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A proposed test system and strategy for the production testing of pulse code modulation codecs.

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2015-02-20
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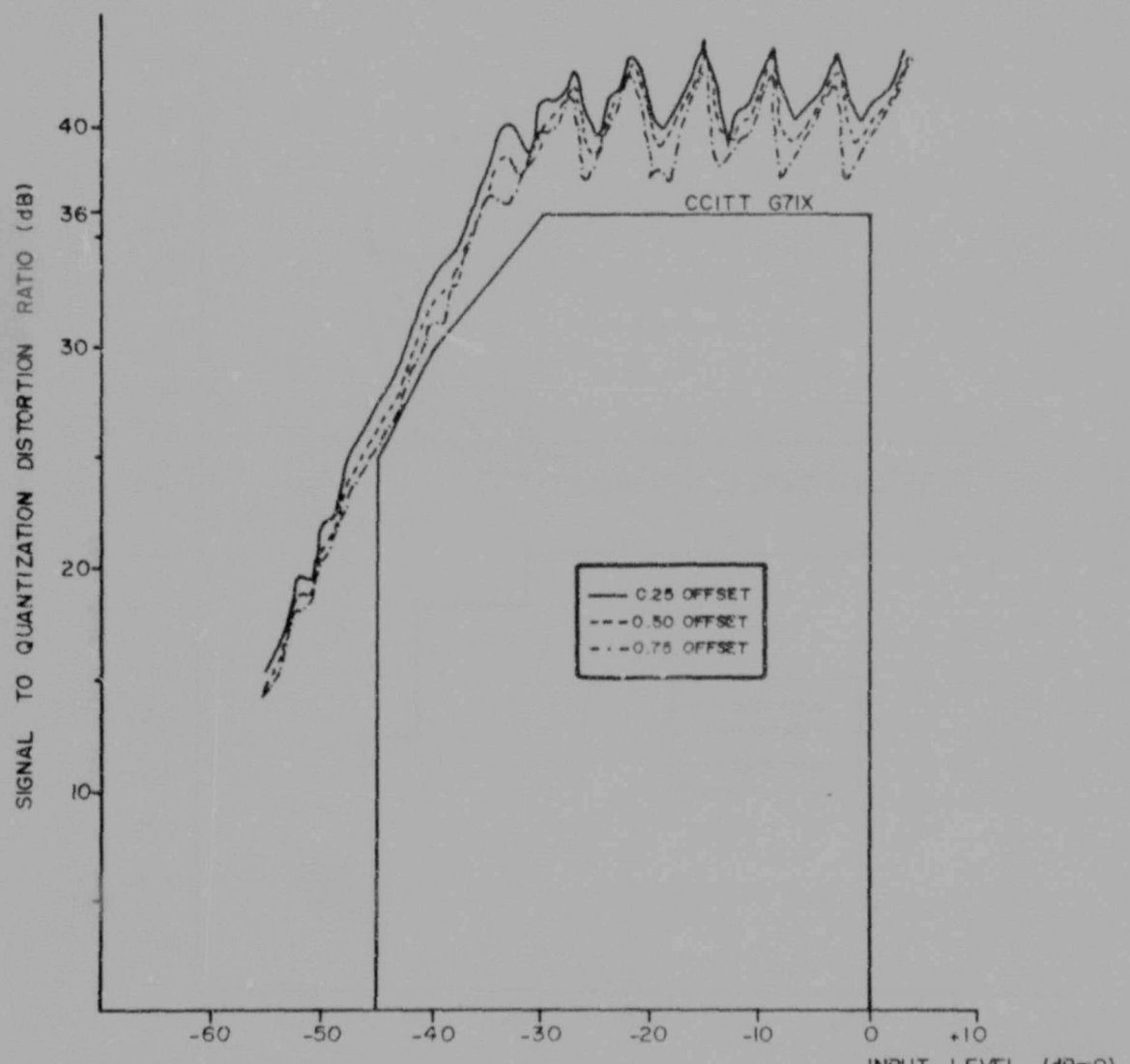

FIGURE 38 SIGNAL TO QUANTIZATION DISTORTION PERFORMANCE FOR A IMPERFECT DECODER WITH A STEP GENERATOR OFFSET ERROR.

