

MODELING AND SIMULATION OF ANALOG AND DIGITAL COMMUNICATION SYSTEMS

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ABSTRACT

In this paper, a complete communication system simulation package is described. The package simulates the digital as well as the analog transmission systems. The digital transmission simulation includes data communication systems with digital filtering, source and channel coding, receiver data demodulation and decoding. The analog transmission simulation is very general; the modulator, the transmitter pulse shaping filter, the communication channel, the receiving filter, and the demodulator are considered. Simulation results regarding predictive source coding, error correcting coding (both random and burst errors) and modulation techniques are presented.

I. INTRODUCTION

The design and performance evaluation of a complete communication chain are, in general, a very difficult task. As a convenient practical approach the simulation of a complete communication system as well as of some of its parts, is required and, in many cases, this is the only practical approach.

In this paper, a complete communication system simulation package is described for both analog and digital transmissions. The package simulates the transmitter, the communication channel, and the receiver. The input signal can be either analog or digitized analog or digital data.

This paper is organized as follows. Sect. II describes the general organization of the simulation system with its input and output capabilities. Sect. III describes in detail the simulation package, which includes the digital and analog transmissions. Finally, as illustrative applications of the simulation package, Sect. IV shows the results obtained for two different communication schemes: an entirely digital communication system including source and channel coding, and a hybrid communication system able to transmit both voice and data in the same bandwidth.

II. GENERAL STRUCTURE OF THE SIMULATION SYSTEM

In this section, a general package for the simulation of communication systems, capable of dealing with both analog and digital transmissions is de-

scribed.

The general structure of this simulation package is shown in Fig. 1. The input signals can be both analog and digital signals. Analog signals are first sampled and quantized in order to be stored and used for the simulation. The sampling frequency and the number of quantization bits are chosen according to the characteristics of the signal and the transmission system which must be simulated. The digitized signal, when the input signal is in analog form, or the data, when the source is digital, are stored in a magnetic disk or tape.

In the case of digital transmission system simulation, the processing section can perform the operations of digital filtering, source coding, and channel coding.

When analog transmission systems must be analyzed, after having received the digitized input signal, the package simulates the transmitter-channel-receiver chain by means of the same processing blocks as shown in Fig. 1. Of course, after the demodulator, in this case, no channel and source decoding operations have to be performed.

The structure shown in Fig. 1 is clearly able to simulate the most general transmission system of interest. Of course, it can be easily adapted when dealing with particular systems having a simpler structure. Any block but the decision blocks can be bypassed without affecting the operations of the other blocks.

Each block of Fig. 1 corresponds to a program package with standardized input and output interfaces to communicate with other blocks. This allows a great flexibility in the actual system configuration and use.

In the following section, the blocks performing the simulation of digital and analog transmission systems are described, respectively, pointing out the capabilities of the program simulation package.

III. THE SIMULATION PACKAGE

A. Digital Transmission Simulation

The simulation of a digital transmission system includes the operations of digital filtering, source coding and decoding (data compression), channel cod-

ing and decoding, and the effects of the digital transmission channel when the simulation of the physical channel is not of interest.

Digital Filtering: Generally, both finite-impulse-response (FIR) and infinite-impulse-response (IIR) digital filters are of interest. From an implementation point of view, several structures can be simulated for either FIR or IIR digital filters [1],[2].

Source Coding and Decoding: Many algorithms for data compression or source encoding are available in the literature [3],[4]. As shown in [1], the simulation structure of these algorithms is quite similar. The subroutine, which performs the compression operation, has as input data the signal samples, the time duration of the signal, the coefficients a_j utilized for the prediction, and the tolerance value Δ . The output data of the subroutine are the compressed vector, which contains non-predicted samples and the time information symbols, and the compression ratio. At the receiver a decompression subroutine is required to reconstruct the signal from the compressed vector.

Channel Coding and Decoding: After data compression or source encoding, a channel coding operation is generally introduced to reduce the effects of channel noise and disturbances. The methods used for channel coding are strictly dependent on the characteristics of the noise introduced by the communication channel.

The memoryless channel is modeled as a binary symmetric channel (BSC), while the channel with memory is simulated using a model proposed by Gilbert [5].

