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A dynamic rate adaptation algorithm using WB E-model for voice traffic over LTE network

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Abstract—This paper presents a dynamic adaptation algorithm of joint source-channel code rate for enhancing voice transmission over LTE network. In order to assess the speech quality, we use the Wideband (WB) E-model. In this model, both end-to-end delay and packet loss are taken into account. The goal of this paper is to find out the best suboptimal solution for improving voice traffic over LTE network with some constraints on allowed maximum end-to-end delay, allowed maximum packet loss and minimum required bandwidth. The best suboptimal choice is channel code rate corresponding to each mode of the AMR-WB codec that minimizes redundant bits generated by channel coding with an acceptable MOS reduction. Besides, this algorithm can be integrated with rate control in AMR-WB codec to offer the required mode of LTE network. Our results show that the MOS degradation is not significant, but the percent of reduced redundant bits to be very considerable. This will require less bandwidth, thus, more mobile users can be served. The algorithm has simple computational operations, it can be applied to real-time voice communications.

Index Terms—AMR-WB, Wideband E-model, VoIP, VoLTE, Source-Channel code rate, Adaptive algorithm

I. INTRODUCTION

In present market of mobile communication in the world, 3GPP LTE (The third Generation Partnership Project Long Term Evolution) is developing strongly and is deployed by the most communications operators. LTE network is based on All-IP network and does not support circuit switching method which is utilized to provide the voice call service in 3G networks. So that, in order to support voice service over LTE network, additional technology has to be included. Voice over LTE network (VoLTE) service was developed to supply voice and video communication and Short Message Service (SMS) on the LTE network [1]. According to [1], there are two types of voice traffic over LTE network, those are VoLTE and VoIP. VoLTE was launched in 2012, and at present, many mobile network operators in the world have provided VoLTE service. VoLTE is VoIP (Voice over Internet Protocol) based multimedia service in which voice call and video conference services can be supplied. VoLTE is signalling protocol that enables carrying of voice packets and is guaranteed of given QoS (Quality of Service) information by LTE network operator [2]. Otherwise, VoIP relies on the internet which is done on a “best effort” basis to deliver voice packets. It is said that VoLTE is basically a subset of VoIP. It is enhanced VoIP over specific access technology (LTE) with given QoS information, but it is still VoIP.

When voice traffic is transmitted over LTE network, the voice signal firstly is compressed at Application layer by AMR-WB (Adaptive Multi-rate Wideband) codec, and then it is packetized into RTP payload. When this payload goes through each layer, it is packetized into the corresponding packet and the header is added. In order to protect the voice packet when it is delivered over a noisy channel, some error correcting technologies are included. The Forward Error Correction (FEC) channel code is widely used in LTE network for data channels is Turbo code. Channel coding reduces Bit Error Rate (BER), so that the speech quality will be improved. Channel coding encodes a k -bits block into a n -bits codeword, thus, the number of redundant bits equal to $n - k$, and so that, the code rate is k/n . This means that the higher channel code rate, the higher speech quality, but this leads to the longer delay and the higher redundancy. Therefore, there needs to be a tradeoff between speech quality and channel code rate.

There are several authors who have proposed techniques to improve the speech quality delivered over a noisy channel. Examples include the works of [3], [4], [5], [6], [7], and [8]. But the closest works related to our paper are represented in [3], [7], and [8]. The authors in [3] present a dynamic joint source channel coding rate adaptation algorithm for VoIP using AMR codec. The algorithm computes the optimal rates allocated to each frame for a set given QoS constraints. The aim of their paper is to find the tradeoff between packet loss recovery and end-to-end delay to maximize perceived speech quality. In [7], the authors propose an optimization issue for supplying unequal error protection of speech frames according to their importance. An optimization framework for identifying the optimal joint source-channel code rate of each voice frame based on the frame perceptual importance is proposed in [8]. In that paper, the quality of the received speech signal is maximized.