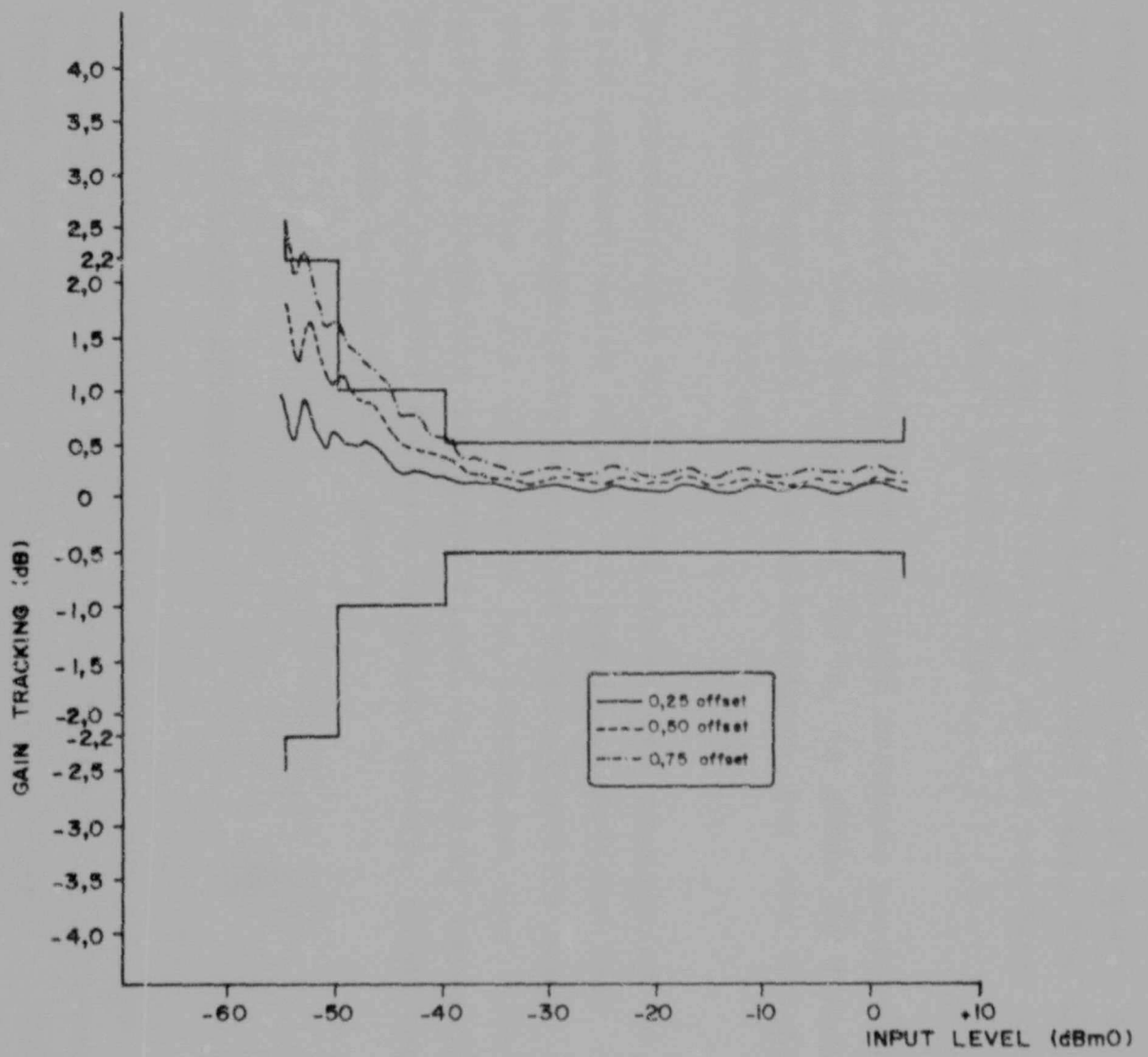

FIGURE 39 GAIN TRACKING PERFORMANCE FOR A IMPERFECT DECODER HAVING A STEP GENERATOR OFFSET ERROR

2.3.3 Step generator gain error

If the levels generated by the step generator referred to above are in error by a multiplicative factor (usually less than 1), the result will be that although all chord endpoints will remain fixed, the first 15 steps of each chord will be smaller than in the ideal case, the last one being larger than the ideal to make up for the fact that the total chord size must remain the same.

The quantization distortion and gain tracking performance of the encoder, for this type of error, is shown in figures 40 and 41 respectively. The performance for the decoder is shown in figures 42 and 43.

The imperfections illustrated above are considered to be a fair representation of some of the imperfections that can be found in commercially available codec circuits under going test. In the above discussion an assumption was made that all chord endpoints remained fixed. A logical extension to this would be chord endpoint location errors, which are not considered in the present work.

2.4 Test system parameters

As mentioned earlier in this chapter, one of the main requirements for the test strategy to meet is a compromise between measurement accuracy and test time. The two quantities are interdependent and it is the purpose here to specify, the test system parameters, input resolution and number of samples at each point, so as to achieve the best possible solution. In addition, the proposed test system makes use of the Fast Fourier Transform (FFT) for distortion analysis in the frequency domain. As a result the frequency

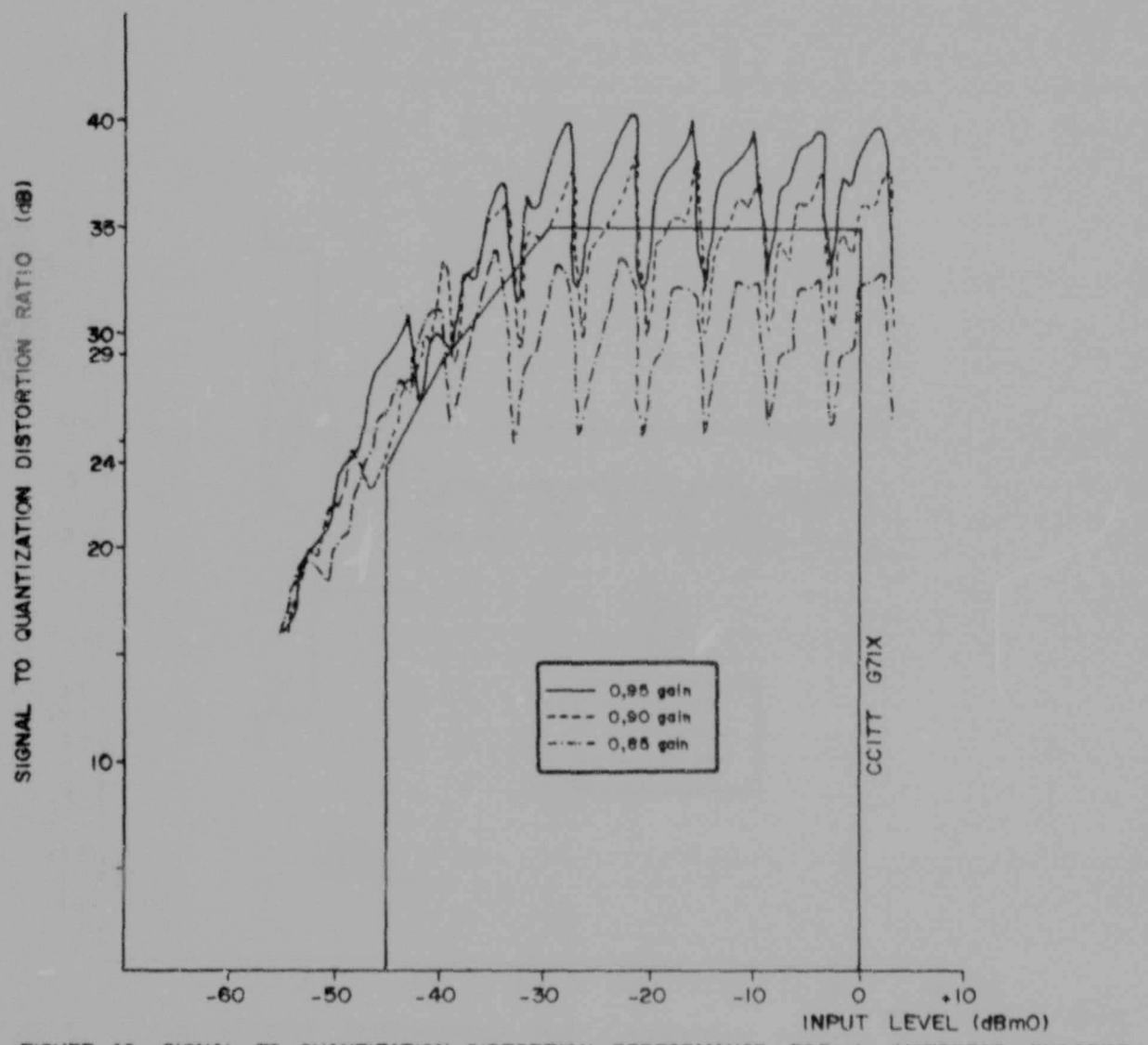

FIGURE 40 SIGNAL TO QUANTIZATION DISTORTION PERFORMANCE FOR A IMPERFECT ENCODER WITH STEP GENERATOR GAIN ERROR.

