A PROPOSED TEST SYSTEM AND STRATEGY FOR THE PRODUCTION TESTING OF PULSE CODE MODULATION CODECS

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DECLARATION

I hereby declare that this dissertation is my own work and that it has not previously been submitted for a degree at any University.

G A DEBBO

ABSTRACT

The growth, in the manufacture of pulse code modulation integrated circuit codecs, in recent years, has been phenomenal. This has mainly been due to the introduction of digital technologies into the present telecommunications network, both in the fields of switching and transmission. Large quantities of these devices are being fabricated daily, and as a result, a method is required in order to evaluate their performance with respect to defined industry specifications laid down by the International Consultive Committee for Telephone and Telegraph (CCITT).

In order to evaluate the performance of a codec, a number of tests are called for by the CCITT. The work reported here is concerned with the two major tests required viz. quantization distortion and gain tracking. The two criteria most important to any test system are test time and measurement accuracy. In terms of these two criteria, the above two tests are the most difficult to implement.

A test system, to carry out the two tests mentioned above, has been proposed, and in the present work examined, in order to determine its feasibility in meeting the two criteria mentioned earlier. Simulating the test system on a computer has resulted in measurement accuracies of $\pm$ 0.5 dB for quantization distortion and $\pm$ 0.01 dB for gain tracking.

In order to limit the test time required, a test strategy has been developed in order to control the test system. The resolution of the input signal used to stimulate the codec under test, which in the present work is sinusoidal in

nature, is controlled by the test strategy. The CCITT specification defines a definite lower limit for quantization distortion and gain tracking performance. Thus for devices where the performance curve falls well above the CCITT lower limit, a much coarser resolution is selected by the test strategy, as the probability of a device failing is very small. However, for devices where the performance curve falls close to the CCITT limit, a much finer resolution is selected, at the expense of test time, in order to prevent bad devices from being classified as good, and vice versus.

The test system layout is such that all signal generation and signal processing is done in software. In addition, all tests are conducted in the half channel configuration, so that the encoder and decoder sections of the codec circuit can be tested separately.

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1 INTRODUCTION

1.1 Background

Pulse code modulation (PCM) was invented in 1937 by Alec Reeves [1965], but because of the complexity of the equipment required to implement the idea, it was not until the introduction of the transistor in the early 1950's that PCM became feasible from an economic as well as practical aspect. Systems of this period were cumbersome occupying a multitude of printed circuit boards. With the introduction of integrated circuits in the early 1960's and the subsequent advancement on this technology leading to large scale and very large scale integrated devices in the late 1970's, the possibility of placing the encoder and decoder functions, required in the pulse code modulation process, on a single chip, became a reality. The term codec was adopted, an acronym made up from the words coder and decoder.

Until recently pulse code modulation has mainly been used as a transmission medium to provide increased capacity on existing physical junction links between exchanges in densely populated areas. With the introduction of electronic exchanges, which are based on PCM techniques, the growth of PCM in the telecommunications network is increasing rapidly. This will be further augmented with the introduction of digital subscriber loops in the future. The size of the installed telecommunications network and current growth rates will make the codec circuit one of the world's largest volume components. Consequently, a need for a comprehensive test strategy for the production testing of codecs exists.

1.2 Purpose and scope

The aim of the present work is to examine a proposed test system layout and to establish a test strategy, in order to perform transmission tests on codec integrated circuits, so as to ensure the proper performance of these components in PCM system applications. The present work encompasses the following:

(a) the computer simulation of a commercially available integrated circuit codec

(b) the study of the operation of a codec with various fabrication imperfections of both the encoder section and decoder section

(c) defining the requirements of the test system and strategy including a study of the CCITT half channel measurement specifications recently proposed

and

(d) the simulation of the proposed test system on a computer.

Pulse code modulation systems are designed to conform to one of two internationally recognized standards. These are:

(a) The European communication standard known as CEPT (the regional European Telecommunication Conference Committee)

or

(b) The North American or Bell standard

The difference between the two standards lies mainly in the type of non-linear companding law used and in the transmission rate used. The European CEPT standard uses A-law companding and a 2.048 Mb/s transmission rate. The Bell standard uses μ-law companding and a 1.544 Mb/s transmission rate. In addition the CEPT system accommodates 30 audio speech channels whereas the Bell system accommodates 24 audio speech channels. The present work deals only with the CEPT standard.

In pulse code modulation a transmit filter is provided at the input to the codec to prevent aliasing by frequency limiting the input speech signal to 4 kilohertz i.e. half the sampling frequency used. In addition the transmit filter attenuates any power line frequency components present at the input to the codec.

