Voice Compression and Communications: Principles and Applications for Fixed and Wireless Channels

by

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Preface and Motivation

The Speech Coding Scene

In the era of the third - generation (3G) wireless personal communications standards - despite the emergence of broad-band access network standard proposals - the most important mobile radio services are still based on voice communications. Even when the predicted surge of wireless data and Internet services becomes a reality, voice remains the most natural means of human communications, although this may be delivered via the Internet - predominantly after compression.

This book is dedicated mainly to voice compression issues, although the aspects of error resilience, coding delay, implementational complexity and bitrate are also at the centre of our discussions, characterising many different speech codecs incorported in source-sensitivity matched wireless transceivers. Here we attempt a rudimentary comparison of some of the codec schemes treated in the book in terms of their speech quality and bitrate, in order to provide a road map for the reader with reference to Cox's work [1, 2]. The formally evaluated Mean Opinion Score (MOS) values of the various codecs portrayed in the book are shown in Figure 1.

Observe in the figure that over the years a range of speech codecs have emerged, which attained the quality of the 64 kbps G.711 PCM speech codec, although at the cost of significantly increased coding delay and implementational complexity. The 8 kbps G.729 codec is the most recent addition to this range of the International Telecommunications Union's (ITU) standard schemes, which significantly outperforms all previous standard ITU codecs in robustness terms. The performance target of the 4 kbps ITU codec (ITU4) is also to maintain this impressive set of specifications. The family of codecs designed for various mobile radio systems - such as the 13 kbps Regular Pulse Excited (RPE) scheme of the Global System of Mobile communications known as GSM, the 7.95 kbps IS-54, and the IS-95 Pan-American schemes, the 6.7 kbps Japanese Digital Cellular (JDC) and 3.45 kbps half-rate JDC arrangement (JDC/2) - exhibits slightly lower MOS values than the ITU codecs. Let us now consider the subjective quality of these schemes in a little more depth.

The 2.4 kbps US Department of Defence Federal Standard codec known as FS-1015 is the only vocoder in this group and it has a rather synthetic speech quality, associated with the lowest subjective assessment in the figure. The 64 kbps G.711 PCM codec and the G.726/G.727 Adaptive Differential PCM (ADPCM) schemes are waveform codecs. They exhibit a low implementational complexity associated with a modest bitrate economy. The remaining codecs belong to the so-called hybrid coding family and achieve significant bitrate economies at the cost of increased complexity and delay.

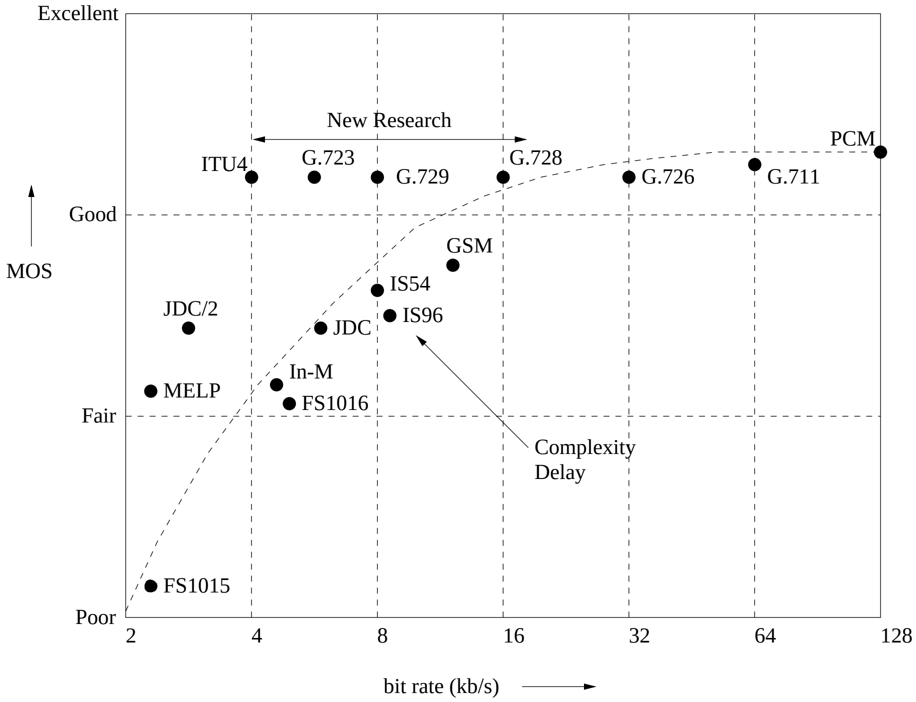

Figure 1: Subjective speech quality of various codecs [1] ©IEEE, 1996

Specifically, the 16 kbps G.728 backward-adaptive scheme maintains a similar speech quality to the 32 and 64 kbps waveform codecs, while also maintaining an impressively low, 2 ms delay. This scheme was standardised during the early nineties. The similar-quality, but significantly more robust 8 kbps G.729 codec was approved in March 1996 by the ITU. Its standardisation overlapped with the G.723.1 codec developments. The G.723.1 codec's 6.4 kbps mode maintains a speech quality similar to the G.711, G.726, G.727, G.728 and G.728 codecs, while its 5.3 kbps mode exhibits a speech quality similar to the cellular speech codecs of the late eighties. Work is under way at the time of writing towards the standardisation of a 4 kbps ITU scheme, which we refer to here as ITU4.