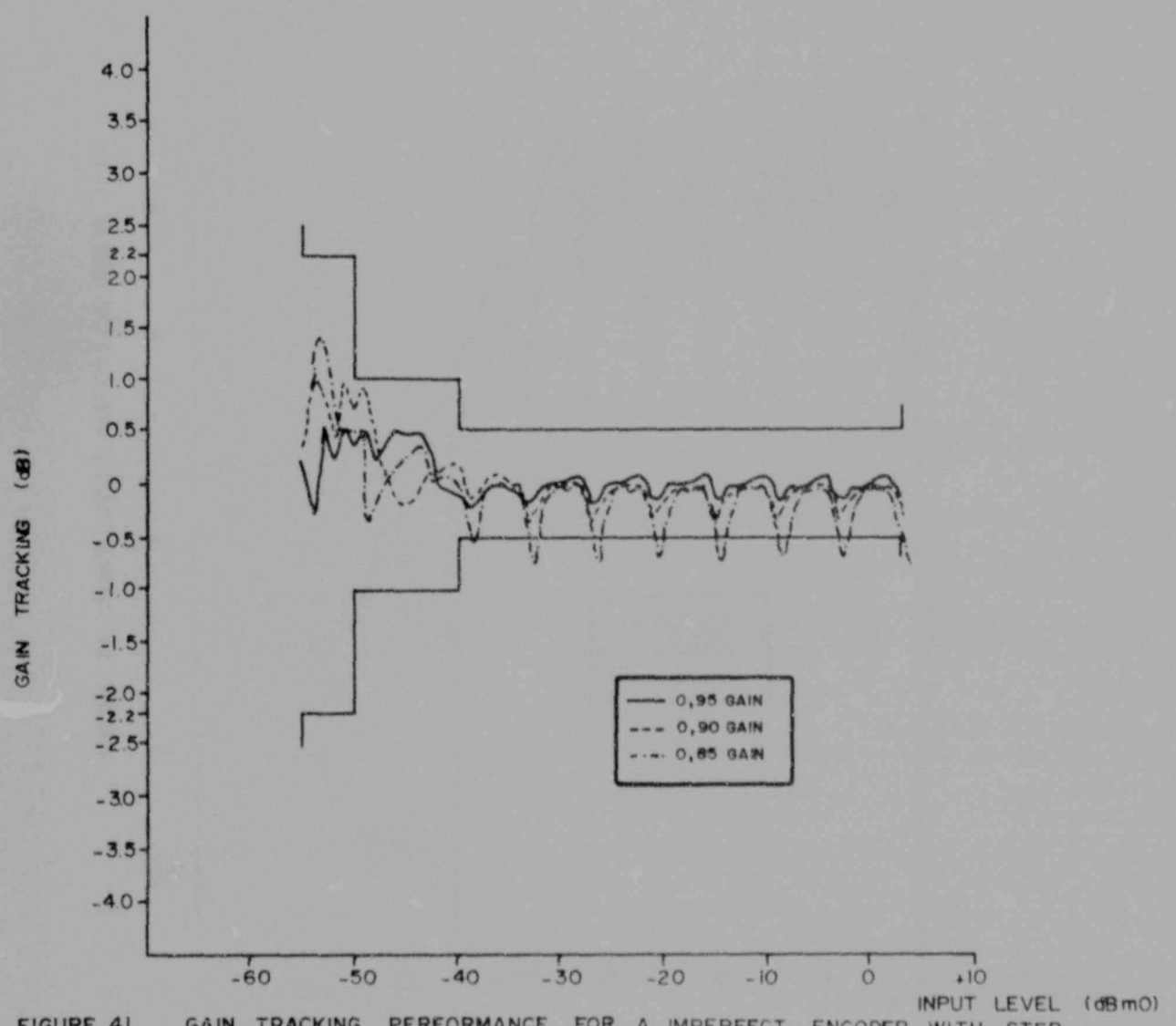

FIGURE 41 GAIN TRACKING PERFORMANCE FOR A IMPERFECT ENCODER WITH STEP GENERATOR GAIN ERROR.