Channel encoding and decoding operations for a block code or a convolution code can be performed by using the generator matrix G or the parity-check matrix H . In our simulation program we utilize the parity-check matrix because when $k > m$ (as in most cases) the matrix H has lower dimensions than the matrix G . The codes considered in our simulation are in a systematic form, where the first k symbols of every codeword are the information symbols to be encoded and the last m symbols are the redundancy symbols.

In order to simplify the structure of the simulation program, the channel encoding operation and the syndrome computation use the same subroutine with different input parameters. The block diagram of this subroutine, called CODDEC, is shown in Fig. 2. If the parameter IDEC is equal to 0, this subroutine is used for the channel coding operation, while if IDEC = 1, it is used for the computation of the syndrome vector.

B. Analog Transmission Simulation

In the simulation of modulated signals, it is often convenient to deal with an equivalent baseband model to reduce computation time and sample

number. In this respect, if $s(t)$ is a modulated signal having a frequency spectrum band limited in the band $(f_0 - B, f_0 + B)$, where f_0 is the carrier frequency and $2B < f_0$, it can be written in the form:

$$s(t) = A(t)\cos[\omega_0 t + \phi(t)] \quad (1)$$

where $\omega_0 = 2\pi f_0$.

In order to reduce the computation time, the corresponding baseband representation of the signal (1) is employed, which is given by:

$$s(t) = x(t)\cos\omega_0 t - y(t)\sin\omega_0 t \quad (2)$$

with

$$\begin{aligned} x(t) &= A(t)\cos\phi(t) \\ &= s(t)\cos\omega_0 t + \hat{s}(t)\sin\omega_0 t \\ y(t) &= A(t)\sin\phi(t) \\ &= \hat{s}(t)\cos\omega_0 t - s(t)\sin\omega_0 t \end{aligned} \quad (3)$$

where $\hat{s}(t)$ is the Hilbert transform of $s(t)$. The signals $x(t)$ and $y(t)$ are the in-phase and quadrature components of $s(t)$, respectively, and are low-pass signals in the frequency range up to B [6].

The modulations considered are binary and quaternary PSK (BPSK, QPSK), FSK and MSK, as shown in Sect. IV.

Transmission Filter, Analog Channel, and Receiving Filter: The block diagram of the program package usually employed in the simulation of analog transmission systems is shown in Fig. 3. The simulation of the transmission filter, the analog channel, and the receiving filter is carried out in the frequency domain through the appropriate use of the fast Fourier transform (FFT) algorithm. The baseband representation of the modulated signal is first transformed in the frequency domain by means of the FFT algorithm; then the operations of multiplication by the samples of the transmission filter frequency response, multiplication by the samples of the analog channel frequency response, addition of the FFT-transformed samples of the baseband representation of the additive channel noise, and multiplication by the samples of the receiving filter frequency response are performed in this order. Finally, the obtained result is transformed again in the time domain by applying an inverse FFT algorithm before supplying the signal to the demodulator input. It is well known that the correct time behavior of the signal from two or more linear systems in cascade can be obtained through the use of the FFT algorithm by applying the overlay-save or overlay-add methods [2]. Either method can be used in the simulation package at the choice of the user.

IV. EXAMPLES OF APPLICATION OF THE SIMULATION SYSTEM

In this section, we present some interesting results obtained from the simulation package previously described. In particular, the results refer to two different communication schemes. The first scheme simulates the digital part of a communication system and includes data compression operations and channel coding. The second scheme simulates a complete communication chain, including the analog section.

The first scheme was considered to evaluate the performance of data compression in noisy conditions and their integration with the channel coding operation [7]. To characterize the performance of these systems using data compression, we have simulated four structures: uncompressed-uncoded (UU), compressed-uncoded (CU), uncompressed-coded (UC), and compressed-coded (CC). For all these cases, we have obtained the compression ratio C_a and the rms error ϵ (as a percentage of the full scale signal). In the CU and CC systems, ϵ value includes the distortion introduced by the compression error and channel noise. To show the different influence of these two error types, we have also computed the rms error due only to the channel noise [1].

