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Original Article

A Review of Challenges and Solutions for Speech Quality Measurement in Low Bandwidth Sensor Networks

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Abstract - *Radio communication has changed the face of communication. It is becoming superior to the landline telephone network. It is mainly famous for rigid voice communication with mobile and remote users using portable devices with good speech measurement quality. With the advent of technology, radio communication has shifted from analogue to digital domain. Low bandwidth Digital Radios are used by Professionals, Emergency service providers like Police and Firefighter to provide immediate, effective communication using portable devices at any remote place. These radios are operated in licensed frequency bands. They used analogue communication technology with sufficient bandwidth (25 KHZ). With this bandwidth, the speech quality in the communication was good. With the increasing demand for spectrum, frequency allocating authority decided to reduce bandwidth from 25 KHZ to 6.25 KHZ. Reducing bandwidth has affected many radios and sensor network parameters, ultimately lowering speech measurement quality. This paper analyses and summarizes those parameters and suggests possible methods for increasing speech measuring quality. It will help original equipment manufacturers, system planners, and engineers design the best possible combination of parameters for good speech quality in Digital Radio. The main affecting parameters are Technology which includes modulation and channel access techniques, Speech coding and quantization techniques, and planning and design of radio networks using Link Budget and Spoken language characteristics.*

Keywords - *Bandwidth, Digital radio, Low bit rate codec, Speech quality, Sensor networks, Quantization.*

1. Introduction

In Emergency services like Police and firefighters, wireless communication plays a vital role. In these services, all commands or orders are issued on wireless media, and the concerned team follows them to complete a task or operation. So good speech quality should be maintained in the wireless network. Also, these services utilize many channels in their different groups to keep security and operation planning. Channel spacing or bandwidth of each channel directly depends on a range of speech frequencies and the type of modulation used. In the old days, for good-quality speech, the range of audio frequency was 0-5000 HZ and Frequency Modulation was used so that each wireless channel requires a bandwidth of 25 KHZ. With the demand for wireless communication, frequency allocating authority decided to reduce channel spacing from 25 KHZ to 12.5 KHZ. This reduces deviation and, ultimately, the range of speech frequency. The reduction in bandwidth demanded using digital wireless communication instead of analogue wireless communication and technology like TDMA to effectively use one channel bandwidth for two channels using time sharing. Due to bandwidth reduction and technological change, many new stages have been introduced in the radio and its network design. These new stages affect speech quality. This paper identified five such areas which are responsible for such degradation of speech. They are a technology used, Speech coder, Quantizer, Link or RF design and Spoken language characteristics. Various stages of digital radio and its network are focused to improve speech quality in low-bandwidth radio. Many researchers have focused on different areas separately to analyse speech quality. This paper integrates all these stages. It is shown that proper configuration of all these stages can give good speech quality despite low bandwidth. The methodology for each section of digital radio to be improved is different. Each section in this paper includes available types, related work carried out, and shortcomings and concludes in the best possible way by integrating that section with other sections of digital radio.

The paper is organized as follows.

1. History and Digital Radio Technologies.
2. Typical speech processing in Digital Radio Transmitter and Receiver.
3. Impact of low bit rate codec.
4. Quantizer types.
5. Link or RF budget.
6. Spoken language characteristics.
7. Conclusion.

1.1. History and Digital Radio Technologies

Demand for wireless communication is increasing due to its flexibility, portability, and quality. This ultimately impacts the bandwidth allocated for a wireless channel. TRAI (Telecommunication Regulatory Authority of India), Network Spectrum and Licensing Division has published a consultation paper [1] and suggested using 6.25 KHZ Bandwidth/Channel Spacing with digital technology for licensed band professional radio. Accordingly, many Open and Private standards [2-8]. They are listed in Table-1. It shows two significant technologies two slot / four slots TDMA with digital modulation and FDMA with digital modulation. It uses 6.25 KHZ bandwidth or channel spacing. Due to this low bandwidth, the deviation has to be reduced. This results in using a speech coder, as shown in the last column of Table-1. Primary technologies used are highlighted.