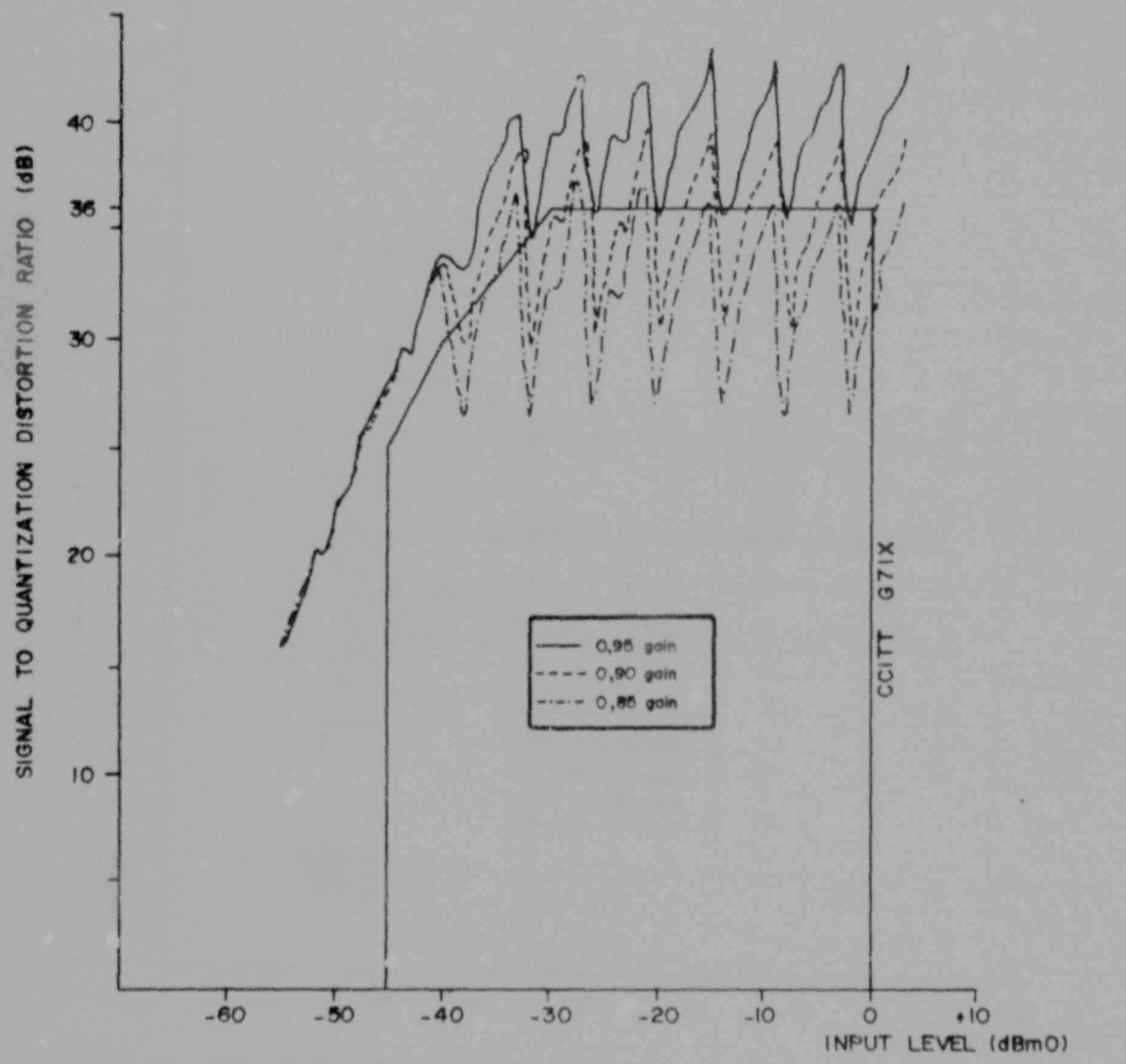

FIGURE 42 SIGNAL TO QUANTIZATION DISTORTION PERFORMANCE FOR A IMPERFECT DECODER HAVING A STEP GENERATOR GAIN ERROR

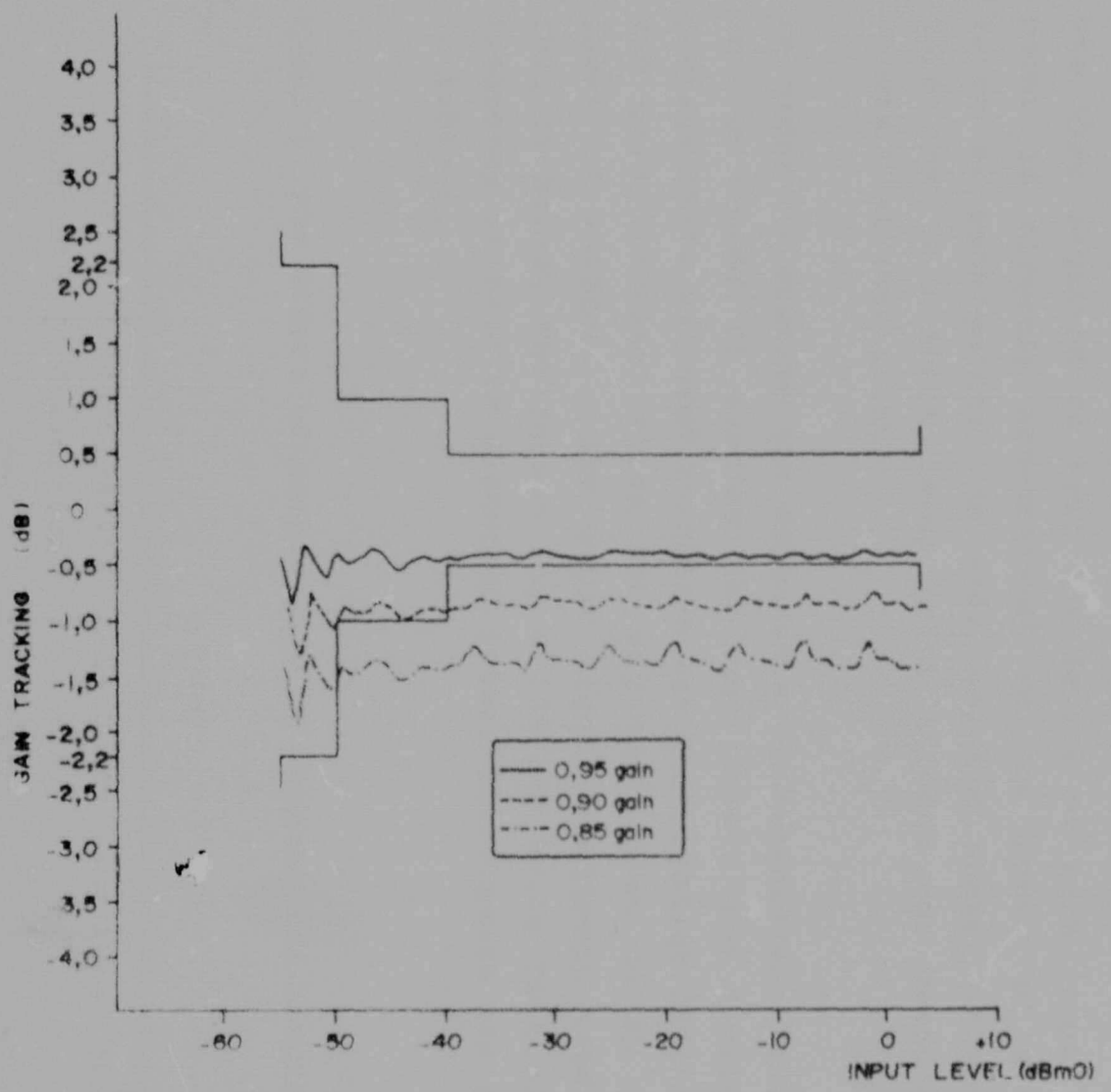

FIGURE 43 GAIN TRACKING PERFORMANCE FOR A IMPERFECT DECODER HAVING A STEP GENERATOR GAIN ERROR

of the input signal used to excite the codec under test has to be chosen with care.

2.4.1 Number of samples at each test point

The time required to test the device at each input level point is directly proportional to the number of samples taken at each point. In addition, the number of samples taken at each input level point affects the measurement accuracy at that point.

In order to examine the dependence of measurement accuracy on the number of samples per test point, the simulation package TESTS was used to obtain the quantization distortion and gain tracking values for an ideal codec at an input level of -10dBm0. The input signal frequency used was 1031.25 hertz. The reason why a 1031.25 hertz test signal frequency is used is discussed later. The results are shown in table 1.

TABLE 1 The effect of sample size on measurement accuracy

Number of samples	Quantization distortion (dB)	Gain tracking (dB)	Test time (mS)
256	42.13	-0.0025	32
128	42.17	-0.0236	16
64	41.49	-0.0243	8
32	41.24	0.0157	4

The sampling frequency used (8kHz) results in samples being taken every 125 microseconds. The test time is calculated from this. The iteration time of the FFT processor is not taken into account, owing to the fact that FFT hardware processors are available which have the capability of performing a 256 point FFT in under 1 millisecond [Rabiner and Gold, 1975].

To determine the quantization distortion and gain tracking with absolute measurement accuracy, a stochastic simulator is needed which provides the mathematical equivalent of an infinite number of samples per test point [Attridge et al., 1979]. With this as a basis, values obtained using a fewer number of samples per test point, can be compared with the absolute measurement value, so as to obtain the best value for the number of samples. This was not done in the present work. Instead, the number of samples suggested by Dell [1981] and Attridge et. al., in the literature, were examined in the light of the information contained in table 1.

Attridge et al. quote their measurement accuracies for the gain tracking measurement. For 400 samples they obtain a measurement uncertainty of 0.01 dB when compared with the absolute measurement value obtained from the stochastic simulator. For 100 samples they obtain a measurement uncertainty of 0.05 dB with a 75 per cent reduction in test time, viz. 12.5 milliseconds. From table 1, comparing the results obtained for gain tracking using a 256 sample size and a 128 sample size, we get a difference of 0.02dB. Thus, as a rough approximation, using 256 samples would improve the measurement uncertainty to 0.03 dB. The test time would now increase to 32 milliseconds.

