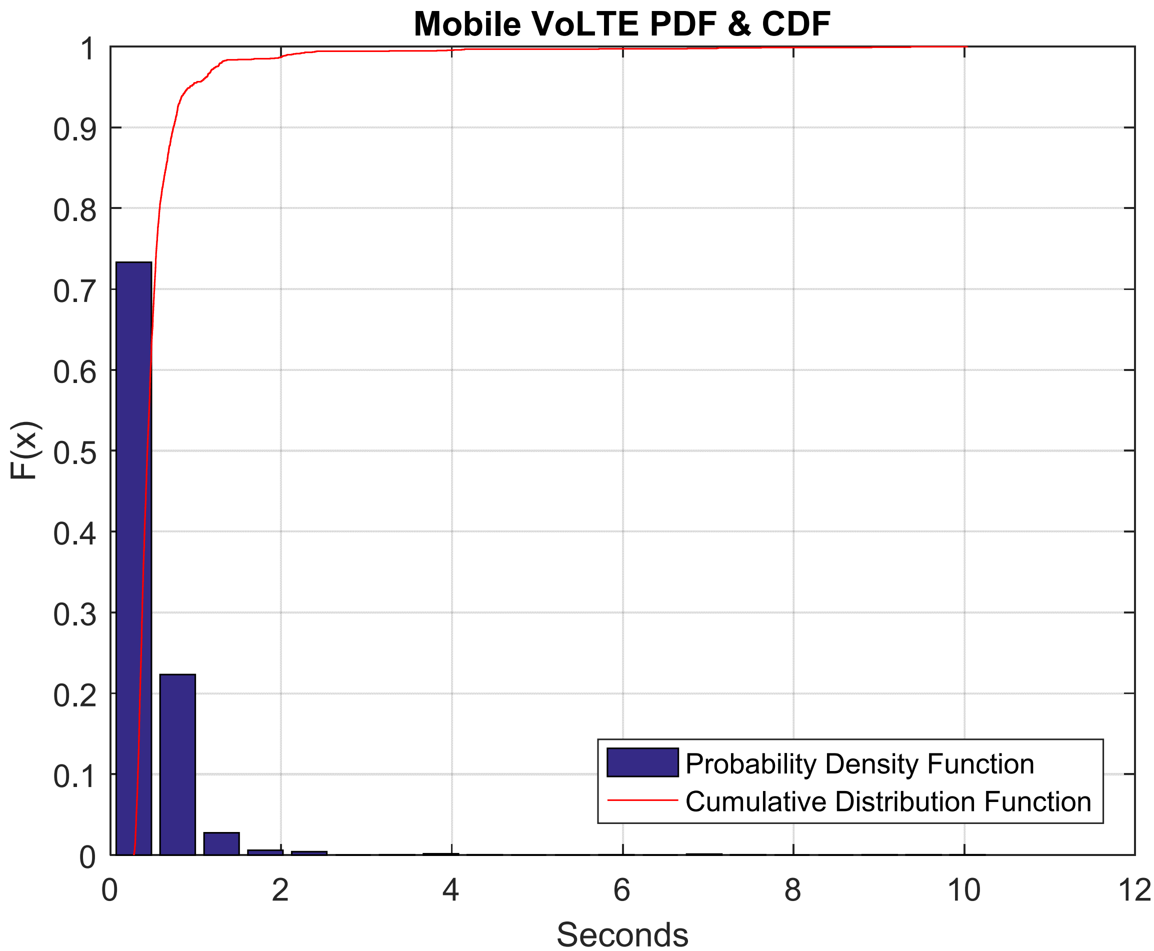

Figure 28. PDF & CDF of Mobile VoLTE Transmission Timing

	PDF X	PDF Y	CDF X	CDF Y
min	2.2739	0	-Inf	0
max	10.04	0.7405	Inf	1
mean	5.155	0.05	NaN	0.5531
median	5.155	0.0004	0.4639	0.5925
mode	0.2739	0.0004	0.2739	0
std	3.04	0.1696	NaN	0.3026
range	9.763	0.7405	Inf	1

Table 21. Statistics based on Figure 28

Simulated results based on VoIP were presented in [9] and the results are compared with the collected VoLTE data, shown in Table 22. Based on this limited trial, VoLTE average transmission time is 1~3 seconds longer than the simulated data and the VoLTE peak transmission time is about one third to one half of the simulated data. Longer average time could be due to the real traffic on the VoLTE channel and the less peak time could be due to the fact that VoLTE uses QCI with GRB during a VoLTE call.

