Perceptual Evaluation of Speech Quality for iLBC Vocoder with Uneven Level Protection over Narrow Band MSK Radio

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Abstract: - Narrow band MSK(Minimum Shift Keying) radio is applied for high receiving sensitive wireless communications like "walky-talky". iLBC (internet Low Bit-rate Codec) is widely used in VoIP (Voice over IP) applications like "Skype". This paper is concerned with the speech quality by using iLBC in narrow band MSK wireless communication. Several ULP (Uneven Level Protection) strategies to protect different part of iLBC coded payloads are evaluated under AWGN (Additive White Gaussian Noise). We utilize PESQ (Perceptual Evaluation of Speech Quality) to evaluate the perceptual speech quality. Simulation results show that significant improvement of PESQ scores are obtained by using appropriate ULP strategies so that our proposed MSK radio system can also achieve high receiving sensitivity.

Key-Words: - MSK, iLBC, VoIP, ULP, PESQ.

1 Introduction

MSK (Minimum Shift Keying) due to its minimum sidelobe spectrum power is often used to achieve more spectrum efficiency of channel usage and to avoid interfering other channel usages in wireless communication systems. In wireless speech communication, there are needs to provide data compression and to provide data correction for the speech payload communicated over noisy or lossy wireless environment. For example, G.729 is used for speech compression and a channel coding is used for error correction of the compressed data in a GSM system. The BER (Bit Error Rate) from the output of the channel decoder can be an indicator of speech quality.

In [1][24], the authors introduced uneven level protection (ULP) forward error correction coding (FEC) schemes to protect different parts of data. The purposes are to provide better channel usage and to provide better protection for the important parts of multimedia data over lossy packet-switched network. The schemes utilizing different protection levels to the multimedia data emphasize in competing packet loss for wired communications.

iLBC (internet Low Bit-rate Codec), a speech compress coding algorithm, is widely used for VoIP (Voice over IP) applications. This paper is concerned with using iLBC in narrow band MSK (Minimum Shift Keying) wireless communication. Besides, we utilize ULP FEC to protect real-time speech data payload over noisy wireless channel. We emphasize in competing BER for wireless communication and evaluate the speech quality of several different ULP schemes through PESQ (Perceptual Evaluation of Speech Quality) scores. The simulation results indicate that our proposed system architecture is suitable for narrow band radio with high receiving sensitivity.

This paper is organized as follows. In section 2, we briefly introduce the basics of iLBC. In section 3, we introduce the basics of ULP. Section 4 states the principle of the PESQ technique. Section 5 is about our proposed system architecture and states the requirement consideration at its RF stages. Section 6 is the simulation results. We conclude this paper in Section 7.

2 iLBC Basics

Internet Low Bit Rate Codec (iLBC) is a royalty free speech codec. It was developed by Global IP Sound (GIPS). It is suitable for VoIP applications, streaming audio, archival and messaging services. The algorithm is a version of block-independent linear predictive coding with the choice of data frame lengths of 20 or 30 ms. iLBC handles the case of lost frames through graceful speech quality degradation. iLBC is defined in [2]. It is one of the codecs used by the Google Talk, Skype and Yahoo! Messenger, etc [3]. The MOS(mean opinion score) performance comparison of several different Codecs is shown in Fig.1.

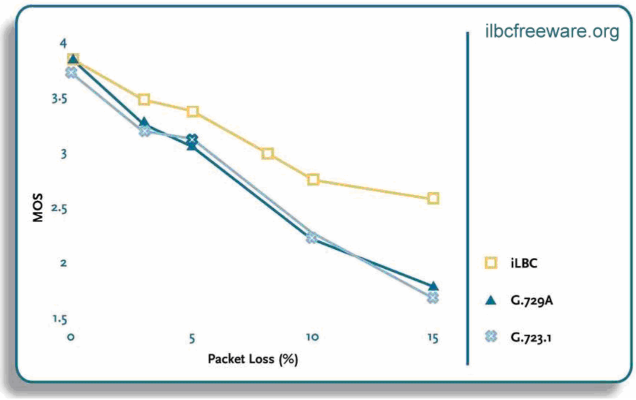

Fig.1, The MOS performance comparison of several different CODECs

The iLBC codec [4] is an algorithm that compresses each basic frame (20 ms or 30 ms) of 8000 Hz 16-bit sampled input speech into output frames with basic frame size of 400 bits for 30 ms and size of 304 bits for 20 ms. (see Fig.2).

20ms frame: 304bit

Class 1/48bit	**Class 2/64bit**	**Class 3/192bit**

30ms frame: 400bit

Class 1/64bit	**Class 2/128bit**	**Class 3/204bit**

Fig. 2, iLBC frame size and the arrangement of three classes.

The codec supports two basic transmission rates: 30 ms at 13.33 kbit/s and 20 ms at 15.2 kbit/s. When the codec operates at block lengths of 20 ms, it produces 304 bits per frame which must be packetized in 38 bytes. Similarly, for frame lengths of 30 ms it produces 400 bits per frame which bust be packetized in 50 bytes.

In the bitstream definition, the bits are distributed into three classes according to their bit error or loss sensitivity. The most sensitive bits (class 1) are placed first in the bitstream of each frame. The less sensitive bits (class 2) are placed after the class 1 bits. The least sensitive bits (class 3) are placed at the end of the bitstream of each frame. This distribution of the bits enables the use of uneven level protection (ULP).