Dell [1981] suggests using 32 samples for both quantization distortion and gain tracking. Comparing the values shown in

table 1, for quantization distortion using 256 samples and 32 samples, we obtain a difference of 0.89 dB. This value of error is too high to accept.

The decision taken is to use a sample size of 256. The resulting test time per input test point is 32 milliseconds. To reduce the overall time for the entire test we need to look at the input level resolution required.

2.4.2 Test signal input resolution and the technique of guardbanding

The input level resolution determines the number of test points that are required over the input range. Difficulty is experienced in determining this owing to the complex nature of the characteristic curves for quantization distortion and gain tracking.

Figures 24 and 26 show the characteristic curves obtained for quantization distortion of an ideal codec using an input resolution of 1 dBm0 and 0.1 dBm0 respectively. The large vertical and horizontal oscillations, shown in figure 24, are caused by the segmented structure of the A-law transfer function. As the input is increased the distortion decreases until each chord endpoint is reached and then increases abruptly to decrease again to the next chord endpoint. The reason for this is that the slope of the companding characteristic changes at each chord endpoint. The small oscillations within the larger oscillations, shown in figure 26, are due to the quantizing steps within each chord segment. An input resolution of 1 dBm0 or bigger is sufficient to track the larger oscillations. A resolution of between 0.05 dBm0 and 0.2 dBm0 is needed to track the smaller oscillations.

The oscillations on the reference curves are due also to the flat topped nature of the sine wave test signal. If one uses a test signal with a guassian distribution, the reference curve does not exhibit this type of oscillation.

The question that needs to be answered is what input resolution do we specify? The object of the test strategy is to give a "go-no go" result with regard to whether the codec under test meets the specification or not. If a coarse resolution was adopted this would not be a problem if the codec characteristic curve fell close to the reference curve obtained for a ideal codec. However, if the codec under test characteristic curve came close to the CCITT limit it is possible to obtain an erroneous "go-no go" result.

This is illustrated in figure 44

Here is shown a section of the characteristic curve obtained for quantization distortion of a decoder having a 0.75 step generator offset error. An input resolution of 1 dBmO and 0.1 dBmO was used. For 1 dBmO resolution, it is seen that the characteristic curve approaches the mask defining the CCITT limit,but does not fall below it. Thus if this resolution was used the decoder would pass the test. However, using a finer input resolution of 0.1 dBmO shows that the characteristic curve obtained falls below the CCITT limit at two points, around -43.10 dBmO and -40.20 dBmO. Thus for a coarse resolution a bad device would be proposed as a good device and this would ultimately effect the yield of the test system as stated earlier on.

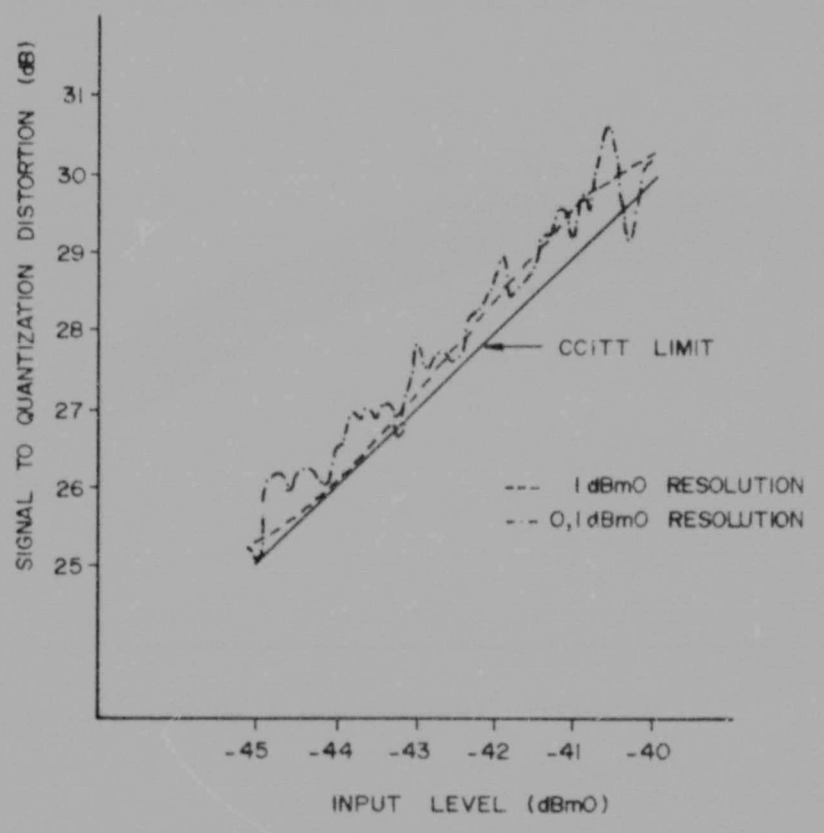

FIGURE 44 SECTION OF QUANTIZATION DISTORTION CHARACTERISTIC CURVE FOR A DECODER WITH A 0,75 STEP GENERATOR OFFSET ERROR

The complexity of the characteristic curve defining the performance of a codec under test and the constraint on the test time suggests the use of an adaptive measurement technique. This technique is used in the test strategy proposed in the present work.

The technique requires guardbanding the CCITT limit. Initially the test on the codec is conducted at a coarse resolution of 1 dBm0. Should the value obtained for quantization distortion or gain tracking fall within the defined guardband, the input changes to a finer resolution

of Ø.1 dBmO and the test is repeated around this point. This is shown schematically in figure 45.

