DELTA MODULATION TECHNIQUES FOR LOW BIT-RATE DIGITAL SPEECH ENCODING

by

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"Nature, as we often say, makes nothing in vain, and man is the only animal whom she has endowed with the gift of speech. And whereas mere voice is but an indication of pleasure or pain, and is therefore found in other animals, the power of speech is intended to set forth the expedient and inexpedient, and therefore likewise the just and the unjust. And it is a characteristic of man that he alone has any sense of good and evil, of just and unjust, and the like, and the association of living beings who have this sense makes a family and a state."

- Aristotle, Politics.

CONTENTS

ABSTRACT

Two new hybrid companding delta modulators for speech encoding are presented here. These modulators differ from the Hybrid Companding Delta Modulator (HCDM) proposed by Un et al[17,24] in that the two new encoders employ Song Voice Adaptation as the basis of the instantaneous compandor, rather than Constant Factor adaptation. A detailed analysis of the performance, both objective and subjective, of these hybrid codecs has been carried out. Results show that overall the two codecs developed as part of this project are better than the HCDM codec. In addition the new codecs offer simpler implementation in digital hardware than the HCDM. A Computer Aided Test (CAT) system has been developed to simplify the design and test processes for speech codecs.

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NOMENCLATURE

ACF	:	Autocorrelation Function.
A/D	:	Analogue-to-Digital.
ADC	:	Analogue-to-Digital Converter.
ADM	:	Adaptive Delta Modulation.
APC	:	Adaptive Predictive Coding.
ATC	:	Adaptive Transform Coding.
BER (or ber)	:	Bit Error Rate.
b_n	:	Error signal - the difference between the input and decoded signals (usually quantized to two levels).
CFDM	:	Constant Factor Delta Modulation.
CODEC	:	A word derived from the combination of CODER and DECODER.
COMPAND	:	A combination of the words COMPRESS and EXPAND.
CVSD	:	Continuously Variable Slope Delta Modulation.
C(n)	:	The autocorrelation function.
D/A	:	Digital-to-Analogue.
DAC	:	Digital-to-Analogue Converter.
DCT	:	Discrete Cosine Transform.
DFT	:	Discrete Fourier Transform.
DM	:	Delta Modulation.
DPCM	:	Differential Pulse Code Modulation.
$E(Y_k - X_k)$	:	Mean square of the difference between Y_k and X_k

(cf. $\langle e^2 \rangle_{av}$).

$E\{ Y_k \mid X_{k-1}, X_{k-2}, b_{k-1}, b_{k-2} \}$: Expected, or estimated, value of Y_k, based on $X_{k-1}, \ldots, b_{k-2}$.

$\langle e^2 \rangle_{av}$: The average of e^2 over all sampling instances.

f_c : Filter cut-off frequency.

FFT : Fast Fourier Transform.

FIR : Finite Impulse Response.

F(n) : This is the value of the continuous function F. It is a function of the independent variable 'n'.

f_s : Sampling frequency.

g(i)T : This is equivalent to g_i (cf. V_n) - used to represent the discretely sampled variable g(i) at the i^{th} sampling instant. . The sampling frequency is 1/T.

HCDM : Hybrid Companding Delta Modulation.

I/O : Input/Output.

kbps : kilobits per second.

LAN : Local Area Network.

LDM : Linear Delta Modulation.

LPC : Linear Predictive Coding.

LSB : Least Significant Bit.

MSB : Most Significant Bit.

MSHCDM : Modified Song Hybrid Companding Delta Modulation.

PAD : Packet Assembler/Dissassembler.

NOMENCLATURE

PCM	:	Pulse Code Modulation.
pdf	:	Probability density function.
PSD	:	Power Spectral Density.
RMS	:	Root Mean Square.
SBC	:	Sub-Band Coding.
SHCDM	:	Song Hybrid Companding Delta Modulation.
SNG	:	Signal-to-Granular Noise Ratio.
SNO	:	Signal-to-Overload Noise Ratio.
SNR	:	Signal-to-Noise Ratio.
SNRSEG	:	Segmental Signal-to-Noise Ratio.
SQNR	:	Signal-to-Quantization Noise Ratio.
SVADM	:	Song Voice Adaptive Delta Modulation.
TASI	:	Time Assignment Speech Interpolation.
V_n	:	This is used to indicate the value of the discretely sampled variable V at the n^{th} sampling instant.
VOCODER	:	A combination of the words VOICE and CODER.
X_n	:	Decoded signal.
Y_n	:	Input signal.
ρ	:	Correlation factor (cf. C(n)).
σ	:	Standard deviation.

CHAPTER 1

INTRODUCTION

The trend in the communications field is to integrate the voice and data services. This has prompted research into developing techniques for achieving low bit rate digital voice communication of an acceptable quality. Low cost digital communication networks have become practically feasible due to advances in the technology for manufacturing integrated circuits. The continued improvement in throughput rates (ie. processing speeds), via improved VLSI (Very Large Scale Integration) techniques and architectures, have enabled complex signal processing algorithms to be realised as high-speed single-chip modules.

Analogue circuits, which are subject to noise and temperature drift problems, can be replaced by digital signal processing methods which are inherently less sensitive to these factors. When considered against the background of increasing research into ways and means of reducing the channel bandwidth (for signal transmission) the application of these digital processing techniques becomes even more significant.

Whereas previously the methods for achieving satisfactory quality using low bit-rate digital voice communications would have been extremely cumbersome to implement, the new generation of digital integrated circuits offer an attractive means to solving this problem.

CHAPTER 1 : INTRODUCTION

One of the most important parameters affecting the efficiency of the processing algorithm is the signal type to be transmitted. For the purposes of this research the message ensemble was assumed to contain only human speech sounds. Although in terms of overall network performance, non-speech signals (such as voice-band data signals) may be significant in their effect on the network, only speech signals were considered when designing for efficient communication. The difficulty, in terms of achieving maximum transmission efficiency, is to design a system that transmits only the perceptually significant information (speech signals contain redundant information which is not readily perceived by the average listener). Two means of achieving this would appear to be, firstly, the incorporation into the transmission system of as many as possible of the characteristics of the speech signal, or secondly, the discarding of that information which is not perceptually relevant. The first method is based largely on the characterization of the sound source while the second stems from the properties of the sound perception mechanism[5].

The nature of the speech production source (ie. the human speaker) makes speech unique in terms of its characteristics. The singular nature of speech has allowed various other factors besides the production and perception characteristics to be used as design parameters. Successful digital speech communication could be

accomplished by means of the analysis of both the physical aspects of the production/perception processes and the semantic/psychological factors.

Initial work involved in digitizing speech communication channels centred around implementing PCM (Pulse Code Modulation) systems. Despite the ever-improving performance levels of alternative systems (such as delta modulation) the large capital investment in PCM has precluded the use of these alternatives. However, the local area networks (LAN's), or local subscriber networks offer significant scope for implementing alternative schemes which can be designed to be more signal-specific and more economical in terms of bandwidth utilisation than PCM.

The use of digital speech encoding techniques means that a single LAN channel can serve both the data and speech requirements of the user. However, in offering the user the speech communication facility it is envisaged that a certain reduction in the subjective speech quality is to be expected. The main object of this research has been to develop a digital speech encoder capable of operating at relatively low information transmission rates (baud rates) while at the same time providing acceptable levels of speech quality.

CHAPTER 1 : INTRODUCTION

In this work a digital encoder is presented which has been found to offer an improvement in speech quality when compared with previously developed techniques[42]. The significance of the encoder design lies not only in the improved performance, but also in the fact that its design is simpler than alternative systems. The latter point is particularly important in that the eventual aim of the research project is to produce a low-cost digital speech codec that can be used in a commercial communications network.

A simulation system has been developed to enable the performance evaluation and optimization of the various speech codecs under consideration to be performed by means of computer modelling techniques. In other words, the hardware designs have been converted to software programs for the purposes of evaluation and optimization.

CHAPTER 2

DIGITAL SPEECH CODING

There are two major areas affecting the design of a digital speech encoder. These are firstly, the statistical properties of the speech signal and, secondly, the characteristics of the transmission system (ie. the communications network). The former is going to have the bulk of the effect on the codec design, but the latter cannot be totally ignored and classified as a network design issue.

2.1 Properties of Speech.

One of our main objectives is, quite simply, to reduce the information transfer rate required for encoded speech data. The characteristics of the speech production and perception processes enable this to be done.

- The first, and perhaps the most obvious, characteristic is the spasmodic nature of the production process - we tend to speak in bursts rather than continuously. Not only are there significant pauses between sentences and phrases, but also between words and even syllables. Characteristically interactive conversational speech contains silences that accomodate over 50% of the elapsed time of the conversation[1]. Hence, a significant band-width saving could be realised if these silences were not transmitted.

- Speech signals contain significant redundancies[4,6]. This is

partly because of the physical mechanism of the vocal tract and partly due to the structure of language. Removal of the redundant information from the signal would appear to be one of the methods for reducing the information rate for transmitted speech. However, reduction of the redundancies also leads to a reduction of the speaker recognition capabilities and a general reduction of the audio quality[7] as it is these apparently useless redundancies which provide the listener with information regarding the speaker's identity, manner and personality. In essence a voice message can be split into two distinct elements - firstly, the content of the message (ie. what was said) and secondly, the peripheral information (ie. who said it, how they said it, etc.). The content of the message, or the intelligence, appears largely in the spectral envelope of the signal[4]. Hence, although the redundant information can be removed without destroying the message content, its removal at the encoder only becomes satisfactory if, at the decoder, it can be recovered from the other sections of the signal.

- The perception of the speech signal is initiated by the human ear in effect performing a crude Fourier transform on the incoming acoustic signal[5]. The presence of high energy signals in the lower regions of the spectrum tend to reduce the sensitivity of the ear to the frequency components in the upper

reaches of the spectrum and hence, noise or distortion added to the signal will not be detected if it is below some threshold[33]. This phenomenon is known as "auditory masking" and it is this that makes noise more perceptable in the presence of high frequency signals than low frequency signals.

o The addition of noise or distortion to a speech signal (by an encoder) is not necessarily disastrous. This is due to the ability of the human brain to extract the pertinent information from the received signal. Owing to the stored vocabulary and its semantic (ie. meaning-related) faculty the brain is able to interpolate the information missing from the speech signal[37].

This last characteristic is however not sufficiently well understood to be exploited in reducing the bandwidth of speech transmissions. It is rather the first three properties which are used in increasing the speech transmission channel efficiency.

The nature of the problem was stated rather succinctly by Flanagan[5] :

"Despite the equivocal aspects surrounding estimates of human channel capacity and speech information rates, it is clear that a mismatch exists between the capacity of the conventional voice channel and the information rate of the

source feeding it. One approach toward improving the match is to incorporate into the transmission system as many as possible of the constraints characterising the production and perception of speech. This information, built into the communication link, is information that need not be transmitted ...The nature of the incorporated constraints therefore influences the form of the coding for the speech information."

2.1.1 A Model of the Speech Source.

There are basically three types of sounds generated by the vocal system, namely "Voiced" sounds, "Fricatives" (or "Unvoiced" sounds) and "Stops"[4].

- Voiced sounds : these are produced by a nearly periodic series of pulses generated by the vocal chords which control the flow of air from the lungs to the pharynx.
- Fricatives : more commonly referred to as "unvoiced" sounds, these are the result of air turbulence at constrictions in the vocal tract and give rise to noise-like sounds.
- Stops : these are essentially an acoustical transient resulting from an abrupt release of pressure from behind an occlusion. An example of this would be the 't' in 'tea' where the tongue is used to block the flow

of air resulting in a pressure build-up which is subsequently abruptly released.

Because stops can be categorized as either voiced ('b' as in 'boat') or unvoiced ('p' as in 'purse') it is common to differentiate between two sound sources (ie. voiced or unvoiced) rather than three (cf. fig. 2.1).

From the model shown in fig. 2.1 the sound source excitation is either noise-like (for unvoiced segments) or pulse-like (for voiced segments). This source signal is then passed through a time-varying acoustic filter (the vocal tract) to produce the

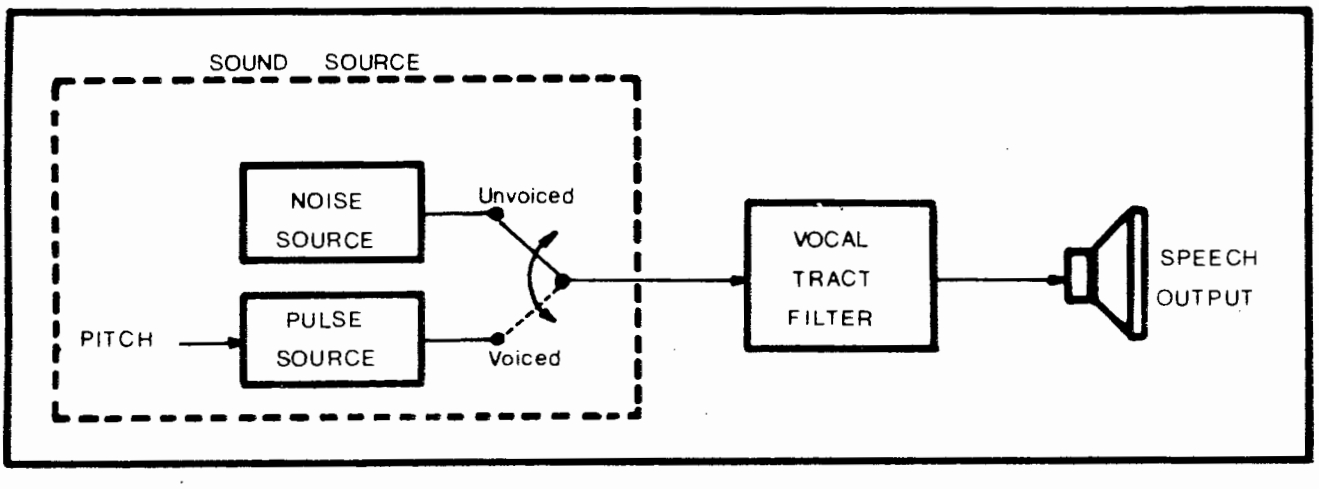


Fig. 2.1 : Model of the Speech Source.

speech output. This model is a first-order approximation that assumes that the two sound sources are linearly seperable, and ignores the possible interaction of the sound source with the

vocal tract and the modulating effect of the nasal cavity, which is a second-order effect[5].

This linear first-order model is going to limit the performance of the coding schemes based on it. However, its main advantage is its relative simplicity which lends itself to widespread use (refer to the sections on time-domain encoders).

2.1.2 Characteristics of Speech Signals.

2.1.2.1 Signal Pass-Band.

In order to digitize the speech signal it is necessary to band-limit it to enable time-sampling at a practical rate. For communications systems the signals are typically bandlimited between 300 and 3300 Hz. The Nyquist condition states that the minimum sampling rate must be twice the highest frequency in the signal pass-band. If the sampling frequency is below this rate then aliasing errors are introduced.

2.1.2.2 Signal Stationarity.

A rather obvious feature of speech is the difference between two speakers' voices - the spectral content of speech signals varies from speaker to speaker (fig. 2.2). The vocal tract has a number of resonant frequencies which tend to accentuate bands of frequencies centred about these resonances, known as

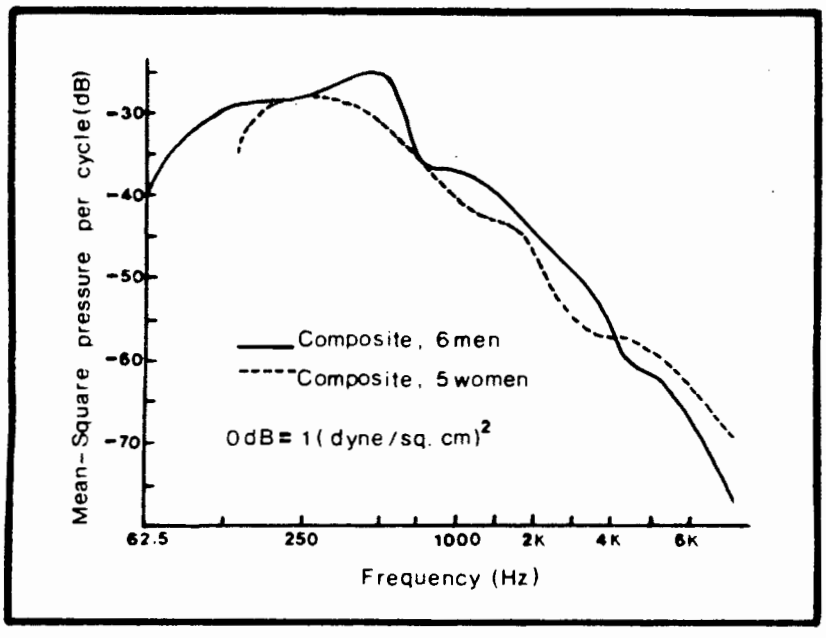


Fig. 2.2 : Spectra for different voices.

"formants"[2]. Typically, there are four formants in the speech spectrum[3]. However, not only do these formants vary from speaker to speaker, but also with time. This is in fact true for the entire spectral image of speech and not only the formants. This gives rise to the non-stationary characteristics of speech - both the amplitude and the harmonic content of the signal are changing with time.

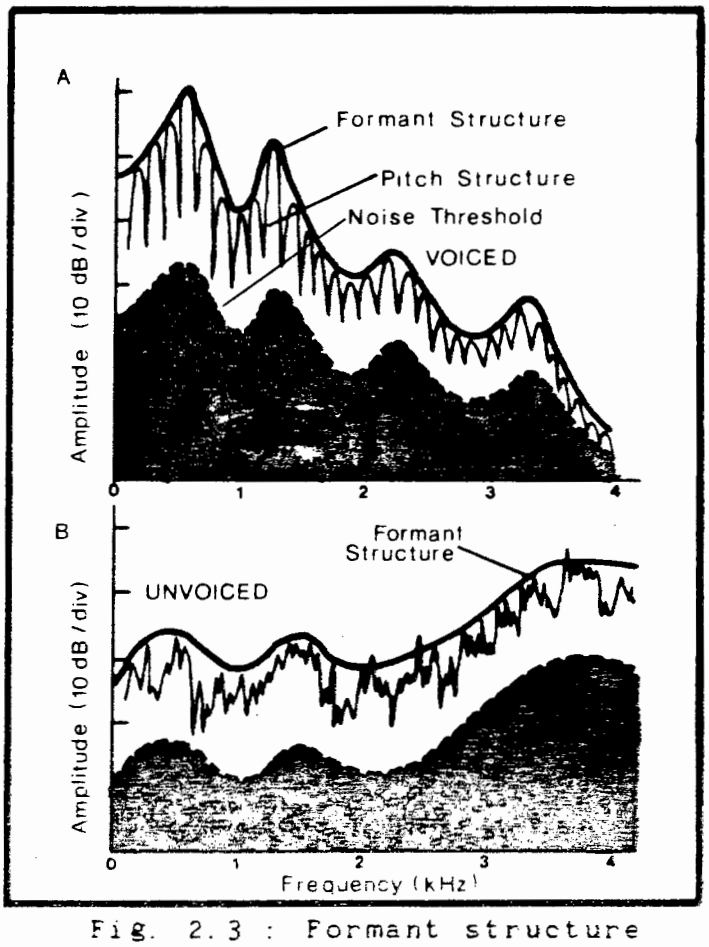


Fig. 2.3 : Formant structure of speech.

Speech signals demonstrate both long-time (0.5 sec) and short-time (20 - 40 msec.) characteristics[3] and it is when viewed over the 'long'

periods that the non-stationarity is most evident whereas over the 'short' period the signal is locally stationary.

2.1.2.3 Amplitude Probability Density Function.

Speech signals typically have a large dynamic range, in the region of 40 dB. As mentioned previously, a large percentage of interactive speech communication time is occupied by silences and hence it is to be expected that the probability of near-zero signal amplitudes would be high. In fact, the probability density function (pdf) of speech is characterized by a high probability of zero or near zero amplitudes which then monotonically decrease towards the very high amplitudes[4]. Speech signals are not random and hence the amplitude pdf is not Gaussian but rather (for the long-time case) the sum of Gaussian and Laplacian pdf's[4].

Because voiced speech segments have a higher probability of occurence (than unvoiced segments) the overall quality of coded speech is going to be determined in part by how well the voiced segments are coded. Added to this is the fact that voiced sounds are more susceptible (from a listener's point of view) to the addition of granular noise (defined later in the section on delta modulation) than unvoiced sounds[4].

2.1.2.4 Power Spectral Density (PSD).

Referring back to fig. 2.2 it can be seen that the high frequency components do not contribute very much to the total energy of the speech signal. In fact, the major portion of the signal energy is confined to the first formant[2]. That is not to say that the high frequency components are insignificant to the information content of the signal. On the contrary, they contain important information[4] and must therefore be adequately coded.

The characterization of the signal according to its harmonic content is given by the PSD and is usually evaluated over a long speech segment. One of the reasons for doing this is that the short-time spectrum could be misleading in that a particular sound may exhibit spectral properties which would result in the poor performance of a particular coder. On the other hand, the long-time spectrum (ie. the average of a series of short-time spectra) could exhibit properties which might be deemed to be favourable for a particular coder. This average measure is likely to be more useful because it is the average performance of a coder that is important to the general user. In other words, the long-time spectrum is a measure of the spectral content of speech signals as opposed to the short-time spectra of specific sounds.

2.1.2.5 The Autocorrelation Function (ACF).

The ACF, denoted by C(n), is an indication of the correlation between two speech samples seperated by 'n' sampling periods. C(0) is the correlation of one sample with itself and is normalised to unity (ie. C(0) = 1). All other values of C(n) are then normalised relative to C(0). The ACF, when taken over a speech segment, will give a measure of the predictability of the speech samples (eg. a value for C(n) of 0.9 would suggest that the subsequent samples could be predicted from the previous ones, while a value of 0.1 would indicate a low degree of predictabilty, which one would expect for random signals). The correlation between adjacent samples, C(1), for speech has a typical average value (for Nyquist sampled speech) of 0.9[43]. The significance of this is that the variance of the difference between adjacent samples of speech is less than the variance of the samples themselves. The relevance of this to low bit rate encoding will be discussed later.

At this stage it is worth mentioning that the ACF and the PSD form a Fourier Transform pair and hence, a coding scheme exploiting the one is essentially equivalent to a scheme exploiting the other, the only difference being that the one works in the time domain and the other in the frequency domain.

2.2 Transmission Issues.

The digital encoder is just one part of a communications system and its design cannot be considered in isolation from the transmission network. Two of the main considerations here are the possible need to transmit non-speech signals (ie. signals other than speech occupying the same spectral band) and the probability of channel errors.

2.2.1 Channel Errors.

The presence of high energy noise fields in the vicinity of the transmission network is likely to induce errors in the digital signal which would result in an erroneous decoder output. Hence, for a practical system it is important that the decoder is capable of recovering from transmission errors. Most speech coding methods can tolerate bit error rates (BER) of the order of 10^{-3} (ie. one error in 1000 bits) and an order of magnitude gain (ie. BER of 10^{-2}) can be achieved by implementing 'robust' coders that utilise built-in error protection techniques[4]. This error protection can take the form of an explicit transmission of code responsible for protecting the signal information (ie. extra data is added to the coded signal to enable the detection of transmission errors by the decoder). Alternatively, using 'leaky' adaption logic, the effects of transmission errors will be 'leaked' out and after a predetermined length of time the effect

of the error will be negligible. The advantage of the second technique is that no extra information need be transmitted, thereby conserving channel bandwidth.

2.2.2 Voice-band Data.

Voice-band data signals are analogue signals that have been created from digital data so as to enable their transmission over an analogue telephone channel. Although they have been modulated into the speech band their statistical properties are considerably different to speech signals (as evidenced by fig. 2.4), and hence it is unlikely that an encoder designed for efficient digitization of speech will be anything but sub-optimal for voice-band data.

Obviously if the transmission channels were all digital voice-band data signals would not exist as it is pointless modulating the digital data into analogue form and then reconverting it to a digital form. However, during the current transition from analogue to digital communications equipment it is common for the communications network to include both analogue and digital channels.

Most voice-band data signals have data rates up to 4.8 kbps compared to the operating bit rates of speech waveform encoders which are in the 16 to 64 kbps range[25,26]. Hence, it would

appear inefficient to use a waveform coder for digitizing the voice-band data as opposed to simply demodulating and multiplexing it with other data. For example, demodulating and multiplexing 4.8 kbps data is nearly 7 times more efficient than using a waveform encoder operating at 32 kbps. However, the actual modulation/demodulation process is dependent on the data rate used[26] and the demodulator must be designed for a particular type of data signal[25]. Waveform coders on the other hand are considerably more versatile, being able to handle a range of voice-band data signals as well as facsimile and speech[25].

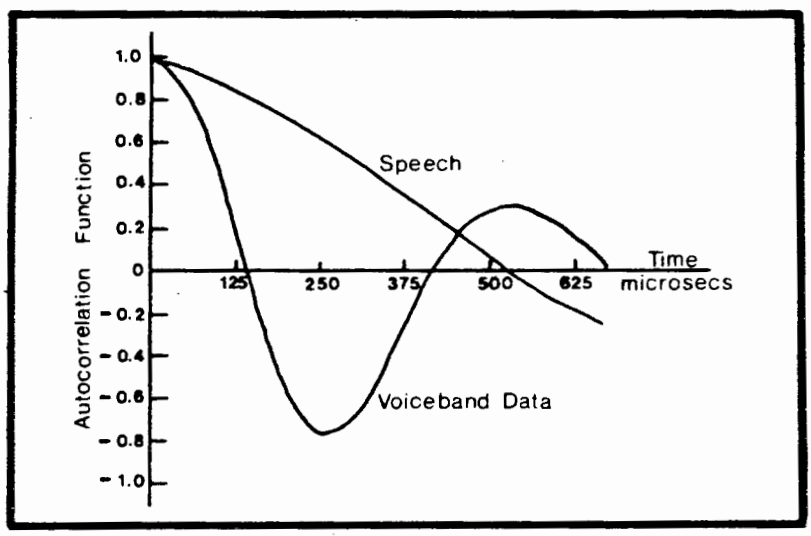


Fig. 2.4 : Autocorrelation functions of speech and voice-band data.

If it is desirable to have optimum performance for both speech and voice-band data it might be necessary to use seperate coders for each. On the other hand, a 'compromise design' might yield satisfactory performance for both speech and data.

2.2.3 Variable-rate Systems.

There are communications systems that exploit the non-stationary intermittent nature of speech while others adapt to the varying demand of the source (not everybody wants to speak at the same time). Examples of these two are TASI (Time Assignment Speech Interpolation) and packet switching networks respectively. The channels for these systems do not operate in a fixed bit-rate mode and hence the encoder/decoder design would have to incorporate some form of interfacing between the codec and the channel.

2.2.4 Tandem Coding.

Depending on the communications link it may be necessary to serially utilise different coder stages[9]. For example, fig. 2.5 shows a link involving one type of encoder operating at 16 kbps interfacing with a second type operating at 2.4 kbps. It is

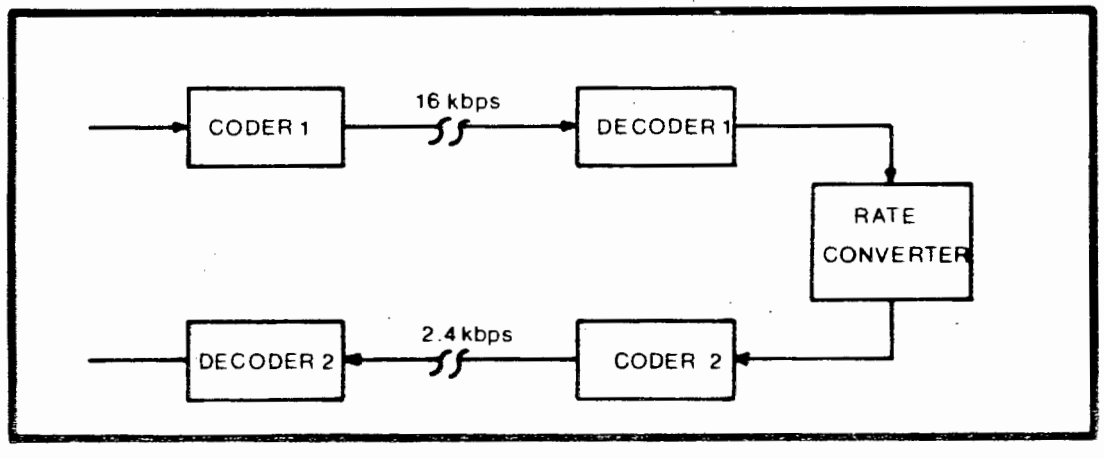


Fig. 2.5 : Tandem connection of codecs operating at different sampling rates.

likely that the use of such tandem connections will result in performance degradation with most of the losses occuring in the first stage[4]. The subsequent losses limit the total number of stages that can be satisfactorily used.

2.2.5 Encryption.

Two methods of accomplishing digital encryption include masking the data bit stream with a pseudorandom binary noise sequence (known by the intended receiver), and permutation of the bit positions within a block of data. Permutation retains a higher residual intelligibility than the masking technique and the effectiveness of the scheme used depends on the type of encoder used[4]. Hence, if a certain type of encryption is already in use it may dictate the types of voice encoders that could be satisfactorily integrated into the communications network.

2.3 Digital Speech Coding Techniques.

Broadly speaking, speech coders can be divided into one of two categories - waveform coders, or source coders. Source coders, generically referred to as 'vocoders' (from the words 'voice coders'), rely for their operation on a priori information on how the signal was generated at the source. This can be achieved by means of a parametric model such as the one shown in fig. 2.1. The second category, waveform coders, attempt to copy the signal

waveform and in general they are signal- (and hence source-) independent, but can be optimized for a specific signal according to the statistical characterization of the signal waveform.

There are coders which tend to make the differentiation between waveform coders and vocoders less distinct. These coders (eg. Linear Predictive Coders) exhibit definite vocoder characteristics in the form of a central parametric model of the speech source and yet are classified as waveform coders because they work by trying to produce a facsimile copy of the input signal.

In the subsequent sections the coders have been grouped according to their overall operating characteristics - do they parameterize the signal (Vocoders), or do they attempt to produce a copy of the signal waveform (Waveform coders) ?

2.3.1 Waveform Coders.

2.3.1.1 Time Domain Coding.

(a) Pulse Code Modulation (PCM) :

Probably the simplest form of analogue-to-digital conversion, PCM entails sampling the signal in time (at the Nyquist rate or faster) and quantizing it to one of a set of discrete amplitudes (ie. the sampled value is rounded off by the quantizer to one of the values in the set). The number of discrete amplitude levels

is determined by the number of bits used in the quantizer (eg. a 12-bit quantizer yields 2^{12} = 4096 levels). The error introduced by the rounding process (the quantization error) depends on the difference between amplitude levels. In linear PCM the quantizer step sizes are all equal and no data compression is possible because none of the characteristics of speech are exploited.

By making the quantization step size vary logarithmically over the range of the signal amplitudes (as in non-linear PCM) it is possible to exploit the redundancy inherent in the amplitude pdf. Linear PCM assumes that all quantizer levels are utilised in an equitable fashion. However, the amplitude pdf of speech suggests that by allocating more levels to the highly probable low amplitude signals and less to the more infrequent high amplitudes would yield better performance. This is because subjectively the quality of coding is likely to be dependent on the quality of the coding of the high-probability signals. For a given encoding quality non-linear PCM can operate at reduced bit rates compared to linear PCM (eg. at a SNR of 35 dB a linear PCM coder requires a 12-bit quantizer whereas a non-linear PCM coder requires a 7-bit log-quantizer).

The long-time amplitude pdf as used in non-linear PCM is a static characteristic. By taking the dynamic speech amplitude variations

into account an increase in the dynamic range of the input signal can be achieved. In adaptive PCM the quantizer step size is varied according to the RMS level of the signal. If the step size is determined from the quantizer output, rather than from the input, the step size information need not be explicitly transmitted to the receiver as it can be recovered from the transmitted signal.