3 ULP Basics

In last section, we state the iLBC has three different classes. Moreover, it implies that we can use ULP technique to protect different sensitive bit streams. In [1], the authors introduce ULP techniques in package payloads. The authors introduce two different types of ULP: single level and multi level.

An example of single level ULP is illustrated in Fig.3. It divides data into two parts. One part is required of protection and the other is not. The data of length L in the beginning of the packets are to be protected by the ULP FEC. As seen in the figure, each packet might be of different length. If the packet is with length shorter than the length L, such as packet #3, then zero padding is needed.

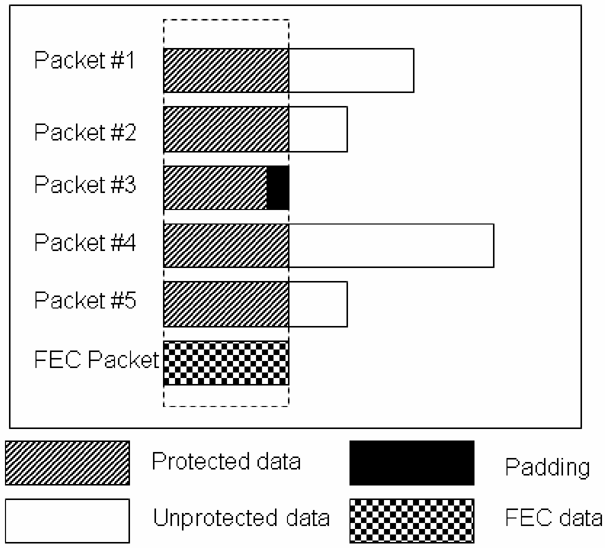

Fig. 3, Single level uneven level protection.

The other example of multilevel ULP is illustrated in Fig.4. In this example, five packets are sent through the channel. The first parts of the data packets #1 to #3 protected by ULP level 0 are encoded into FEC packet #1. The first parts of the data packets #4 and #5 protected by ULP level 0 are encoded into one part of FEC packet #2. At the same time, the second parts of the data packet #1, #2, #4 and #5 protected by ULP level 1 are encoded into the other part of FEC packet #2.

It motivates us to use the schemes to protect speech payload of iLBC packets in wireless applications.

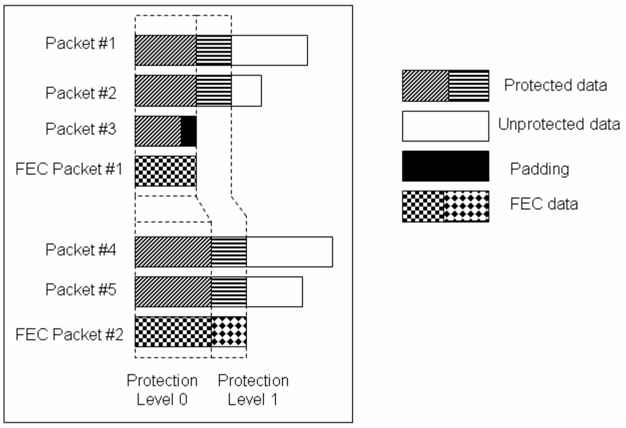

Fig. 4, Multi level uneven level protection.

4 Speech Quality Evaluation and PESQ

4.1 Speech Quality Evaluation

The traditional method of determining speech quality is to conduct subjective tests with "panels of human" listeners. Extensive guidelines are given in ITU-T recommendations P.800/P.830. The results of these tests are averaged to give mean opinion scores (MOS). But such tests are expensive and are impractical for testing in the field [5]. However, MOS is probably the most widely used and simplest method to evaluate speech quality in general. MOS has five-level scales from bad "1" to excellent "5" [6].

4.2 PESQ

For the disadvantage reasons of MOS, ITU recently standardized a new model, PESQ (Perceptual Evaluation of Speech Quality). It automatically predicts the quality scores that would be given in a typical subjective test. An intrusive test as shown in Fig.5 gives the PESQ scores. PESQ provides a rapid and repeatable result in a few moments [5].

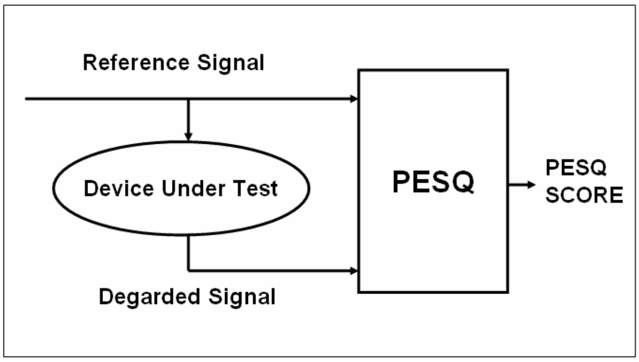

Fig. 5, The usage of PESQ

PESQ is an objective measurement tool that predicts the results of subjective listening tests on telephony systems. PESQ uses a sensory model to compare the original, unprocessed signal with the degraded signal from the network or network element. It is applicable not only to speech codecs but also to end-to-end measurements [7].