A receive filter is provided at the output from the codec. The receive filter is a low pass filter that reconstructs the analogue speech waveform and provides Sin x/x correction.

The physical construction of the integrated circuits implementing the transmit filter, receive filter, encoder and decoder is not well defined. If one takes a look at

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The physical construction of the integrated circuits implementing the transmit filter, receive filter, encoder and decoder is not well defined. If one takes a look at various PCM component manufacturer's [Hayes et al., 1981] one sees these four functions implemented as follows :

(a) individually on four chips

(b) both filters on one chip

(c) the encoding and decoding functions on one chip

(d) all four functions on one chip

In addition, in order to minimize crosstalk, filter/encoder and decoder/filter combinations have been designed.

In the present work, only codec integrated circuits without filtering, whether on-chip or separate, are considered. Thus circuits following the architecture of (a) and (c) above are considered.

In order to ensure that a codec integrated circuit is functioning correctly a number of tests are required [CCITT, 1976b]. Among them are the gain tracking test, which measures the ability of the codec's output levels to linearly track its input levels, the signal to quantization distortion ratio test to measure spurious signals that are added by the device, and the harmonic distortion test which checks for the presence of the second and third harmonics of an input signal. In addition, the idle channel noise test checks for a signal at the codec's output with no input signal, and the crosstalk test checks for noise on a channel with no input signal when there are signals present in the other channels. Codecs, like other integrated devices, must also be checked for their power-supply rejection ratio and frequency response.

The two most critical tests to perform on a codec are gain tracking and signal to quantization distortion. In terms of accuracy and test time required these are the two most difficult tests to make. Consequently the test system and strategy proposed in the present work deals primarily with implementing these two tests.

In order to perform gain tracking and quantization distortion tests, two input signal stimuli are recommended by the CCITT [1976b]. These are band limited pseudo random noise and single frequency sinusoidal signals. The advantages and disadvantages of the use of one stimulus over the other are not a concern of the present work and this is covered adequately by Rolls and Webster [1974]. In the present work use is made of sinusoidal input signal stimuli.

A point of concern at this stage is whether the arguments put forward by Rolls and Webster, which are based on the use of analogue signal stimuli, are valid when using digital signal stimuli in half channel test measurements. As the present work is concerned with evaluating a proposed test system and strategy, it is felt that this is of no immediate importance. However, it should be studied in future work.

1.3 Review of theory

1.3.1 Function of the codec in a pulse code modulation system

To enable voice frequencies to be converted to binary digits and the binary digits to be reconverted back to voice frequencies a device known as a codec is used. The codec comprises a non-linear analogue to digital converter in the transmit path and a non-linear digital to analogue converter in the receive path. The location of the codec in a pulse code modulation system is shown in simplified form in figure 1.

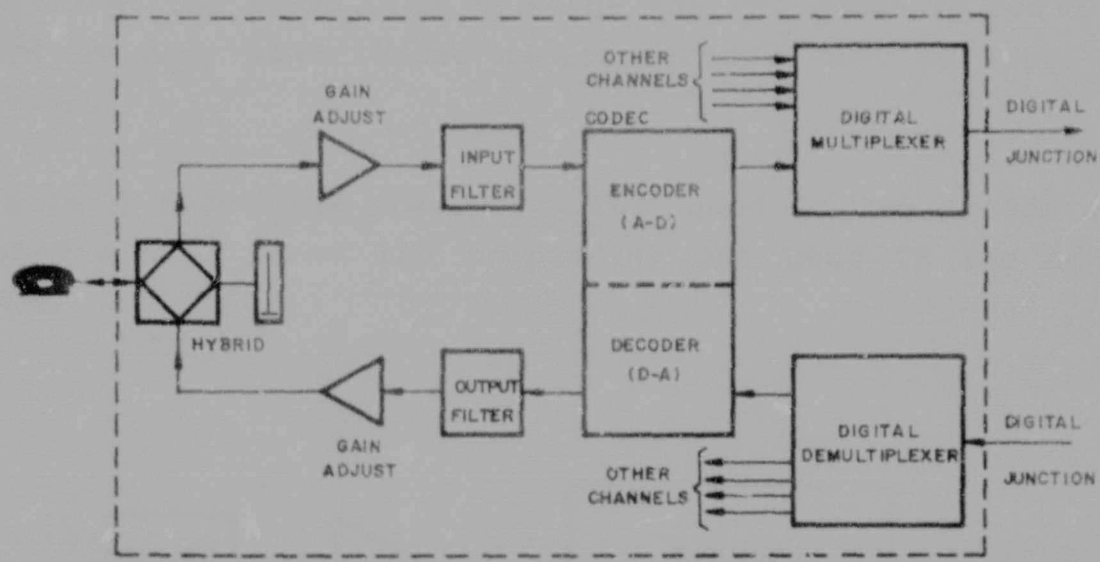

FIGURE 1 SINGLE CHANNEL CODEC IN PCM SYSTEM.