PCM systems are however susceptible to significant performance degradation in the presence of channel errors. The bits in a sample have different weightings and as a result the heavier the weighting of the bit that is corrupted by the channel, the greater is the effect (subjectively) on the receiver output. Non-linear PCM is, at present, the most widely used voice coding technique for telephone networks. The advantage of PCM is that although it may not be optimal for any one particular signal (most notably in the case of linear PCM) it is capable of coding any type of signal (voice-band data, etc.).

(b) Differential Pulse Code Modulation (DPCM) :

The difference between adjacent samples, $D = x_n - x_{n-1}$, has a variance which is less than the variance of the signal itself as long as the correlation between samples is greater than 0.5[8]. The implication of this is that it is possible to quantize this

difference with fewer quantization levels (and hence, bits) than is possible when quantizing the signal itself, and yet maintain the same signal-to-noise level. Essentially the input signal is being discretely differentiated with respect to time, the derivatives are being quantized, coded and transmitted, and at the receiver the input difference samples are integrated to yield the original signal. By coding only the sample-to-sample change in signal level the redundancies have been reduced because, by coding that part of the signal that hasn't changed is a waste of transmission bandwidth as that information is already at the receiver.

As for PCM, an adaptive scheme can be utilised to enable the differential pulse code modulator to respond to the dynamic

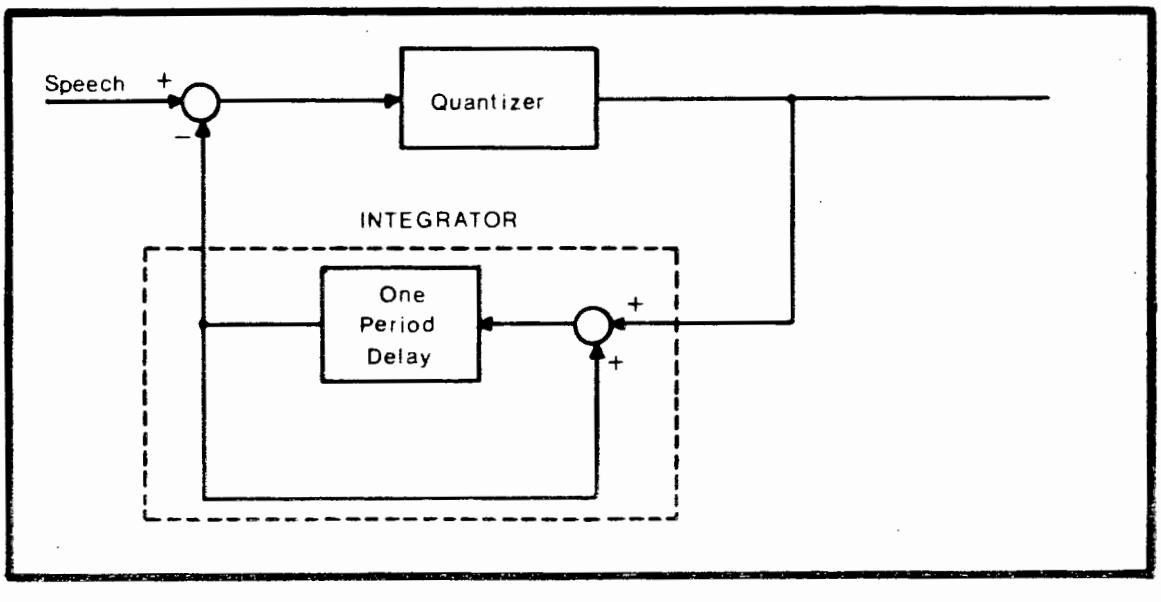


Fig. 2.6 : Block diagram of a DPCM system.

variations of the speech signal amplitudes. Because the transmitted code does not represent the signal itself a different scheme has to be employed in determining the RMS signal value.

(c) Predictive Coding :

A DPCM system is, in effect, a predictive coder. Referring to fig. 2.7 the predictor P would be the integators used in the DPCM system - they predict, based on the previous input sample and their state during the previous sampling time, the next sample value.

If the quantizer input is represented as :

$$D_r = x_r - a.x_{r-1} ,$$

then the variance of D_r is a minimum for $a = C(1)$ (the correlation between adjacent samples)[4]. In the DPCM case the value $a.x_{r-1}$ is the output of the prediction filter and is the first order prediction of x_r. The general predictive coder would essentially be a DPCM of order p. In other words, the p^{th} order prediction of x_r would be derived from the p previous samples, and not simply x_{r-1} :

$$x'_r = \sum_{n=1}^{p} a_n . x_{r-n}$$

In order to control the accumulation of quantizing errors at the receiver, the value X_{r-n} (cf. fig. 2.7) is used instead of x_{r-n}.

This technique is known as Linear Predictive Coding (LPC). The prediction coefficients, a_n, are chosen to maximize the long-time SNR. The error (or noise) is given by :

$$e = x_n - x'_n.$$

$\langle e^2 \rangle_{av}$ is the average of e^2 over all sampling instances. The prediction coefficients for maximum SNR are then found by setting the partial derivative of $\langle e^2 \rangle_{av}$, with respect to each a_n, to zero. The predicition coefficients are then found by solving :

$A.T = V$ A - the vector of optimal prediction coefficients,

T - a symmetrical p X p matrix of the auto-correlation function values,

V - a vector of autocorrelation function values[4,10].

The degree of improvement of this system over linear PCM tends to saturate at a value for p of about 2 or 3[4].

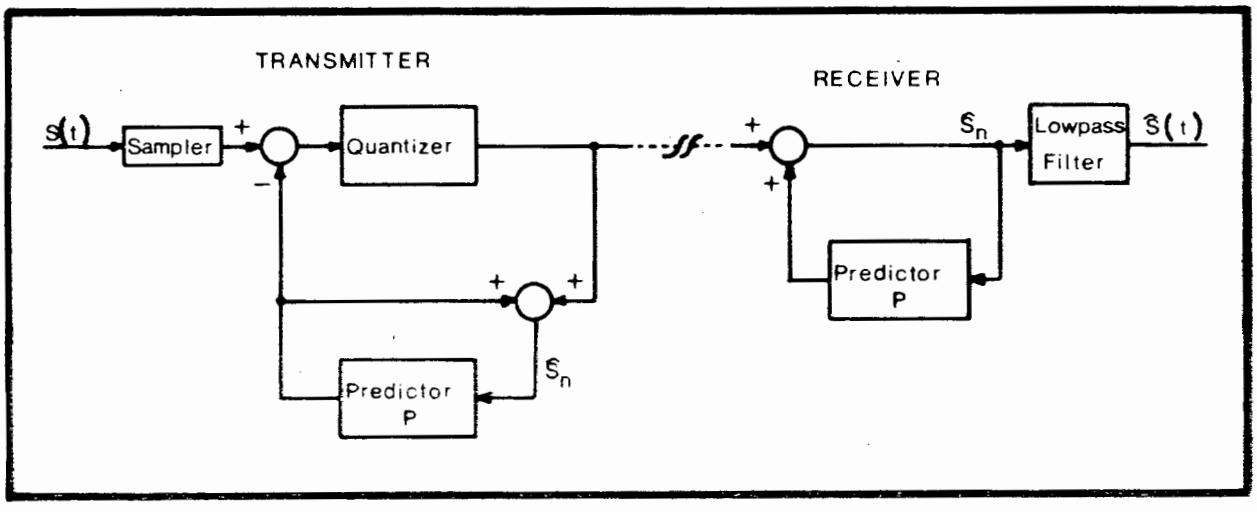


Fig. 2.7 : Block diagram of a predictive coder.

The problem involved with this technique is that, being static (ie. the values for a(n) are optimized once only) the coder's performance is subject to the vagaries of speech non-stationarity and, as a result, may work well for one speech segment or speaker, but not for another.

(d) Adaptive Predictive Coding (APC) :

APC basically involves periodically updating the LPC prediction coefficients to improve the coding efficiency for the general speech input. A common technique utilises two predictors - one is based on the short-time spectral envelope (ie. formant structure) while the other is based on the quasi-periodic nature of voiced speech[4,11] which gives rise to the spectral fine structure as shown in fig. 2.3a.

The periodic updating of the vector A can be achieved by storing a certain length of speech, calculating the auto-correlation matrix T and vector V and then adjusting A accordingly. The net result is that instead of the long-time correlation, the short-time correlation is used to reduce the data rate. This reduction is aided by the fact that the coding of the prediction coefficients can be quite coarse and they can be updated at a relatively slow rate[6].

However, if it is desirable to reduce the data rate even further the receiver can use the incoming data to update the prediction coefficients, thereby eliminating the need to transmit these coefficients, but at the expense of extra processing.

The auditory masking properties of speech would suggest that much of the predictive noise originates from the frequency areas where the signal level is low (eg. outside the formant regions). In other words, subjectively the SNR in the formant regions would be significantly better than other areas of the speech spectrum. By adding an extra feedback network around the quantizer it is possible to shape the quantizing noise spectrum to enhance the subjective quality of the codec[4].

As mentioned previously, predictive coding techniques can straddle the boundary between waveform coders and vocoders. If, instead of prediciton filters, signal source models are used and the information regarding the voiced/unvoiced characteristics and the pitch information are transmitted (instead of the prediction coefficients), then it would be possible to generate the prediction filter excitation locally at the receiver based on the above data. This type of scheme is neither strictly waveform coding nor purely vocoding (to be discussed later) but a hybrid of the two.

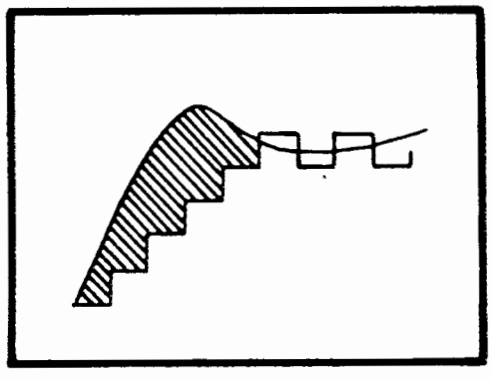
Although the processing overhead is quite high (the solution of a set of simultaneous linear equations involves a large amount of processing in order to determine the prediction coefficients) predictive coding is an important encoding technique at very low transmission data rates (2400 bps or lower) and at least one real-time implementation has been achieved using commercially available components[12].

(e) Delta Modulation (DM) :

In DPCM the signal correlations were used to reduce the number of levels required to quantize the prediction error. If this is taken to the limit case of using a two-level quantizer (ie. the prediction error is either positive or negative) the input signal needs to be oversampled (sampled at more than the Nyquist rate) in order to increase the adjacent sample correlation. This then is Delta Modulation - the 1-bit version of DPCM.

In its simplest form (Linear DM - LDM) the coder approximates the input signal by a ramp staircase with equally sized steps (cf. fig. 2.8). At sample time 'n' the polarity of the error (the difference between the input sample x(n) and the approximation sample y(n-1)) is quantized and transmitted. The staircase function is then updated in the direction of the error.

However, although DM is suitable for speech coding, the simple scheme described above is totally inadequate. Firstly, when the coder is 'hunting' about a constant input signal level so-called granular noise is produced. This is most likely to occur during silences in speech segments and could prove to be irritating to the listener. Secondly, because the slope of the ramp staircase is fixed, if the slope of the input signal is greater than that of the staircase (eg. during the transients produced by 'stops') then the coder produces 'slope overload' distortion - ie. once the frequency-amplitude product (of the input) exceeds a certain threshold the coder goes into slope overload. Obviously, as the



- The shaded area represents 'slope overload' noise.
- The remaining section represents the source of the so-called 'granular' noise.

Fig. 2.8 : Waveforms produced by a Linear Delta Modulator.

signal frequency increases so the maximum possible signal amplitude, before the onset of slope overload, decreases. What this means is that the dynamic range of LDM tends to decrease with

frequency in the same manner as the PSD curve. Granular noise can be reduced by reducing the step size whereas slope overload noise is reduced by increasing the step size and sampling rate.

Again, as with the schemes discussed previously, the static non-adaptive technique tends to fail due to the characteristics of speech.

In order to improve the dynamic range of the DM and to minimise the total noise power adaptive techniques are used - hence 'Adaptive Delta Modulation' (ADM). A number of companding schemes have been developed[13-17], all with one thing in common - the modification of the step size is determined from the encoder output bit stream. The general adaptation rule is given by :

$$S_r = f(S_{r-1}, b_r, b_{r-1}, \ldots, b_{r-n})$$

where

S_r is the new step size, and

b_r is the sign of the error.

Different schemes use different memory lengths and basically there are two categories of companding algorithms. Step size adaptation can be at either the syllabic rate or the instantaneous rate.

- o Syllabic companding uses a short segment (typically about 5 msec in length) of the speech to adjust the step size, thereby utilising the short-time quasi-stationarity charac-

teristics.

- o Instantaneous companding on the other hand updates the step size at each sampling instant.

The main problem with syllabic companding is its inability to handle the transient-like changes present in speech. While instantaneous companding can respond to the transients its fixed basic step size means that, due to the large variation in dynamic range, it may work well for one segment of speech but not another[17]. The performance of an instantaneous compander is also likely to be degraded during silent or steady-state portions of speech. It would seem then that some form of combination of instantaneous and syllabic companding schemes could also be used as a compromise on the shortcomings of the induvidual schemes. The development of a hybrid delta modulator (HCDM) is a fairly recent development[17] and is claimed to be better (in terms of SNR and dynamic range) than the previous methods[18].

Earlier we saw how DPCM was extended from a first order system to a higher order by extending the prediction parameters to include previous values. For DM systems this is equivalent to incorporating the higher order differences and more stages of integration. Double integration includes two integrating stages and it has been found that combining single and double integration, the second

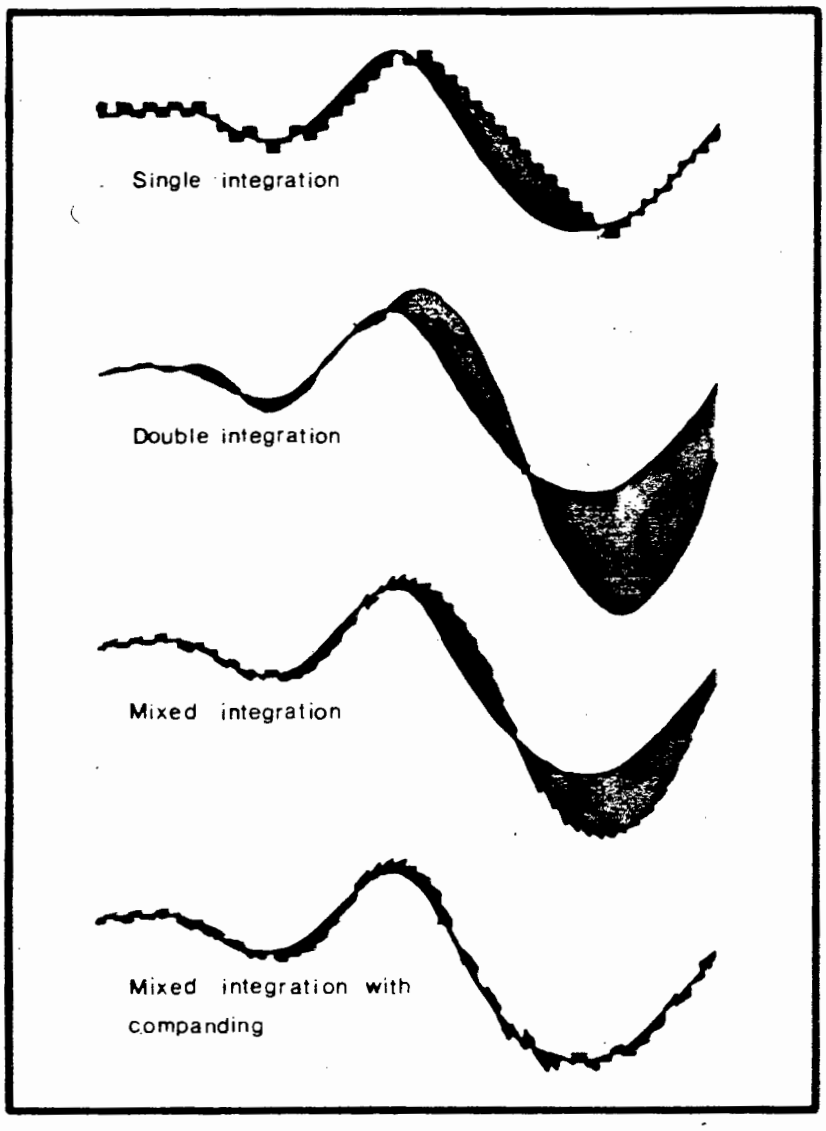


Fig. 2.9 : DM waveforms obtained using different types of integrator combinations.

integrator being progressively engaged above 2 kHz, is an improvement on pure single or double integration (the latter tends to suffer from stability problems)[19]. This combination is referred to as 'mixed' integration and fig. 2.9 shows the types of

waveforms obtained from the different delta modulators.

Whichever form of prediction is used (single, double, etc.) it is still possible to include some form of companding as the relative effects are complementary[37].

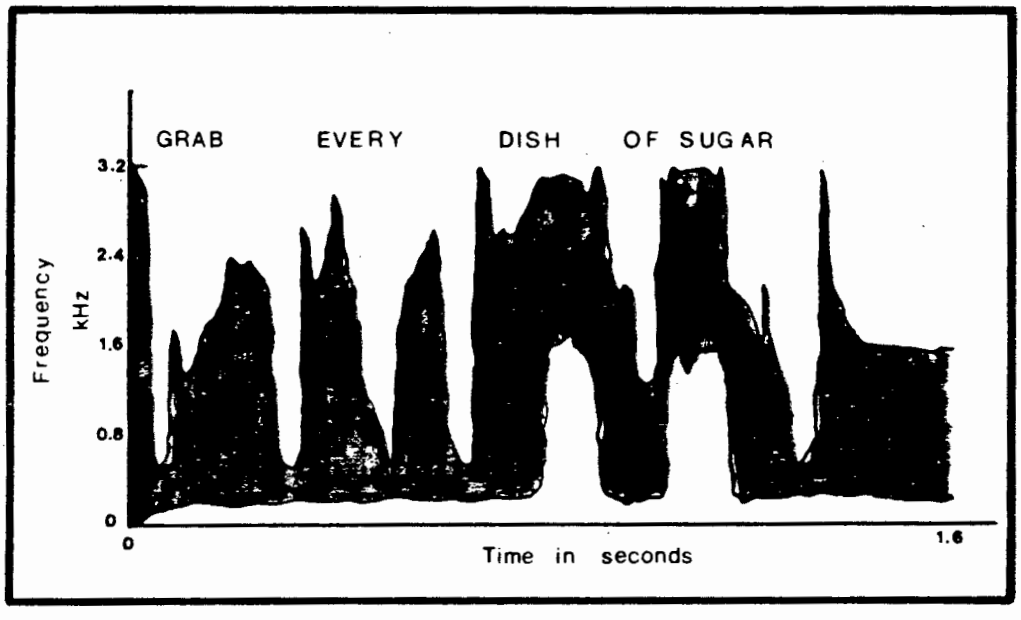


Fig. 2.10 : The variation of the speech signal bandwidth with time.

Although speech is often limited to the 300 - 3300 Hz band, the actual bandwidth of the signal varies with time (fig. 2.10) and does not always fully occupy the allotted band - generally only the lower frequencies are occupied during voiced sounds while the signal is 'high pass' during unvoiced sounds[20]. Commonly a non-adaptive low-pass filter is used on the output to remove coder noise outside the speech pass-band. If instead an adaptive filter is used (ie. one that tracks the actual bandwidth of the speech),

the perceived quality of the low bit-rate coder is significantly improved[20].

2.3.1.2 Frequency Domain Coding.

Instead of treating speech as a single-band signal, as in time-domain coding, the frequency band could be divided into a number of sections with each being coded seperately. The advantage of this is that the encoding accuracy can be determined according to the activity of the seperate spectral bands - eg. bands with very little or no energy need not be coded whereas bands with a high energy signal can be coded accurately. This is a different approach to the problem mentioned at the end of the last section (ie. adaptively filtering the DM output). The main distinguishing feature between the different encoding algorithms is usually, as in the time domain case, the degree of prediction used.

(a) Sub-Band Coding (SBC) :

This technique involves dividing the speech band into a number of sub-bands (by a bank of band-pass filters), translating each band down to zero frequency, and then coding each band using adaptive PCM. At the receiver the process is reversed with each band being modulated back up to its original position and then added to the rest to yield the original signal. The advantage of this method is that each band can be quantized and coded according to the

perceptual properties of that band - bands sensitive to quantization noise (eg. components with low signal energy) can be allocated lower quantizer step sizes to reduce the quantization noise while bands relatively insensitive to noise can be quantized coarsely. The pitch and formant information in the lower frequency bands can be accurately encoded by allocating a large number of bits to these bands while the unvoiced noise-like sounds in the higher frequency bands can tolerate fewer quantizer bits[44]. In this way the masking of one frequency band by the noise in another is prevented.

(b) Adaptive Transform Coding (ATC) :

This technique is considerably more complex than SBC as it involves dividing the input signal into time segments and then transforming these windowed segments into the frequency domain. Each segment is then represented by a set of transform coefficients which are seperately quantized and transmitted. At the receiver these coefficients are then inverse transformed to yield the original signal segment. The transform can be achieved by either a Fast Fourier Transform (FFT) or a Discrete Cosine Transform (DCT) which has been found to be particularly well suited to speech[44].

The performance of this encoder can be enhanced by employing a

dynamically adaptive scheme, based on the spectral properties of speech, to the transform process. The use of a bit allocation algorithm (eg. as described in [4]) can help in controlling the noise spectrum. Because the spectral levels of a speech segment are not known a priori they must be estimated. The information regarding the dynamic properties of the speech spectra is referred to as 'side information' (fig. 2.11). Depending on the added complexity of the system the ATC method can be expected to operate down to about 8 kbps[3].

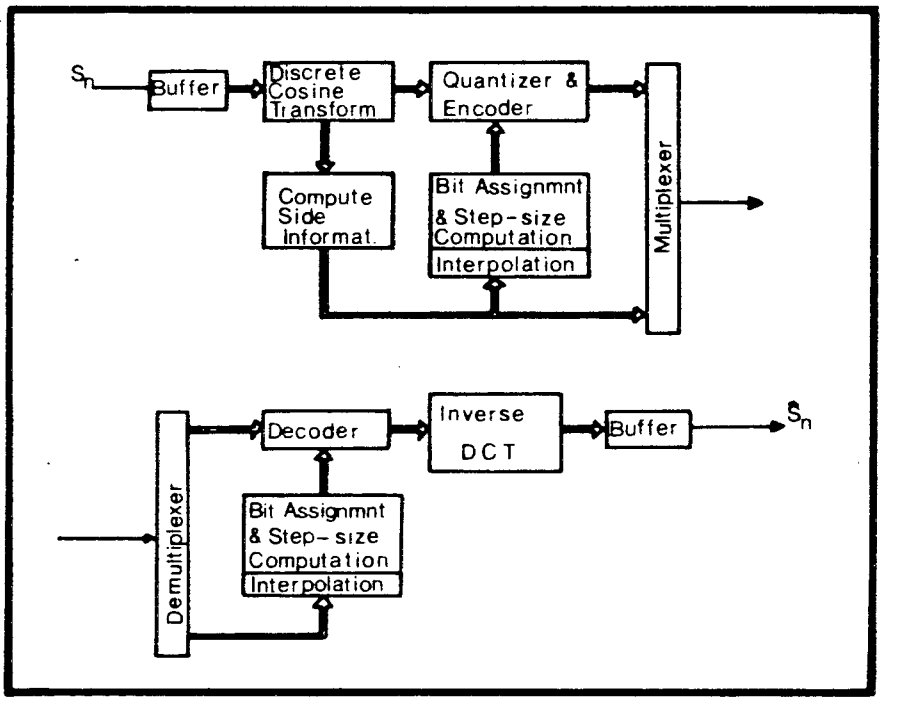


Fig. 2.11 : Block diagram of an Adaptive Transform Coder.

2.3.2 Vocoders.

Source coders are, in general, used for special purpose communi-

cations (eg. where very low bit-rates are required) and are not suitable for use in commercial telephone networks. If one looks at fig. 2.12 this can be clearly seen - telephone networks fall into the 'Toll Quality' category. Vocoders depend for their operation on the source model (shown earlier in fig. 2.1) where the sound generating system is divided into a sound source and a modulator. In the vocoder itself the information to be coded includes the switching information (between the pulse generator and the random noise generator), amplitude signal (for intensity of excitation of the two sources), and the pitch information (specified by the pitch period of the pulse generator). There are a number of possible parametric models that can be used by the vocoder. These vary from the use of linear prediction coefficients describing the spectral envelope (LPC vocoder), to the specification of major spectral resonances (formant vocoder) and a description of the short-time amplitude spectrum for specific frequencies (channel vocoder). Other methods include the specification of samples of the short-time

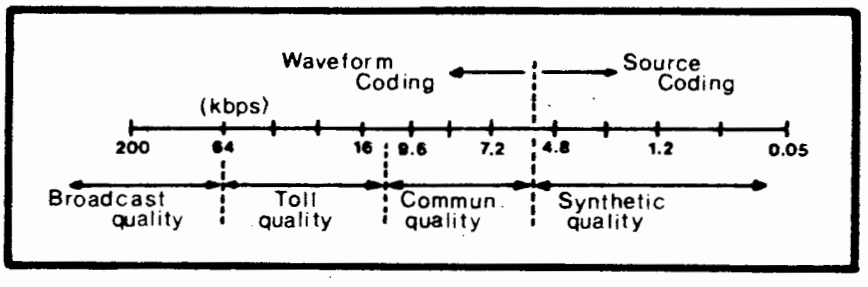


Fig. 2.12 : Communications quality scale.

auto-correlation function (the autocorrelation vocoder), or the use of coefficients of a set of orthonormal functions that approximate the speech waveform (orthogonal function vocoder).

2.3.2.1 Frequency Domain Vocoding.

As was mentioned previously, the human ear performs Fourier analysis of the speech signal. In other words, the perceptual analysis of speech sounds is based on their spectra. It would thus make sense to digitally characterize the signal according to its spectrum. In this way it is possible to drastically reduce the total bandwidth required for transmitting speech when compared to transmitting the signal directly. It has been found that the spectrum can be adequately described by 16 frequency values which are updated every 20 milliseconds[4]. From the Nyquist sampling theorem each value would then require a bandwidth of 1/(2*20 msec) = 25 Hz. The 16 channels would then need 16*25 = 400 Hz. On the other hand, direct transmission of the signal would need about 3500 Hz, or nearly 9 times the bandwidth. If the pitch information, and the voiced/unvoiced data is included this would typically require an extra 300 Hz bandwidth, bringing the total for the vocoder to 700 Hz, which is still considerably less than the direct method.

The channel vocoder consists of a vocoder analyzer at the trans-

mitter (which measures the 16 frequency values, the pitch information and determines the voiced/unvoiced state), and a vocoder synthesizer at the receiver. At the receiver the 16 received channel signals control the frequency response of a time varying filter, in that way reproducing the spectral envelope of the original signal. The filter is excited with a signal which is either white noise or a pulse-train whose period is obtained from the pitch information signal.

A formant vocoder is essentially the same except that instead of 16 frequency values, only the formant frequencies are analyzed. However, the techniques for evaluating the formant data are considerably more complex than those used in the channel vocoder.

2.3.2.2 Time Domain Vocoding.

(a) Orthogonal Expansion Vocoders :

It is possible to expand the signal waveform, or the power (or amplitude) spectrum, or the autocorrelation function into a series of orthogonal functions. Waveform expansions can be done via Laguerre polynomials, while power spectrum expansions are best done by the eigenfunctions of the autocovariance matrix - the so-called Karhunen-Loeve expansions. Another possibility involves expanding the logarithm of the power spectrum into a cosine series to yield the 'cepstrum' (and hence the cepstral vocoder). The

advantage of the cepstrum is that no additional analysis is needed to measure the voice pitch[43].

(b) Autocorrelation Vocoders :

For this type of vocoder the telephone-bandwdth speech is sampled at about 8 kHz - ie. samples are 125 μsec apart. To ensure sufficient spectral resolution a large number of samples would have to be used. Compared to the spectral samples the autocorrelation samples need about twice the quantization resolution (7-8 bits/sample compared to 3-4 bits/sample[43]). By synthesizing the speech signal based on the autocorrelation function yields a signal whose spectrum is the squared version of the original signal[43] thus necessitating spectral square-rooting.

(c) LPC Vocoder :

As with the orthogonal expansion vocoder the LPC vocoder is not purely a time-domain vocoder as it exploits characteristics of the spectral properties of speech as well, and could equally have been classified with the frequency-domain vocoders. The LPC vocoder is basically an APC system except that the prediction residual has been replaced by the pulse and noise sources.

An essential part of all vocoders is the pitch measurement sub-system. This is also one of the more difficult sections of the

analysis of speech, particularly telephone speech which has had its fundamental frequency removed. As yet no completely satisfactory method has been developed owing mainly to the shortcoming of simplistically separating the voiced and unvoiced sounds. A product of this problem has been the emergence of hybrid techniques where part of the spectrum is waveform coded while the rest is vocoded. A major drawback of vocoders is their sensitivity to different speakers, and the synthetic (automaton-like) nature of the output speech, resulting in a substantial degradation of the speaker recognizability. This must however be weighed against the substantial economies in the transmission bandwidth possible with vocoders.

CHAPTER 3

SPEECH CODEC DESIGN

3.1 Choice of Encoding Technique.

Essentially there are three stages to selecting the codec :

(i) the choice between waveform coding or vocoding techniques,

(ii) time-domain or frequency-domain coding,

(iii) the final choice involves the specific technique to be used.

Based on the present state of the art waveform coders are more attractive for commercial communications links than vocoders. This is because of, as was mentioned previously, firstly, the complex techniques involved in overcoming the pitch measurement problems, and secondly, the sensitivity of vocoders to different speech segments and speakers. The synthetic quality of the output speech is also not considered to be attractive for a general purpose link. Hence, the use of vocoding techniques has not been considered in this work.

The frequency domain techniques vary from, on the one hand, a relatively simple concept requiring a considerable amount of hardware, to a much more complex scheme with high overheads in terms of both hardware and processing. At the lower end of the complexity scale SBC requires a bank of band-pass filters with relatively high selectivity. The use of either analogue or digital filters would yield circuits of considerable complexity with a large amount of hardware required. The number of filters

required for ATC is less than SBC but the processing required is considerably more complex. The requirements that the codec used be implementable as a low cost, low complexity integrated circuit tends to gravitate against these techniques.

Primarily we are attempting to improve the transmission channel efficiency by reducing the bit rate while maintaining an acceptable level of speech quality. This immediately eliminates PCM and non-linear PCM as the band-width savings possible with these techniques are relatively minimal compared to alternative techniques. The choice then is essentially between the predictive coders (LPC, APC, etc.) and delta modulation. The complexity and costs of the predictive coders are too restrictive. The implementation of an LPC system requires , for example, three 16-bit microcomputers, another 8-bit microcomputer plus other peripheral elements (memory, etc.) and an approximate estimate of the cost of such a device (in production quantities) is in the region of $1000[1 2].

It was decided then to investigate the use of delta modulation as a means of providing a low cost, low complexity digital speech encoder capable of operating at subjectively acceptable performance levels at low bit rates.

3.2 Delta Modulation Algorithms.

3.2.1 Linear Delta Modulation.

LDM was first described in a French patent in 1946 and although the simplest form of delta modulation, which has been shown to be totally inadequate for digital speech encoding, it has been included in this chapter as it is the foundation from which all other delta modulators have been developed.

The operation of the LDM encoder is very simple. A bandlimited input analogue signal is encoded into a binary signal which is transmitted to the decoder where the analogue signal is recovered. The binary signal at the transmitter is also fed back to the encoder where it is locally decoded and compared to the input analogue signal. The error (ie. the difference between the input and decoded signals) is quantized by a two-level quantizer. It is the sampled quantized error signal that forms the binary output signal. This binary signal is in fact a digital representation of the sign of the error signal.

