

AN INTERACTIVE TOOL FOR BIT ERROR RATE ANALYSIS OF SPEECH CODING ALGORITHMS

Hiren C Bhagatwala, Ted Painter, and Andreas Spanias

Department of Electrical Engineering, Distance Learning Initiative, TRC,
Arizona State University, Tempe, Arizona 85287-7206
spanias@asu.edu

ABSTRACT

A GUI-based software tool that provides a framework for evaluation of different speech coding algorithms is presented. The tool is designed to measure the susceptibility of speech coding algorithms to errors added on the encoded bit-stream during transmission. In particular, the errors can be added individually to the parameters that comprise the encoded bit-stream. This enables a designer of a speech codec to evaluate its performance under adverse or impaired channel conditions. The tool is universally applicable to different speech coding algorithms, by means of a user-defined bitstream definition file. In fact, the tool has been used in the past to evaluate a number of standardized speech coding algorithms. This paper describes the features of the software. In addition, the paper presents sample results generated by this tool during a study of the ETSI GSM Enhanced Full Rate algorithm.

1. INTRODUCTION

In recent years, there have been a large number of enhancements to speech coding algorithms applied in various digital communication systems [1]. In addition, diverse technologies in cellular systems have led to the adoption of several different standardized algorithms for speech coding [2] [3] [4]. Increased research and development in speech coding has brought about a need for tools that enable designers to evaluate and compare the performance of different speech coding algorithms. An interactive GUI-based software that attempts to fulfil this need is presented here.

Performance of the speech coding algorithm under impaired channel conditions forms an important criterion in its evaluation for applicability to a particular system. The amount of degradation in the output of a decoder due to channel errors on the encoded bit-stream depends not only on the error rate, but also on the parameter whose value is being corrupted. A study of this aspect of performance under impaired channel conditions for several speech coding standards was presented earlier by Spanias and Painter [5]. The software presented here provides a framework for analyzing the performance of different algorithms under adverse channel conditions from this viewpoint. Different er-

ror rates can be applied to each individual parameter comprising the encoded bit-stream. The collective or individual effect of a specified error rate in each parameter on the final decoded speech can then be measured by objective or subjective criteria. This is a universally-applicable tool, designed to accommodate different speech coding algorithms. The tool has an intuitive graphical interface that allows developers to experiment rapidly with a variety of speech coding algorithms.

This paper is organized as follows. Section 2 describes the functioning and use of the tool. The section includes a detailed example of how to configure the software for an example speech codec through the user-defined bitstream definition file. Section 3 gives sample results generated by the tool in a study of the ETSI GSM Enhanced Full Rate algorithm. Finally, section 4 describes the use of this tool by graduate students at Arizona State University, and examines possible directions for future work.

2. SPEECH CODEC ERROR ANALYSIS TOOL

2.1. Overview

The objective of this software is to provide a framework for analysis and comparison of different speech coding algorithms using a graphical interface. This tool allows algorithm designers to evaluate the performance of speech coding algorithms subjectively through listening tests as well as objectively through measurements of certain parameters closely related to the speech quality [6]. During the design stages of speech coding algorithms, it is necessary to determine the performance of a speech codec for different bit error rates (BER) in sets of parameters in the bit-stream produced at the encoder end. With this tool, designers can categorize the performance of different speech coding algorithms for varying BERs, through simple interactions with the GUI-based software. Fig. 1 gives a conceptual overview of the software. The next two subsections describe the functionality provided by this tool.

2.2. Parametric BER testing

To use this tool, the developer starts by defining the speech codec. The speech codec definition is in terms of a classification of the distinct sets of parameters for each frame in the encoded bit-stream. As an example, for CELP-based vocoders, these parameters would typically be the

Supported in part by the Distance Learning Initiative at Arizona State University

For questions on this paper or software, please email Andreas Spanias (spanias@asu.edu)

