SYSTEM ASPECTS OF THE 3GPP EVOLUTION TOWARDS ENHANCED VOICE SERVICES

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ABSTRACT

The Enhanced Voice Services (EVS) codec was standardized by 3GPP in 2014. This codec offers significant gains in voice quality, efficiency, channel error robustness over any other existing speech codec and far better music quality. Operators run voice services on a large installed base of 3GPP Circuit-Switched (CS) 2G (GERAN) or 3G (UTRAN) radio networks. These networks offer mobile voice service either as HD voice using the AMR-WB codec, or as traditional narrowband (NB) voice service, based on the AMR codec. Voice over LTE (VoLTE) is currently being deployed throughout the world with HD Voice. The EVS codec will be first introduced as a straightforward VoLTE upgrade. It is also expected that EVS will be deployed over 3G CS networks. This paper describes system aspects relating to EVS codec introduction in VoLTE and CS networks as well as interworking and mobility with legacy systems and services.

Index Terms— EVS, HD voice, VoLTE

1. INTRODUCTION

Since the introduction of GSM mobile telephony, created by the European Telecommunications Standards Institute (ETSI) in the early 1990's, the world has witnessed a tremendous technical revolution in the area of mobile communication that has penetrated everybody's life and changed human society disruptively. Even though mobile telephony triggered this development, users did not experience a lot of changes in this service. However, in reality mobile operators have steadily improved their voice service offerings with optimizations in terms of quality, system capacity and robustness in fast growing mobile systems. The 3rd Generation Partnership Project (3GPP) speech codec standards from GSM Full Rate (FR) [1] over Half Rate (HR) [2] and Enhanced Full Rate (EFR) [3] to the Adaptive Multi-Rate (AMR) codec [4] mark milestones of this development. Only recently users have started experiencing substantial quality improvements. This change came with the introduction of HD (high-definition) voice providing wideband (WB) audio with a frequency range from ca. 50 – 7000 Hz compared to traditional NB voice with about 100 – 3500 Hz. HD voice leads to noticeably enhanced and more natural speech quality. AMR-WB [5], the codec used for HD voice in 3GPP systems, was standardized in year 2001, only three years after AMR. Nevertheless, it took still until year 2009 before its first deployment in a real life network. Reasons were manifold: technical, infrastructural and commercial. This picture changed with the advent of Voice-over-IP (VoIP) and Over-The-Top (OTT) voice services, offering WB speech quality. Mobile operators understood that investments into voice quality were a necessary means to maintain their own service offerings attractive and to avoid customer churn. HD voice now turns into a big success [6].

The mobile industry and operators hope to continue this success story by service evolution towards even higher quality voice services. It is understood that 3GPP voice services have to remain leading edge, even after roll-out of HD voice, in order to succeed in a fierce competition environment. Consequently, in year 2010 3GPP launched the standardization of a new codec for "Enhanced Voice Services", targeted to become the successor of the current HD voice codec AMR-WB in LTE networks [7][8]. The EVS codec standard [9]-[19] was successfully finalized in September 2014, meeting or exceeding the standardization goals. The goals were to enable clear benefits over existing AMR-WB based 3GPP voice services in terms of overall service quality, efficiency and interoperability. The EVS codec offers a toolbox of high performance properties and features comprising: high-quality super-wideband (SWB) and fullband (FB) voice; high-capacity/high-quality NB and WB voice; high-quality music/non-speech signal performance; and high robustness to error-prone VoIP transmission frameworks. SWB voice ranges from about 20 – 15000 Hz, while FB voice extends to about 20000 Hz covering the complete human audible spectrum. In addition, the EVS codec maintains backward compatibility with AMR-WB, thus avoiding interoperability problems or any hard-cut decisions against AMR-WB during its introduction.