2. Literature Review on Typical Speech Processing in Digital Radio Transmitters and Receivers

2.1. Digital Radio Transmitter and Receiver

Analogue speech signal from the microphone is converted to digital format with a 128 kbps rate using Audio Co-dec. It is down-converted to 3-4 kbps using Speech Coders.

This coded signal is packetized and converted to Analog format. It is modulated using Digital Modulator, and modulated signal is RF processed, powered, and transmitted in the air using RF components. Modulated speech signal received through the antenna is first RF processed and converted to IF signal. This analog IF signal is converted to a Digital IF signal. It is then demodulated. The demodulated data is 3-4 kbps and is up-converted to 128 kbps using a speech decoder. Audio Codec is used to get back everyday speech signal from 128 kbps signal. This is the recovered speech signal at the receiver.

1. ETSI –European Telecommunication Standards Institute.
2.APCO –Association Public- Safety Communication Officials, Inc.
3.DMR – Digital Mobile Radio.
4.DPMR – Digital Private Mobile Radio.
5. TETRA- Terrestrial Trunked Radio.
6. NXDN – Next Generation Digital Narrow Band.
7. AMBE – Advanced Multiband Excitation.
8. IMBE-Improved Multiband Excitation.
9. ACELP – Algebraic Code Excited Linear Prediction.
10. RALCWI –Robust Advanced Low Complexity Waveform Interpolation.
11. ASELP - Algebraic Self Excited Linear Prediction.
12. TDMA -Time Division Multiple Access.
13. FDMA-Frequency Division Multiple Access.
14.4 FSK-4 Level Frequency Shift Keying.
15.DQPSK-Differential Quadrature Phase Shift Keying.
16.C4FM-Continuous Four Frequency Modulation.
17.HCPM- Harmonized Continuous Phase Modulation.
18.HDQPSK-Harmonized Differential Quadrature Phase Shift Keying.

2.2. Digital Radio Repeater

The Repeater is used to form Digital Radio Network. Every signal transmitted from Radio Transmitter at the user end is received at Repeater and re-transmitted in the air at a different frequency, which all Radios receive if it is a broadcast call. Repeater also works in Private calls or Group calls. Repeaters can be connected using microwave links to expand the network per user requirements. Digital Repeaters process two or four simultaneous calls on the same frequency. It is called two-slot/4-slot TDMA technology. This paper considers two-slot TDMA technology, but the same discussion applies to 4 slots TDMA or FDMA technology used in Digital Radio.

3. Low Bit Rate Speech Coder Types and Specifications

A speech coder is required in Digital Radio in the interest of low bandwidth requirements. Various types of speech codes are available for low-bandwidth applications. These coders are studied based on bit rate, speech quality in MOS, delay, computational complexity, coder performance in the presence of loss of packets, and robustness of performance to speakers of different ages, sex and native language. Speech codes are broadly classified into six groups: waveform coder, transform coder, subband coder, parametric coder and hybrid coder, and a combination of one or more techniques in these types. The classification of coders, along with their bit rate and speech quality index MOS (Mean Opinion Score), is shown in Table 2. This table is prepared using data in [9]

Waveform coders follow the shape of the input speech signal and convert each sampled signal to digital format. Main waveform coders are based on PCM (Pulse Code Modulation), DPCM (Differential PCM), ADPCM (Adaptive DPCM), DM (Delta Modulation) and ADM (Adaptive DM) principle. These coders produce significantly less loss of input signal while reproducing it but require a high data rate. A comparison of basic waveform coders is made in [10]. It shows that PCM and DPCM suffer from noise generated during the coding and quantization. DM/ADM coders cannot follow rapid fluctuations in the input speech signal and produce unwanted noise, lowering speech quality. Basic 40kbps ADPCM is designed in [11], and [12] shows the design of ADPCM chip for converting speech signal from 64 kbps to 16/24/32/40 kbps. The coder's performance under packet loss conditions is studied in [13] and shown that waveform coders are not much affected by packet loss. The unwanted noise generated during the packet loss process can be reduced by Raised Cosine or Sigmoid function. [14] use logistics function based on the steepest descent algorithm, and [15] uses an optimized parametric function to reduce noise generated during packet loss conditions. It is shown that it helps to improve SNR and hence speech quality.