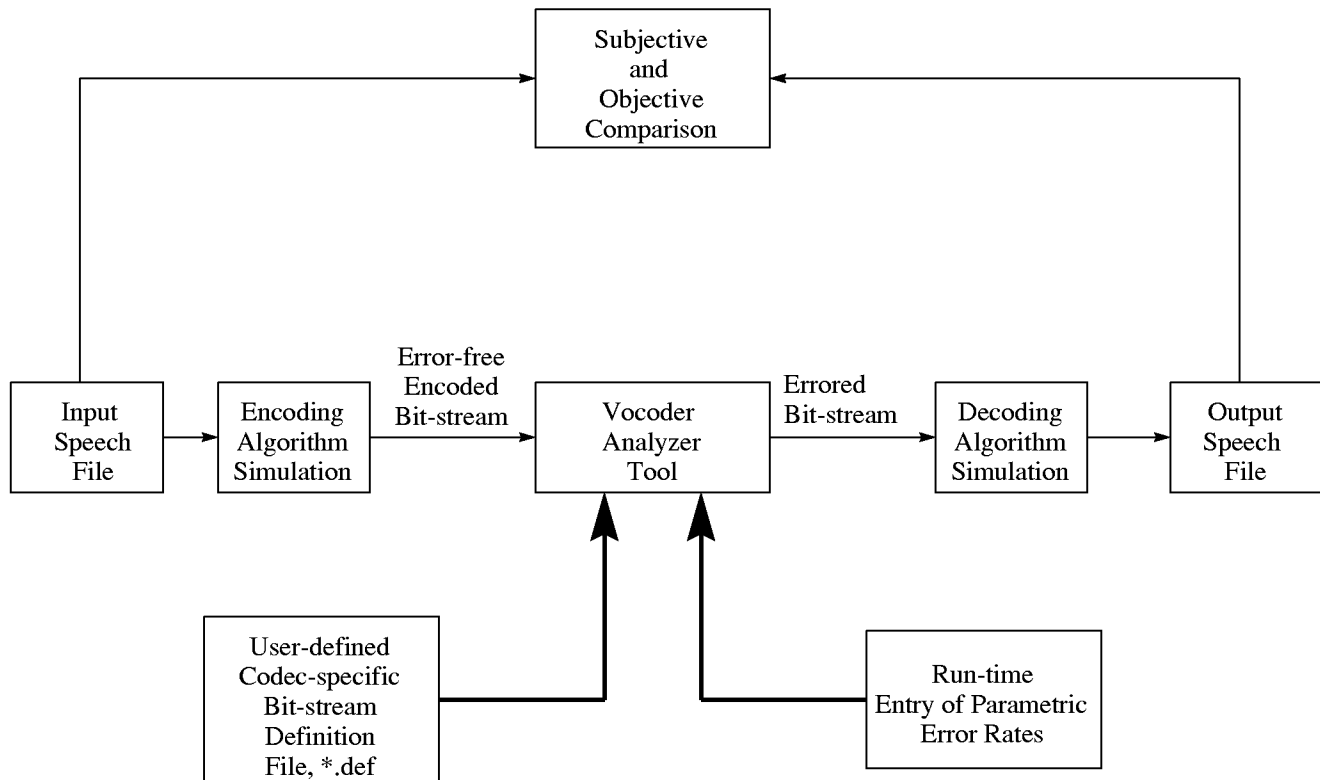


Figure 1: Overview of the speech codec analysis tool

quantization indices for line spectrum pairs, fixed-codebook shape/gain and adaptive codebook index/gain. However, the bitstream definition syntax is flexible and extensible so that developers are free to specify any parameter that may be required by the codec under consideration. The definition file indicates the number of bits for each parameter in each frame of the encoded bit-stream. It also indicates the type of bitstream file the encoder generates, for example, ASCII or BINARY, having packed bits or bits arranged as integers, depending on the type of codec simulation. A special feature enhancing the versatility of this tool is that the definition file could be written for variable rate codecs. For a variable rate speech coding algorithm, the number of bits occupied by each set of parameters for each of the possible rates in the encoded bit-stream is specified in the definition file. The software reads the definition file and presents the user with a graphical interface that lists the parameters on the encoded bit-stream for each frame. Using this graphical interface, the user can then specify the bit error rate to be used for corrupting each of these parameters. The encoded bit-stream, as well as encode and decode commands can also be entered through an intuitive GUI. A snapshot of this graphical interface for the GSM EFR speech codec is shown in Fig. 2. The BERs entered by the user correspond to varying levels of noise being added to the encoded bit-stream. Following this, the tool reads the encoded bit-stream, organizes it in frames as per the frame size specified in the bit-stream definition, and corrupts each set of bits corresponding to a particular parameter with the specified

BER. The degraded output speech is then obtained by decoding the corrupted bit-stream. Users can compare the amount of degradation through subjective listening tests as well as objective measures of speech quality.

2.3. Example Definition File

As an example, we consider here the GSM Enhanced Full Rate vocoder simulation distributed with the ETSI standard. The encoded bit-stream produced by this simulation is represented in terms of the definition file shown in Fig. 3. This definition file represents each frame of the bit-stream as defined in the GSM standard [2]. The 263 bits per frame generated by the encoder simulation are classified according to a set of quantized parameters on the encoded bit-stream. We note an additional feature that can be invoked here divides each frame into sub-frames. It is therefore possible to specify the bit-stream parameters at the sub-frame level. The corresponding GUI representation of the GSM EFR bit-stream definition is shown in Fig. 2. This ETSI GSM-EFR example illustrates how the tool supports arbitrary bitstream definitions. In fact, the tool is fully extensible and universally applicable to any codec, including variable-rate algorithms [7].

