Capacity Enhancement of VoIP over LTE by Stochastic Adaptive Modulation and Coding

Kambiz Homayounfar and Bijan Rohani†

PHYBIT, INC.
AM Building 5F, 2-3-3 Higashi Gotanda, Shinagawa, Tokyo 141-0022, Japan
E-mail: kambiz@phybit.com

†PHYBIT, INC.
10-10 Cendex Center, 120 Lower Delta Road, Singapore 169208
E-mail: bijan@phybit.com

Abstract The Erlang capacity of Voice over Internet Protocol (VoIP) over Long-Term Evolution (LTE) is enhanced by using stochastically adaptive modulation and coding (SAMC). A Channel Quality Indicator (CQI) is derived which makes no assumption about radio channel characteristics and is therefore highly robust against mobility patterns. In effect, SAMC minimizes the number of physical layer (PHY) re-transmissions caused by Hybrid Automatic-Repeat Request (HARQ). Simulation results indicate that SAMC doubles Erlang capacity of VoIP over LTE while providing even lower average delay per user equipment (UE).

Key words LTE, VoIP, Erlang Capacity, AMC, HARQ

1 Introduction

To provide adequate VoIP quality, two conditions must be met for each subscriber. First, the delay from sender to receiver must be as low as possible. In fact, with no other distortion, delays beyond 200 milliseconds (ms) begin to be objectionable and must be avoided. Second, packet loss must be between 1% to 3%, otherwise audible distortions become unbearable and subscribers terminate calls early, resulting in loss of revenues for mobile operators. The success of VoIP in enterprise, private, and public Internet comes largely from the fact that with advanced routers, packet loss and delays in wired networks are kept to a minimum. Mobile wireless networks, however, are just beginning to deploy VoIP services and challenges to ensure adequate and acceptable voice quality still remain. Given that a large portion of operator revenues come from voice telephony, the significance of good VoIP quality in LTE is hard to dispute.

1.1 Defining QoS

The LTE system [1] is an ALL-IP (Internet Protocol) mobile broadband service wherein the transmission of real-time data such as voice and video is carried entirely over packet-switched connections. In LTE, the mechanism for providing end-to-end Quality of Service (QoS) is based on two parameters. First, for each User Equipment (UE) a Layer 2 Packet Delay Budget is specified during each session. This ensures that delay-sensitive packets arrive at their destination on time. Second, a Layer 2 Packet Loss Ratio (L2PLR) is specified. If a connection stays within the limits set by L2PDB and L2PLR, then we can say that the QoS level for a subscriber is satisfactory. For example, a VoIP connection may have an L2PDB of 60 ms together with a L2PLR of 2%. These parameters are not constants. They can be varied by the mobile operator depending on the congestion levels, time of day, or coverage location.

1.2 Sources of Delay and Packet Loss

In wired broadband networks, such as Ethernet or private networks, packet loss comes from congestion. Routers drop packets when their buffers become full or when packets arrive too fast. With giga-rate wired connections, this hardly happens these days. The low delay and packet loss characteristics of wired networks have paved the way for popularity of Skype and Youtube. In wireless networks, however, the situation is different. The radio path between the base-station (Evolved Node B, or eNB in LTE vocabulary) and the UE is the main cause of delay and packet loss. If wireless networks are to compete with wired networks, then the radio link must be made as robust as possible. It is fair to say that all the newest physical layer techniques like Multiple-Input-Multiple-Output (MIMO) schemes, HARQ, and advanced channel coding (Turbo codes, codes with low-density parity check LDPC) are designed to reduce the bit errors and the delay over radio paths. Within the QoS framework of LTE another interesting challenge appears. Meeting the delay and the packet loss limits set by L2PDB and L2PLR by its very nature needs cooperation between Layer 2 (Media Access Control, or MAC) and Layer 1 (the physical layer PHY). It is therefore a cross-layer function.