In parallel to the ITU's standardisation activities a range of speech coding standards have been proposed for regional cellular mobile systems. The standardisation of the 13 kbps RPE-LTP full-rate GSM (GSM-FR) codec dates back to the second half of the eighties, representing the first standard hybrid codec. Its complexity is significantly lower than that of the more recent Code Excited Linear Predictive (CELP) based codecs. Observe in the figure that there is also a similar-rate Enhanced Full-Rate GSM codec (GSM-EFR), which matches the speech quality of the G.729 and G.728 schemes. The original GSM-FR codec's development was followed a little later by the release of the 7.95 kbps Vector Sum Excited Linear Predictive (VSELP) IS-54 American cellular standard. Due to advances in the field the 7.95 kbps IS-54 codec achieved a similar subjective speech quality to the 13 kbps GSM-FR scheme. The definition of the 6.7 kbps Japanese JDC VSELP codec was almost coincident with that of the IS-54 arrangement. This codec development was also followed by a half-rate standardisation process, leading to the 3.2 kbps Pitch-Synchroneous Innovation CELP (PSI-CELP) scheme.

The IS-95 Pan-American CDMA system also has its own standardised CELP-based speech codec, which is a variable-rate scheme, supporting bitrates between 1.2 and 14.4 kbps, depending on the prevalent voice activity. The perceived speech quality of these cellular speech codecs contrived mainly during the late eighties was found subjectively similar to each other under the perfect channel conditions of Figure 1. Lastly, the 5.6 kbps half-rate GSM codec (GSM-HR) also met its specification in terms of achieving a similar speech quality to the 13 kbps original GSM-FR arrangements, although at the cost of quadruple complexity and higher latency.

Recently the advantages of intelligent multimode speech terminals (IMT), which can reconfigure themselves in a number of different bitrate, quality and robustness modes became known in the community, which led to the requirement of designing an appropriate multi-mode codec, the Advanced Multi-Rate codec referred to as the AMR codec. A range of IMTs also constitute the subject of this book. Current research on sub-2.4 kbps speech codecs is also covered extensively in the book, where the aspects of auditory masking become more dominant. Lastly, since the classic G.722 subband-ADPCM based wideband codec is becoming somewhat obsolete in the light of exciting new development in compression, the most recent trend is to consider wideband speech and audio codecs, providing susbtantially enhanced speech quality. Motivated by early seminal work on transform-domain or frequency-domain based compression by Noll and his colleagues, in this field the PictureTel codec - which can be programmed to operate between 10 kbps and 32 kbps and hence amenable to employment in IMTs - is the most attractive candidate. This codec is portrayed in the context of a sophisticated burst-by-burst adaptive wideband turbo-coded Orthogonal Frequency Division Multiplex (OFDM) IMT in the book. This scheme is also capable of transmitting high-quality audio signals, behaving essentially as a good waveform codec.

Mile-stones in Speech Coding History

Over the years a range of excellent monographs and text books have been published, characterising the state-of-the-art at its various stages of development and constituting significant mile-stones. The first major development in the history of speech compression can be considered the invention of the vocoder, dating back to as early as 1939. Delta modulation was contrived in 1952 and it became well established following Steele's monograph on the topic in 1975 [3]. Pulse Coded Modulation (PCM) was first documented in detail in Cattermole's classic contribution in 1969 [4]. However, it was realised in 1967 that predictive coding provides advantages over memory-less coding techniques, such as PCM. Predictive techniques were analysed in depth by Markel and Gray in their 1976 classic treatise [5]. This was shortly followed by the often cited reference [6] by Rabiner and Schafer. Also Lindblom and Ohman contributed a book in 1979 on speech communication research [7].

The foundations of auditory theory were layed down as early as 1970 by Tobias [8], but these principles were not exploited to their full potential until the invention of the analysis by synthesis (AbS) codecs, which were heralded by Atal's multi-pulse excited codec in the early eighties [9]. The waveform coding of speech and video signals has been comprehensively documented by Jayant and Noll in their 1984 monograph [10]. During the eighties the speech codec developments were fuelled by the emergence of mobile radio systems, where spectrum was a scarce resource, potentially doubling the number of subscribers and hence the revenue, if the bitrate could be halved.