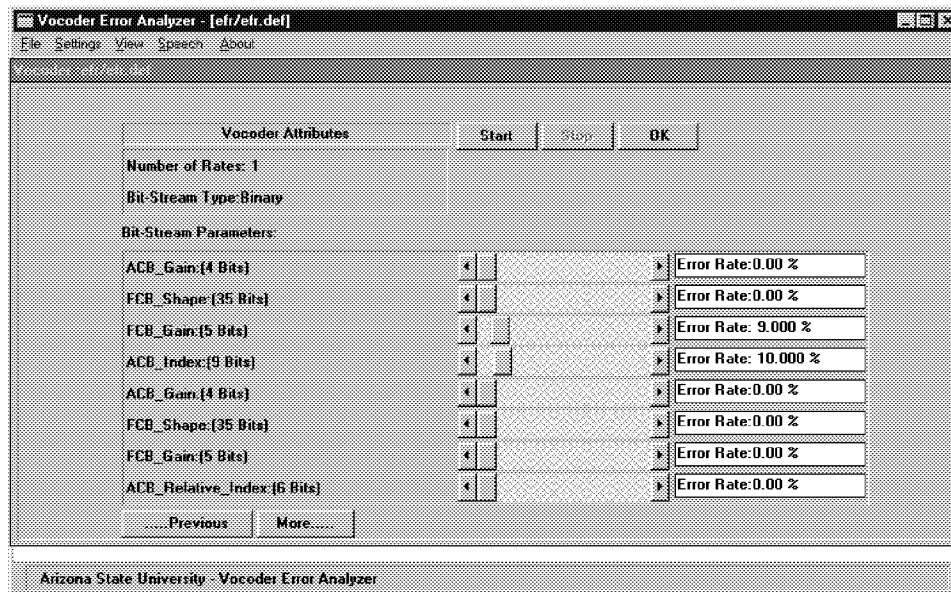


Figure 2: Graphical interface for the error analysis tool

```
Total_Rates=1
Bit_Stream_Format=BINARY
Packing_Format=UNPACKED
Rate=1
Params=21
BFI=1
Lsf=38
ACB_Index=9
ACB_Gain=4
FCB_Shape=35
FCB_Gain=5
ACB_Relative_Index=6
ACB_Gain=4
FCB_Shape=35
FCB_Gain=5
ACB_Index=9
ACB_Gain=4
FCB_Shape=35
FCB_Gain=5
ACB_Relative_Index=6
ACB_Gain=4
FCB_Shape=35
FCB_Gain=5
Blank=16
SID=1
TAF=1
```

Figure 3: Definition file for the ETSI GSM EFR Codec

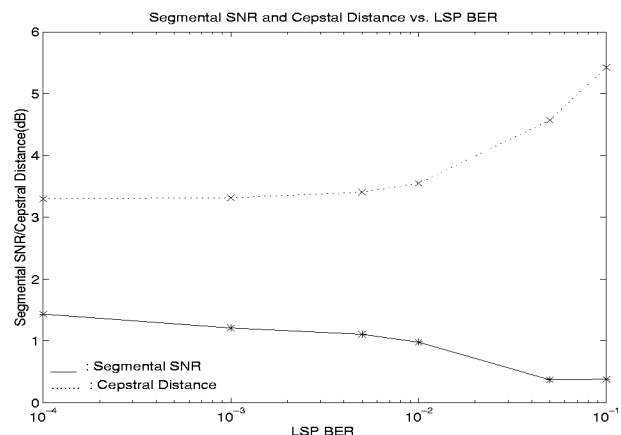


Figure 4: Degradation Associated with LSP Bit Errors, ETSI GSM-EFR

3. SAMPLE RESULTS: EVALUATION OF THE ETSI GSM-EFR ALGORITHM

The vocoder analyzer tool has been used at ASU to compare and evaluate several standardized speech codecs, including the G.729, IS-641, GSM half-rate and the GSM enhanced full rate algorithms. For each algorithm, the speech quality degradation associated with increasing BER in each parameter (e.g., line spectrum frequencies, pitch lags, codebook shapes and gains, etc.), was measured by objective and subjective criteria. As a representative sample of the tool's analysis capabilities, we present here results obtained during a study of the ETSI GSM-EFR algorithm. Figs. 4, 5, 6 and 7, respectively, show the degradation in speech quality caused by varying BER in the LSFs, fixed codebook