At this point the purpose and the contribution of this paper can be stated. We show simple ways to map L2PDB and L2PLR to quality targets that the PHY must meet. This goal is in turn achieved by adapting the Modulation and Coding Scheme (MCS) at the physical layer in order to reduce the number of bit errors. As LTE is heavily dependent on HARQ, reducing the bit errors also reduces the delay of the radio link. Our focus is on the PHY layer, and for clarification, an example of how the PHY delay fits into overall LTE delay budget is shown in Fig. 1.

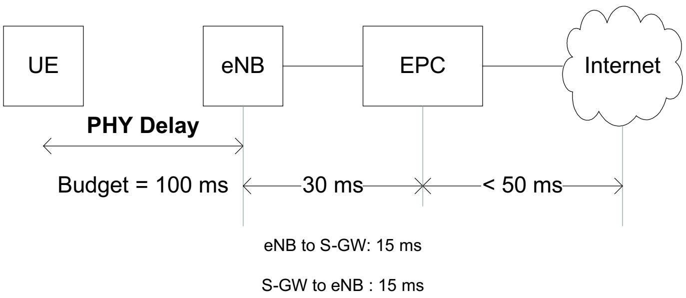

Figure 1: Illustration of Delays in LTE

2 Stochastic Adaptive Modulation & Coding

The usual approach to AMC is select a particular MCS from a list of available MSC schemes. (SAMC shares the same objective.) In most AMC algorithms, however, an estimation of Bit-Error-rate (BER) is mapped to a signal-to-noise ratio (SNR). Quantization of SNR then leads to the choice of MCS. In LTE, there are 31 choices, so a 5-bit Channel Quality Indicator can convey the MCS selection to a remote transmitter. Yet, SNR-based AMC algorithms suffer from two flaws. First, the relation between SNR and BER needs a channel model. In practice, as mobile move around the cell, they experience wildly varying channels, and this variability lowers the performance of SNR-based AMC algorithms. The second flaw comes from limitations on how SNR can be calculated with a reasonable degree of accuracy. For instance, if pilot, or reference signals, are used to measure the SNR, then the result says something about the quality of the pilot bits, and not about the bits destined for the actual subscriber.

2.1 The Idea Behind SAMC

SAMC is about statistical control of radio link quality in order to achieve the lowest bit error rate during transmission. Its principle is shown in Fig. 2. SAMC is in effect a method for AMC which uses a stochastic metric, namely the Bit Error Probability (BEP), instead of the SNR. This metric is readily calculable from the soft output of the channel decoder, the log-Likelihood Ratio (LLR), which we denote by λ_i, by:

$$\varepsilon_i = \frac{1}{1 + e^{|\lambda_i|}} \tag{1}$$

where ε_i denotes BEP (hereafter we use ε_i and BEP interchangeably). The statistical properties of λ_i and ε_i are documented in detail by [2]. To use ε_i for AMC, we must map it to a Channel Quality Indication (CQI) metric which is fed back to eNB by the UE. Several mechanisms are available in LTE for CQI reporting and here we choose the simplest scheme whereby a 5-bit CQI is sent every Transmission Time Interval (TTI). The purpose of Channel Quality function in Fig. 2 is to measure statistical properties of ε_i in order to determine the quality of the channel. One way to do this is to measure the stationarity of ε_i. The more the non-stationarity, the worse the channel quality. Higher order modulation (64-QAM) can be used for stationary channels, while lower-order modulation (4-QAM) are reserved for highly non-stationary channels. No assumptions need be made about mobility patterns or fading characteristics. All that is needed is a quality target (in our case a maximum BER) that must be achieved through the process of CQI feedback and subsequent MCS adjustment. Considering the channel quality as a stochastic process that is stabilized, or made stationary, through quality measurement and CQI feedback provides a simple and effective framework for AMC.