It is recommended that PESQ can be used for speech quality assessment of 3.1 kHz narrow-band handset telephony and narrow-band speech codecs [7]. The resulting quality score is analogous to the subjective MOS measured using panel tests according to ITU-T P.800 [8].

4.2.1 PESQ output

PESQ score ranges form –0.5 to 4.5 [7]. It also offer a function to convert this score to PESQ-LQ (listening quality), which gives a MOS-like listening quality score is between 1 to 5 (see Table I).

Table 1, the score table of MOS

Score	Speech Quality
5	**Excellent**
4	**Good**
3	**Fair**
2	**Poor**
1	**Bad**

5 The Proposed System Architecture

The proposed system architecture is illustrated in Fig. 6. It consists of the following parts: Sample speech database, iLBC speech encoder and decoder, ULP (Uneven Level Protection) schemes, interleaver and de-interleaver, MSK modulator and demodulator. The channel is assumed to be AWGN corrupted. The output of the iLBC decoder is assessed by the PESQ.

Because we focus our system for narrow band MSK radio like "walky-talky" in U.S.A. ISM (Industry, Science and Medicine) bands, we first introduce the related FCC rules in the following subsection 5.1.

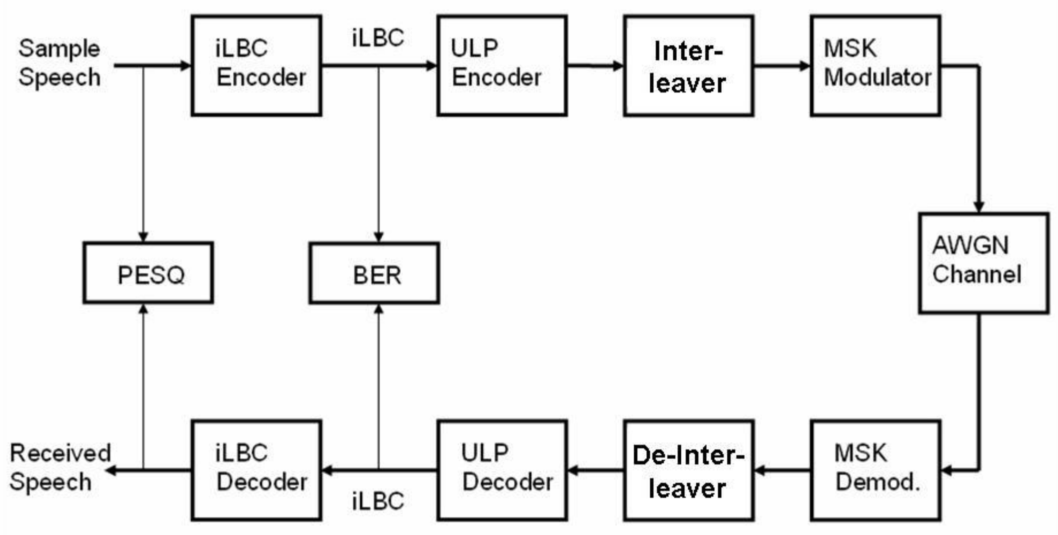

Fig. 6, The proposed system block diagram

5.1 FCC Rules in ISM Bands

The ISM bands are defined by the ITU-R in 5.138 and 5.150 [20] of the Radio Regulation. Originally, the ISM bands are reserved for non-commercial use for industrial, scientific and medical purposes [19]. These bands are also reserved in the world wide, not only in U.S.A. Some countries have little difference on their definition. Basically, ISM bands are license-free. That is, users do not need apply for a license and pay for it to government. Only 915MHz is permitted in North America and Israel. Some countries share this band for GSM or for personal communication usage. The electromagnetic and modulation scheme are defined in Part 15 [9].

According to FCC's regulation 18.304 [17], the well known consumer product use band is 915MHz, 2.4GHz and 5.8GHz. Some of the systems, such as WLAN and Bluetooth, are also deployed in ISM bands, IEEE802.11b and 802.11g in 2.4GHz, 802.11a in 5.8GHz, Bluetooth in 2.4GHz. Except WLAN and Bluetooth products, cordless telephone is also utilized in ISM bands. Note that anyone can use these bands before transmitting signals without monitoring whether they are occupied or not. It is very different to similar applications which occupy in license bands such as 46/49MHz cordless telephone.

In license bands, one should monitor the signal strength before transmitting it because interfering to other equipment is not allowed. In other words, much more interference may exist in the ISM bands while you want to use them. This is another important issue that designer should concern with.

Since the purpose of this paper is to design portable wireless equipment which needs to meet the government's rule, understanding these rules are necessary. As the aforementioned, the regulations of ISM bands are defined in FCC Part [17]. The most important rules, especially in electromagnetic emission, are defined in Part 15 [9].

FCC drafted section 15.247 at 1985, most of the rules of ISM bands which utilize digital modulation techniques such as modulation schemes, bandwidth requirement and hopping frequencies are defined in this section. In these bands, if the modulation scheme is digital, only frequency hopping and direct sequence spread spectrum is allowed. The requirement of frequency hopping system is listed as below:

(a). For the frequency hopping system, "shall have hopping channel carrier frequencies separated by a minimum of 25 kHz or the 20 dB bandwidth of the hopping channel, whichever is greater" [9].
(b). for the frequency hopping system operating at 902MHz to 928MHz, at least 50 hopping frequencies of hopping channel if the 20dB bandwidth less than 250kHz.
(c). each frequency channel should not occupy large then 0.4 second within 20 seconds period.