1. PCM Pulse Code Modulation.
2. DPCM Differential Pulse Code Modulation.
3. ADPCM Adaptive Differential Pulse Code Modulation.
4. STFT Short Time Fourier Transform.
5. DCT Discrete Cosine Transform.
6. MBE Multiband Excitation.
7. LPC Linear Prediction Coding.
8. MELP Mixed Excitation Linear Prediction.
9. CELP Code Excited Linear Prediction.
10. ACELP Algebraic Code Excited Linear Prediction.
11. CS-ACLP Conjugate Structure Algebraic Code Excited Linear Prediction.
12. MP-CELP Multi Pulse Based Code Excited Linear Prediction.
13. LD-CELP Low Delay Code Excited Linear Prediction.
14. VSELP Vector Sum Excited Linear Prediction.

In Transform Coding input signal is transformed into its coefficients using a suitable transform such as Fourier Transform or STFT (Short Time Fourier Transform), DCT (Discrete Cosine Transform) or its subtypes like MDCT (Modified DCT) or Wavelet Transform. The coefficients are then quantized and coded. Its data rate is 14-32 kbps with MOS value approximately 2.0. STFT analysis in [16] shows that STFT magnitude and human ear response are approximately the same. It is possible to reconstruct a signal from STFT magnitude. For this purpose, frame length, shift rate and iteration number are studied [17].

Recent STFT development for speech coding improvement is studied in [18]. Filters are used in coder and decoder circuits to improve their performance. In [19], a filter is designed to be placed in the decoder only to improve coder performance. Speech coding in multichannel environments like surround 5.1 is studied in [20]and shows effective MDCT coding with a bit rate of 80 kbps in a surround 5.1 environment. In transform coding, quantization can be updated with current or future inputs (Forward Quantization) or past inputs (Backward Quantization). Forward Quantization with transform coding is implemented for continuous signal in [21]and realizes more than 4 dB improvement in SQNR. The same author applies the same technique to a discrete signal [22]and observes the same noticeable difference as a continuous signal. Nonlinear Transform Coding (NTC) is studied under [23]and has shown it effectively reduces distortion during coding.

Discrete Wavelet Transform (DWT) is an effective technique, and in [24, 25], it is shown that using DWT compression ratio can easily be varied. It is shown to achieve a high compression ratio with good speech quality. In [26], several transform coding is examined, showing that the DCT coder performs optimally in all coders. The same is expressed in [27]. Sub Band Coding is a type of transform coding. It breaks the signal into different frequency bands using Filters or Fast Fourier Transform and encodes them using techniques like ADPCM or MBE (Multiband Excitation). G.722 is a subband ADPCM operating at 48/56 kbps with an average MOS score of 3.0.

Digital Voice Systems, Inc. has produced new proprietary technology named MBE (Multiband Excitation). Under this heading, two coders are developed named [28] IMBE (Improved MBE) and AMBE (Advanced MBE), which have superior performance than LPC (Linear Prediction Coder) based coders. They produce an MOS of 3.3 at a bit rate of 4.8kbps. In Parametric Coding, Speech is produced in parametric form. The human vocal tract is represented using the parameters of all pole filters. In contrast, input to this filter is represented by an impulse train representing frequency for voiced speech and random noise for unvoiced frequency. These types of coders keep speech quality suitable for the human ear. It consists of LPC (Linear Prediction Coding) and MELP (Mixed Excitation LPC). In LPC order of all pole filters is very important. Speech is divided into many subtypes: voice, UN voice, silence, music, and background. In [29, 30], the neural network decided on the proper LPC order for all these frames and found excellent speech quality at a meagre rate. MELP coders, described in [31-33], work at 2.4 kbps and have good resistance to background noise and channel errors.