This paper provides insights in solutions for the complex task to bridge the gap between legacy voice services and cutting edge EVS, given the heterogeneity of existing 3GPP systems. The paper first describes voice services in 3GPP systems covering legacy CS networks and 4G Packet-Switched (PS) systems with LTE access, i.e. VoLTE [20]. Adaptation principles at call setup and during the call are then addressed as well as interoperability aspects under various system configuration and mobility scenarios. The conclusion of the paper sketches the envisioned smooth migration path towards the new EVS.

2. VOICE IN 3GPP SYSTEMS

2.1. Voice in CS networks

Like the earlier 3GPP speech codecs GSM FR, GSM HR and GSM EFR, AMR was originally developed and standardized by ETSI for CS mobile telephony in the GSM system. The AMR standard for that system (more general GERAN) includes besides the speech codec [21] further components such as channel coding [22], rate adaptation and in-band rate control [23]. The codec comprises 8 modes with net bitrates from 4.75 to 12.2 kbps. AMR rate adaptation allows selecting the most suitable mode for a given radio channel condition, hence choosing the mode that enables best possible quality. Together with FR/HR traffic channel adaptation this allows operators flexibly trading speech quality against system capacity. A cornerstone of the adaptation in GERAN is the feedback channel, allowing signaling codec mode requests (CMR), i.e. adaptation commands, back from the receiving end to the sending end. The AMR concept is discussed in detail in [24].

Based on its merits in comparison with other contenders like G.729 [25], EVRC [26], MPEG-4 Audio [27], and for interoperation with AMR in GERAN systems, 3GPP selected

AMR in 1999 as the mandatory codec for the new UMTS. Compared to GERAN, AMR adaptation in UTRAN serves a different purpose, namely load control [28].

Like AMR, AMR-WB was originally specified for GERAN and UTRAN networks. All principles and mechanisms related to adaptation were retained. The codec includes 9 different modes, from 6.60 to 23.85 kbps. In order to harmonize interworking, 3GPP agreed on one mandatory configuration with a set of modes (active codec set (ACS)) that all GERAN and UTRAN infrastructure must support. This mode set comprises the three lower rate modes of AMR-WB 6.60, 8.85 and 12.65. Even today when AMR-WB is operated in VoLTE networks using its highest mode 23.85, for backward compatibility with CS networks the three low rate modes must always be included in a mode set.

The new EVS codec was originally developed for PS multi-media telephony services over LTE. However, its strong performance has triggered interest to standardize it as well for use in voice services in UTRAN CS networks. Accordingly, 3GPP launched a work item "EVSoCS" [29] in September 2014 with the objective to enable users of 3G services to benefit from the enhanced user experience, system capacity and interoperability provided by EVS. It will be of main importance to make sure that EVS can be introduced into 3G CS systems at minimum system impact. The work item is currently ongoing and is expected to be finalized by December 2015.

2.2. Voice over LTE (VoLTE)

With VoLTE, the operator offers a carrier-grade voice service by ensuring Quality of Service (QoS) with the use of Guaranteed Bit Rate (GBR) bearers for voice. VoLTE makes use of the Real-time Transport Control Protocol (RTCP) associated with the Real-time Transport Protocol (RTP) [30] to monitor voice quality. Although even OTT services could benefit from the excellent quality, efficiency and robustness of the EVS codec, they cannot guarantee a carrier-grade service when using best effort bearers without QoS.

VoLTE enables telephony – including emergency services – and messaging over the 4G networks. The GSM Association (GSMA) specified VoLTE in their PRD IR.92 IMS Profile for Voice and SMS [20]. The VoLTE profile is a subset of the 3GPP IMS standard functionalities [31].

The goal of the VoLTE profile is to accelerate adoption by simplifying the implementation. This is achieved via careful selection of the most appropriate features. The objective is on its way to being accomplished: according to GSA (the Global mobile Suppliers Association) by July 2015: "25 operators in 16 countries have commercially launched VoLTE-based HD voice service, compared to just 3 launched in March 2014. Many more launches are anticipated in 2015" [32].