(d). for the frequency hopping system operating at 902 to 928MHz, if it meet the requirement (a) to (c), the maximum output peak power is 1 watt.
(e). antenna gain does not large than 6dBi. If one uses directional antenna and its gain is larger than 6dBi, the output power should decrease 1dB which corresponds to output power's incensement.

Since our design also tries to adopt frequency hopping technique, the minimum 20dB bandwidth has to be 25kHz. By obeying the rule "frequency hopping system allows 1 watt output power", this is beneficial to design a long-range portable wireless equipment to compete with traditional FM two way radio equipments due to the enough large output power.

The most important parameter is the bit rate requirement. According to FCC's rule, the minimum 20dB bandwidth is 25kHz. Referring to [10], the bandwidth to bit rate ratio of MSK is 1.2. Therefore, our designing bit rate is at least 20.833kHz.

5.2 Sample Speech, iLBC Encoder and Decoder

All of the sample speeches are formatted in 16bit PCM data. The sampling rate is 8 kHz. They contain phonetically English sentences, which provide the ideal samples for our speech quality evaluation. Each speech sample as shown in the Fig.7 contains 316 frames, the frame length is 30ms. One sentence takes 9.48 second.

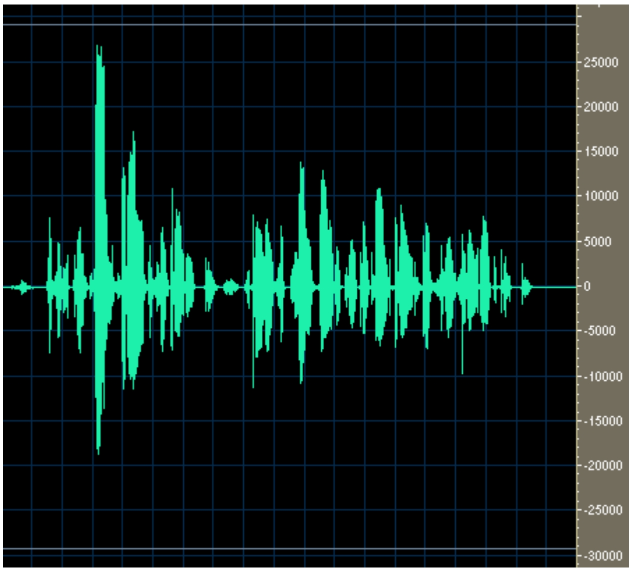

Fig.7, A sample speech(9.48seconds/ 316 frames = 30 ms/frame).

Each speech sample has been coded using the iLBC codec's 13.33kb/s. The frame length is 30ms, which contains 400 bit. The encoded file feeds to the ULP module. At the receiver, the noise degraded iLBC speech bit stream will feed to the iLBC decoder.

For other morphing data, one may refer to [22].

5.3 ULP Schemes

There are eight ULP schemes in this research. In each scheme, several forward error control (FEC) coding schemes (or called channel coding schemes) might be adopted. In this paper, the FEC is convolutional coding (CC). Table 2 presents the eight different schemes in which we adopt three different coding rates: R=1(no coding), R=1/2 and R=2/3. Because of the arrangement, different codeword lengths in the schemes are generated. For example, scheme S1 does not use any coding to protect its data but scheme S3 uses rate-1/2 coding to protect its class 1 data and does not use any coding to protect its class 2 and class 3 data. The total encoded length to scheme S3 is 464 bits.

It is worthy of noting that all "empty indicator", the last bit of one iLBC frame, is force to "0" to indicate this frame is not empty.

Table 2, Eight different ULP schemes

Schemes	Class1	Rate	Class2	Rate	Class3	Rate	Total
S1	64	1	96	1	240	1	400
S2	128	1/2	192	1/2	480	1/2	800
S3	128	1/2	96	1	240	1	464
S4	128	1/2	192	1/2	240	1	560
S5	96	2/3	144	2/3	360	2/3	600
S6	96	2/3	96	1	240	1	432
S7	96	2/3	144	2/3	240	1	480
S8	128	1/2	96	2/3	240	1	464

5.4 FEC Using Convolutional Codes

The FEC coding or channel coding used in this paper is convolutional coding. The used convolutional codes follow the suggestion in IEEE 802.11a [11] which is a well-known industrial and commercial standard. Actually, the codes are also applied for dedicated short range communications (DSRC) [18][21][23]. The generator polynomials to generate a coding rate R=1/2 with constraint length 7 or six shift registers is

$$\mathbf{G1=171_{oct}}, \qquad \mathbf{G2=133_{oct}} \tag{1}$$

Fig.8. depicts the encoder diagram constructed by six shift registers.