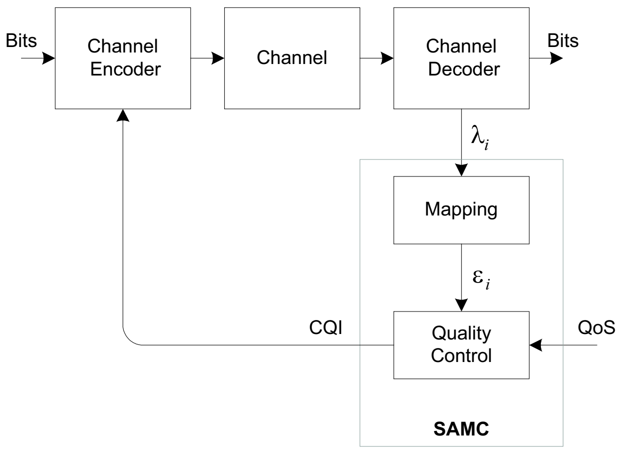

Figure 2: Principle of SAMC

2.2 Cross-Layer SAMC

Cross-layer SAMC is the case where the quality target for Channel Quality function comes from the MAC layer. Its configuration is shown in Fig. 3. We need the quality target to be in the form of a maximum BER. So we now show how L2PDB and L2PLR can be used in adopting AMC policies with the aim to improve overall network throughput. The following are reasonable assumptions:

- UE receives IP packets and requests AMC modes
- UE has the knowledge of the round-trip delay Trrt
- UE measures the packet elapsed life Te since arrival at eNB Packet Data Convergence Protocol (PDCP)
- UE has the QoS parameters L2PDB and L2PLR

UE then uses SAMC for choosing MCS mode based on deriving the CQI from BEP as follows. In [3], it is shown that Pmax can be calculated from:

$$P_{\max} = L2PLR^{\frac{Trrt}{L2PDB-Te}} \tag{2}$$

Pmax itself relates to BER as follows:

$$P_{\max} = 1 - (1 - BER)^N \tag{3}$$

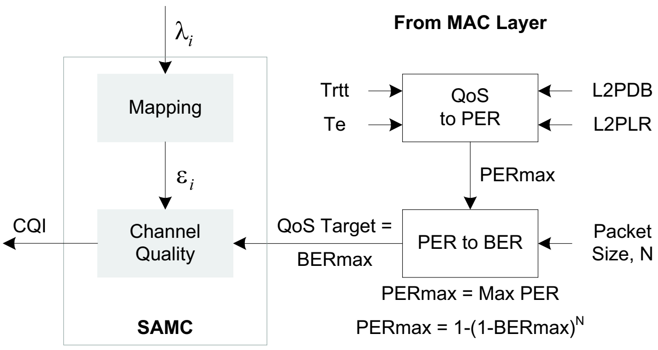

Figure 3: Cross-Layer SAMC

where N is the number of bits in a packet. In SAMC we can update the target value for BEP every time an invalid packet is received. The update becomes necessary as the L2PDB is used up on arrival of every invalid packet. This is accounted for by the term (L2PDB-Te) in (1). The updating of the target for BEP in three steps, as follows:

1. A new value for Pmax is calculated from Eq. 2.
2. The calculated Pmax is used in Eq. 3 to obtain the required BER.
3. The BER is then used as the new quality target for SAMC.

2.3 From ε_i to CQI

Determination of CQI from ε_i can be achieved by empirical methods. We found it effective to count the number of time ε_i exceeds BERmax within a TTI. The higher this number, the worse the channel. So a CQI corresponding to the most robust MCS ought to be fed back. Other ways for mating ε_i to CQI could, for example, include measuring the variations in the mean value of ε_i as a time series. A non-stationary time series has by definition a non-constant mean over a sample realization. (How the mean of a time series varies over time is an excellent indication of non-stationarity.)

3 Simulation Results

Tab. 1 shows the parameters in our system-level simulations. We compared SAMC with a simple SNR-based AMC scheme wherein measured pilot (reference signal) SNR values are quantized to form 5-bit CQI metrics.

Consider the PHY delay distribution of Fig. 4. As the load on the radio link increases from 75 Erlangs to 100 Erlangs, the radio transmission Delay for each UE increases. This is because now the radio resources must be shared between more UEs. Each UE gets lower throughput and experiences more HARQ retransmissions. So it takes longer to transfer its data over the radio link.

SAMC makes the link more reliable by choosing the modulation and coding scheme that gives the lowest error rate. Even when UE traffic doubles from 75 to 150 Erlangs, the SAMC algorithm lowers the delay by reducing the number of HARQ re-transmissions. Similar improvement happens when we increase the traffic load from 100 to 200 Erlangs as in Fig. 5.