The main idea of this paper is to extend results in [3] in context of voice traffic over LTE network. Besides, instead of finding the tradeoff between packet loss recovery and end-to-end delay to maximize the perceived speech quality, our proposal focuses on finding out the compromise between source code rate and channel code rate to minimize the number of redundant bits generated by channel coding with an acceptable Mean Opinion Score (MOS) degradation. It means the proposed algorithm offer a suboptimal solution for the tradeoff between speech quality and redundancy generated by channel coding. We want to offer an other point of view of

choosing the channel code rate corresponding to each mode of AMR-WB codec for voice traffic over LTE network. In order to assess speech quality, we use the WB E-model [9]. In this model, the transmission rating factor (R_{wb}) is used as a measure of subjective quality. This factor is then mapped to the corresponding MOS.

The rest of this paper is organized as follows: Overview of voice transmission over LTE network is described in section II. In section III, we present the proposed algorithm. The simulation results and performance evaluation of the proposed algorithm are analysed in section IV. The conclusion and future work is represented in section V.

II. VOICE TRAFFIC OVER LTE NETWORK: OVERVIEW

A. Voice traffic protocol stack layers

In LTE network, the speech frame is packetized sequentially with network protocols, including Real-time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP). And then, it will be encapsulated with other radio protocols as Packet Data Convergence Protocol (PDCP), Radio Control Link (RLC) and Mac Access Control (MAC). All of these protocols will add their headers into the packetized speech packet.

In order to decrease data overhead of above protocols headers when the speech packet transmitted over a radio channel, Robust Header Compression (RoHC) is used. This will save bandwidth and enhance voice transmission over LTE network. RoHC compresses IP header from 40 bytes (with IPv4) and 60 bytes (with IPv6) down to 1 to 3 bytes [10]. For voice traffic over LTE network, HARQ (Hybrid Automatic Repeat Request) technique is used at MAC layer for retransmission if FEC fails error correction. Each speech packet will be retransmitted at least from one to three times. The retransmission times depends on the error correction or the configured maximum times of retransmission.

B. Speech Codec

Voice traffic in LTE uses AMR-WB as a vocoder. AMR-WB codec is a speech codec which has been developed by ETSI (the European Telecommunications Standards Institute) and applied in the 3GPP LTE network for voice compression and decompression. It is detailed described in [11]. AMR-WB codec uses a sampling rate of 16 kHz, which covers 50-7000 Hz audio bandwidth. It has 9 different codec modes (from mode 0 to mode 8) corresponding to 9 source bit rates in range of 6.6-23.85 Kb/s. Each of them generates encoded 20 ms speech frame and switches among them every 20 ms. The bits in the encoded speech frame to be ordered according to their subjective importance. These bits are divided into three classes with reducing perceptual importance: Class A, Class B and Class C. Total bits of each class depends on codec mode. In this study, we consider the same level of error protection for these three classes. Thus, the bits of these classes are equally protected by channel coding.

C. LTE channel coding

One of the crucial issues of digital communication is error correction. When data is transmitted over a noisy channel, it will be distorted by noise. Thus, the protection of data over noisy channel is mandatory. This so called is channel coding. In LTE network, encoders are used in channel coding including: Block code used for CRC (Cyclic Redundancy Code) and HARQ, Convolutional code used for control channels, with data channels, Turbo code is used. Turbo code is an enhanced Convolutional code. It is a Parallel Concatenated Convolutional Code (PCCC) with two eight-state constituent encoders and one turbo code internal interleaver, with a coding rate of 1/3 [12]. Standard turbo code rates are 1/3, 3/4, and 4/5, where code rate 1/3 is the original code rate. Turbo code rate is chosen based on CQI (Channel Quality Indicator) index [13]. CQI index includes 16 values, where value of 0 is not used. Values of CQI index from 1 to 6, 7 to 9, and 10 to 15 are corresponding to Turbo code rates of 1/3, 3/4, and 4/5. Each CQI index is mapped to a SINR (Signal-to-Interference-plus-Noise Ratio) value. SINR also has 15 values (from -6.7 dB to 22.7 dB) [14] counted by the receiver and sent to the transmitter. Channel coding will map from SINR value to corresponding CQI index, and then chooses the corresponding channel code rate.