In data compression algorithms, time information symbols must be transmitted together with non-predicted samples, to give the exact time position of each received sample. In Fig. 4, we have denoted with index 1 the results obtained by using an algorithm in which time information represents the number of non-transmitted samples [4].

In the following results, the communication channel is simulated using the Gilbert model, which describes approximately the behavior of some channels with memory, as telephone channels. The algorithm utilized for data compression is the ZOP algorithm. The code for burst error correction is a Samoylenko binoid code (90,80) defined in a Galois field GF(31). In the binary transmission, the code is of the type (450,400) and is able to correct all the bursts of length 21 bits or less. The transmitted signal, which is an electrocardiogram (ECG), is first processed using a third-order low-pass digital filter and then compressed by a ZOP algorithm with floating aperture. For a tolerance $\Delta = 2.18$ percent with respect to the full scale, we have obtained an rms error $\epsilon = 1$ percent. It is clear that in the CU and CC cases it is impossible to go below this value. The rms error ϵ versus the channel error probability P_e or signal-to-noise ratio S/N_0 is shown in Fig. 4 for all the simulated systems. For high P_e the error ϵ is mainly determined by the channel noise, while the influence of the compression distortion is negligible. By reducing P_e , the importance of the compression errors becomes higher and higher.

The second scheme is a communication system for the simultaneous transmission of voice and data using the same carrier. The voice signal is transmitted by modulating the carrier amplitude, while the data signal modulates the phase of the same carrier [8].

The digital modulations utilized for data transmission are binary PSK (BPSK), quaternary PSK (QPSK), FSK, and MSK [8].

The bandpass filters in the communication chain were modeled as Butterworth filters with the following characteristics:

- transmit filter: fourth order, with -3 dB single-side bandwidth 7.5 kHz;
- receive filter: eighth order, with -3 dB single-side bandwidth 5 kHz.

The signal was processed in blocks of 2048 samples; the sampling frequency was chosen as 19 200 Hz.

An important problem in this communication system is the interference between the two types of modulations: amplitude and phase. In order to estimate the distortion introduced by data and phase modulation on the voice, the signal at the output of the AM detector $f_1(t)$ is compared with the original signal $f(t)$, which modulates the carrier amplitude. An error signal $e(t) = f_1(t) - f(t)$ is therefore obtained. The simulation program gives some parameters relative to this error signal as the mean power and the signal-to-noise ratio S/N_0 . Some different signals were considered in the simulation as analog transmitted signals [8].

As an example, in Fig. 5 the signal-to-noise ratio S/N_0 versus the bit rate v_s is reported in the case in which the signal modulating the carrier amplitude, is a tone at 937.5 Hz [curves (a)], at 1875 Hz [curves (b)], and the sum of five tones [curves (c)]. The modulation index is assumed equal to 0.8.

V. CONCLUSIONS

A very general computer simulation package for the analysis, design, and test of communication systems has been illustrated. Practically, it can deal with any kind of communication system of interest. Its modular structure allows one to modify any package module without affecting the other modules in order to fit any particular system requirement. Due to its relative complexity, it has been specifically developed for a large computer system. In particular, it has been implemented on a CII10070 computer. The programming language (ANSI Fortran) allows a sufficient degree of portability for use in other computer environments. Recent advances in the technology of minicomputers allow the package, or at least a somewhat simplified version, to be used in this class too.

The application examples of Sect. IV have shown

the capabilities of the package to simulate communication systems of both theoretical and practical interest.

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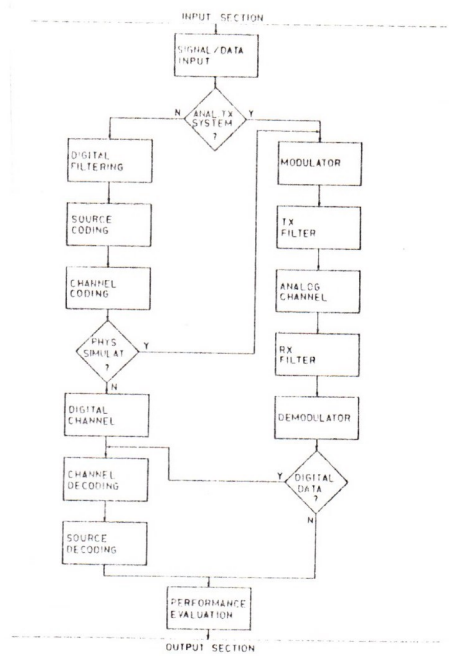