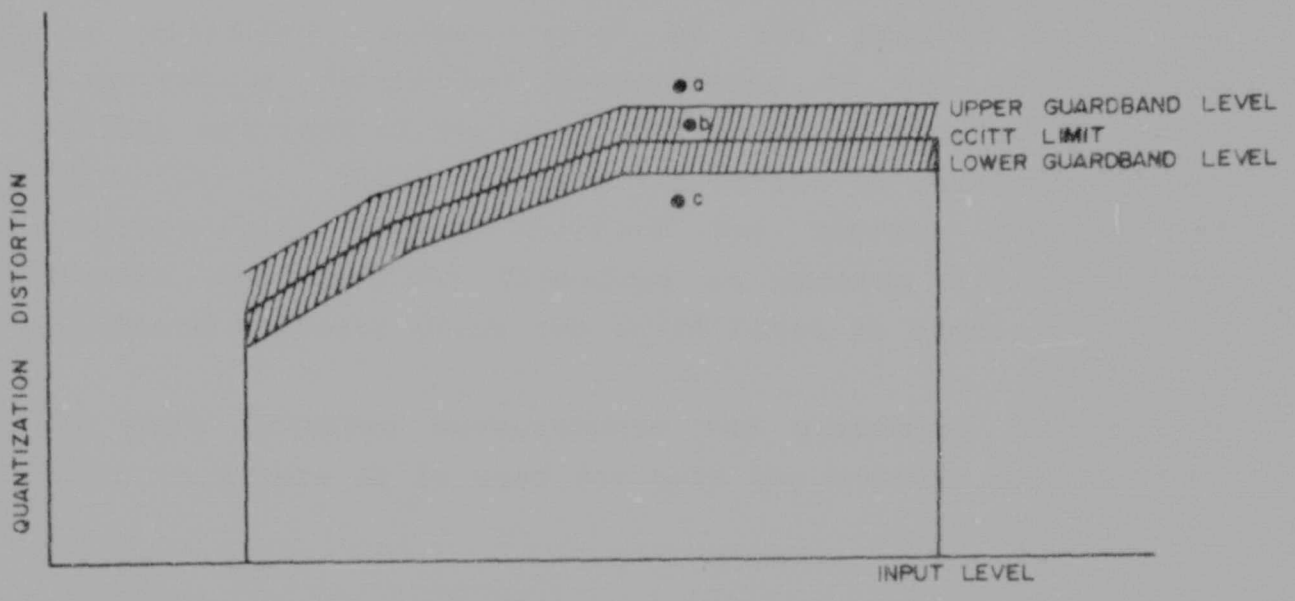

FIGURE 45 THE USE OF A GUARDBANDING TECHNIQUE.

If the value obtained for quantization distortion falls at point a, the device is passed and the test strategy moves on to the next input level point. The input resolution used is 1 dBmØ. If a value falling at point c is obtained the device fails and the test system stops. Thus no time is wasted on carrying out further tests on a device that does not meet the CCITT specification.

If the quantization distortion value obtained falls at point b, the input resolution changes to Ø.1 dBmO and the guardband is removed. The test is then repeated over a range of 1 dB around the input level point which resulted in a value at b being obtained. If the values for quantization distortion obtained over this range fall above the CCITT limit the input resolution changes back to 1 dBmO and the test procedure continues. If at any point within this range the value obtained falls below the CCITT limit

the device fails and the test system stops. The same technique is adopted for gain tracking measurements.

The guardband values used in the present work for quantization distortion measurements of the encoder and decoder sections of the codec are shown in figures 46 and 47 respectively. These values where chosen in the light of the characteristic curves obtained for various encoder and decoder imperfections discussed in section 2.3. A 1 dB guardband centered about the CCITT limit is used.

For gain tracking measurements the guardband combination shown in figure 48 is used for both the encoder and decoder.

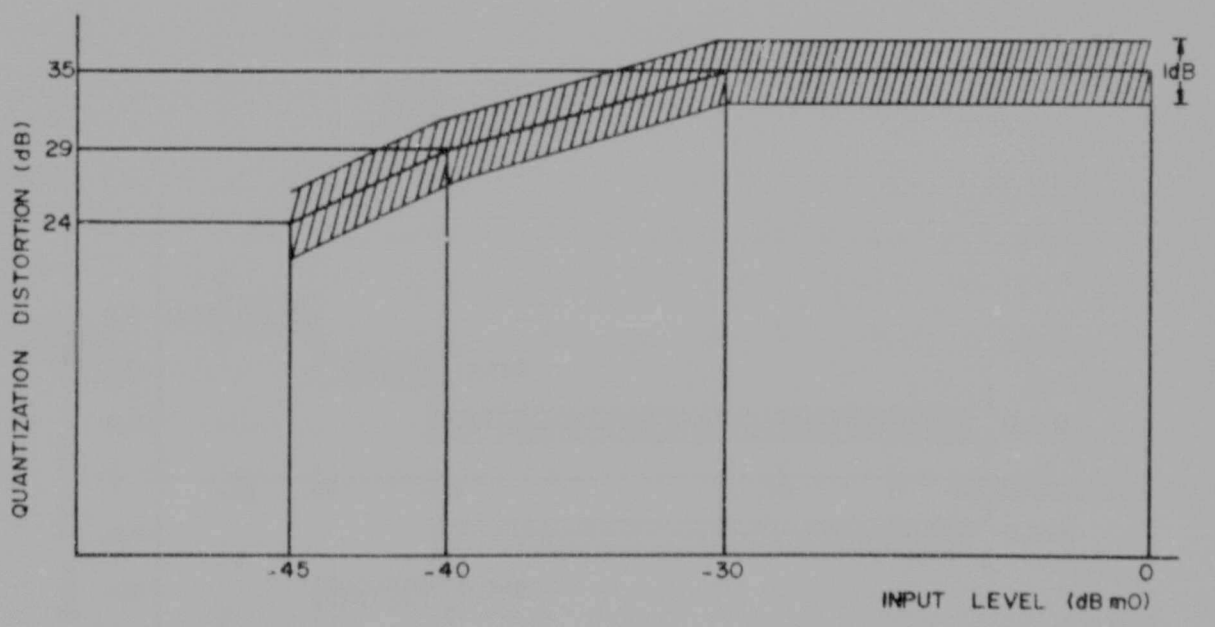

FIGURE 46 CCITT LIMIT FOR QUANTIZATION DISTORTION OF ENCODER SHOWING GUARDBAND

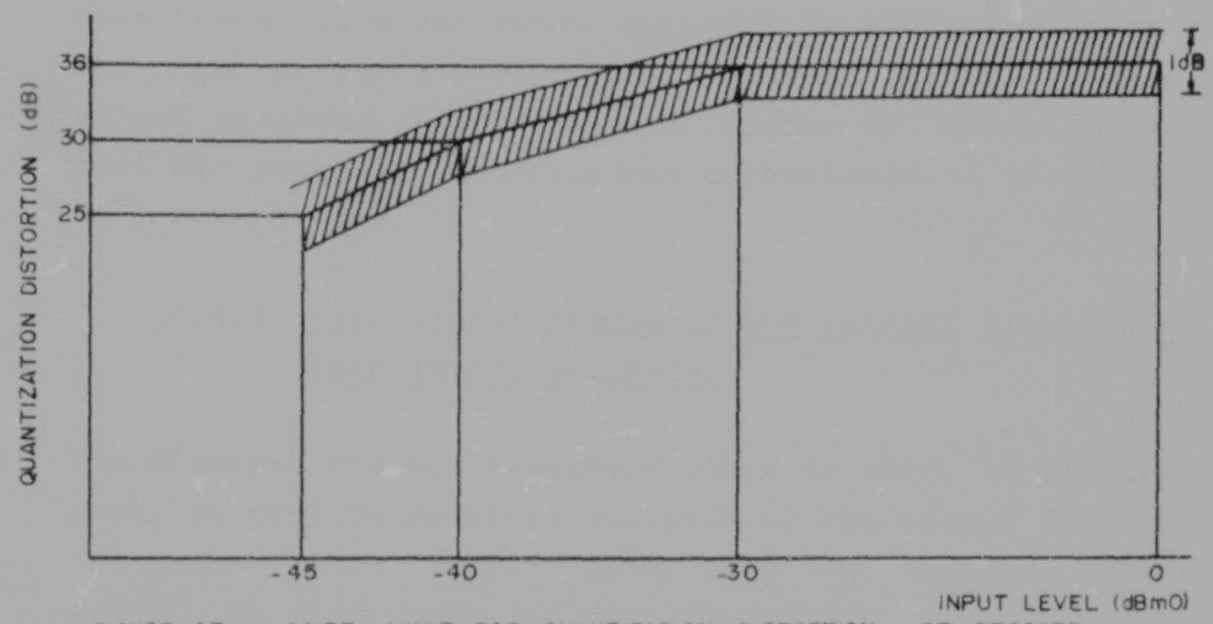

FIGURE 47 CCITT LIMIT FOR QUANTIZATION DISTORTION OF DECODER SHOWING GUARDBAND.