The RPE principle - as a relatively low-complexity analysis by synthesis technique - was proposed by Kroon, Deprettere and Sluyter in 1986 [11], which was followed by further research conducted by Vary [12, 13] and his colleagues at PKI in Germany and IBM in France, leading to the 13 kbps Pan-European GSM codec. This was the first standardised AbS speech codec, which also employed long-term prediction (LTP), recognising the important role the pitch determination plays in efficient speech compression [14, 15]. It was in this era, when Atal and Schroeder invented the Code Excited Linear Predictive (CELP) principle [16], leading to perhaps the most productive period in the history of speech coding during the eighties. Some of these developments were also summarised for example by O'Shaughnessy [17], Papamichalis [18], Deller, Proakis and Hansen [19].

It was during this era that the importance of speech perception and acoustic phonetics [20] was duly recognised for example in the monograph by Lieberman and Blumstein. A range of associated speech quality measures were summarised by Quackenbush, Barnwell III and Clements [21]. Nearly concomitantly Furui also published a book related to speech processing [22]. This period witnessed the appearance of many of the speech codecs seen in Figure 1, which found applications in the emerging global mobile radio systems, such as IS-54, JDC, etc. These codecs were typically associated with source-sensitivity matched error protection, where for example Steele, Sundberg and Wong [23–26] have provided early insights on the topic. Further sophisticated solutions were suggested for example by Hagenauer [27].

During the early nineties Atal, Cuperman and Gersho [28] have edited prestigious contributions on speech compression. Also Ince [29] contributed a book in 1992 related to the topic. Anderson and Mohan co-authored a monograph on source and channel coding in 1993 [30]. Most of the recent developments were then consolidated in Kondoz' excellent monograph in 1994 [31] and in the multi-authored contribution edited by Keijn and Paliwal [32] in 1995. The most recent addition to the above range of contributions is the second edition of O'Shaughnessy well-referenced book cited above.

Motivation and Outline of the Book

Against this backcloth - since the publication of Kondoz's monograph in 1994 [31] nearly six years have elapsed - this book endeavours to review the recent history of speech compression and communications. We attempt to provide the reader with a historical perspective, commencing with a rudimentary introduction to communications aspects, since throughout the book we illustrate the expected performance of the *various speech codecs studied also in the context of a full wireless transceiver.*

The book is constituted by four parts. Part I and II are covering classic background material, while the bulk of the book is constituted by the research-oriented Part III and IV, covering both standardised and proprietary speech codecs and transceivers. Specifically, Part I provides a rudimentary introduction to the wireless system components used throughout the book in quantifying the overall performance of the various speech codecs, in order to render our treatment of the topics self-contained. Specifically, the mobile propagation environment, modulation and transmission techniques as well as channel coding are considered in Chapters 1-4. For the sake of completeness Part II focusses on aspects of classic waveform coding and predictive coding in Chapters 5 and 6. Part III is centred around analysis by synthesis based coding, reviewing the principles in Chapter 7 as well as both narrow and wideband spectral quantisation in Chapter 8. RPE and CELP coding are the topic of Chapters 9 and 10, which are followed by an approximately 100-page chapter on the existing forward-adaptive standard CELP codecs in Chapter 11 and on their associated source-sensitivity matched channel coding schemes. The subject of Chapter 12 is proprietary and standard backward-adaptive CELP codecs, which is concluded with a system design example based on a low-delay, multi-mode wireless transceiver.

The essentially research-oriented Part IV is dedicated to a range of standard and proprietary wideband, as well as sub-4kbps coding techniques and wireless systems. As an introduction to the scene, the classic G.722 wideband codec is reviewed first, leading to various low-rate wideband codecs. Chapter 13 is concluded with a turbo-coded Orthogonal Frequency Division Multiplex (OFDM) wideband audio system design example. The remaining chapters, namely Chapters 14-21 are all dedicated to sub-4kbps codecs and transceivers.

This book is naturally limited in terms of its coverage of these aspects, simply due to space limitations. We endeavoured, however, to provide the reader with a broad range of applications examples, which are pertinent to a range of typical wireless transmission scenarios.

We hope that the book offers you a range of interesting topics, portraying the current state-of-the-art in the associated enabling technologies. In simple terms, finding a specific solution to a voice communications problem has to be based on a compromise in terms of the inherently contradictory constraints of speech quality, bitrate, delay, robustness against channel errors, and the associated implementational complexity. Analysing these trade-offs and proposing a range of attractive solutions to various voice communications problems is the basic aim of this book.

Again, it is our hope that the book underlines the range of contradictory system design trade-offs in an unbiassed fashion and that you will be able to glean information from it, in order to solve your own particular wireless voice communications problem, but most of all that you will find it an enjoyable and relatively effortless reading, providing you with intellectual stimulation.

Lajos Hanzo

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Lajos Hanzo

Part I

Transmission Issues

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