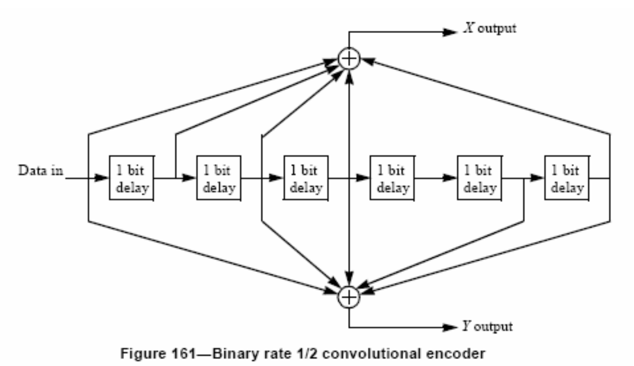

Fig.8, The R=1/2 convolutional encoder diagram.

For generating the rate other than 1/2, the puncturing pattern also follow the IEEE 802.11a [11] as shown in the following Fig.9.

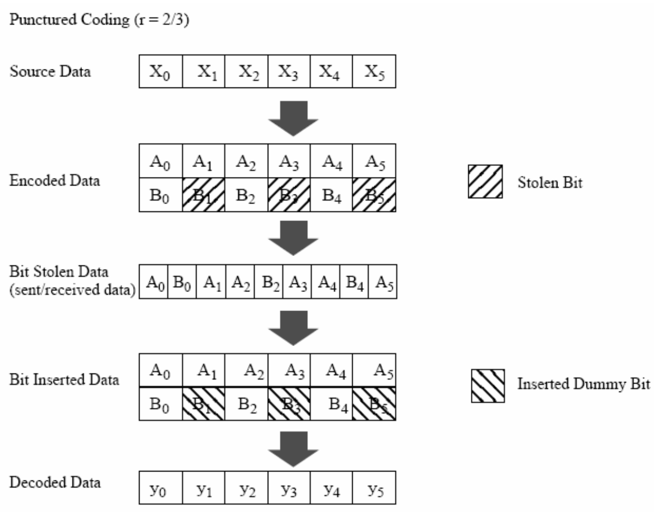

Fig.9, The puncture patern of R=2/3 convolutional code.

The decoding process is done by Viterbi algorithm with hard decision. Viterbi algorithm is a well-known maximum likelihood (ML) decoding algorithm. The simulation in section 6 shows that with hard decision the system has satisfactory performance. Besides, by hard decision, the hardware is much simple than by soft decision.

5.5 Speech Quality Evaluation

The PESQ program is download form ITU web site [12]. The input and output speech-quality "PESQ-MOS" assessment program was in raw data format with 16 bit linear PCM at 8 kHz of sampling rate. In this application program, it actually generates two different scores: our adopted PESQ-MOS and MOS-LQO (listening quality objective) which is a mapping function for transforming P.862 raw data to scores. Note that we adopt PESQ-MOS in this paper.

5.6 Receiving Sensitivity

We assume that the acceptable speech quality is based on bit error rate (BER)=1% under AWGN channel via our tests. For our concern of the transmission distance in the design of MSK radio, we have to calculate the corresponding receiving sensitivity. The following formulas [16] help us calculate the receiving sensitivity:

$$\textbf{Receiving sensitivity = Noise floor + S/N + NF} \tag{2}$$

$$\textbf{Noise floor = -174dBm + 10 log BW} \tag{3}$$

Where NF means total noise figure of RF stages, S/N is the signal to noise ratio and BW is the signal bandwidth. We have

$$\textbf{Eb/No = (S/N) * (BW/SR)} \tag{4}$$

Where Eb is the bit energy of the output to the convolutional encoder No is single sided spectrum density or the single sided variance of white Gaussian noise and SR is the symbol rate. Since we adopt MSK signal, the symbol rate SR = 2.

When devices are connected in series, the composite noise figure for a sequence of n stages is written as below [14]:

$$F = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \frac{F_4 - 1}{G_1 G_2 G_3} + \cdots + \frac{F_n - 1}{G_1 G_2 G_3 \cdots G_{n-1}}, \tag{5}$$

Where Fi and Gi denote respectively the i-th stage noise figure and gain.

The block diagram of our designed RF receiver is shown in Fig.10. The limit amplifier, demodulator and relative blocks that after second mixer are not yet shown there.

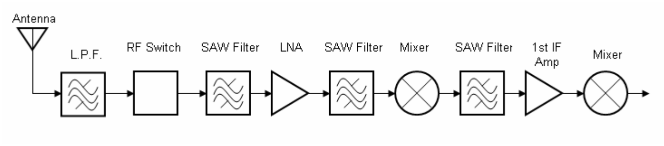

Fig. 10, RF receiver block diagram

We adopt the usual read parameters according to Table 3, the composite NF is calculated to be around 5dB.

Table 3, The usual read power gain and noise figure of each device.

Device	LPF	RFSW	BPF1	LNA	BPF2
Gain (dB)	-0.5	-0.5	-2	24	-2
Noise Figure (dB)	0.5	0.5	2	0.9	2
Device	Mixer1	IF BPF	IF Amp.	Mixer2	
Gain (dB)	13	-4	20	13	
Noise Figure (dB)	9	4	1	9	

6 Simulation Results

6.1 MSK over AWGN

We perform the BER simulation of MSK, which is with smaller bandwidth compared to other modulation. We can see its performance is almost identical to BPSK as shown in Fig.11. By using channel coding, Fig.11 also shows that the performance of rate-1/2 coding and that of coding rate-2/3.