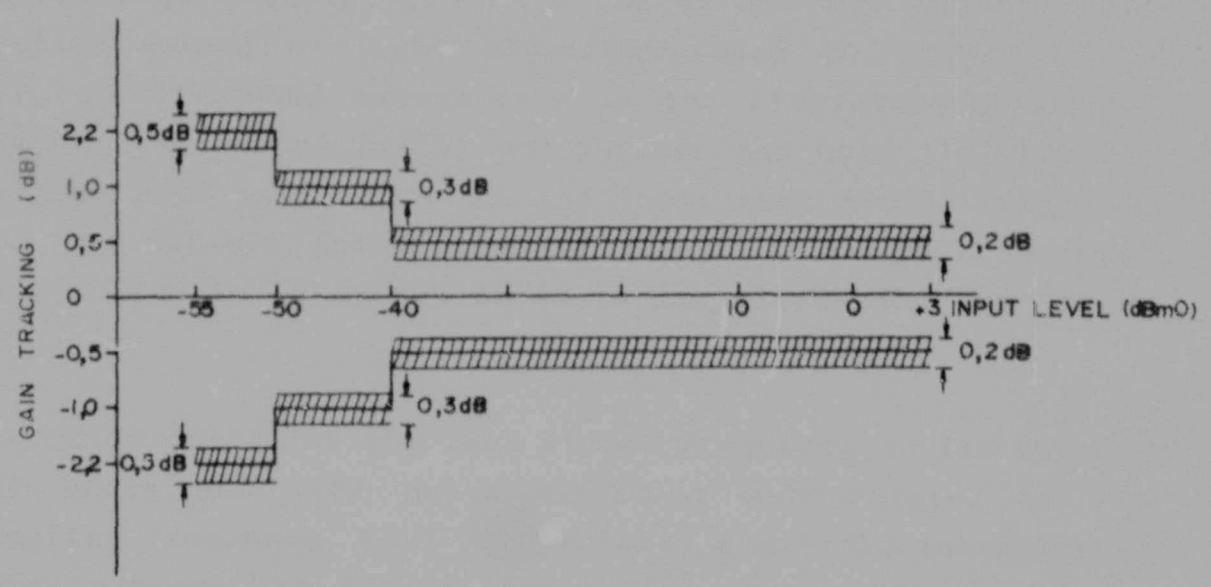

FIGURE 48 CCITT LIMIT FOR GAIN TRACKING FOR BOTH THE ENCODER AND DECODER SHOWING GUARDBAND

The guardband values for both quantization distortion and gain tracking, given above, may seem to be to stringent from the point of yield protection. However, it is felt that the values proposed give a certain degree of safety to ensure that all possible imperfections encountered in practice will be accounted for.

2.4.3 Test signal frequency and cut-off frequencies used in FFT processor

The Discrete Fourier Transform (DFT) is used, in the present work, to perform spectral analysis of the output signal from the codec under test. From this the harmonic content, caused as a result of the quantizing process, can be determined and thus the quantization distortion and gain tracking values can be calculated. The use of Fourier analysis to determine the harmonic content of a quantized signal is mentioned in section 1.3.3.

The theory, pertaining to the use of the DFT and the Fast Fourier Transform (FFT) algorithms used to implement the DFT, is discussed extensively in the literature by Bogner and Constantinides [1975] and Rabiner and Gold [1975]. The purpose here is to show how the input test signal frequency and the cut-off frequencies of the audio channel bandpass filter need to be chosen so as to facilitate the use of the DFT.

The CCITT calls for the test signal frequency to lie between 700 hertz and 1100 hertz excluding submultiples of the sampling frequency. Most commercially available instruments, which provide the single tone option, use a 850 hertz test signal [Hewlett Packard, 1979]. Dell [1981] proposes the use of a 1020 hertz sine wave signal. In addition the audio channel bandwidth, used by all telecommunication administrations, has its 3 dB cut-off frequency points at 300 and 3400 hertz.

The use of the DFT implies a strict format for the data supplied for conversion. It requires exactly 2^n samples and an exact number of complete cycles, as it assumes that the data can be extended to an infinite, continuous sequence. In other words, the data must be interpolated so that it starts and ends on a zero crossing and fits in exactly the appropriate number of samples.

Let us consider the DFT operating on a sinusoidal signal of frequency $f_\emptyset$.

The signal is sampled at a sampling frequency f_s such that

$$f_s \geq 2f_\emptyset \quad \text{(Sampling theorem)} \qquad (2.1)$$

and

$$T_s = 1/f_s \qquad (2.2)$$

The total number of samples taken is given by

$$N = 2^n \qquad (2.3)$$

where n is an integer value

The period of the time window over which the DFT operates is given by

$$T = N\,T_s \qquad (2.4)$$

and the incremental frequency steps given by the DFT in the frequency domain are given by

$$\Delta f = 1/T \qquad (2.5)$$

therefore

$$\Delta f = f_s/N \qquad (2.6)$$

Taking the sampling frequency as 8000 hertz and the number of samples as determined earlier, to be 256, we obtain a frequency increment of

$$\Delta f = 8000 \text{ Hz}/256 = 31.25 \text{ Hz} \qquad (2.7)$$

If the sinusoid has an integral number of periods in the time window, its Fourier transform will lie exactly on one of the lines in the frequency domain which we are capable of observing. In order to accomplish this we need to choose $f_\emptyset$ such that

$$f_\emptyset = m \, \Delta f \qquad (2.8)$$

where m is an integer value

This is shown graphically in figure 49.

$$\Delta f = 1/T \qquad (2.5)$$

therefore

$$\Delta f = f_s/N \qquad (2.6)$$

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$$\Delta f = 8000 \text{ Hz}/256 = 31.25 \text{ Hz} \qquad (2.7)$$

If the sinusoid has an integral number of periods in the time window, its Fourier transform will lie exactly on one of the lines in the frequency domain which we are capable of observing. In order to accomplish this we need to choose f_0 such that

$$f_0 = m \Delta f \qquad (2.8)$$

where m is an integer value

This is shown graphically in figure 49.