Mobile VoLTE		VoIP Profile1		VoIP Profile2		VoIP Profile3		VoIP Profile4		VoIP Profile5		VoIP Profile6	
avg	max	avg	max	avg	max	avg	max	avg	max	avg	max	avg	max
5.155	10.04	2.0632	19.22	2.0143	23.8	2.3471	36.64	3.0135	24.62	4.8047	30.64	1.65	19.76

Table 22. Comparison Data

CHAPTER 6 CONCLUSION AND FUTURE WORK

This experiment creates a testbed for non-speech data transmission over VoLTE channels, formatted in speech-size packets. A few ideas can be suggested for future VoLTE emergency call implementations.

The data packets are short compared to non-IMS IP packets with theoretical delays of less than 100ms. The advantage is guaranteed delivery of data and the disadvantage is low bandwidth efficiency. Modifying the SIP protocol to transmit non-speech data with larger payload sizes may increase the throughput efficiency. Since the wireless LTE coverage is reduced on the edges, the future source coding scheme can dynamically change speech codec between AMR-WB and AMR-NB to ensure the delivery of crucial data, or to remove the speech coded completely.

Time spread introduced by multipath will create Inter-Symbol Interference (ISI). This is more of an issue on the uplink side than on the downlink side due to different modulation methods. The uplink uses a transversal filter channel equalizer to compensate for at most several symbol periods. And this channel equalizer is only effective up to 100 Mbps [46] data rate. The effects of ISI on non-speech data especially at higher throughput need to be investigated. On the downlink side, OFDM uses longer symbol periods in the form of Fast Fourier Transform (FFT) time and Cyclic Prefix (CP) to eliminate ISI. Even though OFDM is much less susceptible to multipath, it is however very sensitive to a large signal PARP and carrier frequency error caused by oscillator offset and Doppler shifts. LTE downlink does not employ PHY Layer preamble to carrier offset estimate, channel estimation, timing synchronization etc. The reference symbols are staggered both in frequency and time within a time slot of 0.5ms and 12 frequency subcarriers. The performance of downlink could be a topic of interest when the vehicle speed exceeds 120km/hour or approaching 300-400km/hour. This could be especially beneficial to future automotive and locomotive applications.

The payload structure will be another topic of interest which would include the size, header structure and source coding that may improve its performance over ISI. This experiment mostly follows the standard VoIP data structure with the minimum header requirement to achieve the best results. Other combinations of protocols can be investigated to compare transmission timing KPI.

The ultimate questions remaining is how to guarantee the VoLTE-based solution in the future to fit safety-critical applications and be compatible with the current legacy system.

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ABSTRACT

INVESTIGATION OF EMERGENCY CALL PERFORMANCE OVER VOLTE

by

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Major: Electrical Engineering

Degree: Doctor of Philosophy

The European emergency call system that's based on 2G/3G (GSM/UMTS) cellular network has been in deployment since April 2018. This system is a two-decade-long effort from multiple parties to ensure vehicular safety. Due to its legacy nature, the system is deployed with circuit-switched technology which is being phased out in many parts of the world. The packet-switched based 4G/LTE technology provides more bandwidths thus faster data speeds and more content for more users. There is a strong discussion in the mobile communications community that 3G will be phased out before 2G where 2G will still be present due to its more stable coverage. Thus, the study of emergency call system performance in the 4G/LTE environment becomes of interest since 2G and 4G/LTE will co-exist

for a very long time. Unlike the United States, the European Union and China keep the current CS network, and several standard bodies have already started the discussion of PS-network based emergency call proposals. The Internet Engineering Task Force group is currently working on the next generation PS-based emergency call system specification within Europe and Automated Crash Notification globally. China is working on its own emergency call system and is scheduled to release as a mandatory standard by the year 2020.

The test setup leverages the 3GPP TS 26.267/268/269 standards and implements data transmission on the VoLTE network. The proposed methods create a testbed to measure the Key Performance Indicator of transmission timing. Based on 5000 error-free data samples, over 99.92% of the transmission can be completed within 4 seconds with a mean of 4.579 seconds with stationary VoLTE and slightly decreased performance with mobile VoLTE when LTE coverage is sufficient enough to maintain a connection.

AUTOBIOGRAPHICAL STATEMENT

John Lu received his B.S. and M.S degrees from Southern Illinois University. He is currently a Ph.D. student at Wayne State University under the supervision of Dr. John Liu. His research focuses on the in-vehicle emergency system, VoLTE physical layer network and statistics. He is also the CTO of a start-up company working on ADAS products within the automotive sector.