In addition to performing analogue to digital and digital to analogue conversion the codec circuit also performs sampling on the incoming speech waveform. As stated previously an input filter is provided in order to prevent aliasing as a result of the high frequency components present in the input speech signal and an output filter is provided to give Sin x/x correction.

1.3.2 Operation of a typical codec

In order to describe the operation of a typical codec, use is made of the companding digital to analogue converter manufactured by Precision Monolithics Incorporated (PMI) [Pastorino, 1982]. The companding digital to analogue converter (DAC) forms the key component in the PMI codec. The DAC is used for both encoding and decoding. The PMI codec system is particularly suitable for the present work because its internal architecture is published in data sheets in great detail. Using the PMI codec as a simulation basis for the present work does not restrict the results obtained. The principles on which the PMI codec is based are used by many other codec manufacturers, such as Intel and Mostek.

Figure 2 shows the companding DAC used in the encode mode and Figure 3 shows the companding DAC used in the decode mode.

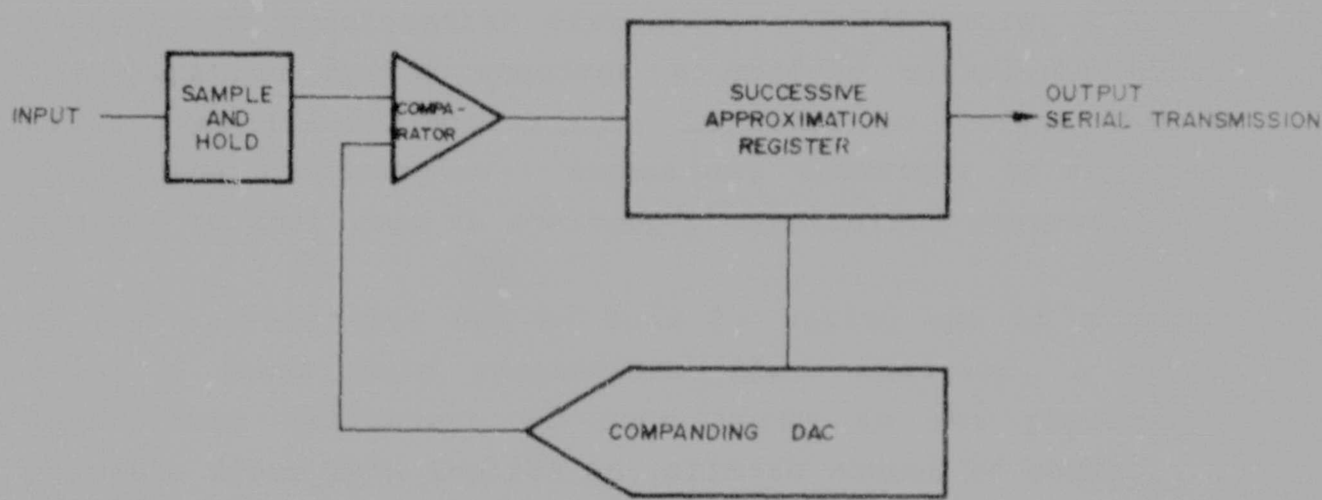

FIGURE 2 ENCODER USING COMPANDING DAC

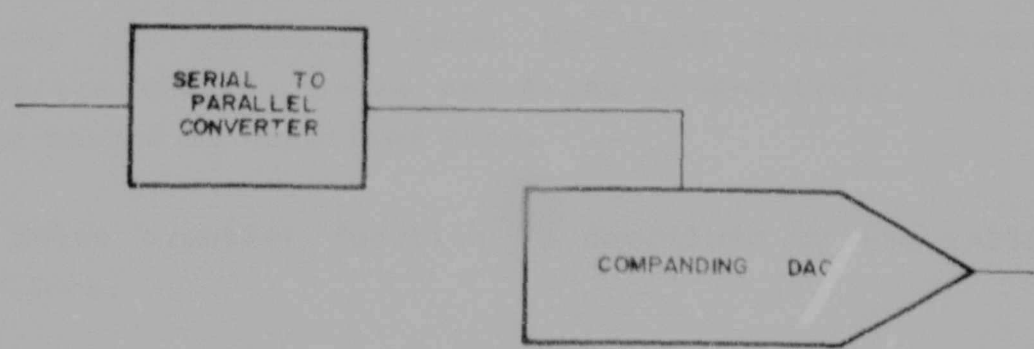

FIGURE 3 DECODER USING COMPANDING DAC

Voice signals in telephony require a system with a very wide dynamic range, typically 62 dB [Hanrahan, 1982, p.63]. If a linear analogue to digital converter was used to digitize the voice signal in a PCM system, it would require a resolution of 12 bits. This would not be satisfactory because of its excessive bandwidth requirements. With present day PCM systems a 64 kb/s data rate is required to transmit each voice channel. The use of a 12 bit linear analogue to digital converter would increase this bit rate to 96 kb/s. This would provide a better accuracy at the expense of excessive bandwidth.