From the first version of the profile, VoLTE terminals must support AMR codec and optionally support AMR-WB codec. From version 9.0 of the profile they also must support the AMR-WB codec; the EVS codec is mandatory for SWB and FB and recommended for NB and WB. However, all VoLTE terminals on the market are HD Voice capable and hence support AMR-WB. HD voice can therefore be enabled where operators have chosen to implement AMR-WB in network entities terminating the user plane (for example, Media Gateways (MGW), Media Resource Function Processors (MRFP), Conferencing Units and Application Servers). Although EVS is not mandatory for VoLTE terminals supporting WB, GSMA is currently in the process of adding EVS operated in WB modes to the list of allowed codecs in the HD voice minimum requirements [33].

The EVS codec includes, besides EVS primary modes, AMR-WB interoperable (IO) modes enabling interworking with AMR-WB implementations. The EVS AMR-WB IO modes offer improved quality over the original AMR-WB modes [34]. Therefore the standard allows a VoLTE terminal to use its EVS implementation operated with AMR-WB IO modes in place of the AMR-WB codec.

The EVS encoder adapts the bitrate of each frame based on a received Codec Mode Request. These rate adaptation operations generally follow the principle introduced with AMR and are described in section 3. VoLTE uses the RTP [30] carriage of user plane information between end-points. Each encoded frame is encapsulated as a payload in the RTP protocol. The RTP payload format for EVS is specified in Annex A of 3GPP TS 26.445 [13]. It actually defines two formats, a compact format and a header-full format. The former offers efficient transmission without any payload header. The latter format in contrast enables encapsulating several frames in the payload and also enables signaling CMR.

The Session Description Protocol (SDP) [35] is used by end-points to describe the session parameters. The session establishment for the VoLTE call requires a negotiation between end-points as their capabilities may differ. End-points could be VoLTE terminals, but also MGWs for transcoding or other Application Servers that may be used e.g. for producing call-progress tones. This negotiation uses the Offer/Answer model [36] that is used between these end-points. In the case of a Mobile Originated call, the VoLTE terminal originating the call sends an SDP offer, describing all its capabilities. This includes in particular media capabilities like the list of codecs supported and associated modes of audio bandwidth, bitrates, type of RTP packetization and desired bearer bandwidth.

It is one important design requirement that the call setup time be minimized so as to improve user experience. Although the Offer/Answer mechanisms can resolve complex heterogeneous cases of media capabilities to result in compatible session characteristics, it is important to minimize the number of handshakes. For this reason it is for example mandatory to support all codec modes for AMR and AMR-WB, respectively, and to offer all of them in a so-called "open-offer". An "open-offer" defines no specific mode set. This enables the distant VoLTE client or MGW to produce an answer compatible to its capabilities in just one handshake. Furthermore, the operator's network can also tailor the offered SDP in a Session Border Gateway (SBG) to apply a particular policy like e.g. limiting the maximum bitrate or restricting to SWB operation. Applying a particular policy in the network, rather than in customized devices, also has the advantage that it becomes applicable to inbound roamers. Finally, sending an "open-offer" improves interworking as described in section 4.

Once the session is negotiated and bearers are established, RTP packets can be transmitted and received over the UDP [37] ports of the VoLTE terminals. Any conversational voice service requires a low end-to-end delay for good interactivity of the conversations. The LTE radio bearers used for voice traffic are configured to minimize delay by setting the RLC (Radio Link Control) layer in unacknowledged mode. This is made possible by the inherent robustness of the AMR, AMR-WB and EVS decoder to lost packets. Furthermore, in a packet based system, packets experience different transmission delays and this delay variation, the jitter, must be handled by receivers. The standard defines minimum performances for the jitter management. Also, the EVS codec comes with a specified jitter manager [16].