The local decoder in the feedback loop of the coder is an integrator, which is a linear network element. The integrator can be implemented using analogue techniques but, because it is common to quantize the input signal in both time and amplitude (using a sampler, zero-order hold circuit and an A/D converter) the rest of

the circuit can be implemented digitally. In this way it is possible to reduce the sensitivity of the circuit to component value fluctuations, and it also offers the facility of using integrated circuit techniques more easily than in the case of analogue implementation.

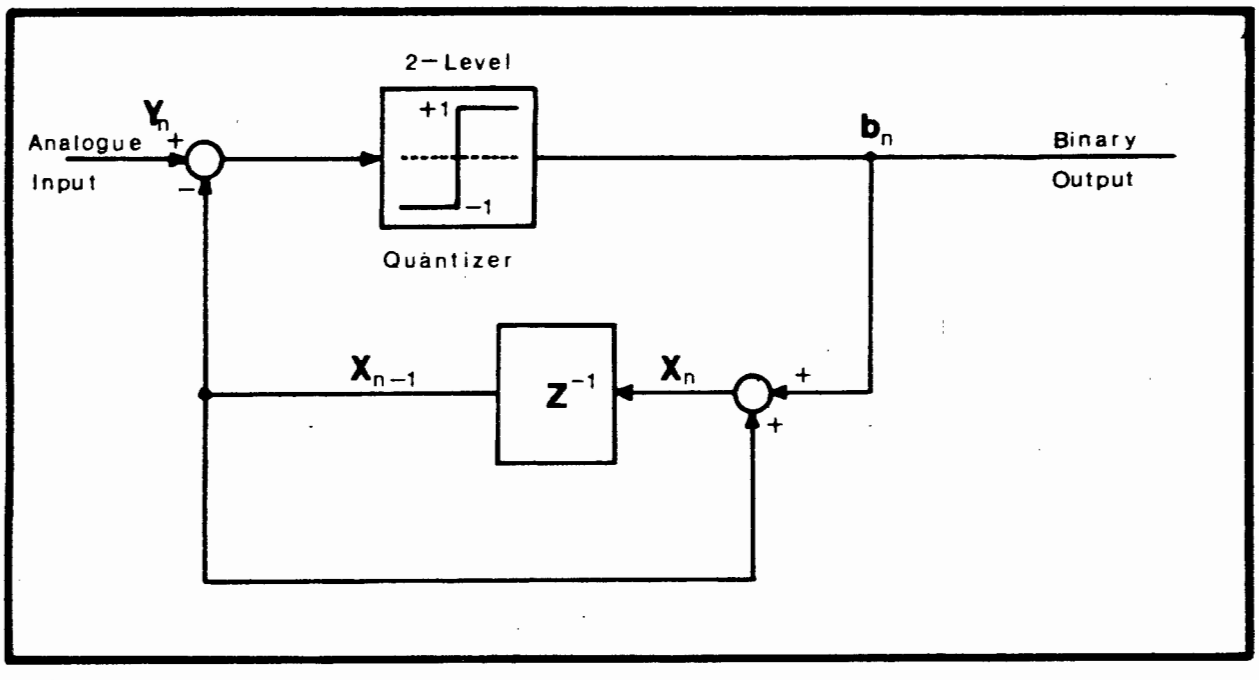


Fig. 3.1 : Block diagram of the LDM encoder.

The algorithm for the LDM encoder shown in fig. 3.1 is given by :

$$b_n = \mathrm{sgn}(Y_n - X_{n-1})$$

$$X_n = X_{n-1} + b_n . S_o$$

where

b_n is the quantized error signal,

Y_n is the quantized input signal,

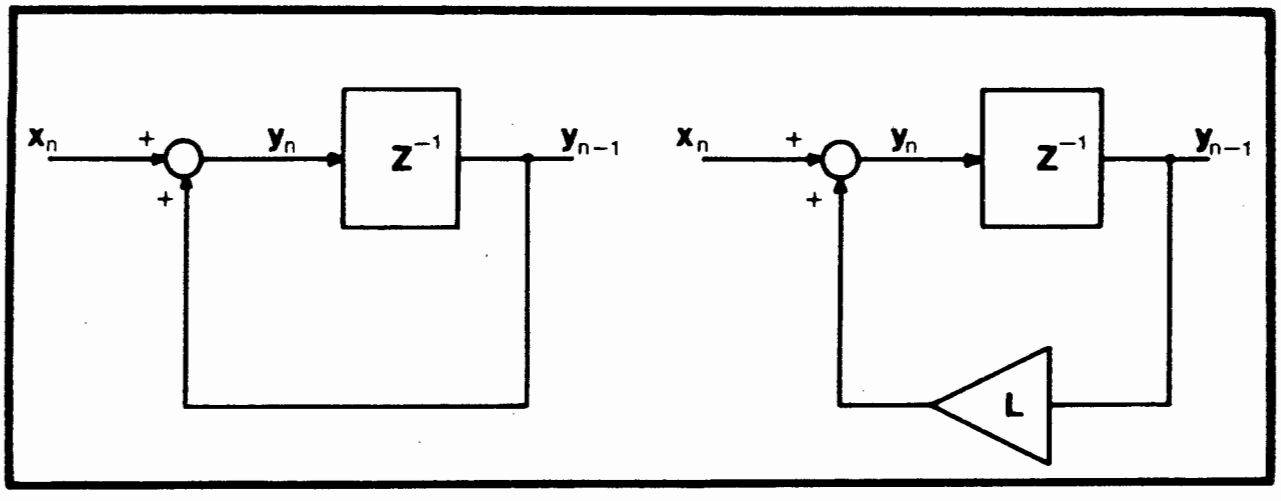
X_n is the LDM approximation signal, and

S_o is the increment (ie. step size) of the approximation signal.

In other words, the coder updates the approximation signal in the direction of the error between the present input signal sample, Y_n, and the value of the approximation signal, X_{n-1}, at the previous sampling time to produce the updated decoder output X_n. From fig. 3.1 it should be clear that the decoder in the receiver is equivalent to the local decoder in the feedback loop of the encoder.

The choice of an integrator for a voice encoding scheme is significant when one considers the ACF for speech. The adjacent-sample correlation, C(1), gives an indication of what percentage of the present sample is contained in the next one. Clearly, only a D.C. signal will have a long-time averaged value of 100% (ie. C(1) = 1) and hence it is unlikely that the perfect integrator, shown in fig. 3.2(a), will be optimum for speech coding. The performance of the encoder is improved if a leaky integrator, shown in fig. 3.2(b), is used instead. Not only does this exploit the characteristics indicated by the ACF, but it also has the added advantage of improving the performance of the codec in the presence of errors induced in the transmission channel. Any error at the input to the decoder will slowly be 'leaked' away by the

leakage factor 'L' and after a predetermined length of time the decoder state will be the same as the encoder to within an acceptable degree of error.



(a) $y_n = y_{n-1} + x_n$ (b) $y_n = L.y_{n-1} + x_n$

$L < 1$

Fig. 3.2 : Discrete integrators - with and without leakage.

The shortcomings of the LDM encoder are well known and have been summarised in the previous chapter.

3.2.2 Continuously Variable Slope Delta Modulation (CVSD).

The problem with LDM is that there is only one level of the input signal at which the signal-to-noise ratio is maximized (cf. chapter 7). One way of improving the performance is to ensure that the input signal level is always close to the optimum level. In order to achieve this for speech the dynamic range of the signal would have to be reduced. This can be done using a

compressor which reduces the large amplitude signal levels relative to the lower levels. The distortion introduced by this technique is compensated for by using a complementary expander at the output of the decoder. Alternatively, this companding effect could be achieved by the delta modulator itself, and more simply than by complementary analogue circuits[13].

Of the adaptive delta modulators that exploit the syllabic characteristics of speech the most commonly used is the CVSD codec[18]. The block diagram of the CVSD coder is shown in fig. 3.3. The main sections are the comparator and quantizer which produces the two-level version of the error signal, a compandor (containing a 'syllabic filter'), and a prediction filter (ie. the leaky integrator described previously). The compandor is made up

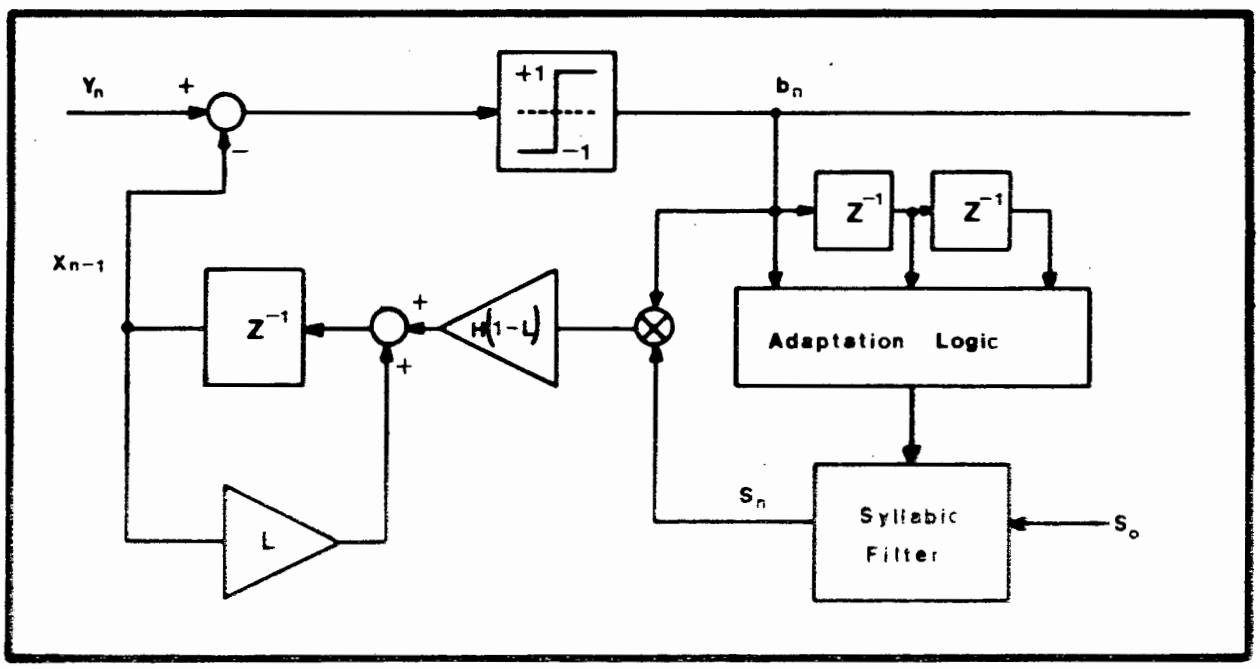


Fig. 3.3 : Block diagram of the CVSD coder.

of a slope detector and a low-pass filter with a 3-dB point ranging between 30 Hz and 160 Hz depending on the particular implementation[18,21,22]. The step- size integrator (a low-pass filter) generates the envelope of the speech signal which is typically between 60 and 100 Hz[21]. The operation of the CVSD is quite simple : the slope detector compares a windowed sequence of the binary output signal (typically three or four bits in length) and generates a pulse if three (or four) consecutive 1's or 0's is detected. This pulse excites the syllabic filter, whose output polarity is controlled by the binary encoder output signal, and is used as the input to the prediction filter, which has a typical value for its time constant of 1 millisecond. The output of the prediction filter is compared to the actual input signal with the difference between the two being quantized to produce the binary output signal.

The algorithm for the CVSD coder is given by :

$$b_n = \mathrm{sgn}(Y_n - X_{n-1})$$

$$S_n = B.S_{n-1} + (1-B).(V + S_n)$$

$$X_n = L.X_{n-1} + H.(1-L).S_n.b_n$$

where

b_n is the quantized error signal at the n^{th} sampling instant,

Y_n is the actual n^{th} input sample,

X_n is the estimate of the input signal,

S_n is the step-size,

S_0 is the minimum step size,

V is the output of the slope detector - V is a constant positive voltage if three consecutive outputs of the encoder are identical, otherwise V is zero,

L is the prediction constant,

B is the step-size leakage constant, and

H is the gain of the prediction integrator.

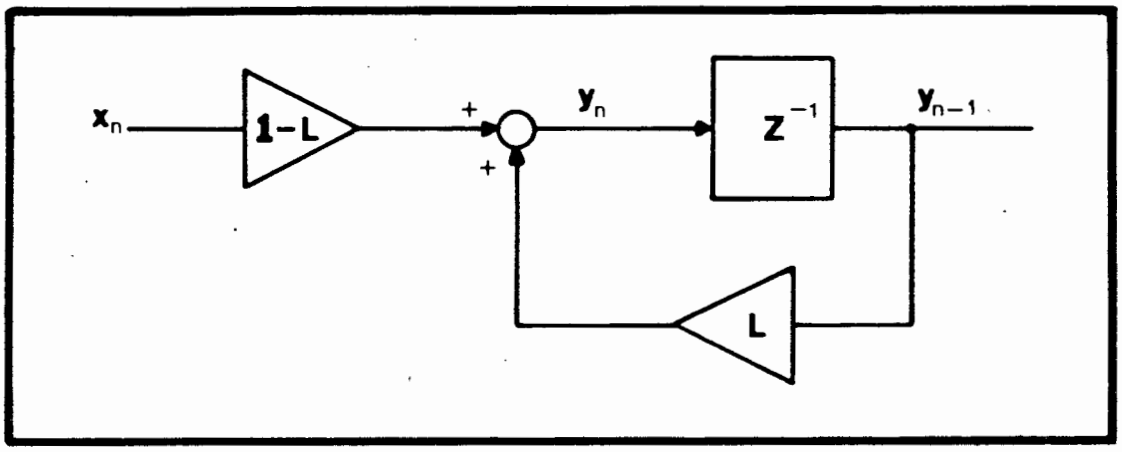


Fig. 3.4 : Modified version of the discrete integrator.

Determining the prediction and step-size leakage constants :

If the integrator shown in fig. 3.2(b) is modified to that shown in fig. 3.4, the time domain equations describing its operation become :

$$y_n = L.y_{n-1} + (1-L).x_n.$$

The Z-domain equivalent :

$$Y(Z) = L.Y(Z).Z^{-1} + (1 - L).X(Z)$$

$$\therefore Y(Z).(1 - L.Z^{-1}) = (1 - L).X(Z)$$

$$\therefore Y(Z)/X(Z) = (1 - L)/(1 - L.Z^{-1}).$$

The poles of this transfer function are given by :

$$1 - L.Z^{-1} = 0$$

$$\Rightarrow Z = L$$

and,

$$Z = e^{j\omega T}.$$

Hence,

$$e^{-j\omega T} = 1/L$$

$$\therefore f_p = (-1/j2\pi T).Ln(1/L).$$

Correspondingly, the RC-equivalent filter has poles at :

$$1 + jwRC = 0$$

$$\therefore f_p = -1/(j2\pi RC).$$

Equating the two :

$$(-1/j2\pi T).Ln(1/L) = -1/(j2\pi RC)$$

$$\therefore L = e^{-T/RC}.$$

So, for an RC-time constant of 1 msec, and a sampling frequency of 16 kHz,

$$L = e^{-1/16} = 0.94.$$

Similarly, for B, the step-size leakage factor; for a leakage time constant of 6.4 msec,

$$B = e^{-1/(16*6.4)} = 0.99.$$

The values for H and V vary considerably in the literature and they can take on values from 3 and 150 respectively[22] down to 1 and 16 respectively[21]. In practice it would be necessary to determine the optimum values for these parameters prior to evaluating the performance of the codec.

3.2.3 Constant Factor Delta Modulation (CFDM).

Whereas the CVSD codec exploited the syllabic characteristics of speech the CFDM codec uses instantaneous exponential adaptation to improve on the performance of the LDM codec.

At every sampling instant the step size of the codec is modified by a specific factor. To do this a one-bit memory is included as the adaptation depends in the immediately past binary output, b_{n-1}, and the present output, b_n. These two outputs are compared to each other and if they are the same then the step size is increased by a constant factor P, else it is decreased by factor Q.

$$b_n = \mathrm{sgn}(Y_n - X_{n-1})$$

$$k_n = P \quad \text{if } b_n . b_{n-1} = 1,$$

$$\quad\; Q \quad \text{if } b_n . b_{n-1} = 0,$$

$$S_n = S_{n-1} . k_n,$$

$$X_n = L . X_{n-1} + S_n . b_n$$

where

b_n is the encoder binary output,

k_n is the multiplication factor for the step-size adaptation,

S_n is the step size, and

X_n is the approximation to the input signal, Y_n, with L being the prediction constant.

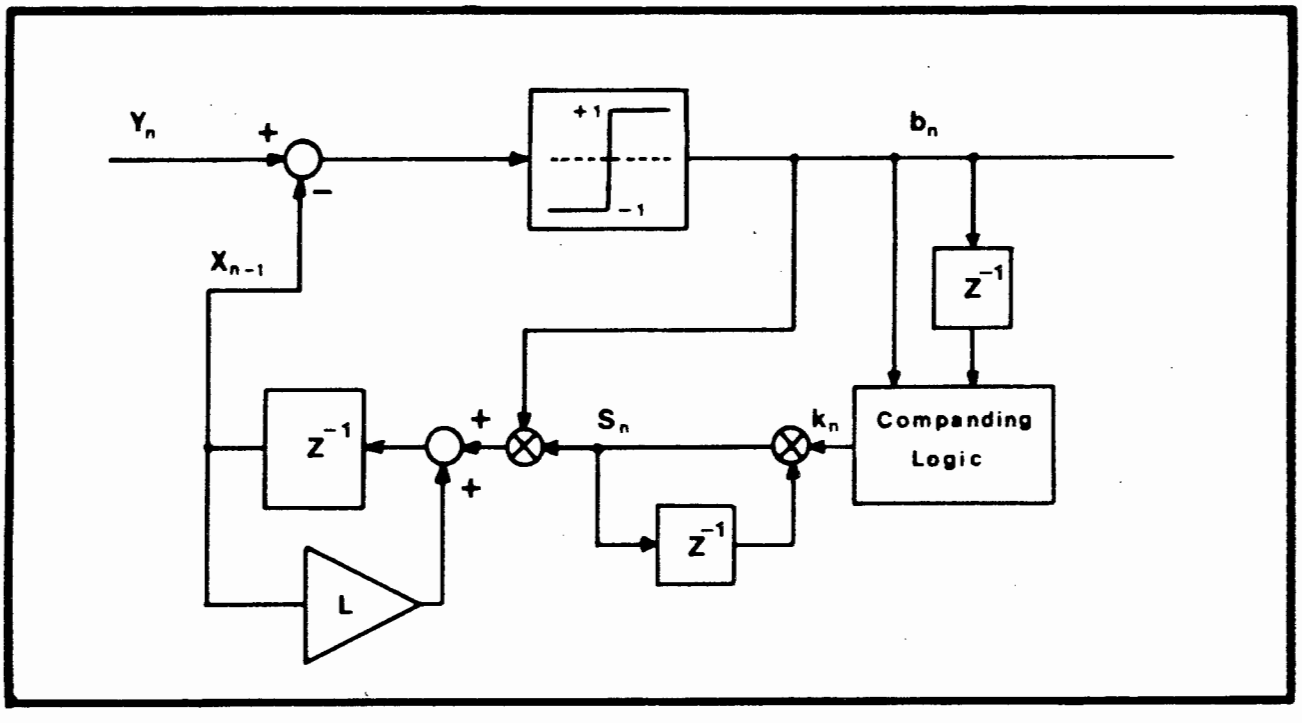


Fig. 3.5 : Block diagram of the CFDM codec.

Obviously the operation of the codec is highly dependent on the values of the time invariant factors P and Q. There are two conditions which must be taken into account when determining the values for P and Q. During the slope overload portion of operation ($b_n = b_{n-1}$) P must be large enough to respond to the input signal, while during the hunting phase ($b_n \neq b_{n-1}$) Q must be

small enough to enable the approximation signal to converge to the input signal. Linear DM, which can neither match the high slopes of some signals nor converge to a steady-state value, is equivalent to a CFDM codec with both P and Q set to unity. In order to reduce the amount of slope overload noise it is necessary that $P > 1$, while in order to enable the signal to converge to a steady-state value it is necessary that $Q < 1$. If the product $P.Q$ exceeded $(1 + \varepsilon)$, where ε is positive, then the adaptation would tend to become unstable. This is because the net effect of the multiplications by P and Q would be an increasing step size (the effect of P is greater than Q). The step size would eventually become so large that the output of the decoder would oscillate between the maximum and minimum values independently of the input signal. Hence, to ensure stability, the product $P.Q$ must be limited to a maximum value of 1. It has been determined[23] that the optimum condition is in fact $P.Q = 1$. It was also found, by means of computer simulation, that the optimum value for P was 1.5 and this was independent of the sampling frequency used[23].

In terms of a practical implementation the reciprocity of P and Q means that the possible step sizes within a certain range can only take on a limited number of discrete values. This means that a relatively compact step size 'dictionary' can be used. If, for example, the values for P and Q were 1.5 and 0.5 respectively,

some of the possible step sizes between 1 and 2.25 (inclusive) are :

1.000 1.125 1.265 1.424 1.500 1.688 1.898 2.136 2.250

whereas the optimum solution yields only three values in this range :

1.000 1.500 2.250.

Fig. 3.6 shows the difference between the two adaptive schemes discussed so far. It can be seen how, under severe slope overload conditions, the step size of the CFDM coder varies at a much faster rate than for CVSD. The CVSD coder represented here has values for H and V of 3 and 150 respectively. It should be remembered that the exact variation of step size for the CVSD coder depends on these values.

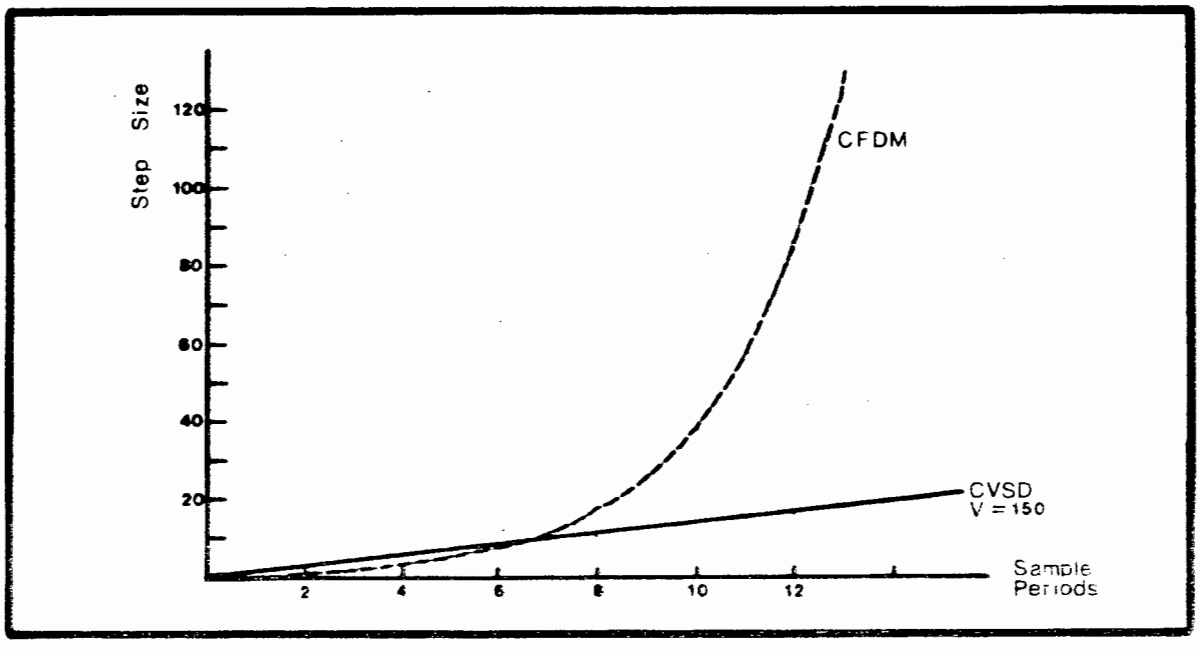


Fig. 3.6 : Variation of step size for CVSD and CFDM encoders.

3.2.4 Hybrid Companding Delta Modulation (HCDM).

Although the previous two methods provide a significant improvement over LDM they both have shortcomings related to the type of companding used (cf. chapter 2). The development then of a hybrid companding scheme that exploits both the syllabic and instantaneous characteristics of speech would appear to provide a potential improvement over both of the methods discussed previously.

The slow response of the syllabic compandors to the sharp changes in the speech waveform could be remedied by increasing the sampling rate, but that would be contrary to what we're trying to achieve. The problem with instantaneous companding is that the step size range is fixed, regardless of the actual speech signal characteristics. As a result of the variation of the input signal power and slope the step size range may be optimum for one phoneme of speech but not another. If the instantaneous compandor is given access to the syllabic information it would be possible for the step size range to adjust to the different speech phonemes. This is essentially what the HCDM encoder does. Instead of using a fixed basic step size the instantaneous compandor has a basic step size that is modified, according to the signal level over the syllabic period, by the syllabic compandor. The basic step size thus produced is then modified by the instantaneous compandor to

produce the actual step size used by the prediction filter. The block diagram of the HCDM encoder is shown in fig. 3.7.

The input sample, Y_n, is compared to the last output of the prediction filter (ie. the estimate, X_{n-1}, of the input). The step size is then modified according to the sign of the difference between the input signal and the estimate which is then increased or decreased using the new step size. The general algorithm for

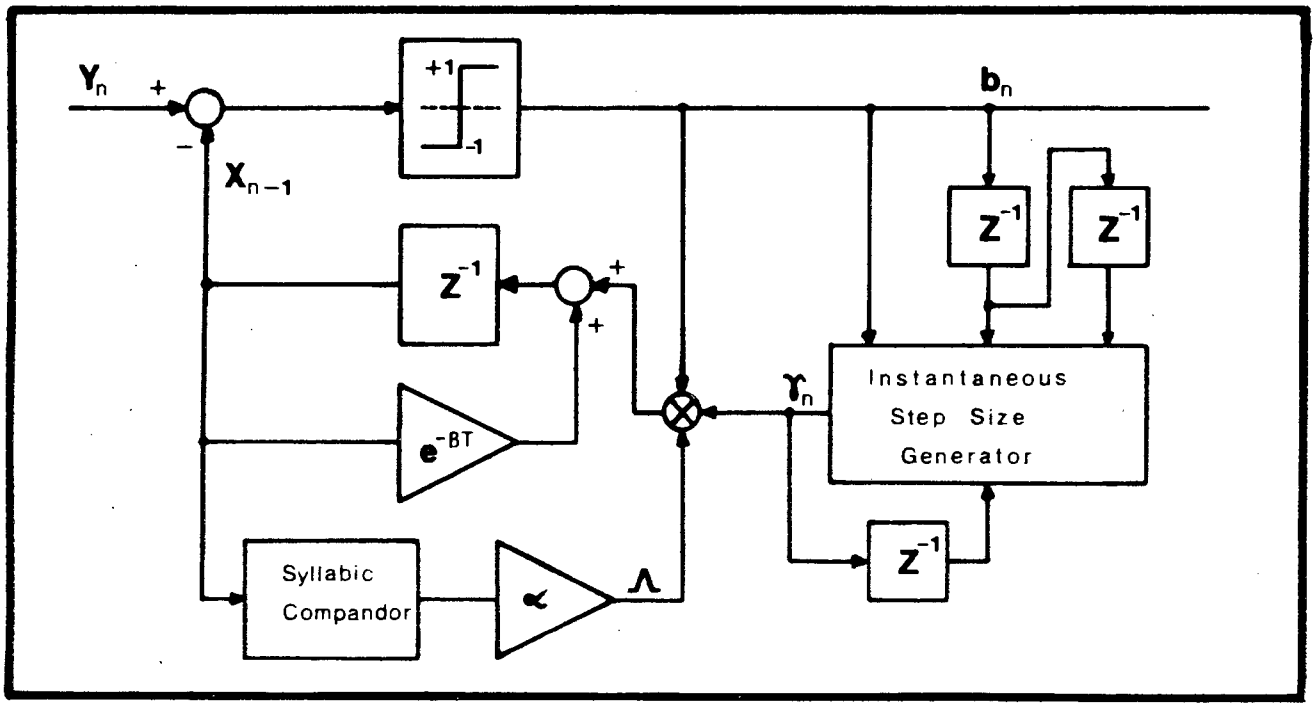


Fig. 3.7 : Block diagram of the HCDM encoder.

the HCDM encoder is given by[17,24]:

- The binary output is given by the difference between the input and estimate signals.

$$b_n = \mathrm{sgn}(Y_n - X_{n-1})$$

o The long-term basic step size is obtained from the RMS slope energy of the low pass filtered decoded signal. It is updated at the syllabic rate (ie. approximately every 5 msec.).

$$E = \left[\sum_{n=1}^{M-1} (X_n - X_{n-1})^2 / (M-1) \right]^{1/2}$$

$$\Lambda = \alpha E$$

where

X_n is the n^{th} predicted speech sample,

M is the number of samples in the syllabic period,

E is the RMS slope energy,

α is the slope energy scale factor, and

Λ is the long-term basic step size.

o The instantaneous step size adaptation is determined from the present output (ie. sign of the error) and the previous two outputs.

$$k_n = f(b_n, b_{n-1}, b_{n-2})$$

$$\gamma_n = k_n . \gamma_{n-1}$$

where

k_n is the instantaneous multiplication factor,

γ_n is the instantaneous step size modification factor, and

$f(b_n, b_{n-1}, b_{n-2})$ is specified by the logic rule shown in Table 3.1. It should be pointed out that the constant factor DM used here is a second order version of Jayant's

CFDM described in [23]. There are alternative versions of the logic rule shown here that, for instance, use the five most recent outputs (ie. b_n, . . . , b_{n-4}), but the difference in performance between these methods is not significant enough to warrant the extra hardware (refer to chapter 7).

Table 3.1 : The HCDM companding logic (instantaneous).

b_n	b_{n-1}	b_{n-2}	k_n
+1	+1	+1	1.50
-1	-1	-1	1.50
+1	+1	-1	1.00
-1	-1	+1	1.00
+1	-1	-1	0.66
+1	-1	+1	0.66
-1	+1	+1	0.66
-1	+1	-1	0.66

o The step size, S_n, is updated by combining the syllabic and instantaneous factors.

$S_n = \Lambda . \gamma_n$

o The estimate of the input sample is then given by :

$X_n = L . X_{n-1} + b_n . S_n$

where

L, the prediction constant is given by :

$L = \exp(-\beta T)$ with β being the inverse of the prediction time constant, and T being the sampling period.

o NOTE : The slope energy, E, is determined from the decoded speech signal. If instead the actual input signal was used

the slope information would have to be transmitted to the decoder. However, it has been found[17] that the advantage offered by the feedforward mode is minimal compared to the performance of the feedback scheme. Secondly, the syllabic companding could be obtained from the variation of the signal amplitude rather than the slope, but because the delta modulator essentially tracks the input slope, as opposed to the amplitude, the use of the slope information was preferred[17].

3.2.5 Song Voice Adaptive Delta Modulation (SVADM).

The SVADM scheme derives from an analytic, as opposed to empiric, approach to optimizing the transmitter and receiver. The design uses the fact that, in general, the optimum predictor (in the transmitter) is different to the optimum estimator (in the receiver)[16]. A set of non-linear equations was derived for a theoretically optimum system and these equations were then reduced to piecewise linear equations to enable a digital implementation of the system. For a detailed derivation of these equations refer to Appendix A. Initially the system was developed for a Markov-Gaussian source and then optimised for voice signals.