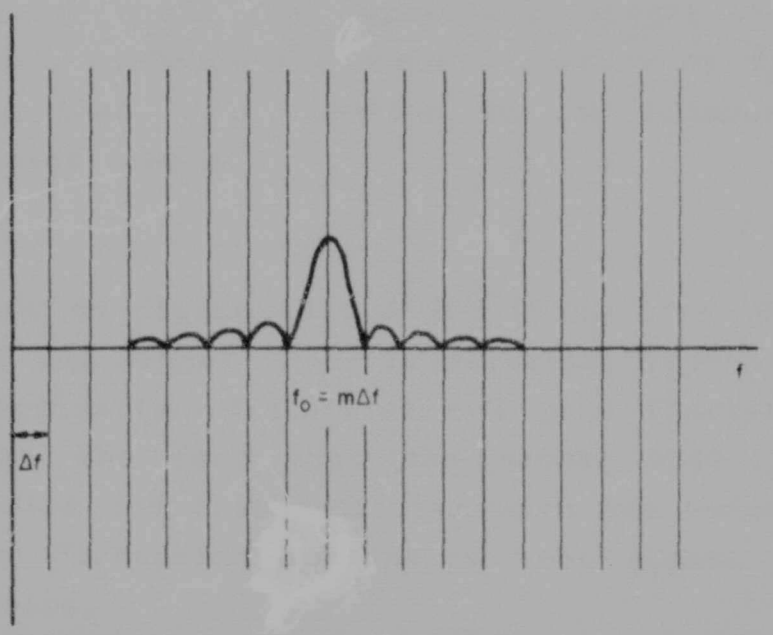

FIGURE 49 FREQUENCY DOMAIN REPRESENTATION OF SINUSOID WITH m AN INTEGER VALUE.

If the sinusoid does not have an integral number of periods in the time window, the equation $f_\emptyset = m\ \Delta f$ is still valid, however m is no longer allowed to be an integer. The frequency domain representation for this condition is shown in Figure 50.

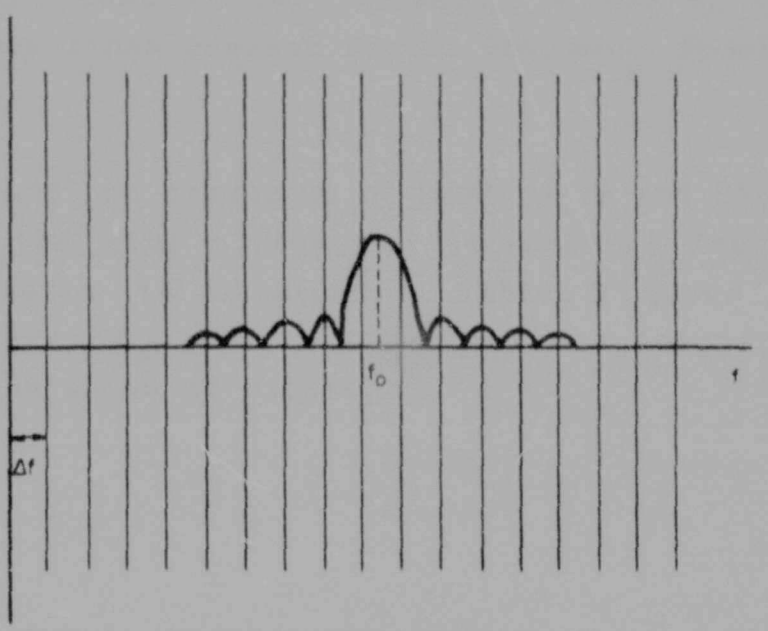

FIGURE 50 FREQUENCY DOMAIN REPRESENTATION OF SINUSOID WITH m NOT AN INTEGER VALUE

The DFT of a sinusoid which does not have an integral number of periods in the time domain will appear at more than one frequency, although the actual location of $f_{\emptyset}$ is uncertain by less than Δf. Leakage to the adjacent incremental frequencies occurs.

Referring to figure 49 and 50, if $f_{\emptyset}$ lies between $m\Delta f$ and $(m+1)\Delta f$ the signal value is affected. This in turn effects the quantization distortion and gain tracking measurement owing to the fact that the signal will leak to other frequencies and this conflicts with the assumption that it is only distortion components that appear at the other frequencies.

One method that can be used to partially eliminate this problem is to make use of a Hanning window. Here the leakage effects can be reduced by obtaining accurate amplitude information at the expense of less precise frequency resolution. This method was not adopted in the work reported here. Instead, the frequency of the sinusoidal signal used to excite the codec under test was chosen so that it fell directly on one of the incremental frequency lines present in the frequency domain generated by the DFT.

For a sampling frequency of 8000 hertz and 256 samples taken within the time window, an incremental frequency, Δf, of 31.25 hertz is obtained. Table 2 gives the frequency components, $f_{\emptyset}$, obtained for various integer values of m, using the above Δf values.

TABLE 2 Frequency component values obtained for various integer values of m

m	$f_{\emptyset}$ (Hz)
8	250.00
9	281.25
10	312.50
32	1000.00
33	1031.25
34	1062.50
35	1093.75
36	1125.00
108	3375.00
109	3406.25
110	3437.50

To choose the test signal frequency to be used in the proposed test system the CCITT constraints must be met, viz that the frequency must lie between 700 hertz and 1100 hertz and that no submultiples of the sampling frequency must be used. The latter constraint excludes the use of 1000 hertz obtained with a m value of 32. From the literature, it is apparent that the present trend is to use a frequency of 1020 hertz as shown in Dell [1981]. In the present work a test signal frequency of 1031.25 hertz was chosen.

As stated earlier, the audio channel band extends from 300 hertz to 3400 hertz, giving a bandwidth of 3100 hertz. In the proposed test system, frequency components as close as possible to the above values where chosen, viz. 312.50 hertz and 3406.25 hertz, giving a bandwidth of 3093.75 hertz. The discrepancy between the ideal bandwidth and the bandwidth

used in the test system causes a negligible error.

In the proposed test system all signal generation and signal processing is done in software. This is discussed later. Thus, using a accurate test signal frequency of 1031.25 hertz and cut-off frequencies of 312.50 hertz and 3406.25 hertz offers no problem from the point of circuit implementation.

2.5 Summary

In this chapter the two main criteria for a test system, viz. measurement accuracy and test time, are identified. It was found that the two criteria are conflicting and that the system parameters need to be chosen so as to obtain a compromise.

A sample length of 256 samples was found sufficient in order to give an acceptable measurement accuracy. In order to reduce time wastage due to lengthy test times, a 1 dBm0 and 0.1 dBm0 input signal resolution is specified. In order to prevent loss of yield the former resolution, 1 dBm0, is used together with a guardbanding technique.

The system test signal frequency, 1031.25 hertz, and the audio cut-off frequencies, 312.50 hertz and 3406.25 hertz, are specified taking into account the use of the DFT for signal processing.

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Author Debbo G A
Name of thesis A proposed test system and strategy for the production testing of pulse code modulation codecs 1984

PUBLISHER:
University of the Witwatersrand, Johannesburg
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