Signal to noise ratio is the most important criterion in a voice communication system and in PCM noise is due almost entirely to quantization distortion. Furthermore, a voice communication system requires a uniform signal to noise ratio over its dynamic range. In order to achieve this a companding (compression - expansion) technique is required similar to that used in analogue communication systems.

In PCM systems this can be done by making use of a codec using a logarithmic compression law. However, a truly logarithmic assignment of code words is not physically possible since this implies an infinite number of codes.

Two methods for generating practical implementations of logarithmic transfer functions have been derived and these have become international standards [CCITT, 1976a]. These methods are generally known by their transfer functions which are called μ-law and A-law respectively. Only the A-law method is described here.

The A-law transfer function is described by the following equations:

$$Y = Ax/1+\log(A) \qquad \text{for} \quad 0 \le x \le 1/A \qquad (1.1)$$

$$Y = 1+\log(Ax)/1+\log(A) \qquad \text{for} \quad 1/A \le x \le 1 \qquad (1.2)$$

This transfer law gives a unique signal-to-distortion characteristic for a particular value of A. At present the CCITT specifies A to be equal to 87.6.

The wideband i.e. unfiltered signal to distortion ratio caused by an ideal quantizer over the dynamic range of a PCM system is shown in figure 4. This figure is taken from Pastorino [1982].

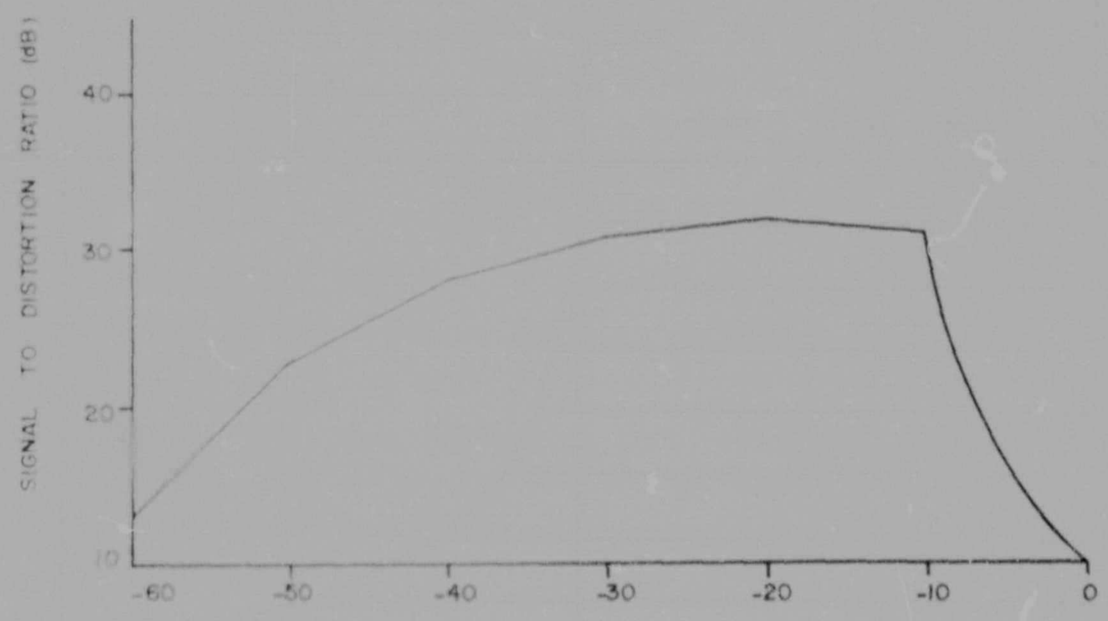

FIGURE 4 SIGNAL TO DISTORTION RATIO FOR AN IDEAL QUANTIZER.