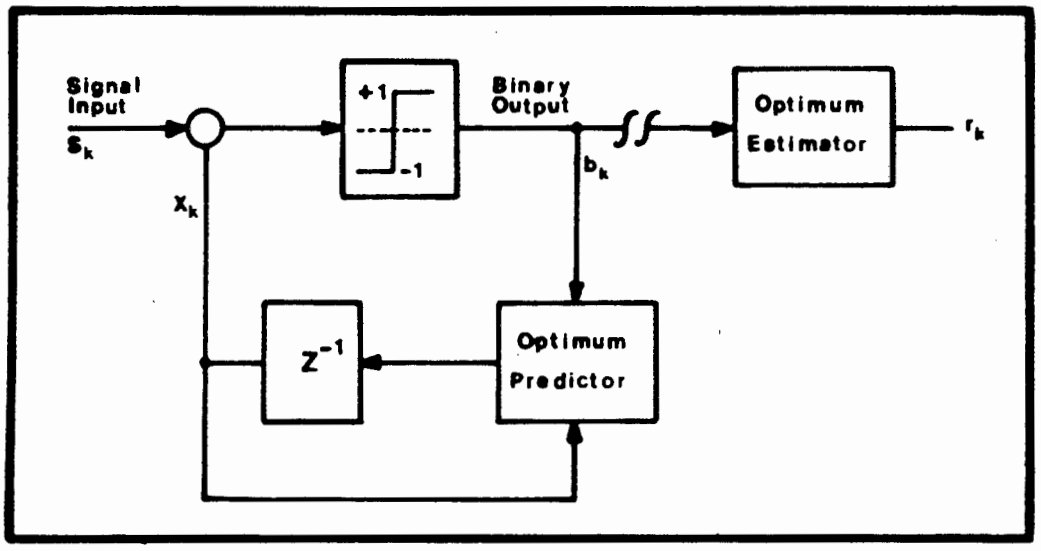


Fig. 3.8 : Optimum adaptive delta modulation system.

The general equation describing the song system is given as :

$$r_k = X_{k+1}$$

$$r_k = X_k + g(b_k, X_k - X_{k-1}) + h(b_{k-1}, X_k - X_{k-1}).$$

From the derivation given in Appendix A,

For $X_k - X_{k-1} > 0$:

$$g(b_k, X_k - X_{k-1}) = +0.08 + U.(X_k - X_{k-1} - 0.08) \quad b_k = +1$$

$$-0.08 - U.(X_k - X_{k-1} - 0.08) \quad b_k = -1$$

For $X_k - X_{k-1} < 0$:

$$g(b_k, X_k - X_{k-1}) = +0.08 - U.(X_k - X_{k-1} + 0.08) \quad b_k = +1$$

$$-0.08 + U.(X_k - X_{k-1} + 0.08) \quad b_k = -1,$$

and

$$h(b_{k-1}, X_k - X_{k-1}) = 0.04 * b_{k-1}.$$

The graphs of the functions g and h are plotted in fig. 3.9.

From fig. 3.9 it can be seen that for speech the step size

function generator is linear except for the region near the origin where the minimum step size is limited to prevent a 'dead' zone. The above set of equations was originally developed for a system using an 8-bit A/D converter at the input with an input signal range of -5V to +5V. This yields a minimum quantization step of 40 mV, and so, these equations can be combined and rewritten for the general system :

$$S_{k+1} = |S_k| . b_k + S_0 . b_{k-1}$$

where

S_{k+1} is the new step size, and

S_0 is the minimum step size (corresponding to the 40 mV step in the 8-bit system).

The updated approximation to the input signal can then be written as :

$$X_{k+1} = X_k + S_{k+1} .$$

These two equations then characterize the SVADM system for speech signals. The form of these equations makes it very simple to implement the circuit using digital integrated circuits (cf. chapter 4).

The system specified by the equations includes the equivalent of a perfect integrator in the prediction filter. However, as has been mentioned before, the use of a leaky integrator serves the dual purpose of enhancing the performance for voice signals as well as

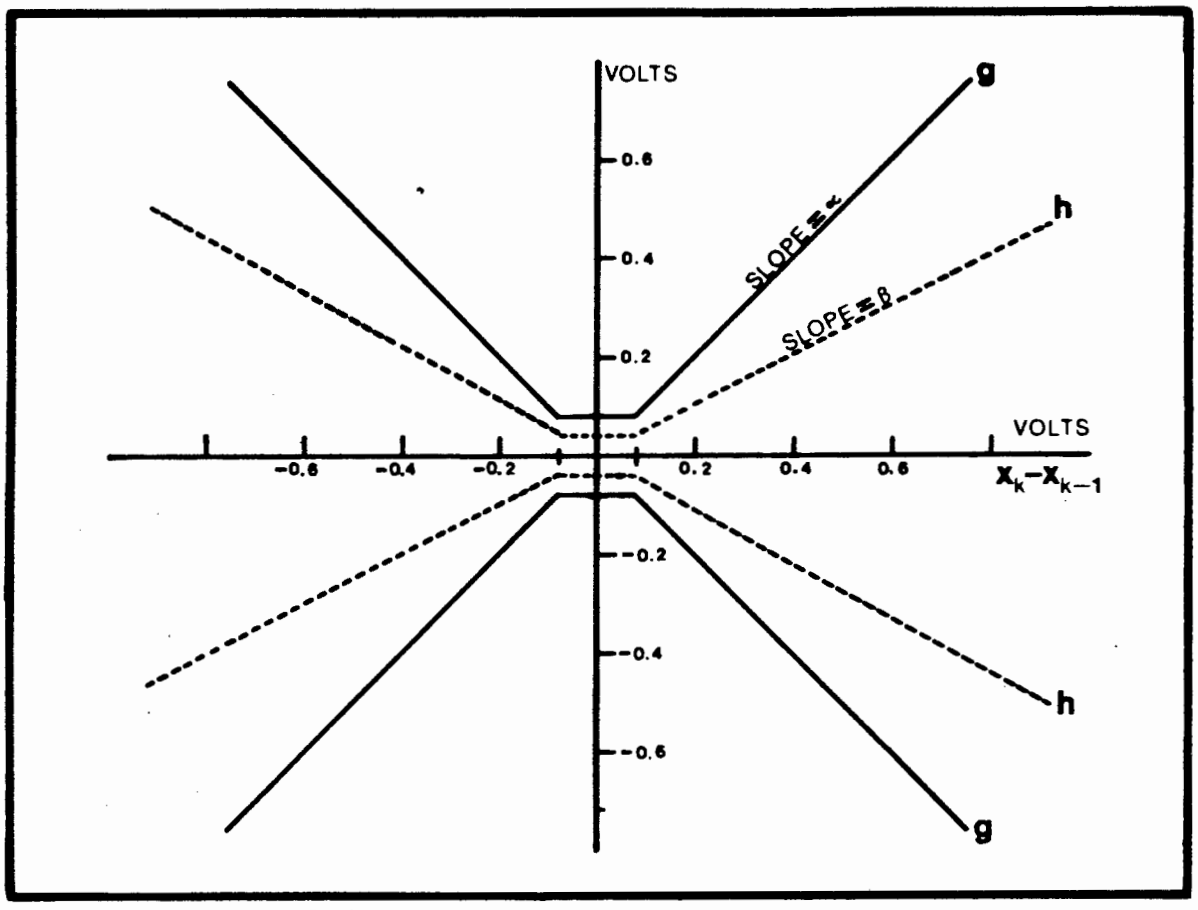


Fig. 3.9 : Graphs of functions g and h.

enabling the decoder to recover from channel errors. The value of L, the prediction constant, is set according to the sampling frequency. However, for a practical hardwired implementation it

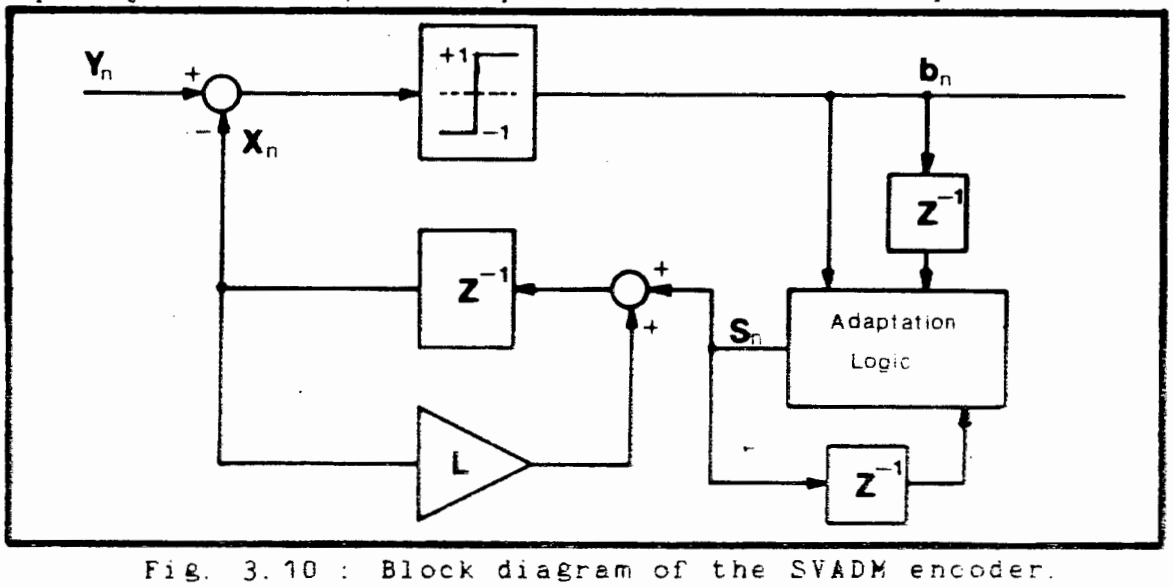


Fig. 3.10 : Block diagram of the SVADM encoder.

is sometimes desirable to choose one value for all sampling rates, as was done by Dhadesugoor et al[21]. Alternatively, the prediction constant can be determined from the RC-equivalent filter as was done for the CVSD, CFDM and HCDM systems.

3.2.6 Hybridisation of the Song Voice Adaptive system.

Essentially the Song Voice ADM is a delta modulator initially designed for a first-order Markov-Gaussian source and then empirically optimized for speech signals. However, the non-stationarity and quasi-periodicity of speech prevent the speech source from being classified as first-order Markov-Gaussian. Just as a combination of syllabic and instantaneous companding was found to improve the performance of the CVSD and CFDM codecs, so it would seem that by incorporating particular characteristics of speech signals into the design of the codec the performance of the Song Voice system could be enhanced. We have investigated means of modifying the design of the SVADM system to include syllabic companding, thereby incorporating the non-stationary characteristics of speech in the codec. In the course of our research we have developed two techniques for achieving this and they are referred to as 'Song Hybrid Companding Delta Modulation' (SHCDM) and "Modified Song Hybrid Companding Delta Modulation' (MSHCDM).

Both of these techniques utilise the same types of companding with the difference being in the method by which the instantaneous and syllabic companding factors are combined. The types of companding used are CVSD for the syllabic, and SVADM for the instantaneous.

3.2.6.1 Song Hybrid Companding Delta Modulation.

The block diagram of the SHCDM encoder is shown below in fig. 3.11.

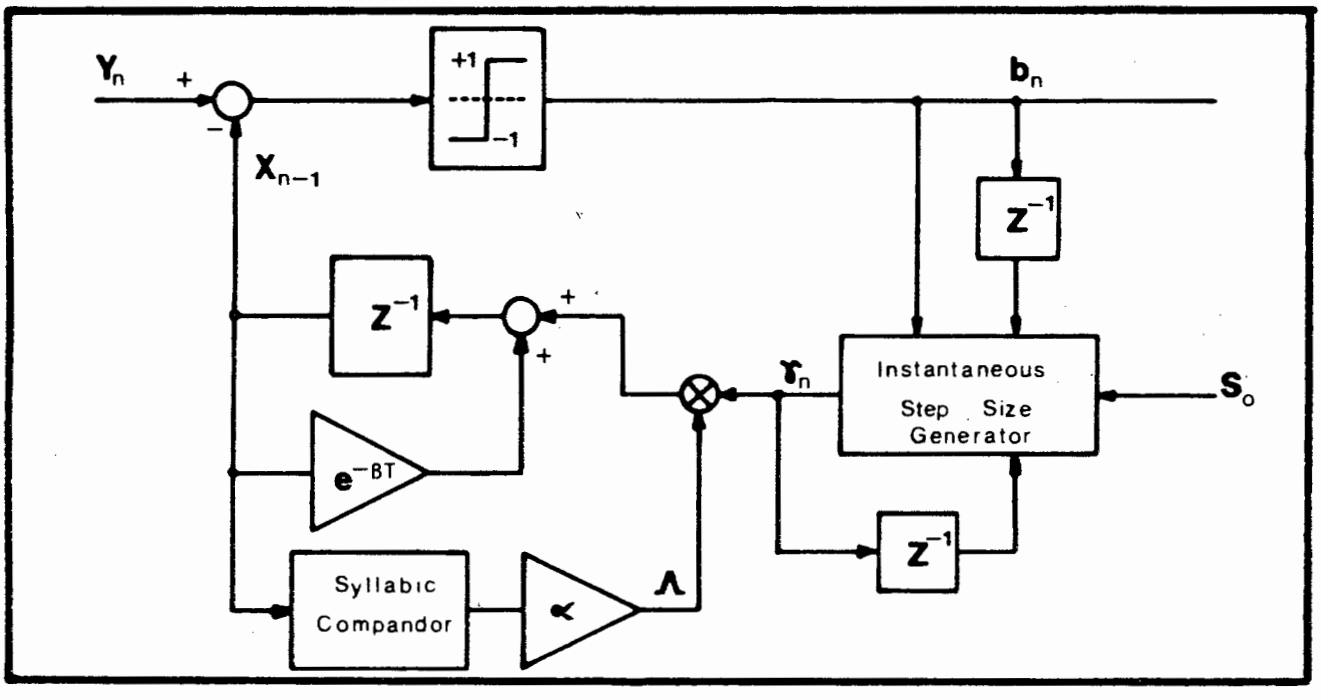


Fig. 3.11 : Block diagram of the SHCDM encoder.

The associated algorithm is given by :

o The binary output is determined from the sign of the difference between the input and approximation signals.

$b_n = sgn(Y_{n+1} - X_n)$.

- o The instantaneous step size adaptation is determined in the same manner as for SVADM.

  $\gamma_{n+1} = |\gamma_n| . b_n + S_0 . b_{n-1}$

  where

  S_0 is the basic step size.

- o The syllabic step size is obtained from the RMS slope energy of the decoded speech signal.

  $$E = [\sum_{n=1}^{M-1} (X_n - X_{n-1})^2 / (M-1)]^{1/2}$$

  $\Lambda = \alpha . E$

  (For an explanation of the symbols refer to the section on HCDM - 3.2.4.)

- o The step size is then obtained by combining the syllabic and instantaneous factors.

  $S_{n+1} = \Lambda . \gamma_{n+1}$

- o The updated approximation to the input signal is then given by :

  $X_{n+1} = L . X_n + S_{n+1}$

  where

  L is the prediction constant.

In this version of the hybridised Song Voice encoder the instantaneous step factor is generated using a fixed basic step (instantaneous) step size. This step factor is then used to modify the

basic step size of the syllabic compandor, thereby generating the hybrid step size.

3.2.6.2 Modified Song Hybrid Companding Delta Modulation.

Whereas the SHCDM encoder uses instantaneous companding to modulate the basic syllabic step size, the MSHCDM encoder uses syllabic companding to modify the basic step size of the Song Voice encoder. From fig. 3.12 it can be seen that the fixed basic step size (S_0) of the instantaneous compandor has been replaced by the syllabically generated step size (Λ).

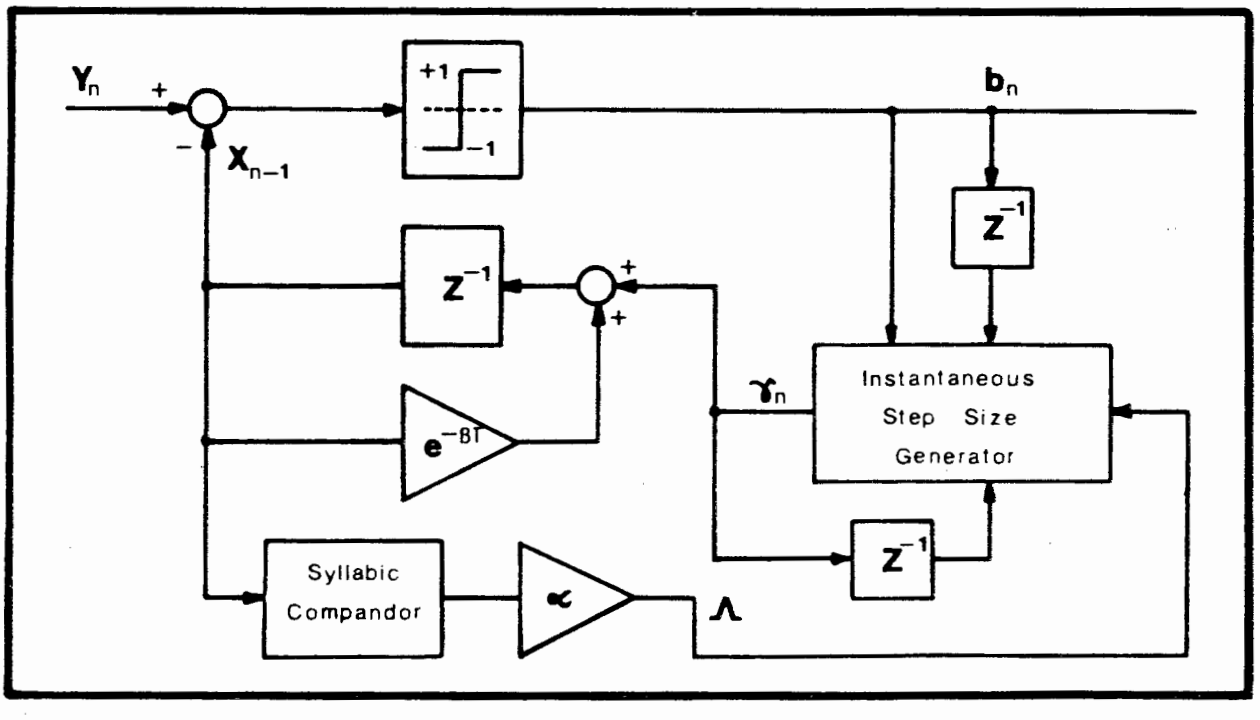


Fig. 3.12 : Block diagram of the MSHCDM encoder.

The algorithm for the MSHCDM encoder is given by :

o The syllabic factor :

$\Lambda = \alpha . E$, as before.

o The step size is then generated using Song Voice companding with the basic step size provided by the syllabic compandor.

$S_{n+1} = |S_n| . b_n + b_{n-1} . (\Lambda . S_0)$

where

S_0 is generally set to unity.

o The approximation signal is then given by :

$X_{n+1} = L . X_n + S_{n+1} .$

CHAPTER 4

HARDWARE IMPLEMENTATION OF A SONG VOICE ADAPTIVE DELTA MODULATOR

During the early stages of this research it was envisaged that the major portion of the work would be concerned with network issues rather than the design of an efficient speech codec (the initial title for the project was "Voice Packet Switching" and involved the development of a suitable protocol). Owing to this our choice of a speech codec was based primarily on simplicity and ease of implementation. As far as was reasonably possible we wished to avoid any design and development work on a speech codec, it being preferable to utilise an already developed and tested algorithm.

Delta modulation was chosen as the means for providing a low cost digital speech codec (the motivation for the choice follows similar lines to those outlined in chapter 3) and in particular, the Song Voice Adaptive Delta Modulator (SVADM) was implemented. This codec was chosen in preference to the more conventional Continuously Variable Slope Delta Modulator (CVSD) owing to its reputed better performance and the fact that in a packet switched network it was found to be better than CVSD[21].

Although the SVADM algorithm has been presented previously (cf. chapter 3, section 2.5) it is repeated here for ease of reference.

$$b_n = \mathrm{sgn}(Y_{n+1} - X_n)$$

$$S_{n+1} = |S_n| . b_n + S_0 . b_{n-1}$$

$X_{n+1} = X_n + S_{n+1}$

where

S_{n+1} is the new step size, and

X_{n+1} is the updated approximation signal.

The difference between the encoder and decoder is that the decoder does not include the circuit to generate b_n, the error signal. Referring to fig. 3.10 the decoder is equivalent to the local decoder contained in the feedback loop of the encoder. In terms of hardware the encoder and decoder are identical except for the comparison section of the error generation circuit (described below).

This algorithm is simple to implement using commercially available digital integrated circuits. As with the codec implemented by Dhadesugoor et al[21], 10-bit arithmetic is used with an input signal range of -10 V .. +10 V. This gives a minimum step size, S_0, of approximately 10 mV. The design of the circuit has been divided into three discrete sections based on fig. 3.10 - these are the error generation and quantization section, the step size generator, and the approximation signal generator.

The main components of the circuit are the 10-bit D/A converter, the adders and latches. Simple logic components, such as

exclusive-OR gates, are also used. In order to implement the basic algorithm a total of 25 integrated circuits (including the ADC, comparator and amplifier chips) were used. Although digital speech coding techniques have been used some portions of the circuit were implemented using analogue techniques (the reasons are discussed in the relevant section of the text). For a detailed circuit design with component values refer to Appendix B.

4.1 Error Generation and Quantization.

The simplest means of achieving this function is to compare the analogue versions of the input signal, Y_{n+1}, and the approximation signal, X_n, by means of an analogue comparator. The analogue comparator is easily implemented using an operational amplifier.

The output of the comparator is sampled and quantized using a D-latch circuit.

An alternative technique for determining the error signal entails using an A/D converter (with its associated Sample-and-Hold module) and a digital comparator. Bearing in mind that 10-bit arithmetic is used and that the A/D converter would have to work at a sampling rate of 8 kHz (ie. the maximum conversion time would have to be in the region of 70 µsec) it was decided, for the initial design, to use the simpler and cheaper analogue comparison

method.

Because the remainder of the circuit is implemented using digital techniques it is necessary to D/A convert the approximation signal, X_n, to facilitate the analogue comparison.

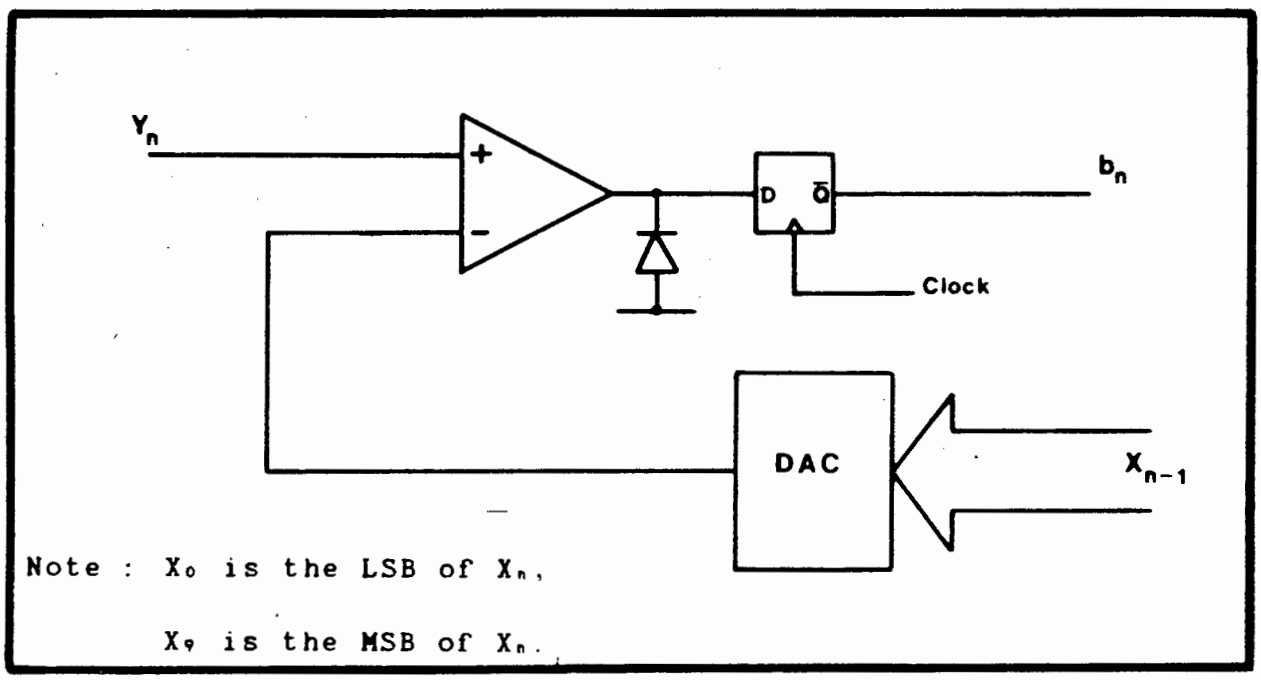


Fig. 4.1 : Error generation circuit block diagram.

The diode on the comparator output is used to convert the output from a bi-polar one to a uni-polar one switching between the positive supply rail and ground. The voltage divider is used to scale the comparator output to a TTL-compatible level.

This circuit is responsible for the generation of the error signal :

$$b_n = sgn(Y_{n+1} - X_n).$$

4.2 Step-size Generation.

The increment to the step size, S_0, is 10 mV, which corresponds to a digital value of '1' (the full range for 10-bit arithmetic is -512 . . . +511). This is multiplied by the sign of the previous error, b_{n-1} (ie. the digital step increment is ± 1). Two's complement arithmetic is used throughout. If the comparator output switches to the positive supply rail (ie. the previous approximation signal was too small) then the step increment is +1, else it is -1.

The most involved function of the step-size generation circuit is the circuit to achieve $|S_n| . b_n$. The possible options for this operation are :

- o $b_n = +1$

  This implies $|S_n| . b_n = S_n$.

  Hence, for $S_n < 0$, $|S_n| = -S_n$

  while, for $S_n > 0$, $|S_n| = S_n$.

- o $b_n = -1$

  In this case, for $S_n < 0$, $|S_n| . b_n = S_n$,

  while, for $S_n > 0$, $|S_n| . b_n = -S_n$.

Hence, using the sign-bit of the step-size word, S_9, and the bit representing b_n, the value for S_n is multiplied by -1 if

$S_9 \oplus b_n = 1 \equiv C$ (cf. fig. 4.2).

NOTE : Because two's complement arithmetic is used a positive sign is represented by '0' and a negative sign by '1'.

ie. b_n = +1 is represented by 0,

b_n = -1 is represented by 1.

In two's complement arithmetic negating a number is achieved by inverting all the bits of the word and adding 1. This is realised by using the control bit, C, and an exclusive-OR gate and an adder.

ie. $-S_n \equiv (C \oplus S_n)$ PLUS 1.

This can be generalised to :

$|S_n| . b_n \equiv (C \oplus S_n)$ PLUS C.

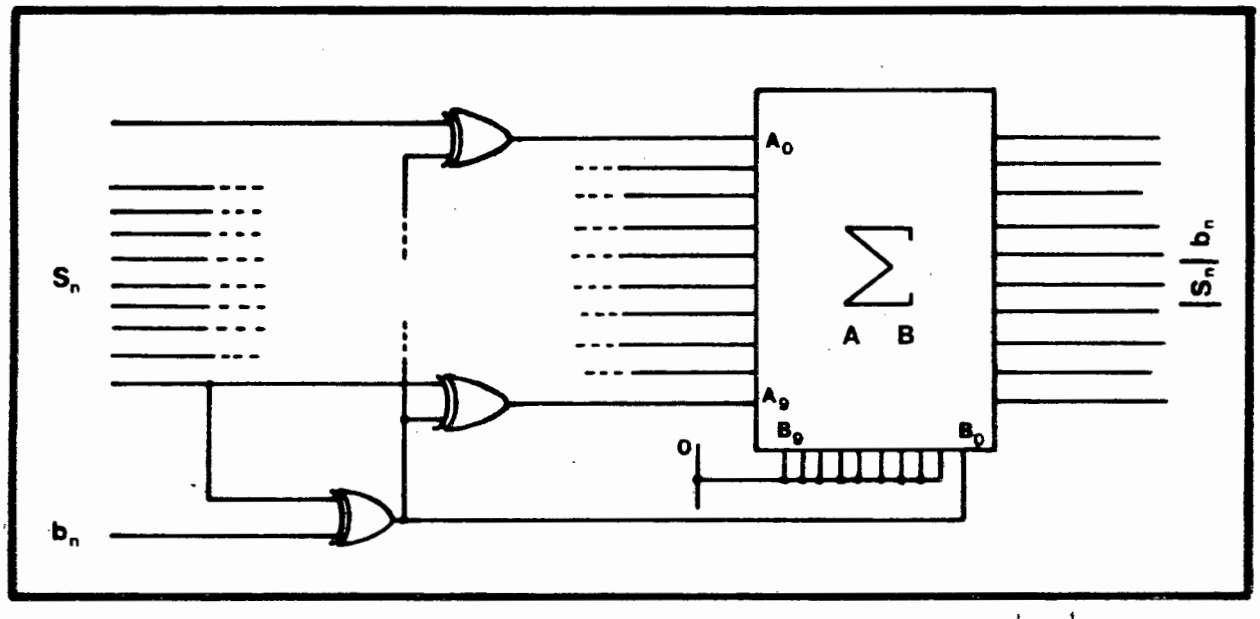


Fig. 4.2 : Circuit block diagram for achieving $|S_n| . b_n$.

The value obtained in this way is then simply added to the increment to produce the new step-size.

It should be pointed out at this stage that although 10-bit arithmetic is used it is possible that certain functions (eg. the step-size generation) could be implemented using fewer bits. The minimum word length would be determined from the maximum value of the particular variable in question. For example, in the case of the step-size, if the maximum value of this variable under normal

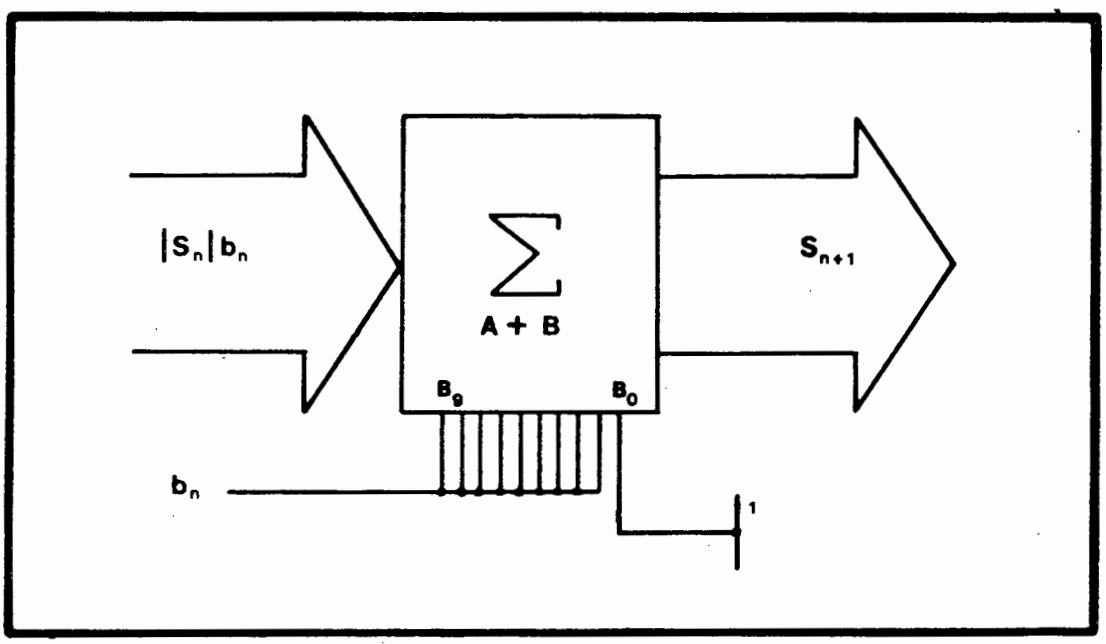


Fig. 4.3 : Generating the new step-size S_{n+1}.

operating conditions did not exceed 128 then the step-size generation circuit could be implemented using only 8-bit arithmetic. This in turn could have an effect on the chip count (eg. two 4-bit adders are needed for an 8-bit adder whereas 3 chips would be necessary for a 10-bit adder). Hence, it would be necessary to study the characteristics of the variables in order

to optimize the hardware design.