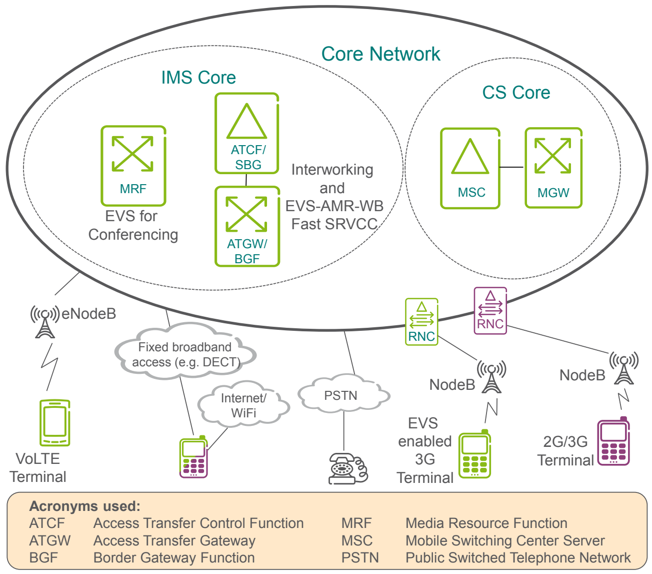

Figure 1: Network impact - EVS codec (green = impacted)

Enabling VoLTE to VoLTE calls with the EVS codec within an existing VoLTE infrastructure requires minimal upgrades. For the VoLTE terminal, besides EVS codec support, there are requirements on the acoustic and speech enhancements functions. These requirements depend on the supported audio bandwidth; more evolved requirements than for HD voice apply if the service shall be operated with SWB or FB audio bandwidth. The network upgrades are limited to the modification of policy functions allowing the negotiation of the EVS codec end-to-end. For example a policy could be to limit the maximum bitrate of operation to 13.2 kbps to benefit from the same network dimensioning as AMR12.2 kbps and AMR-WB 12.65 kbps, yet still allowing to provide excellent SWB voice quality. There is no requirement to implement any particular policy between EVS end-points, but an operator may choose to do so by configuring the policies in the SBG. Further, multiparty conferencing requires upgrade of the Media Resource Function. A system view of the network impact of EVS introduction is provided in Figure 1. The changes required enabling end-to-end VoLTE to 3G Circuit Switched EVS calls and operation with switching to 2G or 3G (Single Radio Voice Call Continuity - SRVCC) are described in section 4.

3. CALL SETUP AND IN-CALL ADAPTATION

3.1. Call setup

Call setup works differently in CS and PS 3GPP systems but follows similar principles. The general purpose is the definition of operation parameters of the used codec(s) for the subsequent call and of the range within which in-call adaptation is allowed.

For AMR and AMR-WB, call setup procedures essentially determine the codec modes (bitrates) among which adaptation is possible during the call. For interoperability reasons, a mode set should be chosen that comprises at least some common low-rate modes that are generally supported, even when roaming within or among various networks.

For EVS applications in VoLTE, TS 26.445 Annex A [13] specifies parameters related to EVS primary and AMR-WB IO mode operation. Bit rate can be specified as a single value or a range of bitrates. Likewise, audio bandwidth can be defined as a single value or a range from NB to a given maximum audio bandwidth. Parameters may be specified individually per call direction or jointly for both directions.

For EVSoCS, the call setup is not yet standardized. It is however envisioned that suitable call control protocol information elements are specified allowing defining similar operation configurations as for PS. Such kind of harmonization will greatly facilitate EVS interoperability between CS and PS (VoLTE) domains.

3.2. In-call adaptation

In this paper, in-call adaptation refers to adaptation of the codec application layer. Adaptation of lower protocol layers (e.g. power control, adaptation of modulation schemes, transport format combination selection based on available power, cell and transport network load adaptations, etc.) occurs frequently. It is system specific for e.g. the radio access technology used. Such kind of adaptation is not covered here.