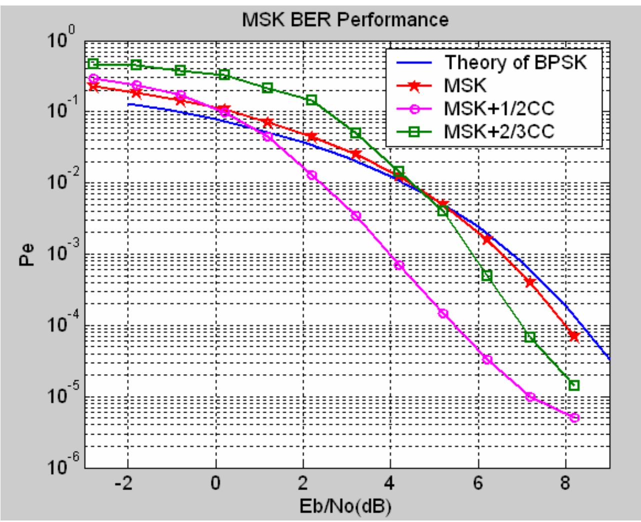

Fig. 11, MSK BER performance over AWGN channel

6.2 Investigation of Degraded Speech Quality

In [15], it states that the most sensitive bits are class 1, the less sensitive bits are class 2 and the least sensitive bits are class 3. In this subsection, we investigate on which class that dominates the speech quality.

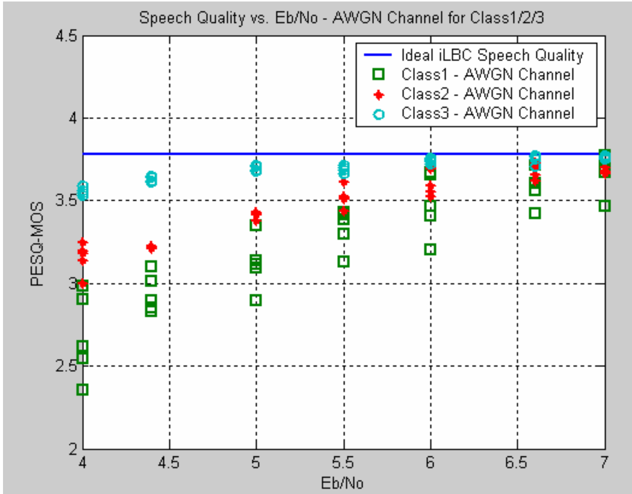

Fig. 12, PESQ-MOS v.s. Eb/No (right) for iLBC class 1, 2 and 3 bit stream over AWGN.

The simulation results are shown in Fig.12. The perceived evaluation speech quality (PESQ-MOS) is plotted as a function of the Eb/No. At class i test, we assume that other bits other than class i are all correctly received, where i=1, 2, or 3. At a given Eb/No, a point means a test speech sample. We can see that the result to each testing speech sample is not a fixed value but a distribution at a given Eb/No.

The result shows that the degraded class 1 bits affect the speech quality a lot. The PESQ-MOS is better than 3 while the Eb/No greater than 5.5 dB but around 2.5 while the Eb/No is 4 dB. The degraded class 3 bits only affect the speech quality a little, even though the BER is worse than 4 dB, the PESQ-MOS is still better than 3.5.

According to our simulation result, the perfect iLBC codec (all bits are correctly received) will get PESQ-MOS score 3.77, as shown the hoizontal blue line in Fig.12.

6.2.1 Scheme S1: Using Rate-1 CC (No Coding) at all classes

The scheme S1 represents that all the iLBC bit streams pass through the AWGN channel without any coding. The results are shown in Fig.13. The PESQ score is below 2.8 under Eb/No worse than 4.5 dB.

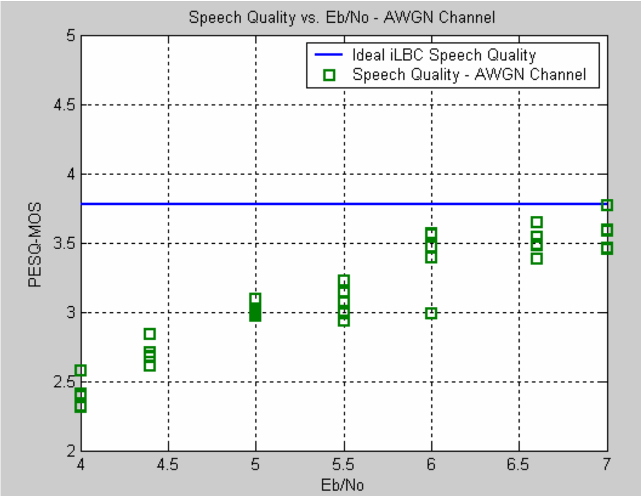

Fig. 13, PESQ-MOS v.s. Eb/No for all iLBC bit streams over AWGN.

6.2.2 Scheme S2,S 3 and S4: Using Rate-1/2 CC and Rate-1 CC (No coding)

As shown in Fig.14 where the green squares represent S2, the red stars represent S3 and the blue diamonds represent S4, the PESQ scores of these three schemes are beyond 3 under Eb/No equal to and greater than 4 dB with paying the prices of rate-1/2 channel coding redundancies.