4.3 Approximation-signal Generation.

The updating of the approximation signal is achieved by simply adding the new step-size to the most recent value of the approximation signal. Although not included in this implementation the performance can be improved by using a leaky integrator to generate the approximation signal (the effect of this on the performance is shown in chapter 7). The approximation signal must then be D/A converted to facilitate the comparison with the next input sample.

4.4 Timing Considerations.

Referring to the detailed circuit diagram shown in Appendix B, it can be seen that three clock signals are used. The reason for this is to prevent race conditions being caused by two interactive circuit units changing state at the same time. For example, if the comparator output is latched at the same time as the D/A converter output is being updated then b_n could take on spurious values unrelated to the actual comparator output.

To prevent these race conditions the comparator output must only be latched once the DAC (D/A Converter) output has been updated. Similarly, the DAC can only be clocked once the approximation

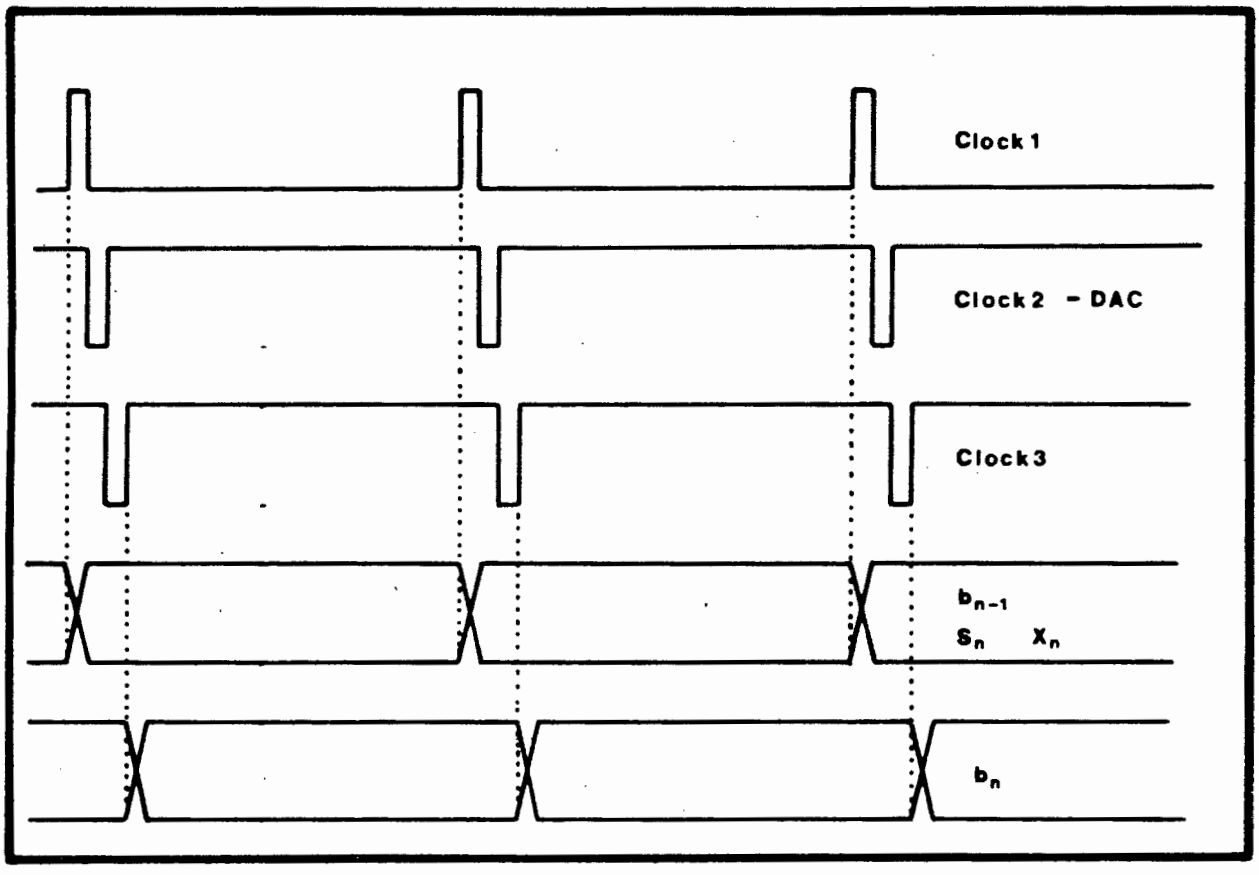


Fig. 4.4 : Codec timing diagram.

signal has been updated. The previous error signal (and previous output), b_{n-1}, must be latched before the new comparator output is latched. If the step-size value and the approximation signal value, S_{n+1} and X_{n+1}, are clocked simultaneously the propogation delay, particularly in the step-size generation circuit, will prevent any race conditions occuring at the X_{n+1}/X_n latch. Hence, there are three sets of functions which must be clocked seperately : Clock 1 : triggers the latches for b_{n-1}, S_n, and X_n.

Clock 2 : triggers the DAC.

Clock 3 : triggers the latch for b_n.

The complete timing diagram for the codec is shown in fig. 4.4.

4.5 Network Interfacing.

The next development stage involved interfacing the codec to a microprocessor as this would facilitate integrating voice communication into a computer communications network. For Local Area Networks (LAN's) in particular the advantages in terms of network flexibility are significant (eg. a single communications link could serve both the voice and data needs of a number of users). The microprocessor would be responsible for the interfacing between the speech codec and the digital communications network (which would involve the packetization and depacketization processes). A block diagram, fig. 4.5, outlines the major components of the interface.

The system shown in fig. 4.5 is a very simple one in that it has the capability of linking only two codec terminals (Tel. A & B) whereas the real system would have to handle a considerable number of telephones. However, the network shown below was sufficient to implement a first-stage design - ie. a simple full duplex two-way link between two speech terminals with a transparent network interface - which would enable the testing of the codec. It was initially intended to extend this system so that the microprocessor be utilised as a PAD (Packet Assembler Dissassembler) and exploratory work was done in this direction[39] (under the author's supervision).

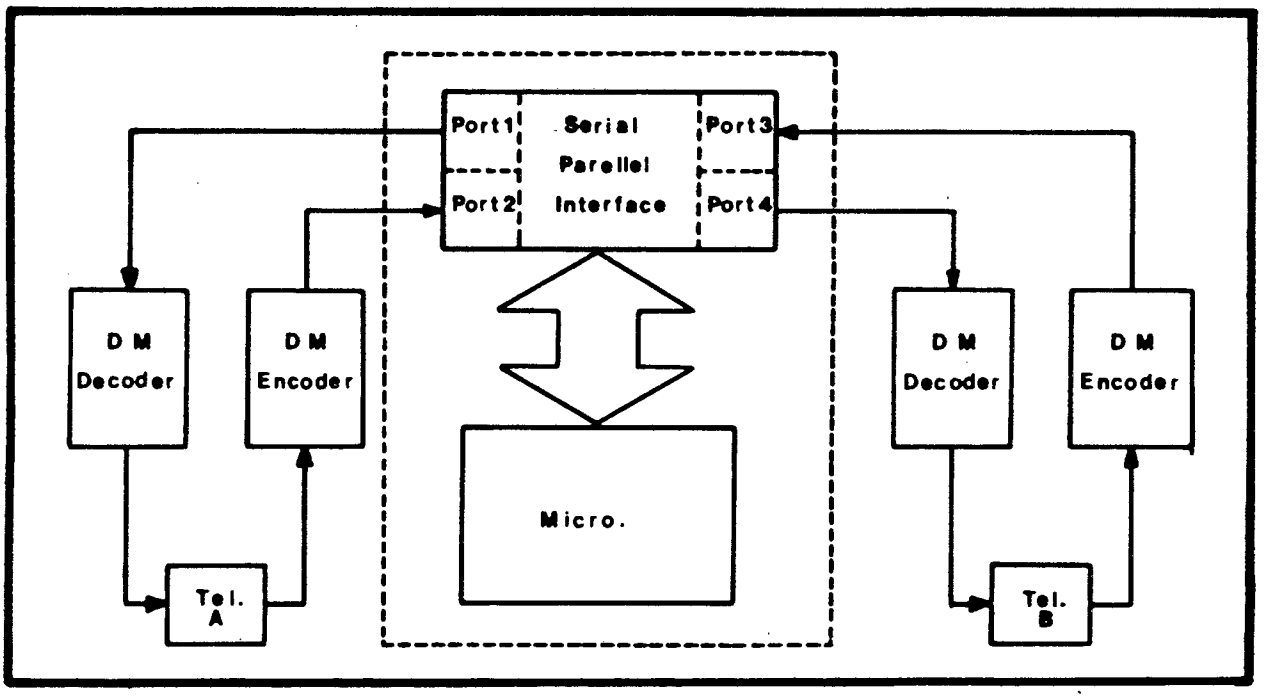


Fig. 4.5 : The communications system used for testing.

4.5.1 Serial/Parallel Interface.

The delta modulator encoder outputs a serial bit stream at a fixed rate. Similarly, the input to the decoder is also a serial bit stream. On the other hand, the microprocessor outputs and inputs parallel bytes of data. The primary function of this interface is to convert between the serial and parallel data modes.

Basically the interface consists of a standard SABus Parallel I/O unit[40] with a few modifications and extensions. The serial/ parallel conversion is achieved using shift registers with a byte synchronised clock controlling the transfer to and from the microprocessor's memory. Two input and two output channels are provided and hence, two codecs can be interfaced. All the timing signals, including the three clock signals required for the codecs

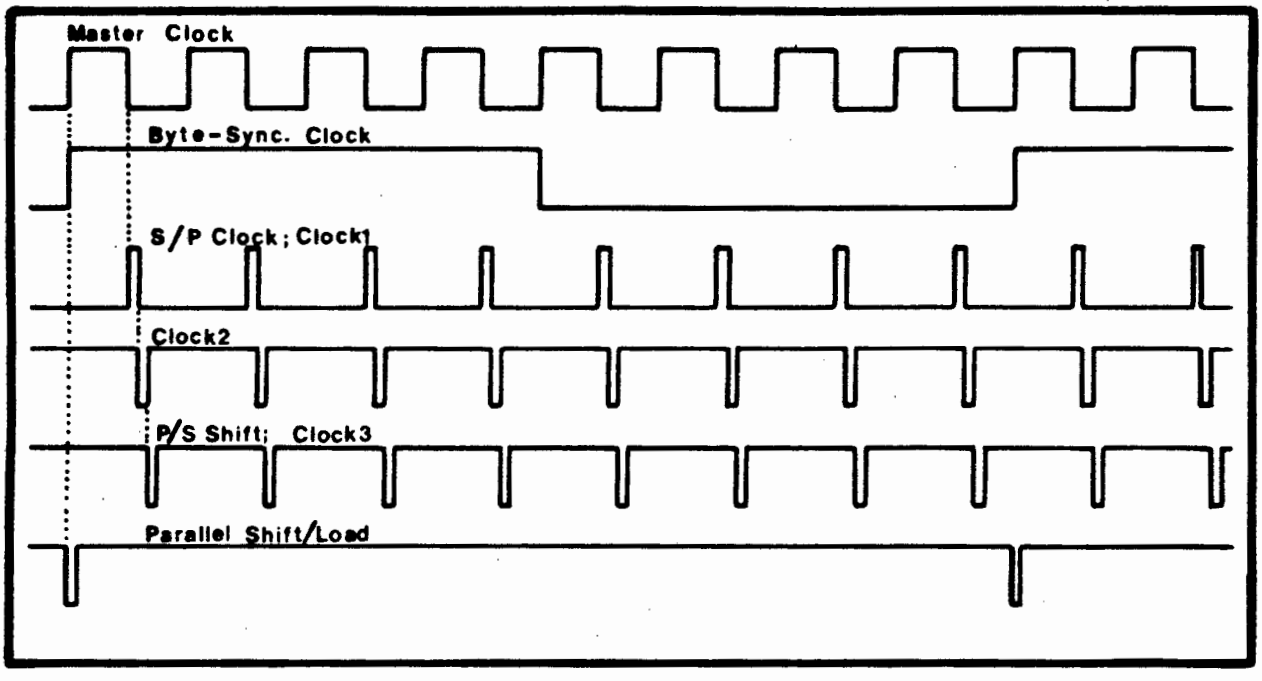


Fig. 4.6 : Interface timing diagram.

(cf. section 4), are generated by this unit.

As with the codecs themselves, the timing specifications are crucial for the interface. The serial/parallel shift registers cannot be updated at the same time as b_n, and neither can they be updated when data is being transferred to/from the microprocessor. The timing diagram for the I/O card is shown below with the block diagram shown in fig. 4.7 (for a detailed circuit diagram refer to Appendix B).

The software required to implement the straight-through full duplex link on the microprocessor was written on the PDP 11/23 computer in 8085 Assembler code and downloaded to the microprocessor. The algorithm flow charts and the control software are

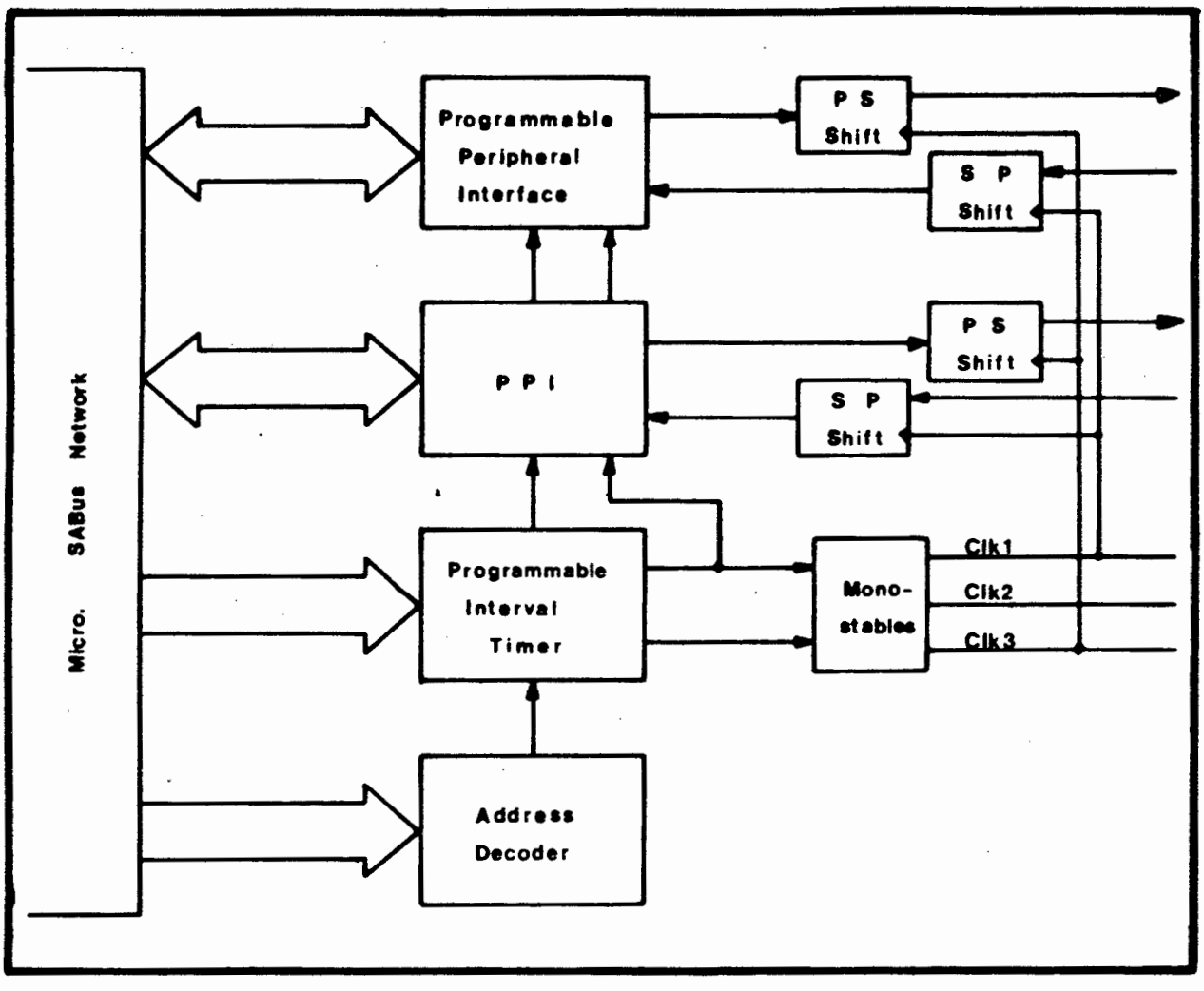


Fig. 4.7 : Block diagram of the interface unit.

listed in Appendix C.

Once the system described above had been implemented it was necessary to test it and to verify the performance of the SVADM codec (as described by Dhadesugoor et al[21]. Because of the dynamic nature of speech signals the determination of the various objective measures (as described in chapter 6) becomes problematic without suitable data storage facilities. At this point the emphasis of this research had swung towards investigating possible

codecs for speech that could be implemented as a one, or two chip low cost delta modulator set. It was therefore necessary to optimize the design with respect to the various circuit parameters, word lengths (eg. the word lengths necessary for the variables, such as the step size, of the design algorithm), etc. Rather than build hardwired versions of the modified codecs it was decided to implement a computer simulation facility which would enable the design to be optimized via the software, and only when the optimum design had been finalised would it be committed to a hardware implementation, and later to a chip.

One of the reasons for implementing an integrated circuit version of the speech codec is that the codec implementation using discrete chips required two SABus cards to contain the circuitry. Given that in a commercial communication network these cards would have to be duplicated at each terminal it is clear that an integrated circuit version would be both more economical and physically more acceptable.

The simulation process has the added advantage of enabling the performance evaluation to be readily achieved using software. The simulation system is described in the next chapter.

CHAPTER 5

COMPUTER SIMULATION

The process of testing and optimising the design of the codecs has been made considerably easier by the implementation of a simulation package on a minicomputer. This simulation package does not incorporate any Computer Aided Design (CAD) facilities requiring the basic design of the codecs to have been established prior to using this system. In essence this is a Computer Aided Test (CAT) station which enables the user to optimise and evaluate the codec design before implementing it in hardware.

The simulation package that was designed for this research project is divided into four main sections.

- o Interfacing : In order to simulate a speech processing system it is necessary to be able to input and output speech signals. This section is responsible for the interfacing between the analogue speech source (the talker/listener) and the digital processing software.
- o Codec simulation : Essentially the central feature of the system, this section simulates the speech codec algorithms which were discussed previously in chapter 3.
- o Measurement : The various objective quality measures, discussed in chapter 6, are determined arithmetically by this section.
- o Peripheral : Because of the speed limitations of the hardware of the simulation system (ie. the computer and

interfacing hardware) a unit to perform digital interpolation of the sampled data was included. Another feature of this section is a digital filtering package.

Fig. 5.1 shows the structure of the simulation package. The four sections listed previously are subsequently individually discussed in detail.

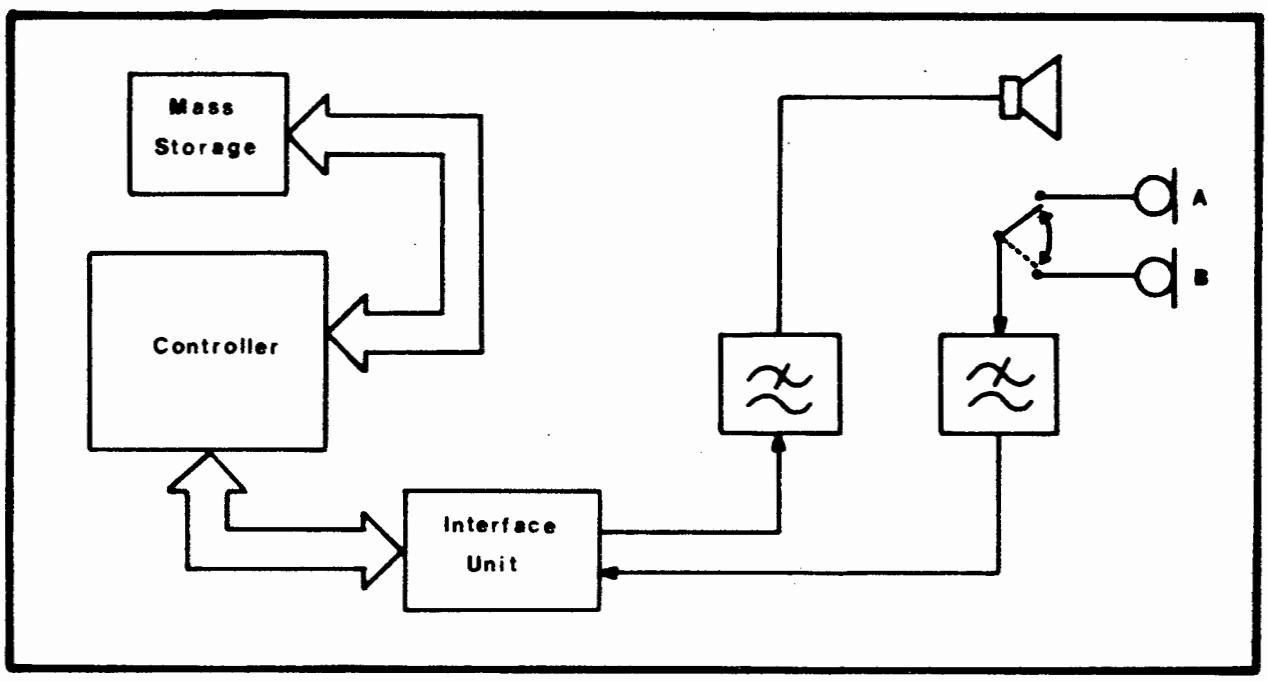


Fig. 5.1 : Block diagram of the simulation system.

The computer used for the simulation was the Hewlett Packard 200 Series HP9836C which is designed around a Motorola 68000 microprocessor. The interface module shown in fig. 5.1 was implemented using the HP6942A Multiprogrammer which has the facility for various plug-in sub-modules (eg. Analogue-to-digital converters, etc.). The hard disc storage unit was used as a mass storage

device (ie. for storing raw and processed speech data). The simulation software was written in BASIC and PASCAL - the control software for the interface unit was written in BASIC because, in order to access the sub-modules in this unit extended addressing is needed and this facility is not supported by the PASCAL available on the computer used for this project. The software responsible for processing the speech data (ie. the software simulating the actual codecs and the software performing the measurement and peripheral functions) was written in PASCAL and compiled into assembler code in order to afford the maximum processing speed. A slight increase in processing speed (at the expense of considerable extra effort) could be achieved if the code was originally written in assembler. This is due to the fact that the PASCAL compiler is not 100% efficient, but because the processing speed is not critical the minimal improvement achieved is not sufficient to warrant this course of action.

The procedure for using the simulation package is described in detail in Appendix D, suffice it to say that it was designed to be user-interactive. The listings of the software programs and their associated algorithm flow-charts are given in Appendix E.

5.1 Interfacing.

Quite simply the interfacing involves converting the speech

information from analogue form to digital form and vice versa. In the first case, the talker speaks into the microphone. The signal from the microphone is then filtered by analogue means, sampled, quantized, digitized and transferred to the computer. Speech data stored in the computer can be sent to a listener via a digital-to-analogue converter, an analogue filter and a loud-speaker.

Two types of microphone were used. For the subjective analysis of quality (cf chapter 6) the mouthpiece of a standard telephone handset was used whereas, for the objective analysis a high quality microphone was used. The frequency responses of the two microphones used are shown in fig. 5.2. For the subjective

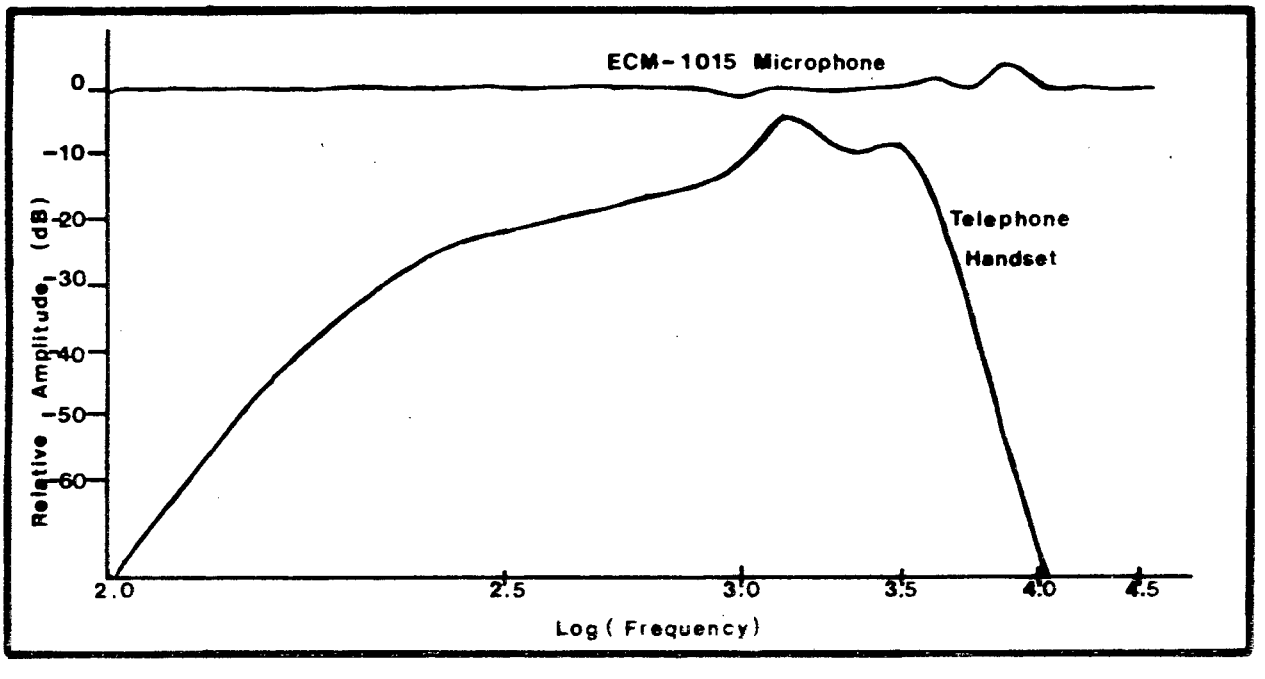


Fig. 5.2 : Frequency responses of the microphones used in the simulation system.

analysis the earpiece of the telephone handset was used to provide a measure of the quality of the different codecs in a realistic system (it is envisaged that the codecs under consideration would be incorporated into a LAN in which the terminals use standard telephone handsets. For the objective tests a high quality microphone was used in order to allow direct comparisons with previously published results.

The filter used on the input was an 8-pole Butterworth filter with a cut-off frequency of 2.5 kHz (f_c = 2500 Hz). On the output a 6^{th}-order Butterworth low-pass filter was used, also with a cut-off frequency of 2.5 kHz. For the subjective tests the input filter was replaced by a 6^{th} order band-pass filter (300 - 2500 Hz) because in public communications systems it is common to multiplex a number of band-pass filtered signals into a single co-axial cable channel and this is typically the pass band used.

The purpose of the input filter is to prevent aliassing effects. The purpose of a band-pass filter on the input is to conform to general communications standards while a low-pass filter on the input enables the objective results to be compared to previously published results.

The output filters serve the dual purpose of removing the

out-of-band noise, produced by the codecs and the A/D converter, and converting the discretely sampled-data to a continuous wave.

The sub-modules used in the Multiprogrammer include A/D, D/A converters, buffer memory and a timer module.

- o Timer module : this was programmed to operate at the required sampling rate of 8 kHz (actually, a sampling rate of 16 kHz was required but the interfacing hardware could not match the requirements, hence the sampling rate specified). All other sub-modules are controlled by the timing signals generated by the timer. The transfer of data from the multiprogrammer to the controller (ie. the computer) is, essentially, controlled by the timer as well.

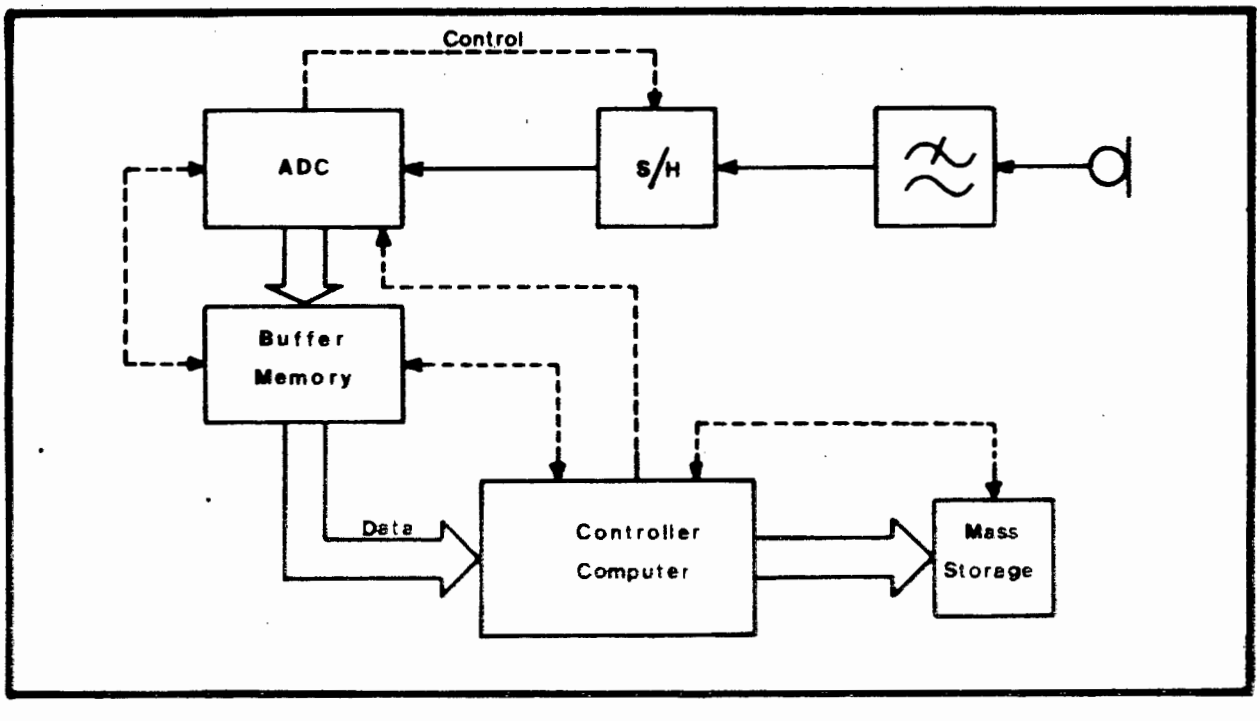


Fig. 5.3 : Block diagram of the A/D system.

o Buffer memory : during the input phase the data from the A/D converter is clocked into the buffer memory by the timer. When a block of data, of a size specified by the software, has been clocked into the buffer an interrupt is generated which then initiates the transfer of data from the buffer memory to the computer. While the transfer is in progress the sampled and quantized speech data continues to be clocked into the remaining memory space available in the buffer. During the output phase the speech data is downloaded from the computer to the buffer memory and clocked out to the D/A converter by the timer.