Table 1: Simulation Parameters

Parameter	Setting
Voice Compression	AMR 12.2 kbps
Voice Activity Factor	50%
Header Compression	ROHC (IETF RFC 3059)
frame Bundling	No
Cell Layout	19 sites, 3 sectors
eNB-eNB Distance	1 km
LTE Bandwidth	20 MHz
Multipath Fading	Ped-B (50%) Veh-A (50%)
Propagation Model	Hata
Shadowing Std. Dev.	10 dB
UE Receiver Diversity	None
MCS Levels	25
Scheduling Algorithm	Proportional Fair
Resource Allocation	Greedy Algorithm
RLC	No ARQ
HARQ Algorithm	Chase-Combining
HARQ Processes	8
HARQ Parameters	Max. 4 Re-trials, Sync.

4 Conclusion

We formulated a method for mapping Layer 2 QoS parameters in LTE to a bit error target that must be achieved by the PHY layer. Through the adjustment of

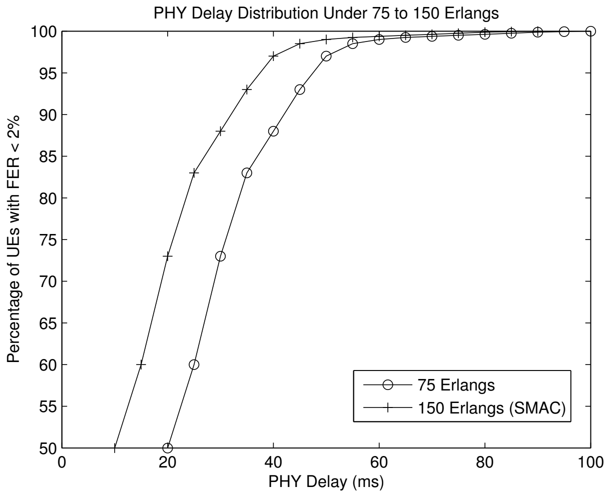

Figure 4: PHY Delay Distribution at 75 and 150 Erlangs with and without SAMC

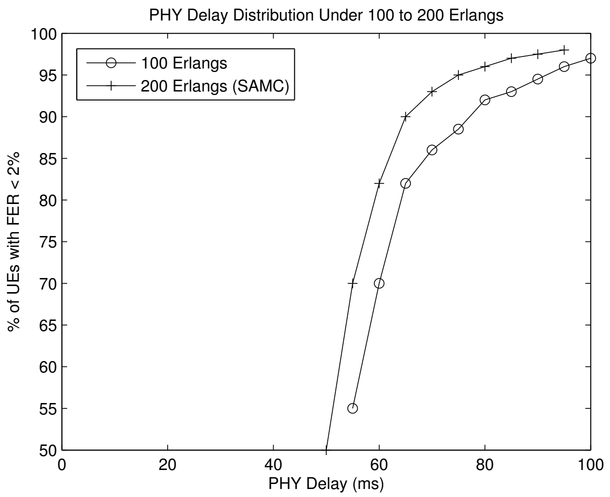

Figure 5: PHY Delay Distribution at 100 and 200 Erlangs with and without SAMC

MCS (achieved by CQI feedback), an effective stochastic AMC loop was derived. This cross-layer cooperation helps to reduce the number of HARQ retransmissions, and therefore reduces the delay over the radio link. Our simulation results show that, compared with SNR-based AMC schemes, the use of SAMC results in lower average delay per UE even under double traffic loads .

References

[1] E. Dahlman. *3G Evolution: HSPA and LTE for Mobile Broadband*. Academic Press, 2008.

[2] B. Rohani and K. Homayounfar. Computation of instantaneous bit error probability from log-likelihood ratio. In *IEICE Technical Report*, volume RCS2008-150, pages 119–123, November 2008.

[3] K. Jun. Packet qos aware amc selection for 3g evolved packet system. *IEICE Transactions on Communications*, E91-B(8):2740–2743, August 2008.