In Turbo Encoder, each information bit stream is a k -bits block. This block size in range of 40-6144 bits. So each block can include one or several speech packets. A k -bits block be encoded into a n -bits codeword. So that, the number of redundant bits of LTE channel coding is $n-k$ bits and channel code rate is k/n .

D. Wideband E-model: Speech quality assessment for Wideband Audio

WB E-model is a computational model developed and standardized by ITU-T [9]. It is used to estimate the MOS for wideband audio quality. The output of the model is R-factor. The values of this R-factor in range of 0-129. And then, it is mapped to the MOS. The R-factor in WB E-model is defined as follows:

$$R_{wb} = R_{0,wb} - I_{s,wb} - I_{d,wb} - I_{e,eff,wb} + A \quad (1)$$

In which:

- $R_{0,wb}$: The basic signal-to-noise ratio;
- $I_{s,wb}$: The simultaneous impairment factor, it is the sum of all impairments which may occur more or less simultaneously with the voice transmission. In this model, this factor is set to 0;
- $I_{d,wb}$: The delay impairment factor, representing all impairments due to delay of voice signals;
- $I_{e,eff,wb}$: The equipment impairment factor, capturing the effect of signal distortion due to low bit rates of the codec and packet losses of random distribution;
- A : The advantage factor, capturing the fact that some users can accept a reduction of quality due to the mobility of cellular networks. In this model, this factor is set to 0.

In above factors, $I_{d,wb}$ and $I_{e,eff,wb}$ are affected by end-to-end delay and packet loss, respectively, while $R_{0,wb}$ and $I_{s,wb}$ do not depend on network performance. The R_{wb} factor is translated into the MOS as follows [9]:

$$R_x = R_{wb}/1.29$$

- For $R_x < 0$: $MOS = 1$
- For $0 \leq R_x \leq 100$:

$$MOS = 1 + 0.035 \times R_x + 7 \times 10^{-6} \times R_x \times (R_x - 60) \times (100 - R_x) \quad (2)$$

- For $R_x > 100$: $MOS = 4.5$

R_{wb} factor is mapped to the MOS using the above equation, and then, the MOS is mapped to the satisfaction level of the users. According to [15], with the wideband audio, the value of $R_{0,wb}$ factor in equation (1) equal to 129. Thus, equation (1) can be rewritten as follows:

$$R_{wb} = 129 - I_{d,wb} - I_{e,eff,wb} \quad (3)$$

In order to compute the R_{wb} factor, we have to count the values of $I_{d,wb}$ and $I_{e,eff,wb}$ factors. The $I_{d,wb}$ factor is determined by the following equation [16]:

$$I_{d,wb} = 0.0024 \times D_{e2e} + 0.11 \times (D_{e2e} - 177.3) \times H(D_{e2e} - 177.3) \quad (4)$$

In which: $H(x)$ is the Heavyside function:

$$H(x) = \begin{cases} 0, & \text{if } x < 0 \\ 1, & \text{otherwise} \end{cases} \quad (5)$$

In equation (4), D_{e2e} represents the total end-to-end delay (or mouth-to-ear delay) of speech packet. It will be described specifically in section III. The $I_{e,eff,wb}$ is determined according to packet loss. In this study, we estimate packet loss probability at the receiver, after FEC. The output bits of AMR-WB encoder will be encoded by channel coding (Turbo code). In fact, the bits in class A, class B and class C can be encoded with different channel code rates. In addition, 8-bits CRC code is applied to protect class A bits. In figure 1, the packet loss represents the average rate of speech frames for which CRC check fails in class A bits. It is determined for each mode of AMR-WB codec. According to [17], $I_{e,eff,wb}$ is determined as follows:

$$I_{e,eff,wb} = I_{e,wb} + (129 - I_{e,wb}) \times \frac{P_{pl}}{P_{pl} + B_{pl}} \quad (6)$$

In which: $I_{e,wb}$: The respective impairment factor without any packet loss. P_{pl} : Packet loss rate. B_{pl} : A codec-specific factor which characterizes its robustness against packet loss.