In-call adaptation has the general purpose, described in section 2.1, to adjust the coding mode to changed system conditions, in order to optimize capacity or service quality. A changed condition requiring selecting a new coding mode may also occur in case of a cell hand-over where the previously used codec mode may not be supported any longer or not be suitable. For AMR and AMR-WB, adaptation is possible with respect to bitrate. For EVS, audio bandwidth is an important further adaptation dimension. Adaptation of the audio bandwidth may e.g. become necessary when the audio front- or backend changes. Adaptation of the encoding audio bandwidth in response to a change of input audio bandwidth is done autonomously by the encoder up to a given maximum audio bandwidth. A change of the output audio bandwidth may on the other hand trigger a CMR from the receiving end to the sending end, accordingly adjusting the maximum audio bandwidth for encoding. A change of encoding bitrate may also result in a changed audio bandwidth since not all audio bandwidths are supported by all bitrates.

A further important adaptation dimension of the EVS codec is the capability to switch between EVS primary modes and AMR-WB IO modes. This kind of adaptation is crucial when a terminal moves between cells with EVS support and cells supporting HD voice but not EVS. Especially during roll-out of VoLTE this may frequently occur since VoLTE coverage may exist only for hot spots while other areas are still only covered by 3G or 2G networks offering HD voice. The possibility to adapt seamlessly between EVS primary modes and AMR-WB IO modes ensures always highest possible service quality and the avoidance of call discontinuities due to codec changes that might otherwise occur. This also avoids the need to use transcoding between networks with EVS and with only HD voice support. Transcoding would degrade quality and require operator investments.

For VoLTE, CMRs are transmitted in-band as part of the RTP payload format or using RTCP. The use of CMR is generally preferable due to lower latency and lower signaling overhead. For AMR and AMR-WB, the RTP payload format [38] specifies a CMR field as part of the payload header. Likewise for EVS, the EVS RTP payload format specifies a CMR field supporting requests for bitrate, audio bandwidth, EVS primary or AMR-WB IO modes and further features.

Codec mode adaptation for EVSoCS is not yet specified. However, it is desirable to allow adaptation along the same dimensions as for EVS in VoLTE. A complication in UTRAN is that rate control is *maximum* rate control. This means that any rate up to the maximum may be chosen by autonomous decisions by the

Radio Network Controller (RNC), depending on the available radio resource, and by the terminal, depending on instantaneous transmit power needs for a codec mode in relation to certain transmit power limits. Uplink rate control commands are signaled to the terminal by means of the Radio Resource Control (RRC) protocol rather than in-band CMR. Similarly, for downlink rate control, RNC sends Rate Control Request (RC-Req) messages to the CS Core. This concept is too limited for EVS with its additional adaptation dimensions. It is therefore envisioned that the existing adaptation signaling is combined with additional in-band CMR signaling. The terminal and likewise MGW in the CS Core would receive both kinds of adaptation signaling messages and merely combine them in a suitable way.

4. INTEROPERABILITY ASPECTS

4.1. Interworking between VoLTE and CS networks

Interworking between public land mobile networks (PLMN) is a strong point of the VoLTE specification. When OTT service providers only create islands of compatible clients, operators can enable their subscribers using VoLTE native client to communicate with any other PSTN phone or 3GPP VoLTE or CS compatible terminal, be it under 2G, 3G or 4G/Wi-Fi coverage (see also Figure 1).

An EVS VoLTE call between two operator's networks requires EVS enabled VoLTE clients, but also that the interworking agreement between operators implements compatible policies. For example, if one operator forces EVS VoLTE calls within its network to negotiate only SWB modes and the other restricts EVS VoLTE calls to only WB modes, no compatible EVS mode can be found to operate transcoder free. The agreement between operators can only operate when VoLTE terminals use "open-offer" at session negotiation. Indeed, an important motivation of the "open-offer" is to guarantee that tandem free interworking can be established with any other remote terminal and network, regardless of whether the remote network is 2G/GSM, 3G/WCDMA, 4G/LTE, WiFi, fixed, non-3GPP or anything else, no matter what configuration that terminal and network allows.