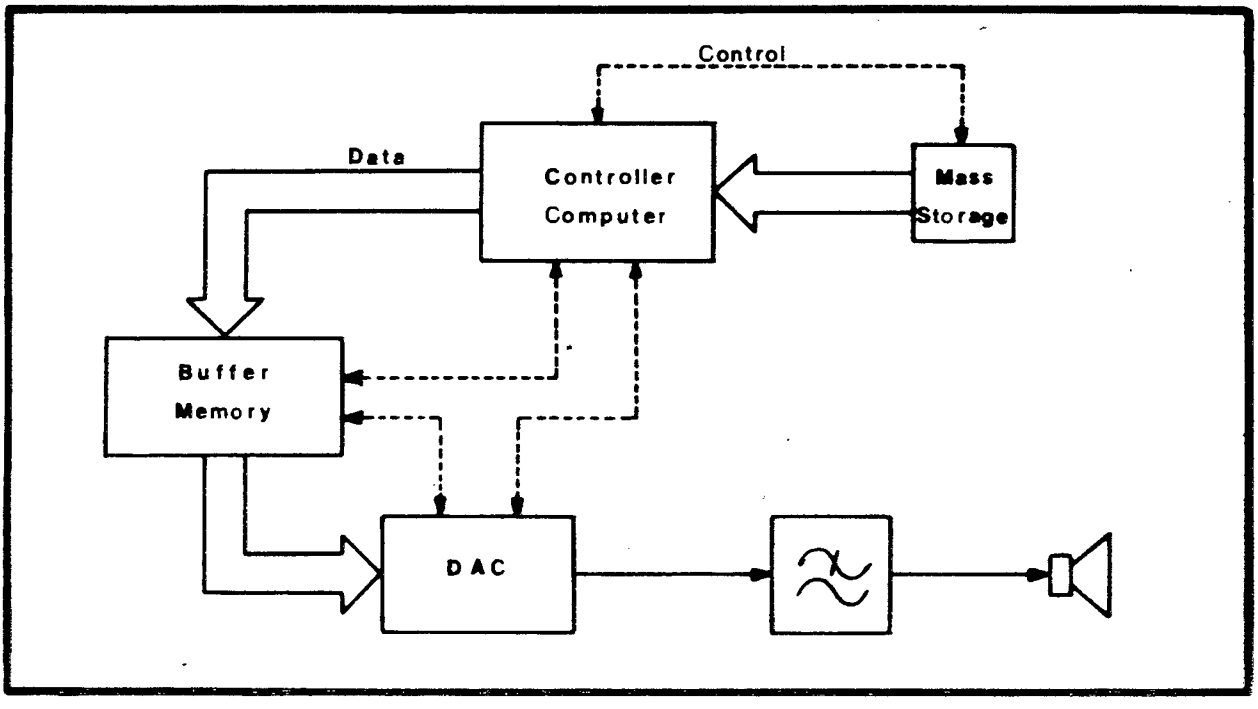


Fig. 5.4 : Block diagram of the D/A system.

o Analogue-to-digital conversion : this is performed by a 12-

bit A/D converter with an input signal range set to -10V . . +10V. This gives a dynamic range for the input signal of over 60 dB. The sampling of data is controlled by the timer.

- Digital-to-analogue conversion : a 12-bit D/A converter with an output range of -10V . . +10V is used for this purpose.

5.1.1 Data input.

As mentioned before, the input of data is interrupt controlled. The optimum buffer size for the chosen sampling rate is first determined and then the memory card in the Multiprogrammer is initialised accordingly. The timer card is set to operate at the required sampling rate. The commencement of data aquisition is then initiated by the user, who has previously specified the length of the speech segment to be recorded (the maximum allowable length of the segment is 10 seconds). The rest of the operation is done under the control of the computer. Once the segment has been stored in the computer's memory the RMS level and the maximum and minimum levels are displayed. The user then determines whether or not to store the data on the mass storage device and indicates the choice, in response to a prompt from the controller (the computer).

5.1.2 Data output.

The selected speech segment (stored on the mass storage device) is transferred to the computer's memory. The user then selects which portions of the segment he/she would like to listen to. The segment (or part thereof) is then transferred to the buffer memory and clocked out to the D/A card at the operating rate of the codecs (ie. 16 kHz). It is possible for the specified segment of speech data to be output once only or repeatedly. In other words, if the user so desires, a segment of speech data can be continuously repeated until interrupted, via the keyboard of the controller, by the user.

5.2 Codec simulation.

A total of seven different codecs have been simulated. From the discussion in chapters 2 and 3 it should be clear that only the encoder need be simulated because an ideal channel has been assumed and the local decoder in the encoder is equivalent to the decoder at the receiver. Four of these have been included purely for reference while the remaining three are the hybrid codecs that are central to this work. The seven codecs are those discussed in chapter 3 - ie. LDM, CVSD, CFDM, SVADM, HCDM, SHCDM, MSHCDM.

The simulation of the codecs cannot be done in real time because firstly, the time overheads involved in transferring the data

between the computer and the Multiprogrammer are excessive, and secondly, the actual time taken to process each sample is longer than the sample period. Hence, the data, once having been processed by the codecs, is stored on disc and then, at a later stage, it can be downloaded to the Multiprogrammer.

The user selects which codec is to be used to process the speech data. The data is then passed to the routine simulating the codec. In all cases where maximum processing speed is required the software has been implemented as a compiled PASCAL module. Realising the simulation software entailed translating the algorithms for the codecs (as listed in chapter 3) into an equivalent PASCAL form.

The codec simulation serves two purposes. Firstly, the speech data processed by the codecs can be stored on disc and subsequently used for subjective listening tests. Secondly, the various objective measures can be evaluated with or without the processed data being stored on disc.

5.3 Measurement.

There are a number of objective quality measures (discussed in chapter 6) which have been included into the simulation package. Besides being able to process speech data and listen to it the

user is provided with the facility for determining these objective quantities.

Basically the method for assessing the quality (objectively) is as follows :

The recorded speech data is processed by the desired codec and the difference between the corresponding samples of the input and processed signals is stored. This difference signal is then filtered, using a digital filter simulated on the computer (cf. section 4.1 of this chapter), to remove the out-of-band components of the signal. The respective powers of this signal and the input signal are then compared yielding the signal-to- noise ratio.

5.4 Peripheral.

The term 'peripheral' has been used here to describe those functions, essential to the simulation, that cannot be included in any of the other sections. The two functions included in this section are filtering and interpolation.

5.4.1 Filtering.

A non-recursive digital filter operating at 16, 24, 32 or 64 kHz has been implemented. A detailed description of the filter design procedure and its resultant characteristics is given in Appendix

F. The purpose of the filter is to remove the out-of-band components of the noise signal so as to enable the calculation of the various signal-to-noise ratios. This filter does not replace the analogue filter at the codec output but simply simulates it for measurement purposes.

5.4.2 Interpolation.

The sampling frequency of the A/D card is limited to a maximum value of 11.4 kHz (88 μsec sampling period). Above this value the buffer size becomes insufficient for a sentence-length speech segment (the buffer size is determined by the required sampling frequency and the data transfer rate using the HP-IB). This problem can be overcome by digitally interpolating the sampled data to a higher effective sampling rate.

Essentially, interpolation involves fitting a curve to the discrete sampled data values. In order to interpolate to any multiple of the base sampling rate a continuous curve could be fitted to the available data. This could be done using Lagrange polynomials :

$$p_{n-1}(x) = \sum_{i=1}^{n} \left[\prod_{r \neq i} (x - x_r) / \prod_{r \neq i} (x_i - x_r) \right] y_i$$

where n is the number of samples in the set, and

x_i and x_r are the known sample values.

With a known set of samples the polynomial $p_{n-1}(x)$ could be calculated and then all values on the fitted curve could be determined. When the number of samples in the set becomes very large the amount of processing becomes excessive and hence this method would be very costly in terms of processing time.

In the case of digital communications some of the common sampling rates are 8, 16, 32 and 64 kHz and hence for the purposes of the simulation it would only be necessary to interpolate to the integer multiples of the base sampling rate. This can be achieved using a digital filter.

In fig. 5.5 a waveform is sampled at 8 kHz. If null samples (ie. samples with value zero) are inserted between each actual sample then the information contained in the sampled waveform has not been changed. The time between samples though has been halved, thereby effectively doubling the sampling rate. If this signal is then passed through a digital filter operating at the new sampling rate the effect will be to average out the signal. In order to restore the RMS level of the signal to the pre-insertion value (ie. before the null samples

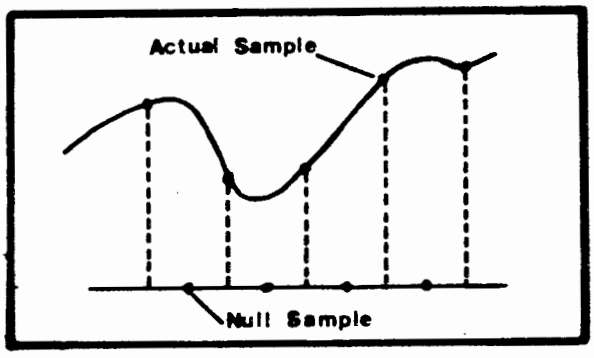


Fig. 5.5 : Sampled waveform.

were inserted) the samples are multiplied by a constant factor whose value is determined by the ratio of the new-to-original sampling rates (ie. if the effective sampling rate is doubled then this factor has a value of 2).

A basic value of 8 kHz has been used as the practical sampling frequency. If a higher effective sampling frequency is required then the 8 kHz sampled data is digitally interpolated to the desired rate. It should be noted that in this package only integer multiples of the base rate are attainable (ie. sampling rates of 8, 16, 24, 32, 64, etc. are attainable).

The data to be interpolated is read off the mass storage device and passed to the interpolation routine (this data was previously stored on the mass storage device during the input phase).

5.5 Special Considerations.

There are a number of differences between the simulated version and a hardwired implementation of the codec circuits. The advantage of the simulated version is that the circuit performance can be evaluated with or without these variations included.

5.5.1 Signal Range.

The first of these differences involves the maximum range of the

decoded speech signal. At the input the signal range is limited by the range of the A/D converter and the power supply voltages for the pre-sampling filter. Numerically this range, for a 12-bit ADC (Analogue-to-Digital Converter), is -2048 . . . +2047. In the simulation software the arithmetic signal range is limited by the word size of the computer. In this case a 16-bit word is used for integers (ie. a range of -32768 . . . +32767) while the 32-bit floating point arithmetic limits the range for real numbers to -10^{+308} . . . $+(10^{+308} - 1)$. Hence, it is possible to let the decoded speech signal, as represented in the computer, have a range which is larger than the range of the input signal.

In order to evaluate the performance of the codec algorithm without the constraints introduced by a practical implementation the decoded signal range was only limited by the numeric range of the computer. On the other hand, to simulate the practical codec the decoded signal range is limited to match the range of the input signal (ie. numerically the decoded signal would be limited to values between -2048 and +2047).

5.5.2 Step Size.

Another parameter which could be limited in practice is the step size or step size dictionary. For example, in the analogue implementation of the HCDM coder given by Un et al[17] the

Constant Factor step size dictionary is limited to eight values (1.5^0, 1.5^1, 1.5^2, . . . , 1.5^7). The reason for doing this is obvious : it is not possible to use an infinite number of gain-setting resistors in the circuit. The significance of this is that the step size range produced by the constant factor compandor is limited to a maximum and a minimum value.

Again, in order to evaluate the performance of the algorithms no explicit limits were put on the step size ranges.

However, in the case of the codecs using constant factor adaptation (ie. CFDM and HCDM) a limit was set for the minimum value of the step size multiplication factor. During the 'hunting' phase (when the input signal is in a steady state, or near-steady state condition) the locally decoded signal oscillates about the input signal which in turn causes the constant factor modulator step size to decrease (ie. $k_n = 2/3$ and $\gamma_n = \gamma_{n-1} * 0.66$). If the hunting phase continues for long enough then γ_n becomes small enough to degrade the performance of the codec. This is caused by the response of the encoder being impeded by the very small step factor. In other words, when there is a change in the input signal level after a period of inactivity the indefinitely small step factor prevents the encoder from tracking the signal and a severe case of slope overload is experienced. If

the minimum step factor is limited then it does not take as long for this value to recover from the steady state condition, thereby preventing dead-bands from occuring in the decoded speech signal output.

Similarly, the value of γ_n can be limited to a maximum value in the software, but for the objective performance evaluation this was not done.

Thus for the simulations comparing the relative performance of the HCDM, SHCDM and MSHCDM, the signal value and step size ranges were allowed to follow the algorithm exactly without any of the constraints introduced by a practical hardwired implementation of the codec circuit. In chapter 7 the effects of these constraints on the various codecs' performance have been studied.

CHAPTER 6

CODEC PERFORMANCE ANALYSIS

The assessment of speech coder performance can be divided into two discrete sections - objective and subjective testing. The objective measures, on the one hand, are not sufficiently well established to provide an unambiguous indication of a codec's performance while, on the other hand, there does not exist a generally applicable method for subjectively evaluating the quality of the codec. Compounding these problems is the fact that the correlation between the results of the objective and subjective tests is not well understood and as a result there does not exist one well-defined empirical method for determining the quality of a speech processing system. Due to this anomaly our codec performance evaluation has been based on both objective and subjective test procedures, with the emphasis being placed on the subjective techniques.

6.1 Objective Performance Evaluation.

It has been established that acceptable voice communication is reliant on the preservation of the short-time amplitude spectrum of the signal[43]. In the case of waveform coders the quality could be characterized by a signal-to-noise ratio (the noise being the difference between the original input signal and the eventual output signal). The use of this measure can be justified on the basis of the operation of the waveform coders - ie. the waveform coder essentially attempts to preserves the amplitude spectrum of

the signal by producing a replica of the time domain waveform.

In the case of sampled data signals the signal-to-noise ratio (SNR) can be defined as :

$$SNR = 10 \log\left(\sum_{i=1}^{N} Y^2(i) \; / \sum_{i=1}^{N} e^2(i) \right)$$

where

N is the number of samples in the speech segment used for the test (typically the summations are over the duration of a sentence-length segment),

$Y(i)$ is the i^{th} input signal sample, and

$e(i)$ is the corresponding noise sample.

$e(i) = Y(i) - X(i)$

where

$X(i)$ is the output of the codec.

However, although this is probably the most commonly used method for measuring coder distortion, it does have one significant drawback. The effect of high energy components of the speech segment is to swamp the contribution to the SNR measure of the lower energy segments (eg. some unvoiced sounds). An alternative measure, which has been found to exhibit better correlation with subjective measures[22,28], is the segmental signal-to-noise ratio which assigns a more equitable weighting to the high and low

energy components of the signal. The segmental SNR computes the SNR for short portions of the speech segment (typically 15 - 20 msec in duration) and then averages these short-time measures over the duration of the full segment.

The segmental signal-to-noise ratio is then defined as :

$$SNRSEG = \left(\sum_{m=0}^{M-1} 10 \log\left(\sum_{i=1}^{N} Y^2(i + Nm) \Big/ \sum_{i=1}^{N} e^2(i + Nm) \right) \right) / M$$

where

M is the total number of short-time segments,

N is the number of samples in the short-time segment, and

Y and e are as defined previously.

As has been mentioned before, there are basically two types of noise produced by a delta modulator, namely slope-overload noise and granular noise. The relative effects of the two types of noise on a listener's subjective response have been found to be different[29] (ie. the one type of noise is less objectionable than the other). There are two measures which describe the amounts of distortion due to one or the other of these noise types - Signal-to-Granular Noise Ratio (SNG) and Signal-to-Overload Noise Ratio (SNO). If e(i) is the i^{th} noise sample and $e_g(i)$ and $e_o(i)$ the i^{th} granular and overload noise samples respectively, then the SNG and SNO are defined as :

$$e_g(i) = \begin{cases} e(i) & \text{if } e(i).e(i+1) < 0 \\ 0 & \text{otherwise;} \end{cases}$$

and

$$e_o(i) = \begin{cases} e(i) & \text{if } e(i).e(i+1) > 0 \\ 0 & \text{otherwise.} \end{cases}$$

With this classification of the noise, SNG and SNO are then defined by :

$$SNG = 10 \log\left(\sum_{i=1}^{N} Y^2(i) \; / \sum_{i=1}^{N} e_g^2(i) \right)$$

and

$$SNO = 10 \log\left(\sum_{i=1}^{N} Y^2(i) \; / \sum_{i=1}^{N} e_o^2(i) \right)$$

As in the case of the total signal-to-noise ratio there are corresponding segmental values for SNG and SNO.

The segmental signal-to-granular noise ratio :

$$SNGSEG = \left(\sum_{m=0}^{M-1} 10 \log\left(\sum_{i=1}^{N} Y^2(i+Nm) \; / \sum_{i=1}^{N} e_g^2(i+Nm) \right) \right) / M$$

The segmental signal-to-overload noise ratio :

$$SNOSEG = \left(\sum_{m=0}^{M-1} 10 \log\left(\sum_{i=1}^{N} Y^2(i+Nm) \; / \sum_{i=1}^{N} e_o^2(i+Nm) \right) \right) / M$$

(The symbols have the same interpretation as before.)

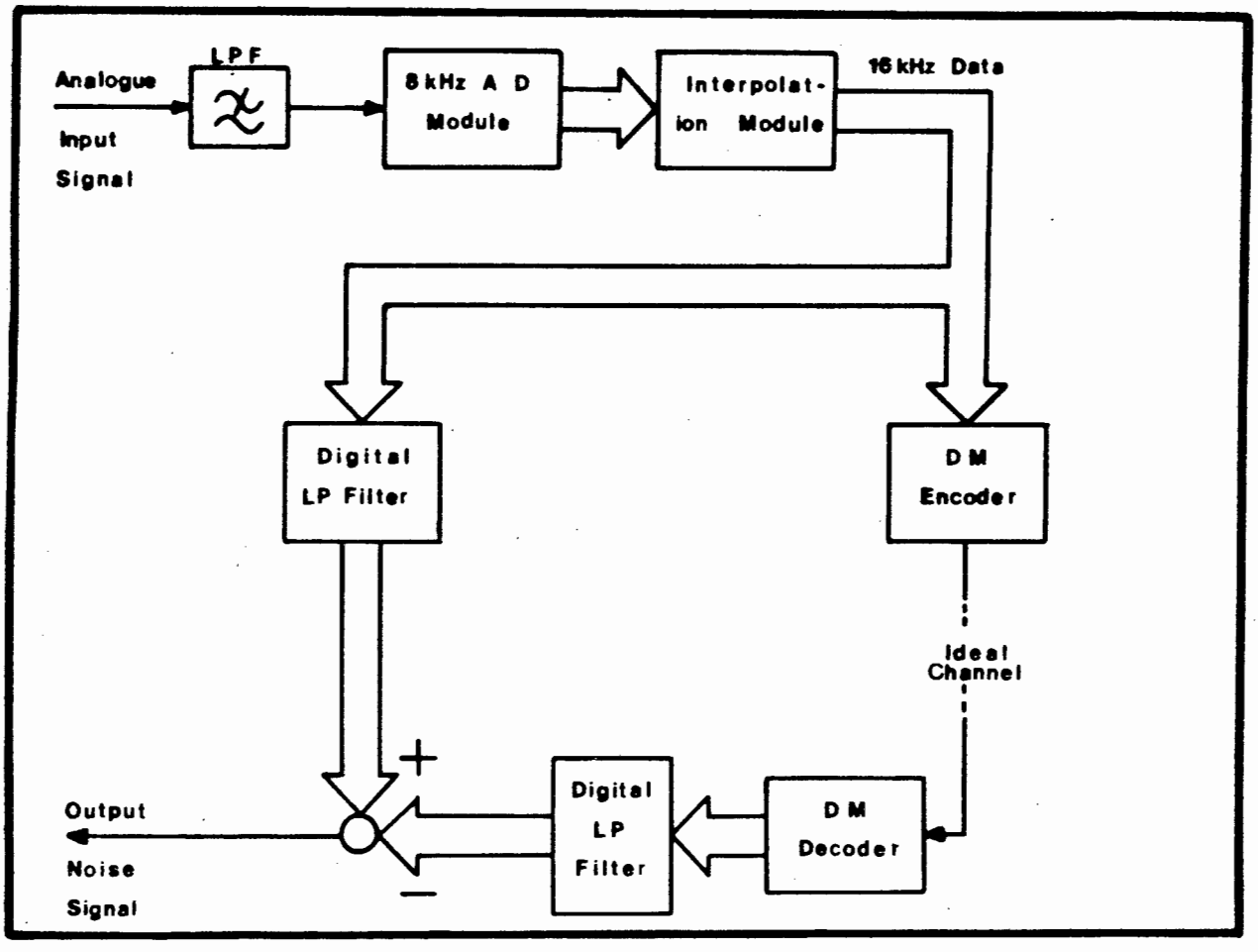


Fig. 6.1 : The objective test configuration.

The test set-up for measuring these quantities is shown above.

6.1.1 Dynamic Range Test.

This test involves computing the signal-to-noise ratios (SNR, SNRSEG, etc) for different levels of the source signal. Having established the peak value of the signal-to-noise ratio over the range of input signal levels, the dynamic range is then defined as the range of signal levels for which the signal-to-noise ratio is greater than, or equal to a value 3 dB below the peak value.

6.1.2 Speech Material.

Real speech, spoken by male and female voices, was used as the source excitation. Short phonetically balanced sentences were spoken by the different speakers with the sentences being taken from the Harvard set of sentences[27].

6.2 Subjective Quality Evaluation.

Because the objective measures described previously fail to give an absolute measure of a coder's quality, and because the system users' assessment of performance is subjective by nature (based on the total auditory impression on the listener) formal judgements on coded speech quality are based on subjective testing.

The use of speech intelligibility alone as a measure of the quality of a speech communication system is not sufficient[30-32]. Two codecs, for example, may exhibit similar word/sentence intelligibility scores while listeners find one more acceptable than the other. The main problem lies in the fact that there are invariably a large number of parameters (eg. received signal volume, intelligibility, noise level, etc.) affecting the listener's evaluation, and it is difficult to estimate the relative importance of the different parameters, either singly or in combination with each other. Essentially what is needed is a one-dimensional scale which would enable a speech transmission

system to be rated on the basis of listener preferences, thereby enabling the communication system to be compared to other systems regardless of the physical parameters of each.

Hecker and Guttman[32] have categorized the subjective testing procedures as either 'analytic' or 'utilitarian'.

- o Analytic techniques : these techniques attempt to establish a circuit's performance criteria on the basis of the various psychological components affecting a listener's judgement (eg. the effects of low-pass, band-pass or high-pass filtering of the signal on the assessment of the coded speech's quality; or the difference between listener responses due to signal distortion as opposed to the presence of background distortion).
- o Utilitarian techniques : these are concerned with obtaining an over-all measure of the speech quality without investigating the psychological factors. Despite the problems associated with using a unidimensional scale for assessing speech quality the utilitarian techniques are preferred (from an engineering point of view) for quality measurements[27,35] as this allows direct comparisons to be made between codecs. In other words it would allow a choice to be made between different types of codecs based simply on how they rate on a single quality scale.

Basically the utilitarian methods can be divided into two main categories (after Munson and Karlin[33]) - direct, and indirect comparisons (cf Table 6.1). In the case of indirect comparison the listener is only presented with the system in question and must rely on previous experience with other systems in order to make a judgement. Direct comparisons involve presenting the listener with two circuits in close succession and the listener then evaluating the one relative to the other.

Table 6.1 : Classification of speech quality assessment methods[33].

Indirect comparisons	Category tests o Loudness o Quality o Effort
	Intelligibility tests o Words o Sentences o Immediate appreciation
	Articulation tests o Sounds o Syllables
	Repitition count
	Message rate
Direct comparisons	Fixed reference, variable unknown Variable reference, fixed unknown Variable reference, variable unknown

(a) Indirect comparisons.

o Category judgement : these tests are usually concerned with a particular aspect of the system (ie. loudness, quality or effort required to use the system). The listener is required to evaluate the circuit according to the categories listed.

Typical terms used are :

excellent, good, fair, poor, bad - for 'quality';

much too loud, too loud, satisfactory, too soft, much too soft - for 'loudness';

no effort, no appreciable effort, moderate effort, considerable effort, impossible to understand - for ease of use of the system.

- o Intelligibility tests : this could involve evaluating the words or sentences correctly interpreted and identified by the listener. Alternatively, a simple YES-NO decision as to whether the sentence was understood (without the listener reviewing what was heard) could be used (ie. immediate appreciation).
- o Articulation tests : similar to intelligibility tests, these involve identifying the sounds received by the listener.
- o Repitition count : this test determines the number of times a speech message must be repeated to effect satisfactory transmission.
- o Message rate : this test involves a two-way link over which two communicants attempt to solve a problem as efficiently as possible.

These various tests are heavily dependent on factors outside the

control of the tester. For instance, different listeners are likely to have widely differing interpretations of the meanings of terms such as 'good', 'poor', 'too loud', etc (cf category tests). Learning factors on a new circuit could influence the outcome of the tests (ie. the listeners' responses could change as they become more familiar with the characteristics of the circuit). A particular problem with the 'immediate appreciation' test is the lack of control or confirmation - a listener may think he/she has interpreted the sentence correctly whereas, in fact, a mistake has been made.

(b) Direct comparisons.

The general procedure for this type of testing involves pair comparisons between two circuits. The one circuit would be a reference, the value of whose parameters are well established, while the second would be a test circuit whose parameters are to be assessed. Basically the test involves varying the operating conditions of either the test or reference circuits (or both) and establishing the point at which the test subject (ie. the listener) judges the two circuits to be equal. The test could be used to evaluate a particular aspect of a system's performance (such as loudness), or simply to assess the over-all preferences of the listeners. A significant point about this test procedure is that the test subject is not required to specify the reasons

for their judgement.

As shown in Table 6.1 there are three basic modes of comparison.

- o Fixed reference : the test circuit is varied until it is judged equal to the fixed reference. The only subjective attribute commonly tested using this method is the loudness balance[33]. This is because the effects of the physical circuit parameters on the various attributes, such as quality, are not always easily assertained.
- o Variable reference : the complement to the above method, this test involves varying the parameters of the reference circuit and comparing the effect with the fixed test circuit.
- o Isopreference method (discussed in more detail shortly) : this test method involves varying both the test and reference circuit conditions.

6.2.1 Isopreference Test.

Whereas in the past the value of comparisons between circuits based on a single parameter (eg. intelligibility) were limited[34] the aim of the isopreference method is to implicitly take the effect of all parameters into account. As implied by the above discussion, the isopreference test (devised by Munson and Karlin[33], and modified by Tedford and Frazier[34]) involves varying both the test and reference conditions - a basic reference

signal has varying controlled amounts of noise added to it with the resultant signal being compared to a test signal whose energy level is varied over some range. Isopreference contours are constructed by finding speech level/noise level combinations which are equally preferred. For example, in fig. 6.3 the point 'A' represents a certain signal condition (ie. a certain long-time averaged signal and noise level). From preference tests involving direct comparisons, points 'B', 'C' and 'D' (representing three other signal/circuit conditions) were found to be equally preferred to the condition specified by 'A', enabling those points to be linked by a contour. Similarly, using condition 'E' as a starting point, 'F' and 'G' were found to be equally preferred to 'E', enabling a second contour to be drawn. The distances between the contours can be used to provide the so-called transmission preference unit scale (TPU) which was intended to be used as a unidimensional speech quality measure.

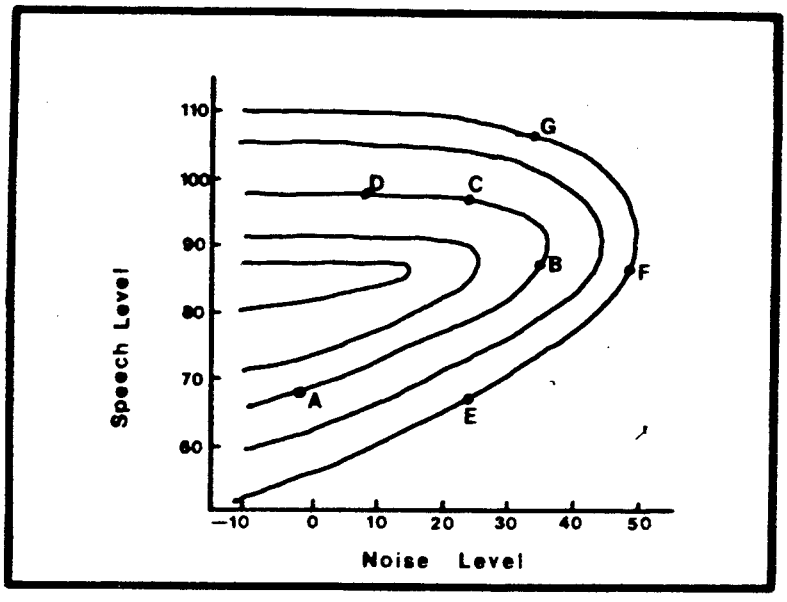


Fig. 6.3 : Isopreference Contours[333].

A simplified measure, derived from the isopreference procedure,

involves calculating a subjective signal-to-noise ratio. Essentially this test involves comparing the test system with a reference system which has been corrupted by a controlled amount of noise, with a known spectral shape (the signal level and noise level of the reference are both known). The subjective signal-to-noise ratio of the test is then defined as the SNR of the reference system which was equally preferred to the test system. This signal-to-noise ratio was recommended as a speech quality measure by the IEEE subcommittee on subjective measurements[27].

In an attempt to consolidate the issue of subjective quality measurements the IEEE subcommittee recommended three measures as opposed to one single test procedure.

6.2.2 IEEE Recommendations.

Some of the more important attributes of the testing scheme are outlined below.

- Quality attributes other than Preference must be neutralized - eg. intelligibility should not be an issue as it should be well established.
- Results should be quantifiable on a unidimensional scale.
- Testing methods should allow for the evaluation of one-way communications systems only as two-way systems introduce

behavioral indices (the speakers modify their speech characteristics in order to accomodate the system's properties - eg. very noisy communications channels may induce the speakers to talk louder).

- o The language used, and the semantic content thereof should be familiar to the listener.
- o The speech material used for the test and reference signals should be of the same type. Narrative material (eg. extracts from novels, etc) or short homogeneous sentences (eg. the 'Harvard Sentences'[27]) can be used.
- o The listener group should be a subset of the intended users of the system being analyzed.
- o If the system specifications require trained operating personnel then the test group (of listeners) must be similarly trained.

The three tests recommended by the IEEE subcommittee are the category judgement test, isopreference test and the relative preference test. Whereas the isopreference test involves using speech signals corrupted by different levels of noise the relative preference method uses different types of speech distortion. In this work the relative preference test has been discarded because the different delta modulators under consideration essentially produce the same type of noise. Because of the problems

associated with defining terms such as 'good', 'excellent', and 'poor' (in terms of speech quality) the category judgement tests were not performed. This problem becomes particularly relevant when attempting to compare the results of the category tests for different speech encoders because of the inevitable variation in interpretation by different listeners of the descriptive terms.

6.2.3 Subjective Test Procedure.

The test procedure that was used for subjectively assessing the quality of the processed speech involved two seperate stages. Firstly, a combined speaker recognition/intelligibility test was performed. This was then followed by a preference test. Based on section 6.2.2 the following test conditions applied.

- Speech material used : a set of sentences was selected at random from the 1969 set of 'Harvard Sentences'. In this way it was possible to ensure that phonetically balanced sentences of approximately the same length were used throughout. The tests were conducted in English and all listeners were screened to ensure that their mother-tongue was English. During one full test sequence (eg. the isopreference test) no speech material was repeated.

- Talkers : to prevent the tests being biassed by a talker-system

interaction (ie. a particular talker's voice may highlight certain characteristics of one system) a variety of talkers were used. Two males and two females were used to read the sentences mentioned above. For both sexes two age groups were used (20 - 35 years, and 35+ years). More specifically, the ages of the talkers were as follows :

Male 1 : 23 years old,

Male 2 : 46 years old,

Female 1 : 24 years old, and

Female 2 : 48 years old.

o Listeners : because it is envisaged that our systems would be in general usage (eg. in a commercial communications network) the listener group was chosen at random from the staff and students of the university where this research took place. No tests for hearing deficiencies were carried out as we were primarily interested in the response of the average listener. A group of 50 listeners was used with only one member of this group having had any prior experience with any form of subjective listening test.

o Training : an untrained group of listeners was used. Prior to performing the three tests the testing procedure was explained

to the listener.

- Test equipment : all speech material was recorded via a standard telephone handset. The speech signal was then band-pass filtered (300 - 2500 Hz), sampled at 8 kHz and digitally stored on magnetic disc. It was then processed by either the simulated hybrid delta modulators or the reference signal-generator (the interfacing and simulation is described in detail elsewhere in this work). Playback of the speech material was via a low-pass filter (cut-off at 2500 Hz) and a standard telephone handset. Because it has been found that the background noise level of the room is not critical to the tests[34] a sound proof room was not considered necessary and instead a suitably quiet office was used as the venue for the tests, with listeners being tested one at a time.