III. THE PROPOSED ALGORITHM FOR MINIMIZING REDUNDANT BITS GENERATED BY CHANNEL CODING

As mentioned above, in the WB E-model, the value of the MOS depends on both impairment $I_{d,wb}$ and $I_{e,eff,wb}$ factors and they have direct relationship with end-to-end delay and packet loss. If increasing end-to-end delay leads to decreasing the MOS while reducing packet loss leads to increasing the MOS. Therefore, finding out the suboptimal joint source-channel code rate solutions is very essential. In this paper,

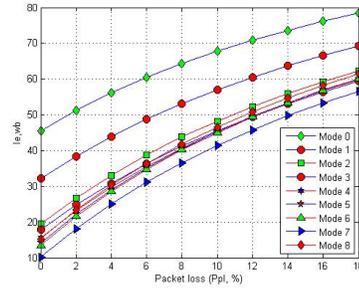


Fig. 1. $I_{e,wb}$ vs. Packet loss for nine modes of AMR-WB codec

we offer another viewpoint of choosing the suitable channel code rate corresponding to each mode of AMR-WB codec for minimizing the number of redundant bits generated by channel coding with an acceptable MOS reduction.

A. The calculation of the delay impairment factor

In order to calculate this factor, we have to compute the end-to-end delay. According to [18], the end-to-end delay can be counted as follows:

$$D_{e2e} = D_{enc} + D_{network} + D_{play} \quad (7)$$

Where:

- D_{enc} : The delay time caused by encoding and packetizing at the AMR-WB encoder, $D_{enc} = k \times T \times f + l_a + D_{pack}$ With: T : The speech frame size, $T = 20$ ms. f : The number of frames of a speech packet. l_a : The look-ahead delay, $l_a = 5$ ms for all modes of AMR-WB codec. D_{pack} : The packetization delay for grouping f frames into one speech packet, i.e. $D_{pack} = (f - 1) \times T$.
- $D_{network}$: The sum of transmission delay, propagation delay and queuing delay at each hop h in the network path from the transmitter to the receiver. The transmission delay (T_h) is computed using the following equation [18]: $T_h = (n - k + 1) \times (T \times f \times R_s + H_{overhead}) \times \sum_h \frac{1}{B_h}$

In which: R_s : AMR-WB bit rate before channel coding. $H_{overhead}$: The number of overhead bits introduced by RTP/UDP/IPv6/PDCP/RLC/MAC headers and 24 bits introduced by CRC at Physical layer. B_h : The bandwidth at hop h .

The propagation delay (P_h) depends on the distance from the source to destination. It is negligible if within a local area. For intra-continental calls, the propagation delay is in the order of 30 ms and for inter-continental calls, it can be as large as 100 ms [18]. It is clear that for a given voice connection, the only random component of voice delay (that is the only source of jitter) consists of queuing delay in the network [18], $Q = \sum_h Q_h$.

- D_{play} : The playback delay, voice packets are usually delayed in a jitter buffer and the fixed playback delay must be equal to at least two speech frame length [19], i.e. $D_{play} = 2 \times T$.