The effect of distortion on the performance of MELP is demonstrated in [34] and shown that the more distortion more MELP coder is affected, but voicing decision and pitch estimation remain unaffected. Hybrid coding is a mixture of Waveform coding and Parametric Coding. CELP (Code Excited Linear Prediction) and its variants like ACELP (Algebraic CELP), CS-ACELP (Conjugate Structure ACELP), LD-CELP (Low Delay CELP), MP-CELP (Multi Pulse CELP), VSELP (Vector Sum Excited

LPC). CELP coders divide an input speech signal into frames and sub-frames. LPC coefficients are adjusted for each subframe using a codebook such that the predicted signal closely matches the original signal. [35] tested the CELP coder with two codebooks, namely the Gaussian and Fixed codebook, and found that the fixed codebook performs better than the Gaussian codebook. In [36], CELP coders are tested with Gaussian, Fixed and Fixed-Genetic codebooks and found that it performs will with Fixed-Genetic codebooks. [37]. It is also pointed out that the importance of codebook optimization and fast searching algorithm for codebook plays a vital role in deciding the CELP coder's performance. A psychoacoustic filter for noise is developed in [38] to improve CELP coder performance. Codebooks can be implemented in the transform domain for increasing the data rate of CELP [39], or instead of using codebooks, MLCC (Mel Cepstral Coefficients) can be used [40] to get better results. For getting wideband CELP, either a multiband bank of offline filter excitation codebooks [41] or a single pulse codebook is to be used instead of a Gaussian codebook [42]. CELP coders are affected by the loss of frames. This problem can be resolved using the proper FEC (Forward Error Correction) technique [43]. CELP coders working at 4.8 kbps designed [44] produces good result against channel errors and noisy environment. CELP with different rates in the range of 4-8 kbps in real time is developed by [45], and in the range of 9.6-16 kbps is developed by [46]. Both these papers have got good results in SNR and MSE indexes

Table 1. Open and private standards

Sr.	Name of Technology	Description	Developed by	Standards listed	Speech coder used
1.	DMR	2 Slot TDMA with 4 FSK Modulation	ETSI	ETSI TR102-398 and TS 102 361-1to 361-4 [2]	AMBE+2
2.	DPMR	FDMA with 4 FSK Modulation	ETSI	ETSI TR102-658 ver. 2.6.1 [3]	4 coders are selectable 1. AMBE+2 2. RALCWI 3. ASELP 4.AMBE+2C
3.	TETRA	4 Slot TDMA with pi/4 DQPSK Modulation	ETSI	ETSI EN 300 392-2 ETR 300 and TR 102 300 EN 300 394 to 396 [4]	ACELP
4.	APCO 25 PI	Analog Digital and Mixed Mode with C4FM Modulation.	APCO	TIA 102 [5]	IMBE
5.	APCO 25 PII	2 Slot TDMA with HDQPSK Modulation for downlink and HCPM Modulation for Uplink.	APCO	TIA -102 [5]	AMBE +2
6.	Open Sky	4 Slot TDMA	Harris Corporation	Open Sky standards by Harris	Enhanced AMBE
7.	NXDN	FDMA with 4 PSK Modulation	ICOM and Kenwood	ETSI DPMR	AMBE+2

Table 2. Speech coder types and essential parameters

Sr.	Type	Sub Type	Standard	Data Rate Kbps	MOS value
1.	Waveform Coder	a. PCM based	G.711	64.0	4.3
		b. DPCM based		64.0	3.5
		c. ADPCM based	G.723.1	5.6/6.3	3.8
2.	Transform Coding	a. STFT based		14.0	2.0
		b. DCT based		4.8	2.1
		c. Wavelet based		16.0	2.4
		d. Adaptive Transform		16.0	2.5
3.	Sub Band Coding	a. ADPCM based	G.722	48/56	3.0
		b. MBE based	AMBE	4.8	3.3
			IMBE	4.8	3.4
4.	Parametric Coding	a. LPC based	LPC 10e	2.4	3.1
		b. MELP based		10.0	3.4
5.	Hybrid Coding	a. CELP based	G.729A	8.0	4.0
		b. ACELP	G.722.2	4.7-24	3.2
		c. CS-ACELP	G.729	8.0	3.48
		d. MP-CELP		4.2-7.5	3.0
		e. LD-CELP	G.728	16.0	4.0
		f. VSELP		8.0	3.45
6.	Harmonic Sinusoidal coding	a. codec-2		2.550	**