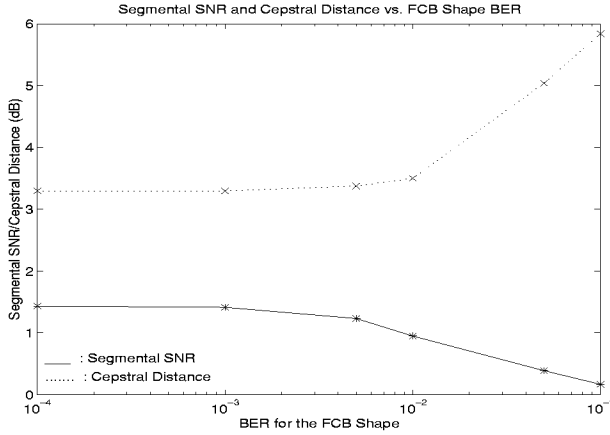


Figure 5: Degradation Associated with FCB Shape Bit Errors, ETSI GSM-EFR

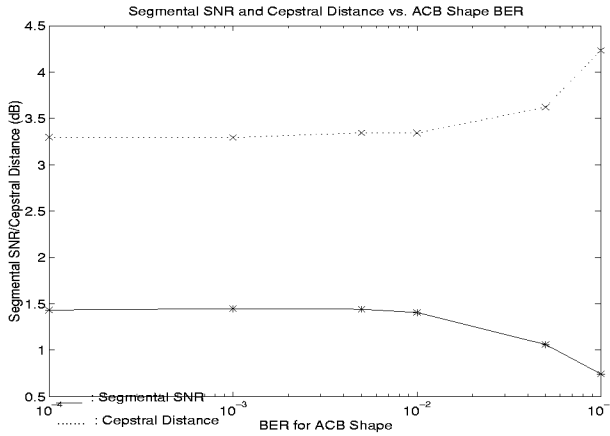


Figure 6: Degradation Associated with Adaptive Codebook Bit Errors, ETSI GSM-EFR

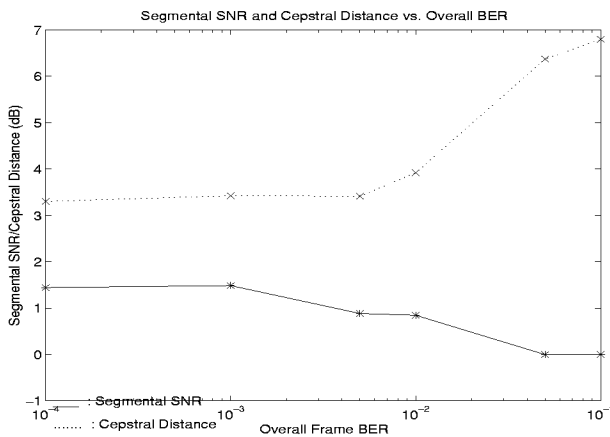


Figure 7: Degradation Associated with Overall BER, ETSI GSM-EFR

shape, adaptive codebook index and the overall channel. The degradation was measured in terms of the cepstral distance and segmental SNR between input and output speech. This information could be used to optimize the quantization required for each parameter in the encoded bit-stream, or to assist in the allocation of error protection bits in an unequal error protection (UEP) or other channel coding scheme [3].

4. CONCLUSION

This tool has been used by graduate-level students at Arizona State University to evaluate different speech coding algorithms based on subjective and objective criteria. The ability to define the speech codec under consideration and the interactive GUI aspects of this tool assumed significance in this setting.

Future enhancements to this tool for parametric BER testing include porting it to a Java-based application, wherein users across the Internet could test their own algorithms and evaluate the results. The use of streaming media in this aspect will also be explored.

5. REFERENCES

- [1] A. S. Spanias, "Speech coding: A tutorial review," *Proceedings of the IEEE*, vol. 82, pp. 1541–1582, October 1994.
- [2] European Telecommunications Standards Institute, *GSM: Digital cellular telecommunications system; Enhanced Full Rate (EFR) speech transcoding (GSM 06.60)*, 1996.
- [3] Telecommunications Industry Association, *TIA/EIA/IS-641: TDMA Cellular/PCS - Radio Interface - Enhanced Full-Rate Speech Codec*, 1996.
- [4] R. Salami, C. Laffamme, B. Bessette, and J.-P. Adoul, "ITU-T G.729 Annex A: reduced complexity 8 kb/s CS-ACELP codec for digital simultaneous voice and data," *IEEE Communications Magazine*, vol. 35, pp. 56–63, September 1997.
- [5] A. S. Spanias and E. M. Painter, "A software tool for introducing speech coding fundamentals in a DSP course," *IEEE Transactions on Education*, vol. 39, pp. 143–152, May 1996.
- [6] S. R. Quackenbush, T. P. Barnwell III, and M. A. Clements, *Objective Measures of Speech Quality*. Prentice Hall, 1988.
- [7] Telecommunications Industry Association, *TIA/EIA/IS-127: Enhanced Variable Rate Codec, Speech Service Option 3 for Wideband Spread Spectrum Digital Systems*, 1997.