6.2.3.1 Intelligibility and Speaker Recognition Test.

The purpose of this test is to establish that the intelligibility and the speaker-identity preservation for the codecs is at an acceptable level, thereby providing a basis upon which to conduct the isopreference tests.

Although the two characteristics (speaker recognition and intelligibility) are commonly considered separately, the nature of these

two tests enabled them to be combined into one.

Sentence intelligibility, as opposed to word intelligibility, was used as it matches the practical communications characteristics more closely (a telephone conversation is more likely to consist of sentences and phrases than single words). The identification of speakers by human listeners is regarded as an important aspect of codec performance and is, in fact, an additional subjective measure of the quality of the coded speech[43]. However, it is generally a more important issue for vocoders than waveform coders[43]. Basically, the listener was presented with a sentence processed by one of the three hybrid delta modulators and then asked to repeat (vocally) the sentence and to identify the speaker. Because the four speakers used were not familiar to the listeners, before each processed sentence the listener was presented with two recorded sentences spoken, one each, by either the two male speakers or the two females (it is unlikely that a

Table 6.2 : Example of the test sequence for the Intelligibility & Speaker Recognition test.

Male 1 Male 2 Test sentence	Reference sentence Reference sentence spoken by either Male 1 or Male 2, processed by one of the three hybrid delta modulators.

male voice would be mistaken for a female voice). These two sentences are used as references so that the listener is aquainted with what each speaker's voice sounds like. An example of the test sequence heard by a listener is given in Table 6.2. The listener then indicates which speaker said the test sentence, and repeats the sentence. An example of the test form that was to be filled in by the listener is shown in fig. 6.4. A 'PARTLY CORRECT' sentence is one for which the listener got the basic semantic content correct, but some of the words incorrect.

	Did you understand the sentence ? YES☐ NO☐ PART'LY☐ Identify the speaker. 1☐ 2☐ Not Sure☐
	Did you understand the sentence ? YES☐ NO☐ PART'LY☐ Identify the speaker. 1☐ 2☐ Not Sure☐
	Did you understand the sentence ? YES☐ NO☐ PART'LY☐ Identify the speaker. 1☐ 2☐ Not Sure☐

Fig. 6.4 : Intelligibility & Speaker Recognition test answer form.

A total of 15 processed sentences were presented to the listener (5 each for the three hybrid DM's). The sequence of speakers was randomised as is the sequence of the codecs. This randomization

process is to prevent the listener from detecting a pattern or sequence in the test procedure, or from being able to predict the speaker for the next sentence.

6.2.3.2 Isopreference Test.

The procedure used here was based on the recommendations of the IEEE subcommittee[27], and not the full test specified by Munson and Karlin. In essence the test involves deriving the psychometric curves[33] for the preference tests (for each of the three hybrid delta modulators), and then, from these curves, determining the subjective signal-to-noise ratios[35].

Following the intelligibility/speaker recognition test the listener was presented with a preference test. Basically the listener hears two sentences (a reference and a test) and is required to indicate which of the two they would prefer as a source of information. In total the listener was presented with 24 such reference-test pairs (8 each per hybrid codec).

The reference signal was generated by adding a controlled amount of noise to the original speech signal. It has been found[27] that in the case of digitally processed signals multiplicative, as opposed to additive, random noise is most suitable. From the definition given by Schroeder[36] the reference signal r(i) is

given by :

$r(i) = (s(i) + n(i))/k$

where

k is a power normalisation factor given by :

$k = (1 + a^2)^{1/2}$

where

'a' is the coefficient specifying the SNR,

$n(i) = a.s(i).e(i)$,

e(i) being a random variable taking on values of +1 or -1 with equal probability,

s(i) is the sampled input speech signal, and

n(i) is the pink noise signal.

In other words, the reference signal is generated by taking a controlled proportion of the randomly inverted input signal and adding it to the original input signal.

Once again, the sentences (24 reference and 24 test) were spoken by the four speakers. The 24 test sentences were processed (8 each) by the three codecs. The sequence in which the test sentences were presented was randomized to prevent the listeners identifying the different codecs, and the allocation of speakers to the different codecs was also randomized (each codec had at least one sentence spoken by each of the four speakers). The

eight reference sentences, for each codec, were given different SNR values (-6, -3, 0, 2, 4, 6, 8, 10 dB) and the sequence in which they were presented to the listener was randomized. This was done to prevent the listener detecting a progression in the amount of degradation added to the references.

A test consisted of a repeated reference-test signal pair. In this way the listener can compare the reference (B) to the test signal (A) as well as the test to the reference. The full isopreference test is illustrated in fig. 6.5.

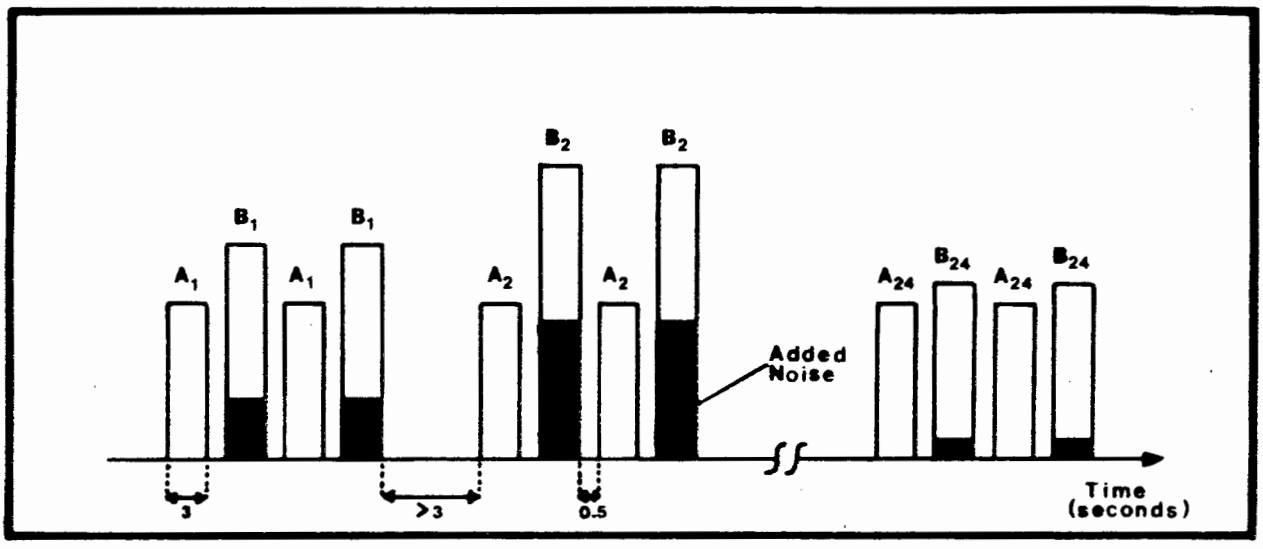


Fig. 6.5 : The Isopreference test sequence.

After each A-B-A-B sequence the listener is required to indicate their preference. An example of the form that was to be filled in by the listener is shown in fig. 6.6.

Test Sequence Number	Preference		Test Sequence Number	Preference		Test Sequence Number	Preference	
(1)	A B UNSURE		(2)	A B UNSURE		(3)	A B UNSURE	
(4)	A B UNSURE		(5)	A B UNSURE		(6)	A B UNSURE	
(7)	A B UNSURE		(8)	A B UNSURE		(9)	A B UNSURE	
(10)	A B UNSURE		(11)	A B UNSURE		(12)	A B UNSURE	
(13)	A B UNSURE		(14)	A B UNSURE		(15)	A B UNSURE	

Fig. 6.6 : Isopreference test answer sheet.

CHAPTER 7

SYSTEM PERFORMANCE

This chapter has been divided into two major sections - objective and subjective quality measures. All the results presented here have been obtained using the techniques described in chapter 6. For the subjective results the tests, exactly as described in the previous chapter, have been used. For the objective results the major tests are the same as those described in chapter 6. However, for some of the tests (eg. section 7.2.4 which shows the effect of using the feedforward mode of calculating the syllabic factor for the hybrid codecs) slight modifications were made to either the algorithms or the measurement techniques. In cases where the techniques differ from those described previously the changes have been detailed prior to presenting the results.

7.1 Autocorrelation Function of a Speech Signal.

Fig. 7.1 shows the autocorrelation function for the speech segment used in the tests detailed here. The purpose of this is simply to confirm that the correlation between adjacent samples is in fact within the limits given in chapter 2 (ie. $C(1) > 0.5$).

7.2 Objective Results.

Before the codecs could be evaluated it was necessary to optimize the parameters affecting the performance of the circuits so as to enable direct comparisons between the three to be made.

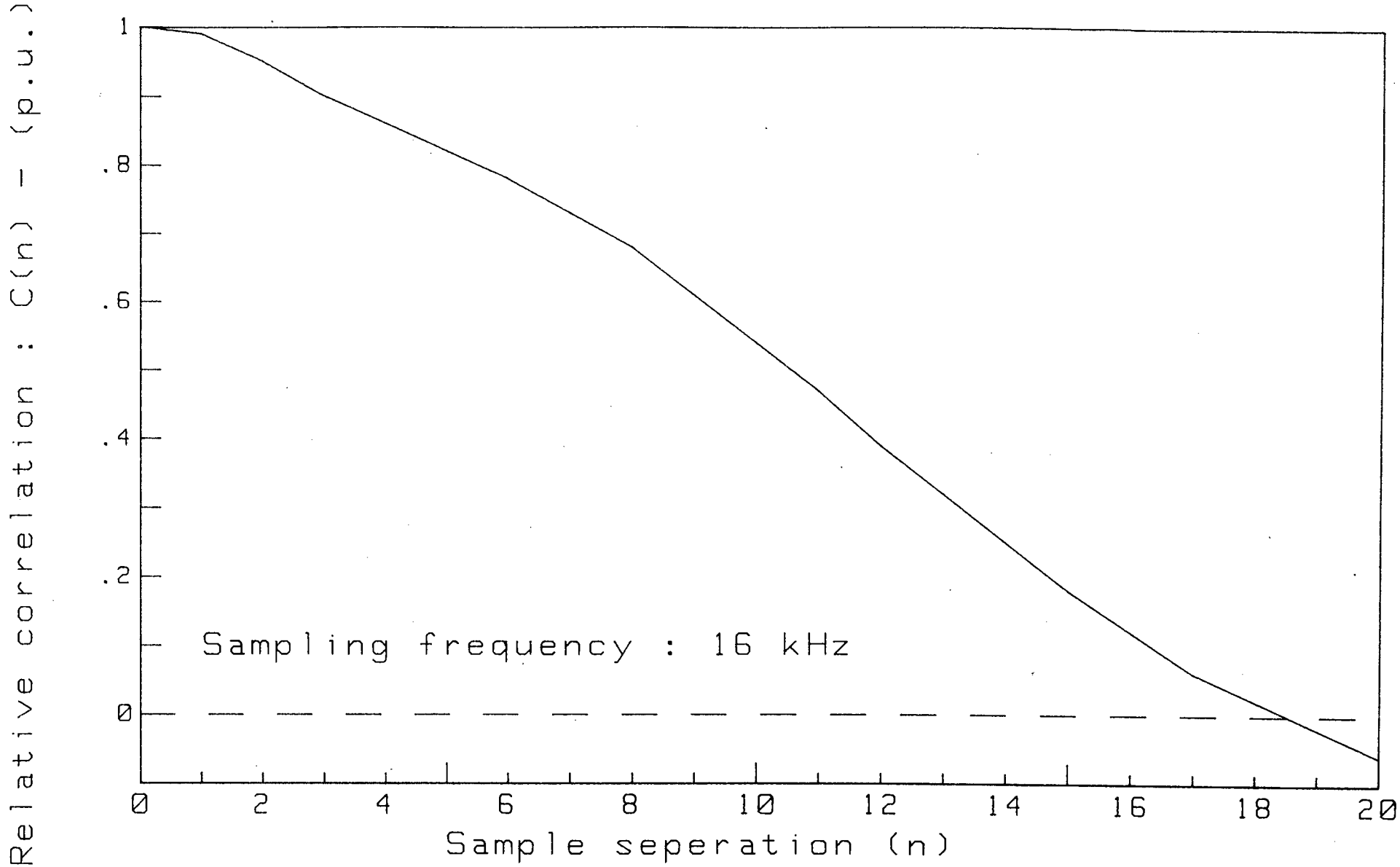


Fig.7.1 : Autocorrelation Function of sampled speech.

7.2.1 Circuit Optimization.

The basic procedure for evaluating the optimum value of a circuit parameter involves setting all others to a fixed value and then varying the parameter in question over some range. The parameters are optimized with respect to the SQNR. Initially it would appear that, based on the fact that slope overload noise is subjectively preferable to granular noise[29], the SNG would be a better measure of codec quality. However, the SQNR is used as the quality measure for optimization purposes because, when a delta modulator is tracking the input signal optimally, all the noise is granular (this stems from the nature of the operation of the delta modulator). By optimizing with respect to SNG it would be possible to achieve an encoder which would be virtually permanently slope overloaded. Hence the use of SQNR.

The parameters which have the main influence on the system performance are the quantizer minimum step size, the time constants of the prediction and syllabic filters, and the scale factor of the syllabic compandor.

In all cases (unless otherwise specified) the simulation runs were made using a real 10 second segment of male speech and a noiseless channel.

7.2.1.1 Syllabic Gain Factor (α).

The scale factor α for the long-term basic step size controls, to a large extent, the quantization noise produced by the encoder. If α were too small, the step size would be too small and the encoder would operate in the slope overload mode for most of the time. On the other hand, if α were too large, the output of the local decoder would oscillate about the input signal as a result of the large basic step size. In the extreme case this could lead to the encoder becoming unstable.

Fig. 7.2 shows the variation of the signal-to-noise ratio as a function of α for the three hybrid codecs.

For the HCDM encoder an optimum value for α of 0.8 correlates with the findings of Un et al[17,18], while the peak SQNR's for the SHCDM and MSHCDM encoders occur at values of 0.2 and 0.25 respectively.

An interesting feature of the curves in fig. 7.2 is the relative insensitivity of the SQNR's of the SHCDM and MSHCDM encoders to the variation of α. The advantage of this is that a value of α other than the optimum value is likely to be less drastic, in terms of performance degradation, for these two encoders than for the HCDM encoder. This would enable the values for α to be chosen

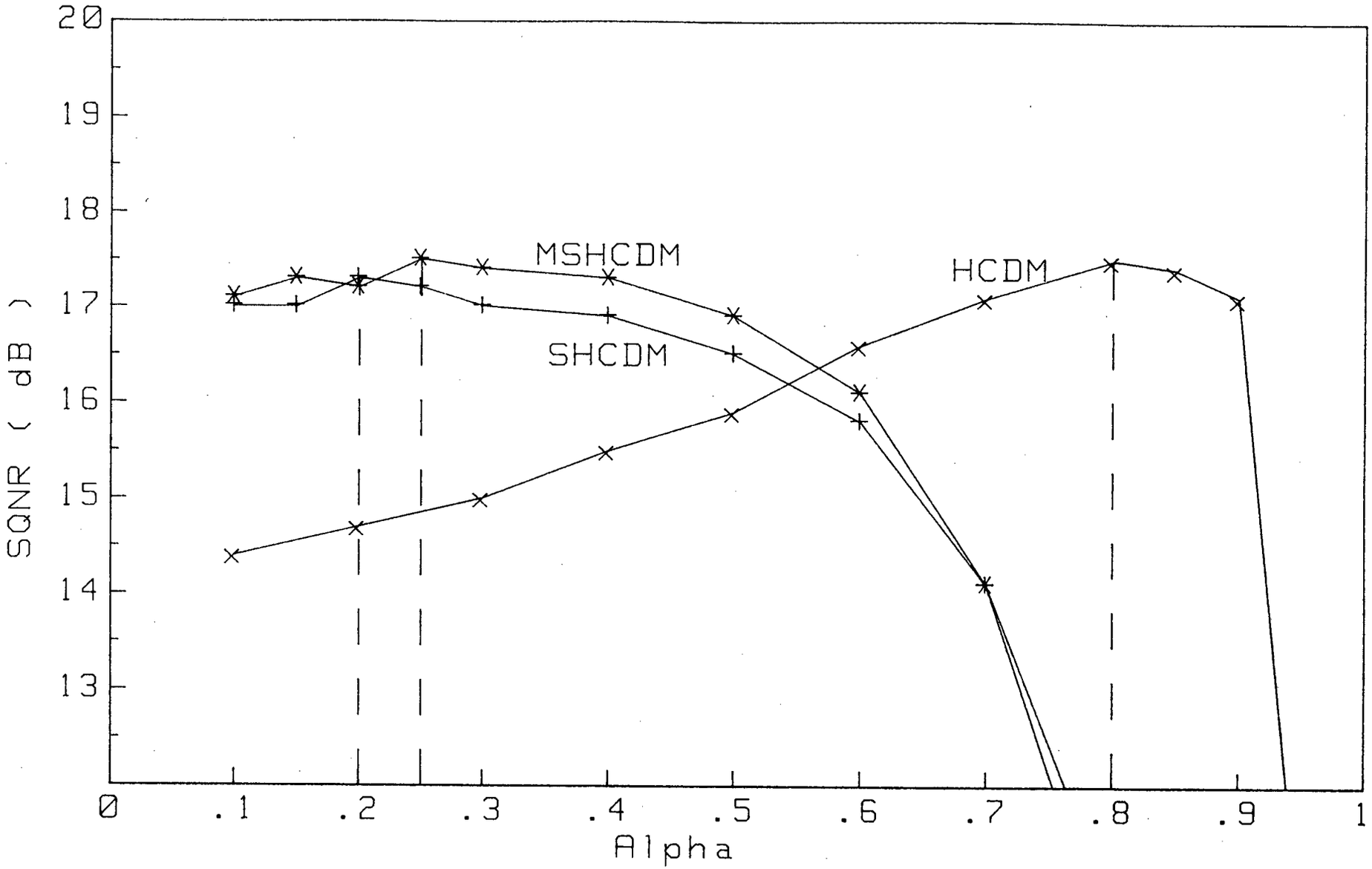


Fig.7.2 : SQNR as a function of Alpha

based on other considerations as well (eg. ease of implementation).

Based on the preceding discussion, it would appear that a value for α of 0.25, instead of 0.2, for the SHCDM encoder would be satisfactory. For both the SHCDM and MSHCDM encoders the digital implementation of the multiplication by α is easily realised as a two-place arithmetic shift. However, a value of 0.8 (for the HCDM encoder) is not very easily realised using digital methods.

Referring to fig. 7.3, the variation of the segmental signal-to-noise ratio as function of α, the peak of SNRSEG for the HCDM encoder occurs, not at 0.8 but, at 0.7. As the segmental measures have been found to correlate better with the subjective results[43] it would appear that this would be a better value to use. Alternatively, a compromise value of 0.75 would have the advantage of simplifying the digital implemetation of the multiplication operation (0.75 could be realised by dividing by four, a process requiring a two-place arithmetic shift, and subtracting the result of this from the original value). However, the SNRSEG measure was not used by Un et al[17,18] and so no comparison with published findings can be made in this regard. In the case of the SHCDM and MSHCDM encoders the results of the SNRSEG tests serve to confirm the findings of the SQNR tests.

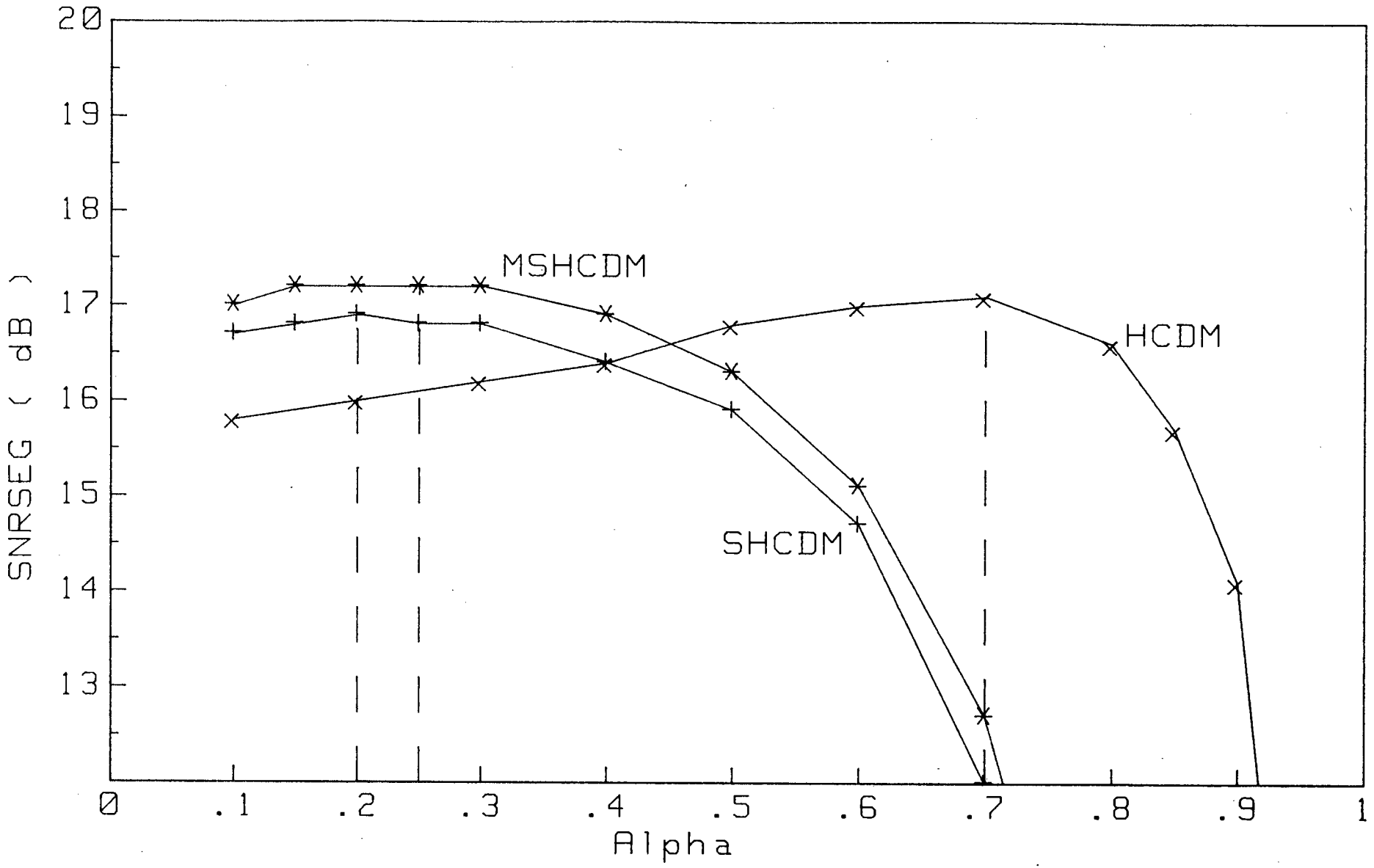


Fig.7.3 : SNRSEG as a function of Alpha

7.2.1.2 Prediction Time Constant ($1/\beta$).

The high degree of correlation between adjacent samples of speech sampled at 16 kHz (cf. fig. 7.1) suggests that the optimum value for the leakage factor of the prediction filter would lie somewhere between 0.8 and 1.0. The leakage factor L is related to the prediction time constant $1/\beta$ by :

$$L = \exp(-\beta T).$$

Fig. 7.4 shows the variation of SQNR as a function of L. These curves were determined using the values of α established in the previous section. The syllabic time constant was set to 5 msec and the minimum step size to 5 mV. From these curves it is evident that a value for L of 0.96 would suffice for all three encoders. This is not entirely surprising as it should be remembered that this leakage factor is derived more from the characteristics of the speech signal than from those of the encoding algorithms (although there is some inter-relation). From this value it is possible to calculate the value of the prediction time constant.

$$\beta = -(1/T) \, Ln(L)$$

$$\therefore \beta = - 16.10^{3} \, Ln(0.96)$$

$$\therefore \beta = 1/(1.53 \text{ msec}).$$

NOTE : In fig. 7.4 the SQNR curve for the SHCDM encoder was only

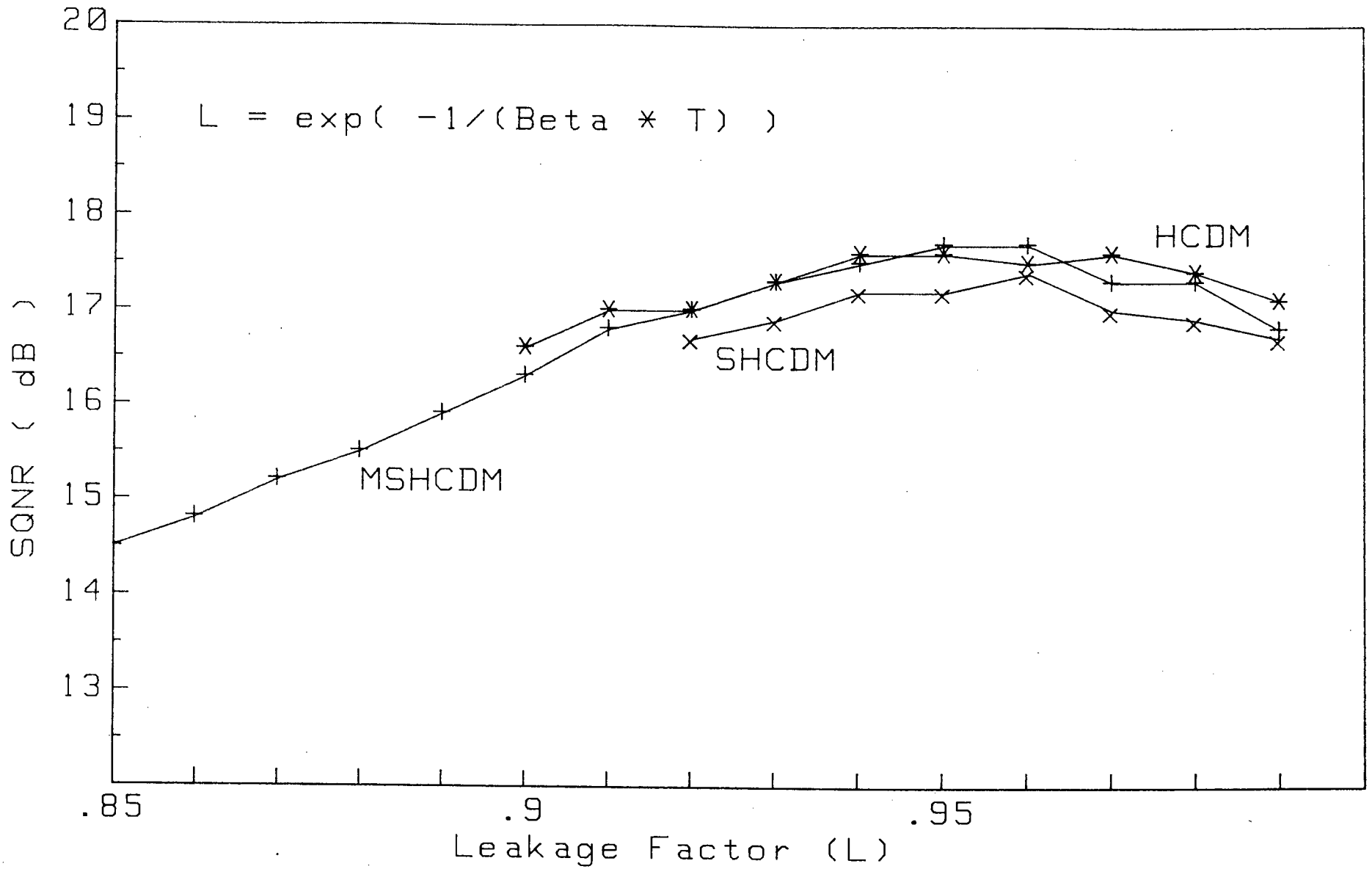


Fig.7.4 : SQNR as a function of the Leakage Factor (L)

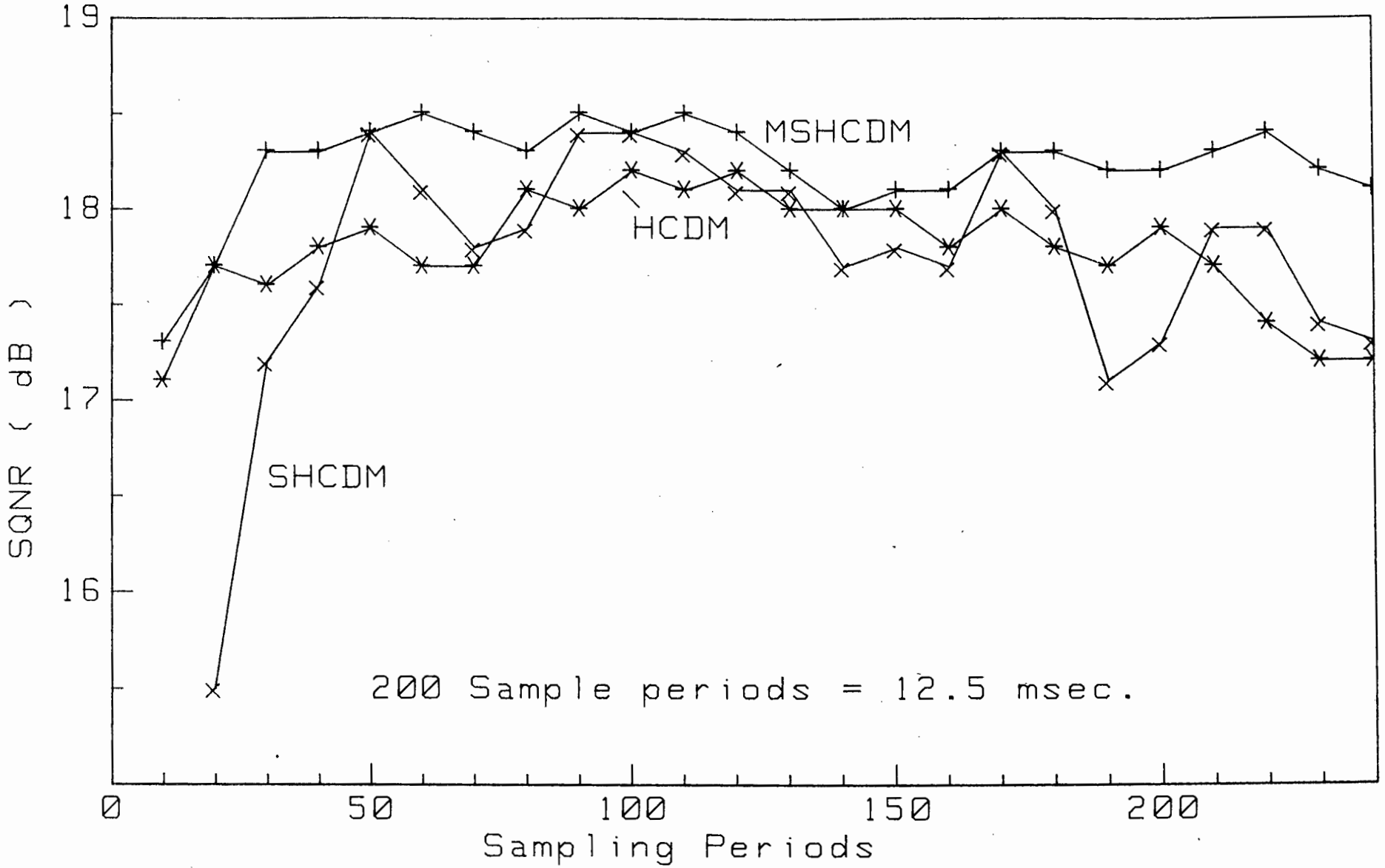


Fig.7.5 : SQNR as a function of the Syllabic Period

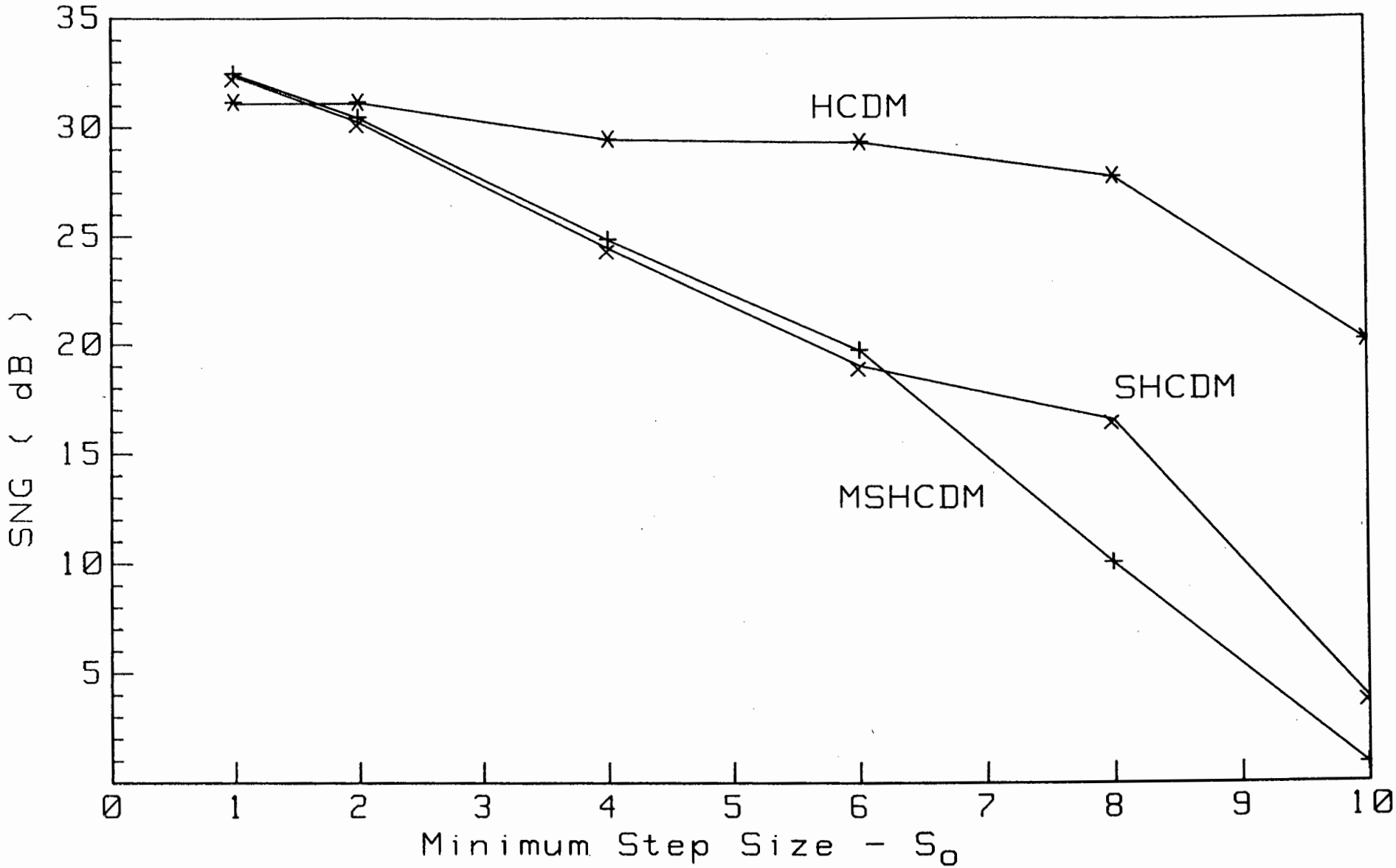


Fig.7.6 : SNG as a function of the Minimum Step Size

plotted for values of L between 0.92 and 1.00 as the curve for the MSHCDM encoder was used to estimate the likely range within which the peak value of SQNR would occur.