From above analysis, we get the end-to-end delay determined as the following equation:

$$D_{e2e} = k \times T \times f + l_a + (n - k + 1) \times (T \times f \times R_s + H_{overhead}) \times \sum_h \frac{1}{B_h} + \sum_h (Q_h + P_h) + 2 \times T + D_{rohc} + D_{harq} \quad (8)$$

With D_{rohc} : The delay of RoHC processing time at PDCP layer. According to [20], RoHC should not noticeably add to the end-to-end delay and according to [21], this delay is not very significant, approximately 10 - 67 μs /packet to compress and 12 - 51 μs /packet to decompress RoHC packets. Thus, in this study, we consider the delay caused by RoHC equal to 0. D_{harq} : The delay due to retransmission at MAC layer by HARQ. Each voice packet is retransmitted at least one times. According to [22], because the RTT (round trip time) of HARQ is fixed and because of the higher priority for retransmissions, the HARQ delay is normally within 10 ms.

Normally, FEC also can cause the delay. However, a number of authors have pointed out that FEC does not introduce any delay unless there is packet loss. So that, in this study, we do not mention the delay caused by FEC. After calculating the end-to-end delay, we can obtain the impairment factor $I_{d,wb}$ from equation (4).

B. The calculation of the equipment impairment factor

In order to compute this factor, we have to count the packet loss rate after FEC schemes try to recover errors. We assume that the estimates for the packet loss rate P_{pl} on the end-to-end network path is available at time an adaptation decision is being made. In order count the P_{pl} , we assume a random loss model, the relationship between the parameters (k, n) of FEC schemes, "raw" packet loss rate on the end-to-end network path p_r and the packet loss rate P_{pl} is described as follows [20]:

$$P_{pl} = \sum_{i=n-k+1}^n \binom{n}{i} \times p_r^i \times (1 - p_r)^{n-i} \times \frac{i}{n} \quad (9)$$

Equation (9) shows that, we can attain the current p_r from the given measurement P_{pl} and a pair of (k, n) . This p_r value is counted once per adaptation period and is utilized in the proposed algorithm as shown in the next subsection. Figure 1 shows $I_{e,wb}$ versus packet loss(%) after FEC. Thus, we can obtain $I_{e,wb}$ by referring to figure 1.

C. The proposed algorithm

In this algorithm, we solve the problem of choosing suitable source and channel code rate within some constraints on maximum allowed end-to-end delay, maximum permitted packet loss and minimum required bandwidth for minimizing the number of redundant bits generated by channel coding with an acceptable MOS reduction. This algorithm is located at the transmitting side. The inputs of the algorithm include:

- QoS information: As path packet loss P_{pl} , path bandwidth B_h and congestion Q_h . These values can obtain by QoS estimation module used in the network.

- QoS constraints: Maximum allowed end-to-end delay D_{max} , maximum permitted packet loss rate P_{max} and minimum required bandwidth BW_r .

The outputs of the algorithm will be a decision on the choice of source code rate R_s and channel code rate R_c . Besides the most suboptimal choice of channel code rate for every AMR-WB codec mode, the algorithm also offers the best suboptimal choice of channel code rate for all of AMR-WB codec modes. In order to describe the algorithm, we start by setting two the following constraints:

$$\begin{cases} D_{e2e} \leq D_{max} \\ P_{pl} \leq P_{max} \end{cases} \quad (10)$$

For each mode in AMR-WB codec modes, we change k and n over ranges $k = 1, 2, \dots, k_0$ and $n = k + 1, \dots, n_0$ where k_0 and n_0 are maximum available values of channel code rate in LTE network ($n_0 > k_0$). In order to find the most suitable suboptimal pair of (k_{subopt}, n_{subopt}) for all values of R_s , we define the MOS reduction ($MOS_r, \%$) and the percent of decreased redundant bits ($G_r, \%$) as follows:

$$MOS_r = \frac{MOS(k_{best}, n_{best}) - MOS(k_{subopt}, n_{subopt})}{MOS(k_{best}, n_{best})} \times 100 \quad (11)$$

$$G_r = \frac{(n_{best} - k_{best}) - (n_{subopt} - k_{subopt})}{n_{best} - k_{best}} \times 100 \quad (12)$$