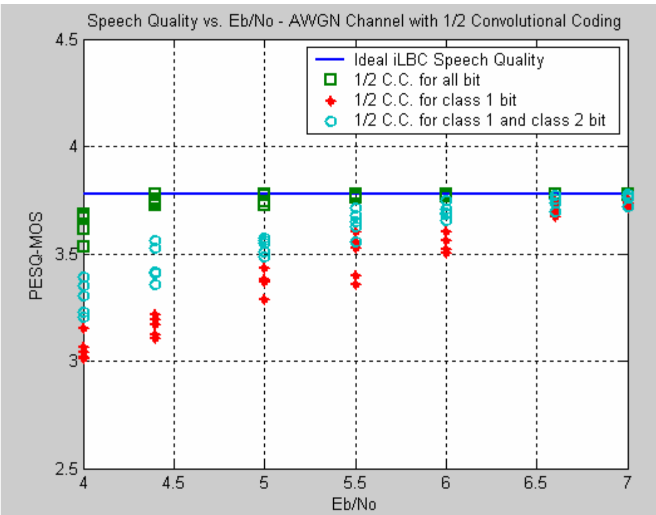

Fig. 14, PESQ-MOS v.s. Eb/No for all iLBC bit streams over AWGN with 1/2-CC.

6.2.3 Scheme S5, S6 and S7: Using Rate 2/3-CC and Rate-1 CC (No coding)

As shown in Fig.15 where the green squares represent S5, the red stars represent S6 and the blue diamonds represent S7, at Eb/No less than 4 dB the PESQ score of scheme S5 is less than 3 and at Eb/No less than 6 dB, the PESQ score of scheme S6 is less than 3. The PESQ score of scheme S7 is less than 3 under Eb/No less than 5 dB. We conclude that the rate-2/3 convolutional code is not good enough in protection. It may be improved if we use soft Viterbi decoding.

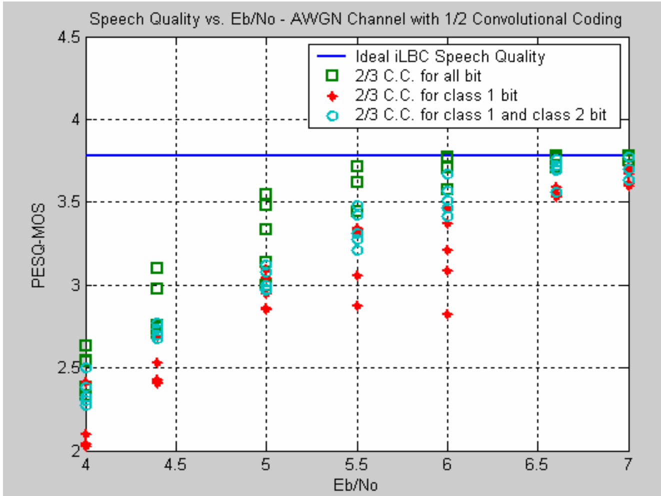

Fig. 15, PESQ-MOS v.s. Eb/No for all iLBC bit streams over AWGN with 2/3-CC.

6.2.4 Scheme S8: Using Rate-1/2 CC at Class 1, Rate-2/3 CC at Class 2 and Rate-1 CC (No Coding) at Class 3

In Fig.16, the PESQ scores of scheme S8 which uses rate-1/2 CC at class 1, rate-2/3 CC at class2 and no coding is applied at class 3 indicate the ULP arrangement is appropriate.

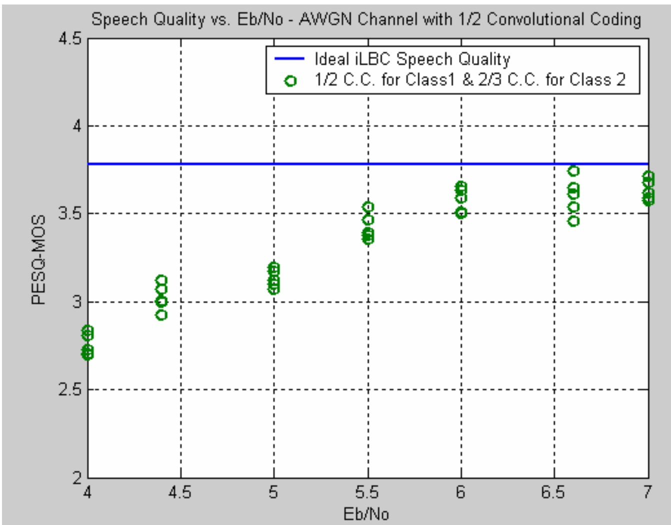

Fig. 16: PESQ-MOS v.s. Eb/No for iLBC bit streams over AWGN with 1/2-CC for class 1 and 2/3-CC for class 2.

6.3 Comparison of the Schemes on Receiving Sensitivity and Speech Quality

The comparisons of these schemes from S1 to S8 on receiving sensitivity and speech quality are listed in Table 4. We assume each frame should add 54 head and tail bits for message synchronization and PLL settling time, and so on. The designed MSK radio follows the FCC part 15.247 [9]. In ISM (industrial, scientific and medical) band for frequency hopping spread spectrum application, the minimum bandwidth is 25kHz. Accoding to [10], the bandwidth (20dB) to bit rate ratio is 1.2.