When operating between VoLTE and CS networks, an upgrade is required to the Media Gateway to handle transcoding from and to different codecs. When EVS is supported on both sides, transcoder free interworking can be achieved. This is only possible if the subset of EVS modes operated on the CS side is compatible. By sending an "open-offer", the VoLTE terminal allows the network to limit the EVS modes to the ones compatible with the CS side. In order to achieve backwards compatibility and interworking with CS networks with transcoder free operation (TrFO), for optimal performance and resource utilization, it is required that the three lowest AMR-WB codec modes are included in the VoLTE mode set. Backwards compatibility with transcoding is always a possibility.

When operating between two CS networks with EVS, it can be a challenge for operators to agree on a single codec set. The standard can limit the relevant sets but should not limit the flexibility offered to operators to run their networks at their preferred operating points. One proposal that can improve the situation is if the EVSoCS mode sets are constructed such that the lowest bitrates are always included and when an audio bandwidth is offered, all smaller audio bandwidths are also offered. With this principle, it is always possible to find a compatible set enabling TrFO between CS networks.

4.2. Mobility and AMR-WB backward compatibility of EVS

Mobility is an essential feature of mobile communication systems and continuity of voice calls is expected by moving users. The introduction of LTE made the CS fallback and SRVCC handovers from VoLTE to 2G/3G systems essential. Typically, the SRVCC procedure requires the addition of a transcoding function in the path. This comes with speech interruptions and with added costs.

As described in section 2.2, a subset of the EVS modes is compatible with the AMR-WB codec. In addition to allowing an EVS implementation to efficiently replace an AMR-WB implementation, this also enables transcoder free operation between an end-point operating EVS and the other operating AMR-WB under the condition that a compatible mode set is used.

In the case of SRVCC during an EVS VoLTE-VoLTE call, assuming the used codec in the target CS network is known, it is well possible to avoid any transcoding function and minimize speech interruptions in the following way: If the target codec is EVS over CS, the gateway can – prior to the switch – control the EVS codec rate used by the VoLTE terminal such that it is compatible with the target EVS set. During the switch, no interruption due to incompatibility of speech modes is experienced. Likewise, if the target codec is AMR-WB, the gateway would – prior to the switch – make sure that the VoLTE terminal uses AMR-WB IO modes that are compatible with the target AMR-WB mode set.

5. CONCLUSION

Today 3GPP voice services are globally implemented in a large variety of 3GPP systems that have been continuously evolving like the service itself. Improvements to service quality and system efficiency/capacity have been the driver of that evolution under the fundamental constraint to warrant interworking with legacy systems and devices. HD voice is now broadly deployed and VoLTE is taking off. 3GPP has standardized a new codec for EVS over LTE and launched the standardization process to make EVS available even for 3G CS networks. This will take the evolution to its next stage.

Despite large system heterogeneity the evolution towards EVS proceeds along a smooth migration path. It turns out that underlying principles of codec adaptation at call setup and in-call have not substantially changed since they were first time applied with AMR. This means that there are many commonalities which facilitate a smooth evolution. Careful design of suitable hand-over mechanisms supporting efficient interworking and service continuity in various mobility and roaming cases is another essential component of this. One important aspect with the introduction of the new EVS is that it can be done with minimum system impact. In case the system already supports VoLTE based HD voice no other network upgrades than just adding EVS codec support to the control plane are necessary. Further upgrades of SBG/BGF will enable transcoder free interworking with CS 2G/3G HD voice deployments and transcoding to other HD voice codecs in e.g. the fixed-network or Internet realms. Migration towards EVS will be smooth even in terms of radio network planning. EVS can operate in its high-quality SWB modes providing natural voice sound unprecedented in mobile telephony at rates no larger than required for existing HD voice. The deployment in an operator-controlled network with QoS mechanisms in place ensures guaranteed telecom grade service quality. This is a substantial advantage over OTT voice services that, in contrast, cannot guarantee good voice service quality for the users.

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