Fast searching and ease of codebook design problem give rise to the use of algebraic code with good statistical properties is explained in [47], and control loss of speech quality due to loss of packets is controlled by proper use of FEC technique is developed in [48]. CS-ACELP uses one codebook instead of two by codebook partition and label assignment [43].

A new Algebraic codebook is designed in [49] to reduce code search time. The depth-first tree structure reduces the search load by 75%. CS-ACELP MATLAB Simulink model is developed in [50] and implemented on the DSK kit to evaluate its performance. It is shown that processing time is reduced by 8.564 microseconds, and memory use is reduced by 8 % with the same data rate. LD-CELP aims to decrease delay and noise to improve speech quality [51]. In [52], both LPC predictor and excitation gain are back-adaptive using an excitation vector of 5 samples. It reduces delay to less than 2 msec. Multi-stage excitation coding achieves flexible bit rate and bandwidth capability [53]. VSELP described in [54] protects against channel errors with less computational complexity. Various researchers concentrate on one particular aspect (for example, speed, speech quality etc.) of the coder and improve it[55]. Other aspects related to the coder are the language used for communication, RF link design which includes the design and selection of Modulator and Channel coding, and Link budget are essential and need to be considered when the speech quality of total digital radio is considered. Even though AMBE or IMBE-based coders have good performance for speed and speech quality, facing the problem of change in speech quality as language changes or change in speech quality as received signal strength at receiver input reduces. Different coders are suitable for different applications. Digital radio should consist of various types of coders, and coders should be user selectable also coder codebook should be easily alter-able as per change in language. With the change in climatic conditions, they received signal strength at receiver input changes, so it is suggested to use Automatic change in modulation and channel coding as per change in environment. Software Defined Radio (SDR) technology can be its one solution.

4. Quantizer Types

Quantizer follows the speech coder stage. It has two types – scalar and Vector. Scalar Quantization deals with one sample at a time. Spacing between quantizing levels is either fixed or changed as per a fixed rule or changed dynamically from past samples. Vector Quantization deals with a group or block of samples at a time and works on a code book. Its sub-types [9] are based on searching techniques and the formation of code vectors, numbers and codebooks. They are listed in Table-3. Vector Quantization parameters include Distortion Parameters, mean square estimation, Signal Quantization Ratio (SQNR), and compression ratio [56]. The code book's increasing size increases the compression ratio and computational time [57].

The use of the Lattice structured codebook solves this issue. It can be further improved using the extending technique mentioned in [58]. The code book can be altered depending on the input vector. This change is based on the similarity of difference between successive frames or the like hood of adjacent frames. Such dynamic codebook reordering techniques are provided in [59-62] and [63]. Reordering can be static or dynamic. [64] Carried out the successful experiment of dynamic reordering of code book to improve the efficiency of vector quantization. [65] Proved Tree-structured vector quantization is efficient from the point of view of efficiency and memory requirements. This shows that the central heart of the quantizer is the codebook design. Codebook design and its database change as per language change. For this purpose, either quantizer should be able to adopt all languages, or they should be selectable as per spoken language.

5. Link or RF budget

In Radio Network Design, RF Link Budget decides desired signal strength at receiver input at a specified signal-to-noise ratio or BER (Bit Error Rate). Depending on this signal strength, the quality of the speech signal varied. This is shown in [66, 67]. It is shown that for the fixed data rate of AMBE+2 coder, as the percentage of BER changes from zero to three, the per cent MOS value changes from 3.6 to 2.5 for 3600 bps codec,3.5 to 2.45 for 3300 bps codec and 3.7 to 2.6 for 7200 bps codec, with changes in Modulation and Channel Coding type, bandwidth changes, affecting speech quality. For this purpose, Digital radio should provide changing modulation and channel coding as per user need. It should incorporate changes in channel coding and modulation per change in environment.