Table 4, The comparison of the schemes in receiving sensitivity and speech quality

Scheme	iLBC Length (bit)	Final Frame Size (bit)	Data Rate (bit/sec)	BW (kHz)	Noise Floor (dBm)	Sensitivity (dBm)	PESQ -MOS
S1	400	454	20833.33	25	-130.02	-121.42	2.611
S2	800	854	28466.67	34.16	-128.66	-120.06	3.724
S3	464	518	20833.33	25	-130.02	-121.42	3.103
S4	560	614	20833.33	25	-130.02	-121.42	3.411
S5	600	654	21800.00	26.16	-129.82	-121.22	2.713
S6	432	486	20833.33	25	-130.02	-121.42	2.403
S7	480	534	20833.33	25	-130.02	-121.42	2.674
S8	464	518	20833.33	25	-130.02	-121.42	2.926

7 Conclusion

This paper has shown that employing iLBC with ULP for narrow band MSK radio can not only improve the speech quality but also increase the receiving sensitivity by our proposed system architecture. The architecture is very suitable for high sensitivity receiving applications like “walky-talky”.

We have investigated that the performance of MSK which is almost identical to that of theoretical BPSK but with smaller signal bandwidth. Our simulation shows the significant phenomenon that the degraded class 1 bit stream affect the PESQ-MOS a lot. We have compared several uneven level protection (ULP) schemes by using perceptual evaluation of speech quality (PESQ) score. Protecting class 1 bitstreams with rate-1/2 convolutional code reaches significant improvement. We also conclude that using rate-2/3 convolutional code to protect class 1 is not good enough. It is propable to improve the PESQ if we use soft Viterbi decoding.

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References:

[1] Li, A.H., Fahlen, J., Tao Tian, Bononi, L., So-Young Kim, Jeong-Hoon Park and Villasenor, J., Generic Uneven Level Protection Algorithm for Multimedia Data Transmission over Packet-Switched Networks, *Computer Communications and Networks*, 2001, Proceedings, Tenth International Conference on 15-17 Oct. 2001 , pp.340–346.

[2] Internet Low Bit Rate Codec (iLBC), *Network Working Group, Request for Comments: 3951 (RFC3951)*, The Internet Society 2004.

[3] http://en.wikipedia.org/wiki/ilbc

[4] F. Hammer, P. Reichl, T. Nordström, and G. Kubin, Corrupted Speech Data Considered Useful: Improving Perceived Speech Quality of VoIP over Error-Prone Channels, *Acta Acustica Special Issue on Auditory Quality of Systems*, Vol 90, No. 6, Nov/Dev 2004, pp.1052-1060.

[5] PESQ: An Introduction, http://www.psytechnics.com, September 2001.

[6] http://www.acoustics.hut.fi/publications/files/theses/lemmetty_mst/chap10.html

[7] Perceptual Evaluation of Speech Quality (PESQ): An Objective Method for End-to-end Speech Quality Assessment of Narrow-band Telephone Networks and Speech Codecs, *ITU P.862*, 2006.

[8] http://www.malden.co.uk/pesq.htm

[9] *Title 47 Part 15 of the Code of Federal Regulations*, USA

[10] K. Kuchi and V.K. Prabhu, Power Spectral Density of GMSK Modulation Using Matrix Methods, *Military Communications Conference Proceedings*, 1999.

[11] IEEE STD802.11a-1999, Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, High-speed Physical Layer in the 5 GHz Band, *IEEE Standard - 802.11a*, 1999.

[12] http://www.itu.int

[13] Bernard Sklar, *Digital Communications Fundamentals and Applications*, Prentice Hall, 1988, pp.174.

[14] [13] pp.218, (4.39)

[15] A. Duric and Telio S. Andersen, Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech, *Network Working Group A, Request for Comments: 3952 (RFC 3952)*, Aalborg University, December 2004.
[16] Behzad Razavi, *RF Microelectronics*, Prentice Hall Inc. 1998.
[17] *Title 47 Part 18 of the Code of Federal Regulations*, U.S.A.
[18] S. K. Lee, H. W. Chen and M. H. Lin, A Robust CD3-OFDM Receiver Architecture for DSRC Systems, *WSEAS Trans. on Communications*, pp.1400-1406, August 2006.
[19] http://en.wikipedia.org/wiki/ISM_b
[20] *ITU regulation - R5.138, 5.150, and 5.280*
[21] S. K. Lee, Y. A. Kao, H. W. Chen, Performance of A Robust Inner Receiver with Frequency Domain LMS Equalizer for DSRC Systems, *IWCMC'06*, 2006.
[22] Orphanidou C., Moroz I.M. and Roberts S.J., Wavelet-based Voice Morphing, *WSEAS Journal on Systems*, Vol.10, No.3, December 2004, pp 3297-3302.
[23] Shih-Kai Lee, Han-Wei Chen, Meng-Hsiang Lin and Yung-An Kao, A Robust OFDM Receiver for DSRC Systems, *WSEAS 10th Conference on Communications*, 2006.
[24] Minh Hung Le, Nikos E. Mastorakis; Variable Length Codes with Unequal Error Protection for Wireless Communication Systems, *WSEAS Transactions on Communications,* 2006..