Fig. 1 -

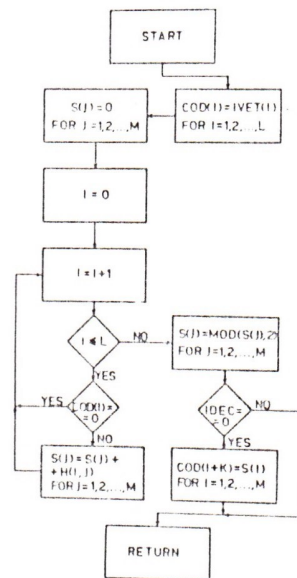


Fig. 2 -

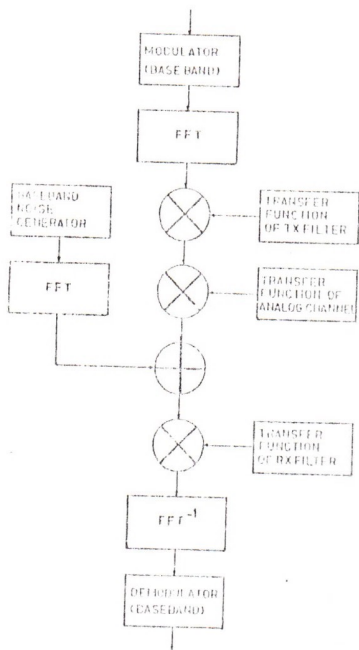


Fig. 3 -

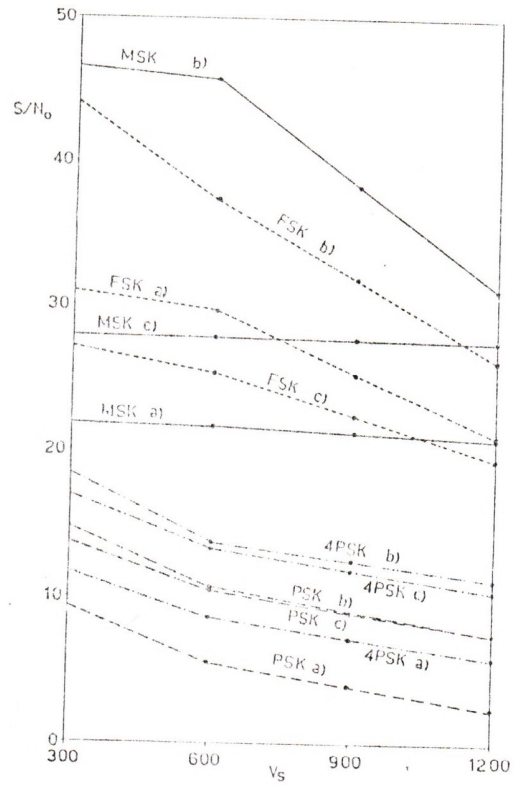


Fig. 5 -

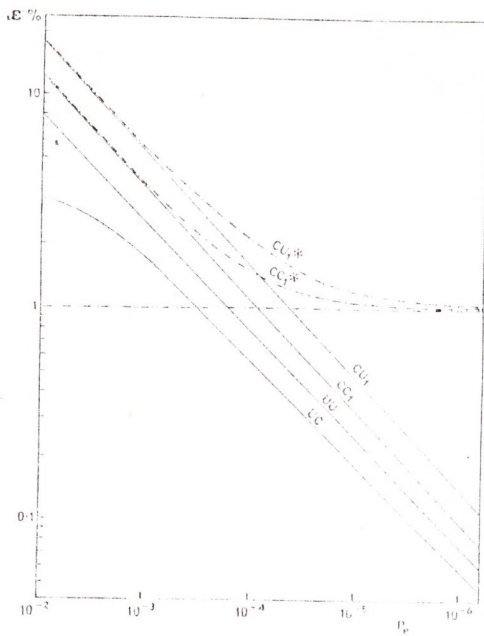


Fig. 4 -