6. Spoken Language Characteristics

Each speech coder performs differently for different languages. This is because each language has a different architecture than others. A few characteristics of language that affect coder performance are listed below.

- Number of vowels and consonants, and syllables.
- Vowels are analyzed using Duration (short or long), pitch, formant frequencies, Number and position of vowels in syllables. Duration of vowels and pitch for words in Tamil are studied in [68]. [69] proved that vowel duration and the short-to-long ratio of vowel duration in words is responsible for getting the correct meaning. Different languages like Standard Arabic, Japanese and Thai are studied by [70] to test their vowel characteristics and effect on speech quality.
- Consonants are analyzed for their duration.
- Syllable duration, the position of vowels in syllables and grouping of vowels and consonants in syllables. The effect of syllable type on speech quality is studied by [71].

In [72], Vowel quality is directly related to vowel duration. In the Uigur language, Vowel duration is directly related to the position of the vowel in a syllable (word-initial or word-final) and the number of syllables in a word. Formant frequency stability is analyzed by [73], concluding that vowel duration and format frequency are important in English for better voice clarity. Consonant and repeated consonant duration for the Arabic language is studied by [74] and proved that for VCCV and VCV syllables, the duration of a consonant could prove whether it is single or repeated. Specific consonant clusters in the Hindi language are studied by [75], and found essential properties of those clusters. American, Arabic, English and British English are tested for ACELP and MDCT coders [76,77], and it found that the performance of all coders is not the same for these three languages. It is shown that ACELP and MDCT coders perform better for American English than the other two languages.All these papers deal with vowels, consonants, and syllables characteristics of males and females for different languages. It is concluded that speech quality depends on language constructs and the response of a speech coder to a language. A speech coder must be designed to produce reasonable responses to all languages, or they should be selectable as per language.

Table 3. Quantization types

Sr.	Quantization	Method	Subtype	Explanation
1.	Scalar Quantization	PCM	Uniform PCM	Equal Spacing between quantizing levels.
			Non-Uniform PCM	Spacing between quantizing levels depends on signal amplitude.
			APCM	Spacing between quantizing levels approximated from past coded samples.
		DPCM	Simple DPCM	Encode the difference between sample values.
			Delta Modulation	Encode the difference between sample values using a single bit.
2.	Vector Quantization- VQ	Codebook based design	Search technique	Full search Vector quantization (FSVQ) is based on single quantization based on a tree or binary tree search.
			Multistep VQ	Cascading of VQ
		Long Code Vector	Vector Sum Quantization	Formation of long code vector using many small code vectors.
		Signal and gain code book.	Gain Shape VQ	Separate code book for signal and gain.
		Dynamic code book	Adaptive Code Book (ACB)	Codebook size changes as per signal. It can be forward-adaptive or backwards-adaptive.

7. Results and Conclusion

For designing Digital Radio and its Network, spoken language and the response of low data rate codec are very important. Besides other Radio Network design tools, Digital Radio designers should find outspoken language characteristics. In future, speech coders in Digital radio should be selectable as per the spoken language used. In radio, using a speech-to-text converter will be essential if bandwidth is further planned to reduce. Suppose the Digital Radio receiver sensitivity is specified as -119 dbm. at 5 pm BER, the mini-mum signal at the receiver should be designed to be far greater than approximately -100 dbm. at five ppm BER using link budget calculations. Strong RF signal, Adaptive Modulation and Channel Coding, Selectable speech coder and quantizer as per user spoken language, easily tenable or intelligent code book for adopting user language, and adjustable gain parameters for compensating human speech level to radio transmitter input will be possible solutions for improving speech quality in low bandwidth digital radio. This section describes the Marathi Alphabet and its components, syllable or Akshar and their types, Compound Syllable and their types and Union of words and their types. These are basic constructs of the language. The analysis carried out in section 4 is based on these constructs.

Conflicts of Interest

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