Example T.O.C.

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CHAPTER 2

WAVEFORM CODING

In this chapter, the basic definitions and concepts of waveform coding will be reviewed. The chapter is intended to provide definitions and concepts that will be used in subsequent chapters. The treatment of waveform coding here is not intended to be exhaustive. The reader is referred to the excellent survey text by Jayant and Noll [5] for a more thorough treatment of the theory and practice of waveform coding. The chapter begins with a review of the elements of digital coding. Next, scalar quantization is discussed in some detail. The term scalar quantization is used to differentiate quantization processes which operate on single samples (scalars) from processes which operate on blocks of samples (vectors). Block or vector quantization is discussed at length in the following chapter. This chapter concludes with a description of several waveform coders: PCM, APC, ATC, and SBC.

2.1. Digital Coding

A simplified block diagram of a digital transmission system is shown in Fig. 2.1. The analog waveform $s(t)$ is presumed to be strictly bandlimited such that $|S(\omega)| = 0$ for all $|\omega| > \omega_H$ where $S(\omega)$ denotes the Fourier transform of $s(t)$. If $s(t)$ is not bandlimited (or approximately bandlimited), then $s(t)$ must be low-pass filtered prior

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protocol of the performance comparison. The formal comparison in terms of the four criteria ΔP_b, SQR, R_c, and γ (DSD) is then presented. A graphical performance comparison using the DSD criteria is given in Section 5.2. The issue of codebook mismatch is discussed in Section 5.3.

5.1. Performance Comparison

5.1.1. Methodology and Protocol

The comparison data presented here were obtained by computer simulation. The test modem signals were generated, coded, and demodulated using simulation software. The simulation process is illustrated in Fig. 5.1. A pseudo-random sequence of bits $\{b_T(m)\}$ is generated and input to the modulation routine. The modulator computes the signal sequence $\{s(n)\}$ which is equivalent to a sampled-data version of a modem waveform $s(t)$. That is

$$s(n) = s(t)\Big|_{t=nT}, \quad \forall\ n. \tag{5.1}$$

For some empirical tests, the sequence $\{s(n)\}$ is corrupted with additive noise $\{w(n)\}$. In particular, the ΔP_b data are obtained by corrupting $\{s(n)\}$ with an AWGN sequence $\{w(n)\}$ to produce some nominal bit-error probability for signal $\{x(n)\}$. The signal sequence $\{x(n)\}$ is processed by the waveform coder under test. The waveform coder produces the sequence $\{x_Q(n)\}$ which is equivalent to a sampled-data version of the reconstructed, coded waveform $x_Q(t)$. The sequence $\{x(n)\}$ serves as the input to the demodulator which produces the bit sequence $\{b_R(m)\}$.

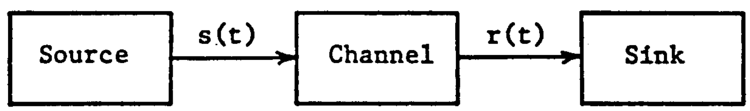

Fig. 1.1. Simplified model of a communication system.

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