The practical implementation of the A-law transfer function is accomplished by making use of a piece-wise linear approximation. The transfer function is implemented in chords or segments were the transfer function within any one chord is a linear staircase. The A-law transfer function for the encoder section is shown in figure 5 and figure 6 gives, in finer resolution, the transfer characteristic about the origin showing the linear staircase function within a chord.

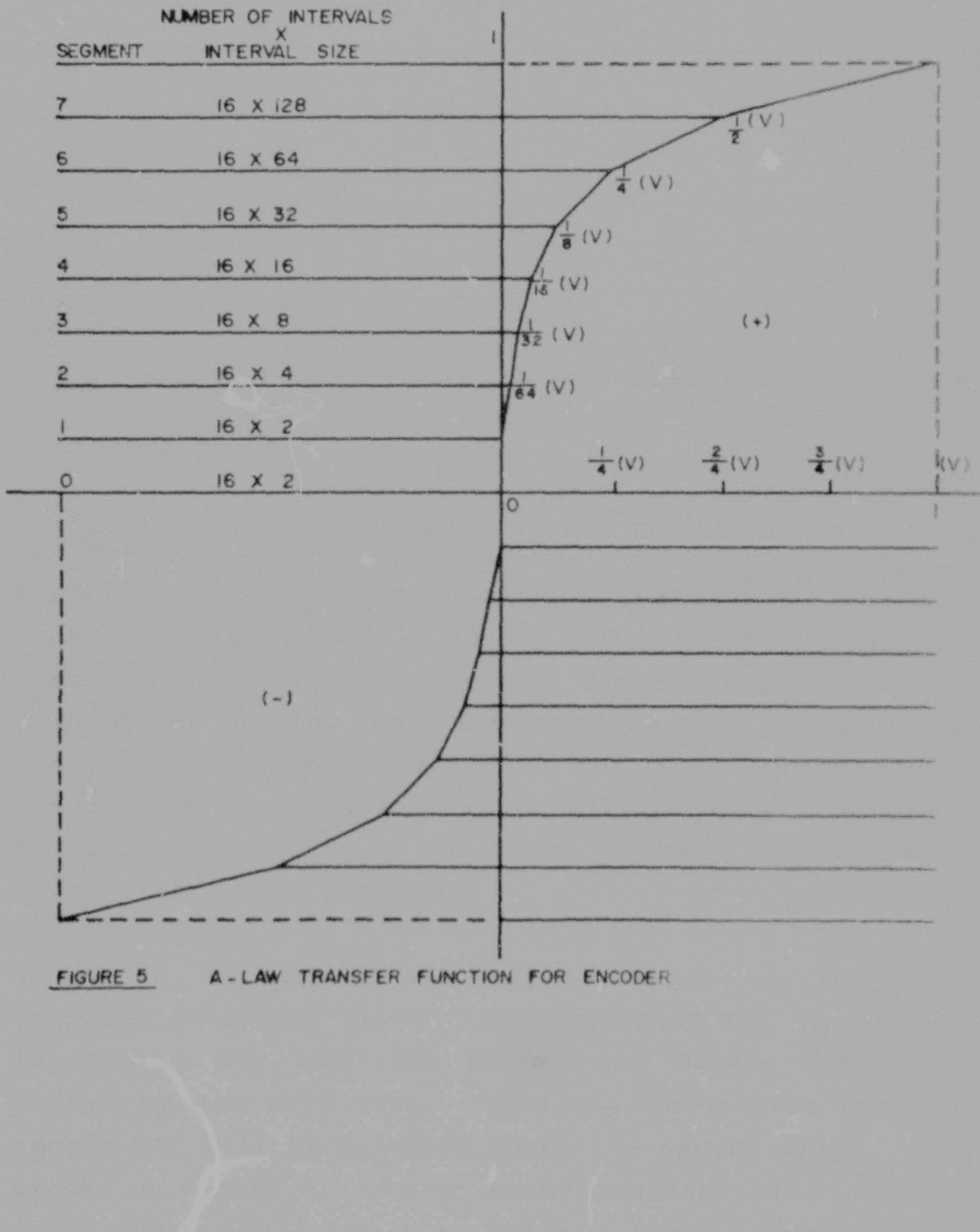

FIGURE 5 A-LAW TRANSFER FUNCTION FOR ENCODER