We see that if the larger MOS_r , the lower speech quality, and the higher reduced redundant bits, the higher BER (Bit Error Rate) and this leads to the lower speech quality. So that, in order to find out the tradeoff between MOS_r and G_r , we propose the criteria for this as follows: In the pairs of (k_{subopt}, n_{subopt}) of all modes, the algorithm firstly finds a pair which has the maximum value of G_r meeting $MOS_r \leq 1\%$ to ensure that this MOS is very close to the highest MOS. In the case there are many the same maximum values of G_r , the algorithm will secondly choose a pair of (k_{subopt}, n_{subopt}) which has the lowest value of the MOS_r . If there are many the same maximum values of MOS_r , then the algorithm will thirdly choose a pair of (k_{subopt}, n_{subopt}) which has the lowest value of BW_{subopt} where BW_{subopt} is the required bandwidth corresponding to the pair of (k_{subopt}, n_{subopt}) . The required bandwidth of each mobile user (BW_r) must be met the following condition:

$$BW_r \geq \frac{n}{k} \times (R_s + \frac{H_{overhead}}{T \times f}) \quad (13)$$

In which: all factors are as same as ones described in the previous equations.

Finally, the proposed will offer the best suboptimal solution for minimizing the redundant bits generated by channel coding with an acceptable MOS reduction. The steps of the proposed algorithm are described in Algorithm 1.

IV. SIMULATION RESULTS AND PERFORMANCE EVALUATION

In order to simulate the algorithm, we assume simulation parameters as follows:

- The number of speech frames per packet: $f = 1$

Algorithm 1: A dynamic rate adaptation algorithm using WB E-model

Step 1: For all $R_s \in \{\text{mode } 0, \dots, \text{mode } 8\}$:
 For $k = 1$ to k_0
 For $n = k + 1$ to n_0

- Compute the end-to-end delay using equation (8).
- Compute the packet loss rate using equation (9).

Step 2: Find all triples $(R_{s,i}, k_i, n_i)$, $i = 1, \dots, t$ meeting the conditions in equation (10).
 Step 3: Compute $I_{d,wb}$, $I_{e,eff,wb}$, R_{wb} , MOS , and BW_r using the equation (4), (6), (3), (2), and (13) respectively.
 Step 4: Find the highest MOS of each AMR-WB codec mode corresponding to each pair of (k_{best}, n_{best}) .
 Step 5: Find the suboptimal selection for each AMR-WB codec mode for reducing redundant bits of channel coding:
 For $k = 1$ to k_0
 For $n = k + 1$ to n_0

- Find the last pair of (k, n) which has the highest MOS, then mark $k_{subopt} = k$

For $k = 1$ to k_{subopt}
 For $n = 1$ to n_0
 if $n > k$ then

- Find the first pair of (k, n) which has the highest MOS, then mark $n_{subopt} = n$.
- Find the suboptimal MOS corresponding to each pair of (k_{subopt}, n_{subopt}) .

Step 6: Let $S = (R_{s,i}, k_i, n_i)$, $i = 1, \dots, 9$ denote the set of nine suboptimal solutions for nine AMR-WB codec modes. Finding the best suboptimal solution in set S which meets sequentially all the following constraints: (1) has the highest value of G_r , (2) has the lowest value of MOS_r , and (3) has the lowest value of BW_{subopt} . In the case three these conditions are met, we propose to choose the triple that has the lowest AMR-WB mode because normally the lower AMR-WB mode, the less required bandwidth.

