3GPP TSG-RAN WG2 Meeting #72bis **R2-110334**
Dublin, Ireland, Jan 17-21, 2011

Agenda Item: 7.7
Source: Motorola
Title: Delay and Voice Capacity
Document for: Discussion and Approval

1. Introduction

At the last meeting RAN WG2 received an LS from SA WG4 requesting input from RAN WG2 on the relation ship of voice capacity and end-to-end delay.

2. Discussion

During Rel-8, RAN1 and RAN2 spent substantial effort to define mechanisms that ensure high capacity for voice services in varying radio channel conditions. In order to meet the rather stringent requirements of voice, specifically:

1. maximum delay from mouth-to-ear of 200 ms (for a UE-UE call), and
2. cell edge outage of no more than 2%,

RAN has defined SPS and TTI bundling. These mechanisms were proved to be necessary in order to support the three different coverage scenarios of most interest to operators:

1. Case 1: inter-site distance of 500m, 20 dB penetration loss, 3 km/h
2. Case 2: Inter-site distance of 500m, 10 dB penetration loss, 30 km/h
3. Case 3: inter-site distance of 1.7 km, 20 dB penetration loss, 3 km/h

In case of Case 3, even with TTI bundling and the small packet sizes of AMR 12.2, as many as 20 re-transmissions are seen to be necessary to ensure no more than 2% blocking at cell edge. TTI bundling is necessary in this case to meet the coverage and stringent delay requirements.

EVS is being defined in order to improve the quality and coding efficiency for conversation applications. Annex A is an extract of the Objective of the WI in SA WGs. Thus while non-speech signals are to be supported to enhance user experience, the primary goal continues to be improved conversational user experience with higher quality and coding efficiency.

It is expected that the inter-arrival rate of conversational media frames will continue to be 20 ms. In addition, the maximum mouth-to-ear delay of 200 ms still has to be met to ensure appropriate user experience. Given the varying radio environments used during the definition of Rel-8, the codecs need to be operational not only in varying propagation conditions but also in varying network conditions supporting different end-to-end delays.

Thus it can be seen that any new codec defined, needs to operate within the delay bounds supported by the RAN mechanisms defined in Rel-8. SA WG4 indicated in their LS that the TB sizes of interest for EVS vary from 144 to 2664 bits, the maximum packet size reflecting an almost 10-fold increase in packet size over AMR 12.2. These large packets would need significant retransmissions and cause a large hit to capacity. In case of Case 3, such packet sizes are most likely not feasible. If any delay budget would be available to steal from the network, then this would be needed simply to enable more retransmissions of these larger packets and even with this increase the outage criterion would not be met.

3. Conclusion

It is proposed to reiterate to SA WG4 that any new codec to be defined under the EVS WI in support of conversational applications needs to continue to be bound by the radio interface limitations of Rel-8. In particular the delay currently budgeted for the radio interface cannot be modified due to the need to support different coverage scenarios and the impact on capacity. If the EVS codec is to support larger packet sizes, the situation is even more challenging since Rel-8 has been optimized for small packet sizes (AMR 12.2). From the analysis of current network deployments there does not seem to be any delay budget available in the network to steal for higher algorithmic delay in the codec or increased retransmission count. If any delay budget would be available this would need to allocated for the increased retransmission count to be supported for the larger packet sizes of interest to SA4.

4. References

[1] R2-106676, LS on impact of LTE air interface transmission delay on LTE system capacity for voice services, source SA WG4

[2] LTE, The UMTS Long Term Evolution: From Theory to Practice, Stefania Sesia, Issam Toufik, Matthew Baker (eds)

5. Annex A

Extract from SP-100202 – WID: Codec for Enhanced Voice Services

4 Objective

The overall objective of this work item is to develop a codec suitable for the Enhanced Voice Service in the EPS. The following objectives should be achieved with the new codec:

- *<u>Enhanced quality and coding efficiency for narrowband (NB) and wideband (WB) speech services, leading to improved user experience and system efficiency</u>. This should also be achieved in interoperation with pre-Rel-10 systems and services employing WB voice.*
- *Enhanced quality by the introduction of super-wideband (SWB) speech, leading to improved user experience.*
- *<u>Enhanced quality for mixed content and music in conversational applications</u> (for example, in-call music), leading to improved user experience for cases when selection of dedicated 3GPP audio codecs is not possible.*
- *Robustness to packet loss and delay jitter, leading to optimized behaviour in IP application environments like MTSI within the EPS.*
- *Backward interoperability to the 3GPP AMR-WB codec by having some WB EVS modes supporting the AMR-WB codec format used throughout 3GPP conversational speech telephony service (including CS). The AMR-WB interoperable operation modes of the EVS codec may be either identical to those in the AMR-WB codec or different but bitstream interoperable with them.*

These objectives shall be achieved while meeting all design constraints and performance requirements set forth in TR 22.813. It is further desirable that the codec fulfills needs for enhanced voice services in other 3GPP systems, such as CS.
The developments under this work item should lead to a set of new specifications defining among others textual description of the coding algorithm and the VAD/DTX/CNG scheme. Following 3GPP practice, fixed-point and floating-point C code and associated test vectors should also be part of this set of specifications. The included AMR-WB interoperable coding format may become an alternative implementation for AMR-WB operation, provided that the enhancements are consistently significant. Jitter buffer management and packet loss concealment should be specified as part of the set of EVS specifications.
The EVS Service Requirements will be defined by SA1 into TS 22.173 "IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1" using the EVS Service Requirements identified in TR 22.813.
Potential architectural impacts through the introduction of the EVS codec into 3GPP systems will be addressed by involving and liaising with the responsible 3GPP working groups such as SA2.
Note: It is envisioned that subsequent work outside this work item will address suitable acoustic requirements, considering the extended audio bandwidth beyond wideband that currently standardized test equipment does not support.

Annex B:
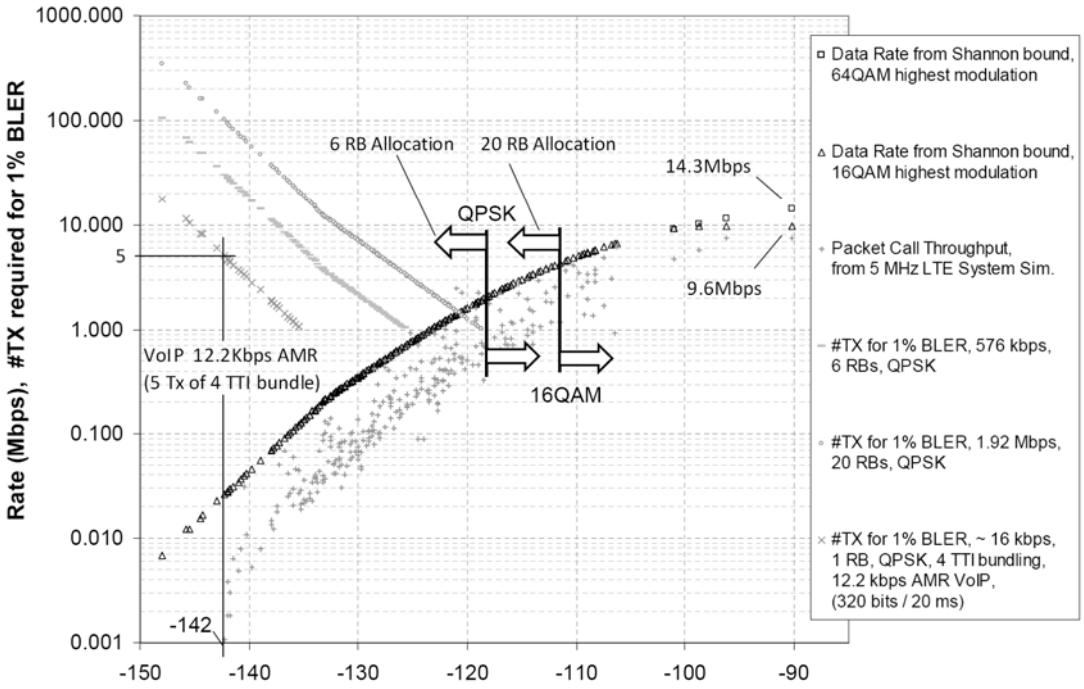
Figure showing bit rate/number of retransmissions as a function of path loss [2]