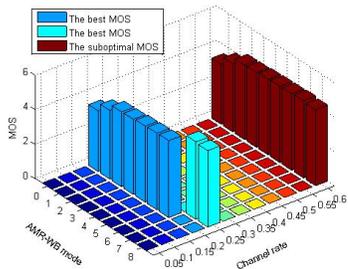


Fig. 2. MOS vs. AMR-WB mode and channel code rate when p_r is fixed equal to 10%

TABLE I
THE DETAILED RESULTS OF FIGURE 2

Mode	k_{best}	n_{best}	MOS_{best}	k_{subopt}	n_{subopt}	MOS_{subopt}	$MOS_r(\%)$	$G_r(\%)$	$BW_{subopt}(\text{kbits})$
0	1	5	3.51	3	5	3.47	1.14	50.00	17.26
1	1	5	3.98	3	5	3.95	0.75	50.00	21.10
2	1	5	4.33	3	5	4.32	0.23	50.00	27.59
3	1	5	4.35	3	5	4.33	0.46	50.00	30.32
4	1	5	4.40	3	5	4.39	0.23	50.00	33.05
5	1	5	4.43	3	5	4.42	0.23	50.00	37.15
6	1	5	4.45	3	5	4.43	0.45	50.00	39.88
7	1	4	4.48	3	5	4.47	0.22	33.33	45.34
8	1	4	4.42	3	5	4.40	0.45	33.33	46.70

- The network path includes 15 hops, with 13 hops are the fast core network links: 10 hops at 622 Mb/s, and 3 hops at 1.8 Gb/s, and two are in the eNodeB cell at 50.4 Mb/s and 25 Mb/s, respectively.
- $P_h = 0.06$ ms per hop. Q_h is random between 0 and 1 ms.
- $D_{max} = 150$ ms. $P_{max} = 10^{-2}$
- $k_0 = 4$ and $n_0 = 5$

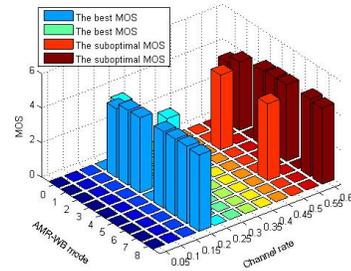


Fig. 3. MOS vs. AMR-WB mode and channel code rate when p_r is randomly changed in range of 0..15%

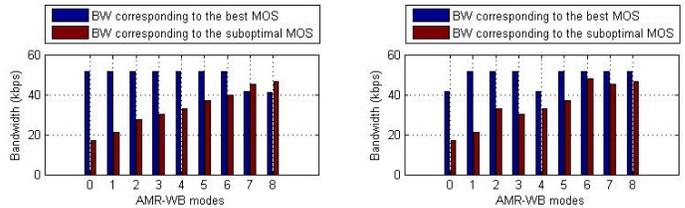


Fig. 4. AMR-WB mode vs required bandwidth when p_r is fixed equal to 10% (left figure) and when p_r is randomly changed in range of 0..15% (right figure)

TABLE II
THE DETAILED RESULTS OF FIGURE 3

Mode	k_{best}	n_{best}	MOS_{best}	k_{subopt}	n_{subopt}	MOS_{subopt}	$MOS_r(\%)$	$G_r(\%)$	$BW_{subopt}(\text{kbits})$
0	1	4	3.51	3	5	3.47	1.14	33.33	17.26
1	1	5	3.98	3	5	3.95	0.75	50.00	21.10
2	1	5	4.33	3	4	4.32	0.23	50.00	33.11
3	1	5	4.35	3	5	4.33	0.46	50.00	30.32
4	1	4	4.40	3	5	4.39	0.23	50.00	33.05
5	1	5	4.43	3	5	4.42	0.23	33.33	37.15
6	1	4	4.45	2	4	4.44	0.22	50.00	47.85
7	1	5	4.48	3	5	4.47	0.22	50.00	45.34
8	1	5	4.42	3	5	4.40	0.45	50.00	46.70

Figure 2 and table I show the results in the case “raw” packet loss is fixed equal to 10%. This demonstrates that when the “raw” packet loss is fixed, the best MOS depends on only two channel code rates, and one for the suboptimal MOS. Figure 2 also shows that with the “raw” packet loss is fixed equal to 10%, all of AMR-WB codec modes obtain the best MOS with channel code rate equal to 1/5 and 1/4, and the suboptimal MOS with channel code rate equal to 3/5. The results show that the suboptimal MOS is very close to the best MOS while the redundant bits generated by channel coding decreased significantly. In this case, the G_r of all triples have two different values (50% and 33.33%). There are only modes of 1 to 6 that have the highest values of G_r , thus, the algorithm will choose according to the lowest values of MOS_r . There are modes of 2, 4, and 5 that have the lowest values of MOS_r (0.23%), the algorithm will choose according to the third condition that has the lowest value of BW_{subopt} . So that, the triple $(R_{s,2}, 3, 5)$ is the best suboptimal solution because it has the lowest value of MOS_r with the suboptimal MOS equal to 4.32 while the best MOS equal to 4.33. Figure 3 and table II show the results in the case the “raw” packet loss is varied in an instant. These results demonstrate that when the “raw” packet loss is changed, the best MOS depends on

only two specific channel code rates (1/4 and 1/5), while the suboptimal MOS depends on three channel code rates (3/5, 3/4, and 2/4). The results also show that there are 7 triples which have the same highest value of G_r (50%), those are two triples of $(R_{s,1}, 3, 5)$, $(R_{s,2}, 3, 4)$, $(R_{s,3}, 3, 5)$, $(R_{s,4}, 3, 5)$, $(R_{s,6}, 2, 4)$, $(R_{s,7}, 3, 5)$ and $(R_{s,8}, 3, 5)$ which meet the conditions. Similar to the case of the fixed pr, the algorithm will firstly choose triples that have the highest values of G_r , secondly the lowest values of MOS_r and thirdly the lowest values of BW_{subopt} . So that, in this case, the best suboptimal solution is triple of $(R_{s,7}, 3, 5)$ which has the highest value of G_r (50%), the lowest values of MOS_r (0.22%), and the lowest values of BW_{subopt} (45.34) with the suboptimal MOS equal to 4.47 while the best MOS equal to 4.48.

In order to demonstrate that the proposed algorithm saving the bandwidth, we calculate the minimum required bandwidth for two cases of the best MOS and the suboptimal MOS. These values can be obtained from the equation (13) through two couples of (k_{best}, n_{best}) and (k_{subopt}, n_{subopt}) . Figure 4 shows the charts of the minimum required bandwidth of each mode of AMR-WB codec corresponding to two cases of the best MOS and the suboptimal MOS. It is clear that, when we choose according to the suboptimal condition, the system will save the bandwidth very significantly. This means the system will serve more mobile users at a moment.

V. CONCLUSION

In this paper, we present an adaptive algorithm for dynamic joint source-channel code rate for voice traffic over LTE network. The proposed algorithm permits choosing a suboptimal solution for minimizing the number of redundant bits generated by channel coding. The output of the algorithm is a value pair of source code rate and channel code rate resulting in minimizing the number of redundant bits generated by channel coding with an acceptable MOS reduction based on some constraints on the maximum allowed end-to-end delay, maximum permitted packet loss and minimum required bandwidth. The simulation results show that in the case of the fixed "raw" packet loss, AMR-WB codec mode 2 offers the suboptimal solution with channel code rate equal to 3/5. Otherwise, when the varied "raw" packet loss, AMR-WB codec mode 7 offers the suboptimal solution with channel code rate equal to 3/5. The simulation results also demonstrate that the proposed algorithm always finds out the suboptimal solution which meets some constraints on given QoS information for the tradeoff between speech quality and the number of redundant bits generated by channel coding. This means there is a slightly reduction of MOS ($\leq 1\%$) when compared to the best MOS while we can obtain the percent of decreased redundant bits generated by channel coding up to 50%. This will lead to saving the bandwidth and so that the system can server more mobile users at the same time. In this study, we assume the random loss model for calculating packet loss. For the future study work, we will consider packet losses are

For the future study work, we will consider packet losses are bursty.

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