7.2.1.3 Syllabic Companding Period.

Based on the findings of Un et al[17,18] it was not expected that the value for the syllabic time constant would be critical. Fig. 7.5 demonstrates that a value of between 2 msec and 20 msec would suffice.

The tests to obtain the curves shown in fig. 7.5 were conducted with α and β set to the values determined previously. The minimum step size was limited to the numerical value of 1 (ie. 5 mV), and a 3 second speech segment was used.

Owing to the fact that, between the limits mentioned, the value for the syllabic time constant is not critical a value of 5 msec was chosen, thereby enabling a direct comparison with the findings of Un et al to be made. Because the syllabic compandor is common to all three of the hybrid codecs, the factors influencing their relative performances are not likely to be affected by this section.

7.2.1.4 Minimum Step Size (S_o).

The measure used to investigate the effect of using a minimum step size other than 1 was the SNG. If a large minimum step size is used then, during steady state portions of the input signal especially, a large granular noise component is to be expected. Fig. 7.6 shows the variation of the SNG as a function of the minimum step size. No limit was placed on the maximum step size and α, β, etc. were set to the values determined previously.

Based on the results plotted in fig. 7.6 a minimum step size of 1 (ie. 5 mV) was used for all subsequent simulation runs and tests.

7.2.2 Dynamic Ranges of the HCDM, SHCDM and MSHCDM Encoders.

As was discussed in earlier chapters, the problem with instantaneous companding is that, for different speech signals the optimum quantizer step-size ranges may be different (cf. fig. 7.7). The hybrid companding schemes serve to adjust the step-size adaptation range according to the RMS input signal level. Fig. 7.8 shows the variation of the SQNR as a function of the RMS signal level for a 10 second segment of speech. The input signal level was varied over a range of 80 dB and a step size range of 60 dB was used.

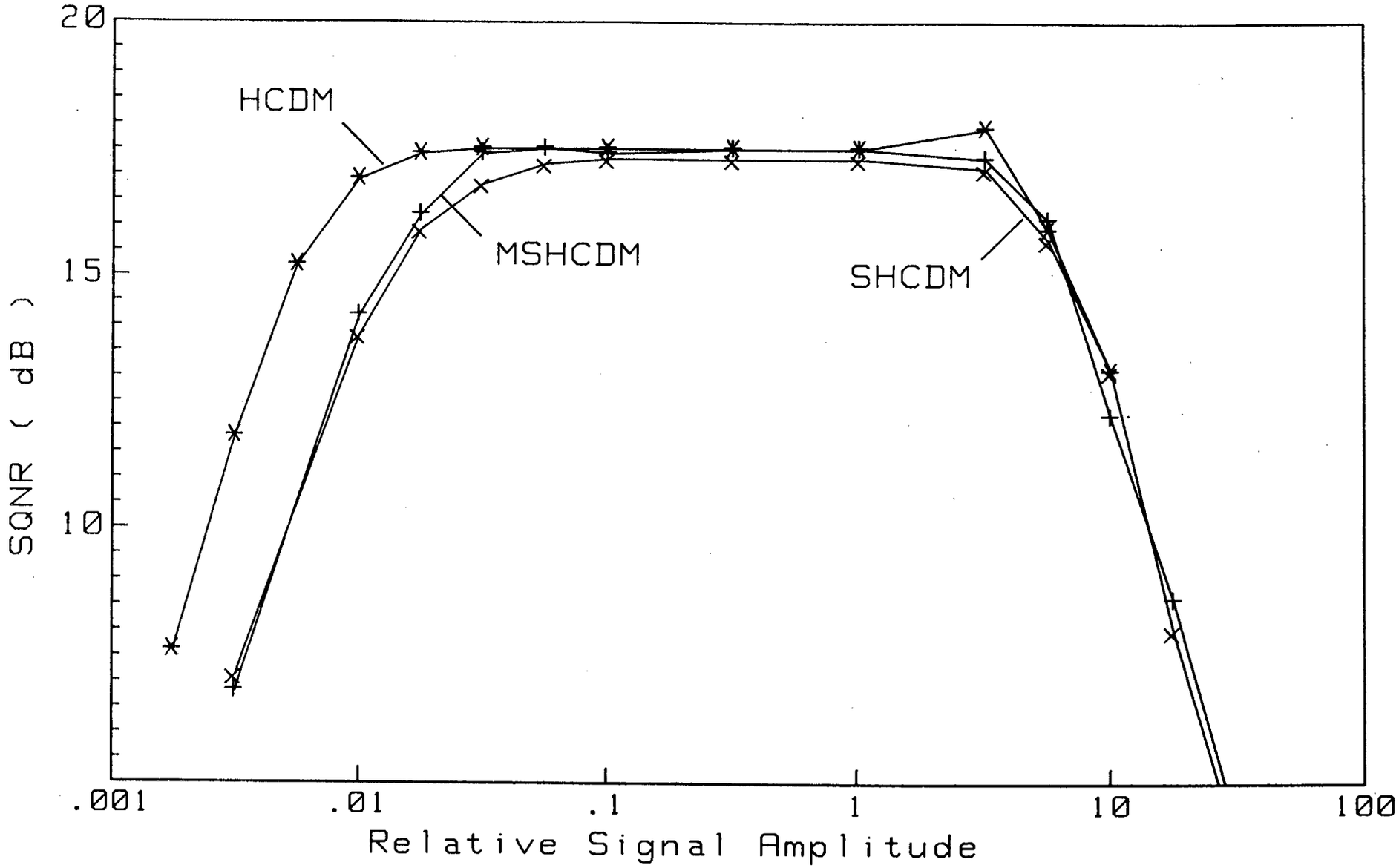


Fig.7.8 : Dynamic Range of the hybrid encoders.

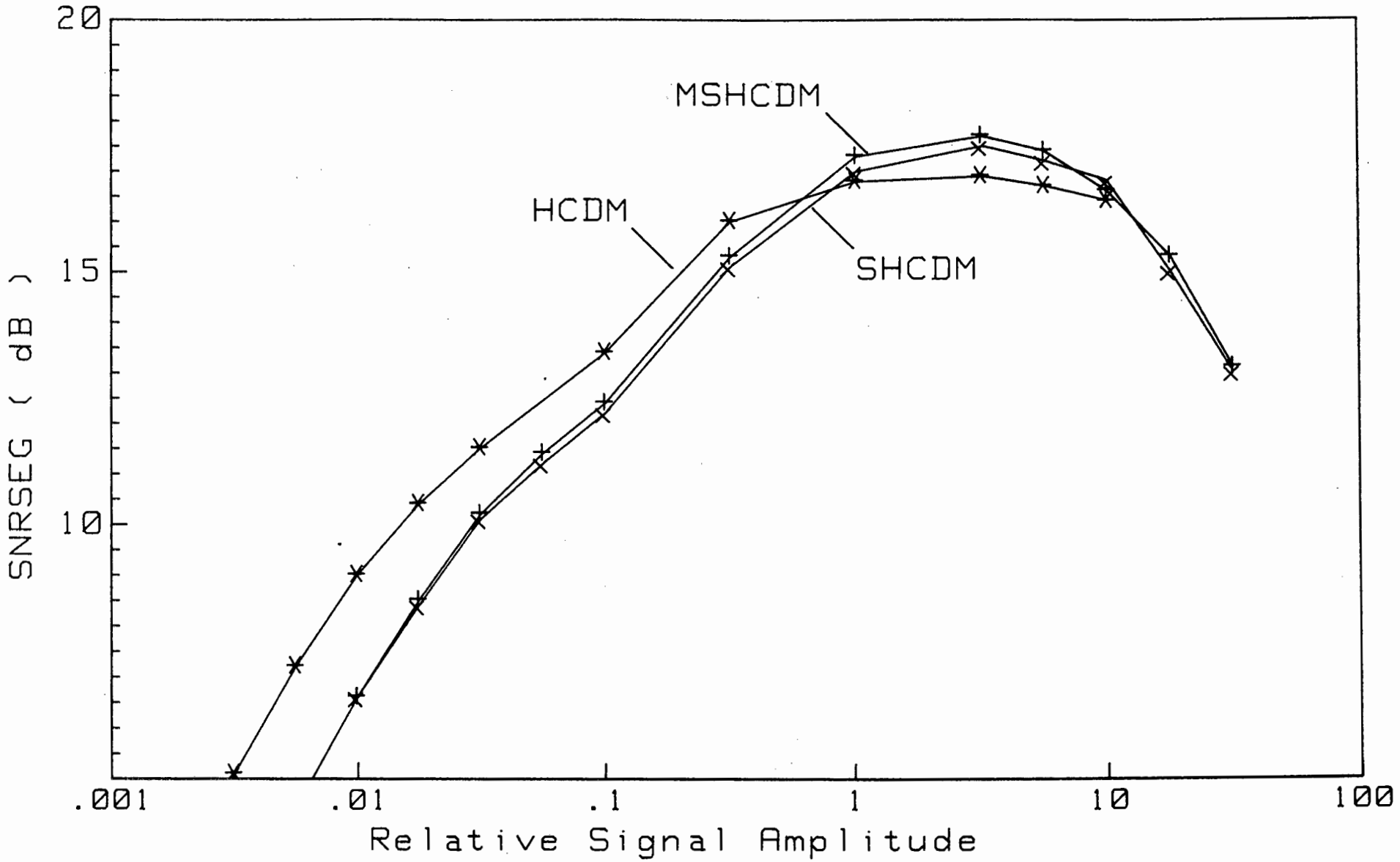


Fig.7.9a : SNRSEG as a function of Signal Amplitude.

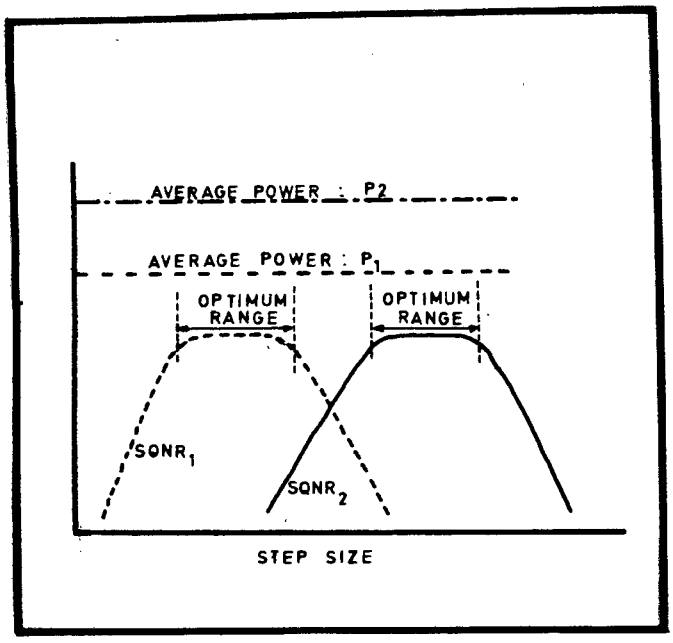


Fig. 7.7 : Dynamic range curves for different step-size adaptation ranges.

From the curves shown in fig. 7.8 it can be seen that there is no appreciable difference in the SQNR's of the three codecs over the dynamic range. The results presented for the HCDM encoder correlate very closely with those published by Un et al[17,18]. Although the HCDM encoder has a marginally higher SQNR it does not necessarily imply that it will be subjectively better than the other two hybrid codecs. Because the segmental values of SQNR show a better correlation with subjective results[43] it was decided to plot these curves to obtain a better indication of what might be expected of the subjective tests. The variation of the segmental signal-to-noise ratio (SNRSEG) as a function of the RMS signal level is shown in fig. 7.9a.

From the segmental results it would appear that the two new codecs (SHCDM and MSHCDM) might offer an improvement in performance over that of the HCDM codec.

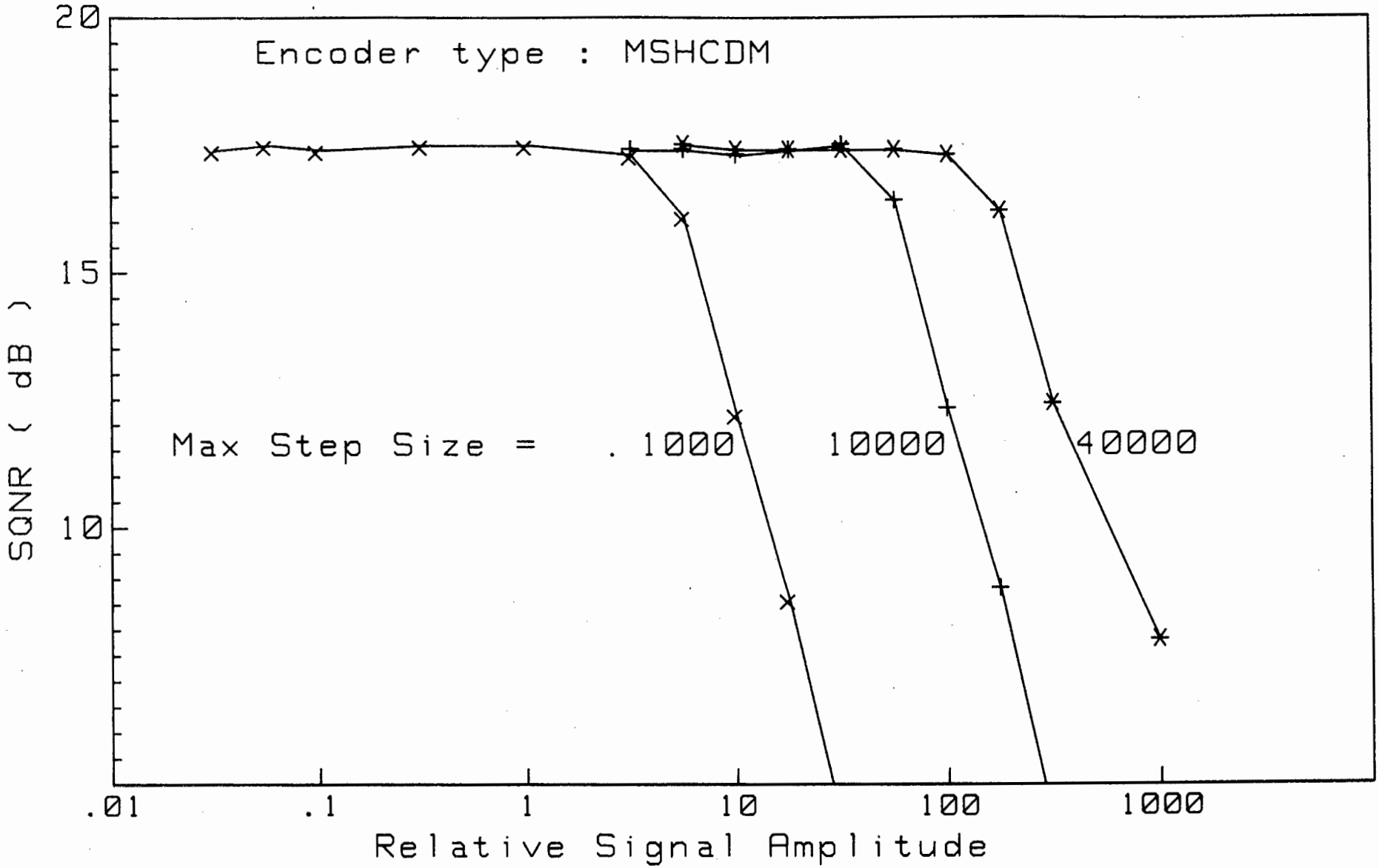


Fig.7.9b : Variation of Dynamic Range with maximum step size.

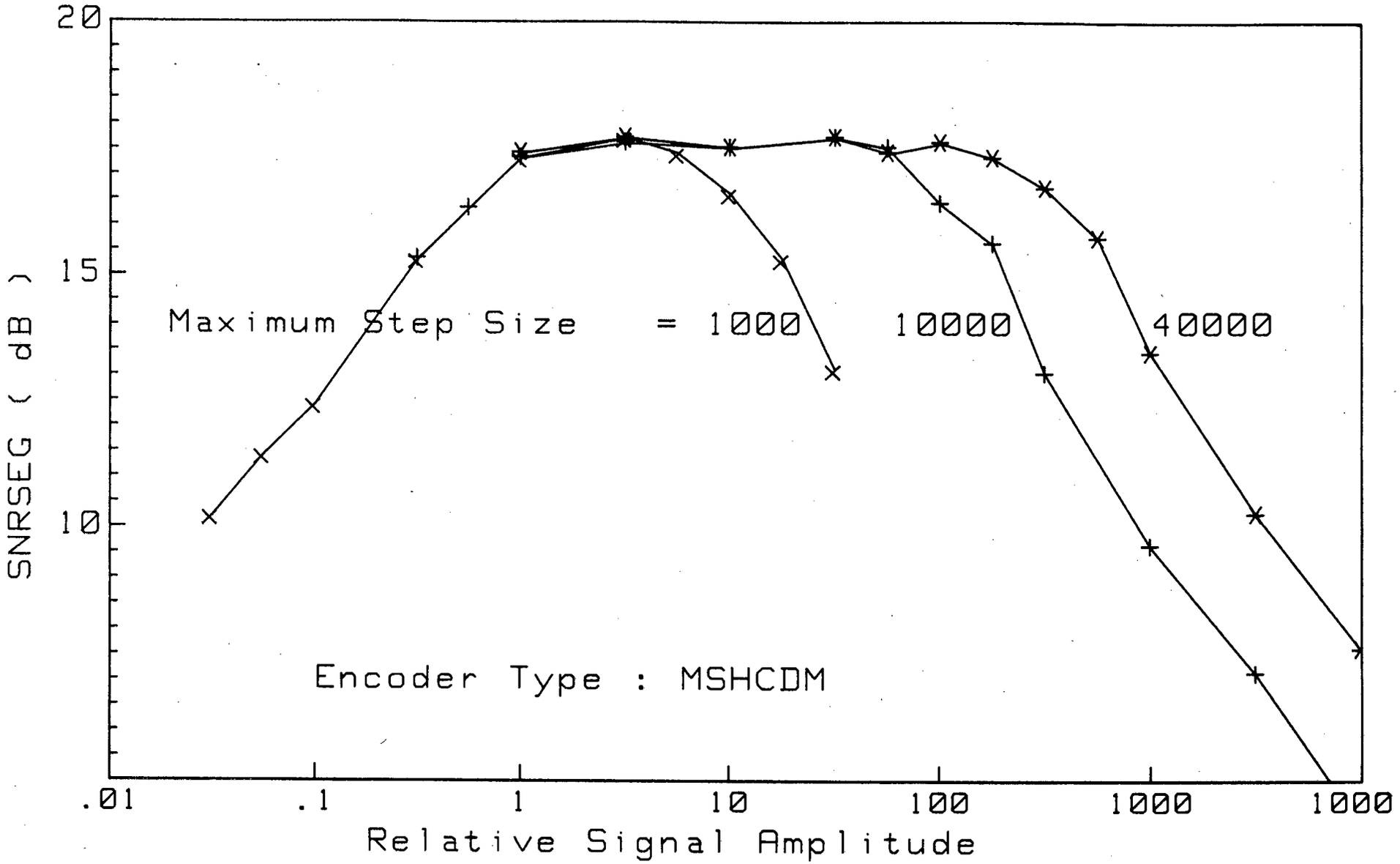


Fig.7.9c : Variation of SNRSEG with maximum step size.

7.2.2.1 Variation of the Step Size Range.

The roll-off of the SQNR at higher values of the RMS signal level is caused by the limit placed on the step size range. If no limit were placed on this parameter the SQNR curves would exhibit a roll-off only at the lower end of the RMS signal level. By increasing the step size range the level at which the "upper" roll-off point occurs can be increased. This effect is illustrated in figs. 7.9b and 7.9c (the effect is shown only for the MSHCDM codec).

7.2.2.2 Variation of Granular SNR.

Based on the previous discussion relating to SNG as a performance measure, it should be noted that, having optimized the codecs with respect to SQNR, this measure (ie. SNG) could be used to give some indication of the relative performances of the codecs. Using the fact that slope overload noise is preferable to granular noise[29] the variation of SNG as a function of the input signal level is plotted in fig. 7.10. From these results, as well as the segmental signal-to-granular noise results (SNGSEG) shown in fig. 7.11, it can be seen that both the SHCDM and MSHCDM codecs offer better performance than the HCDM codec over the bulk of the dynamic range. This assessment is particularly relevant in the light of the fact that, particularly for adaptive systems, the SNG and SNGSEG measures are the best predictors of overall subjective

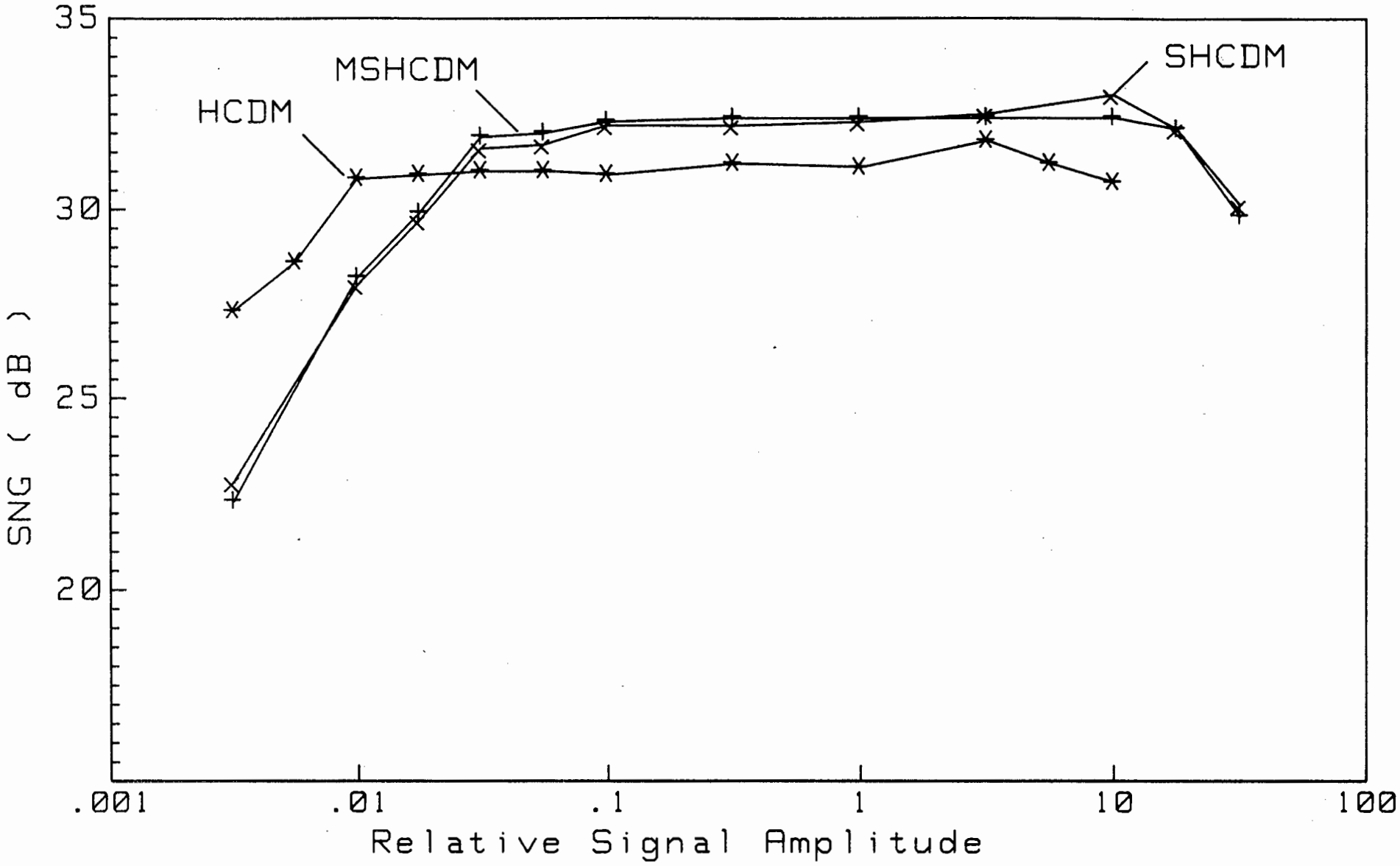


Fig.7.10 : SNG as a function of Signal Amplitude.

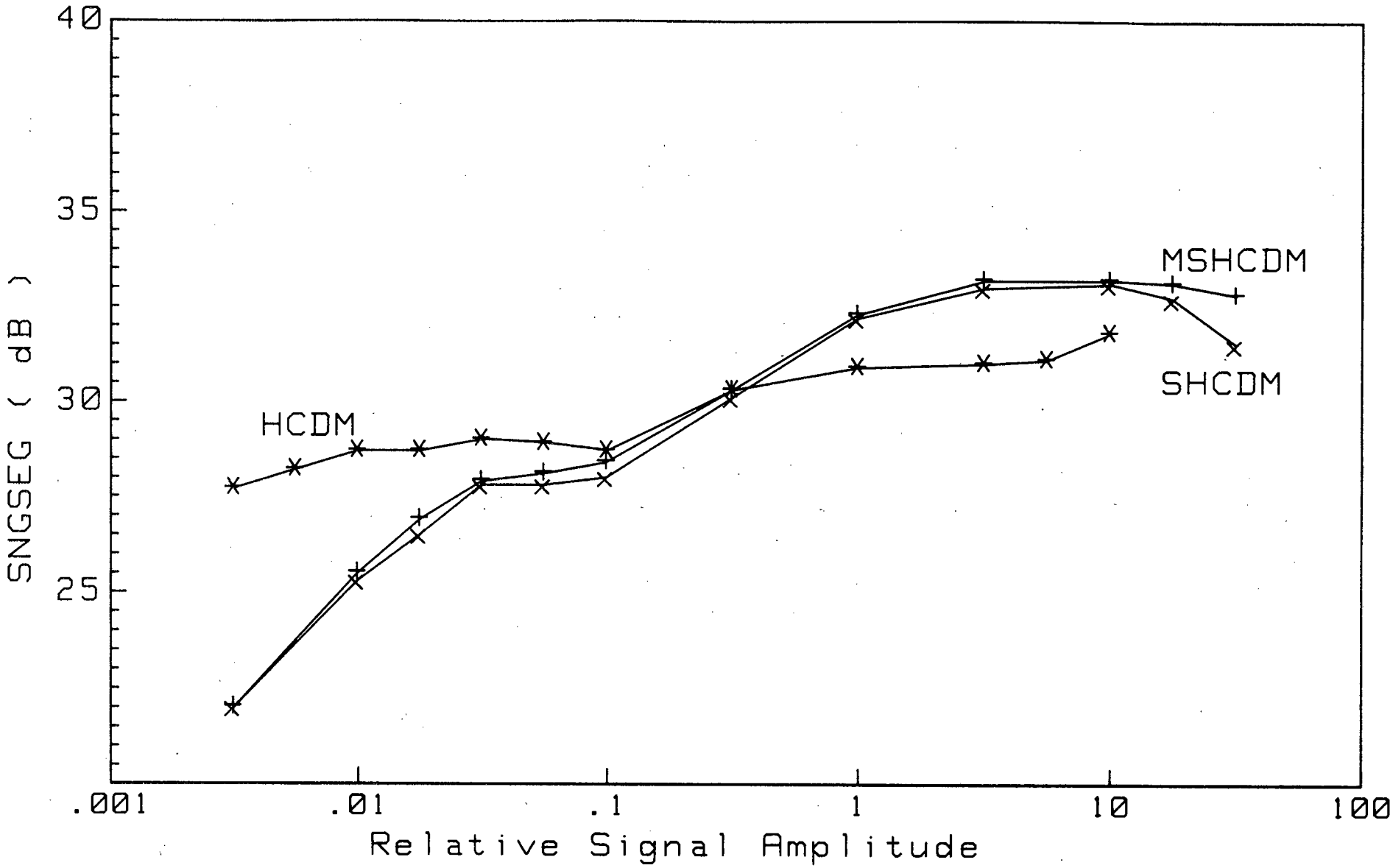


Fig.7.11 : SNGSEG as a function of Signal Amplitude.

quality[35,43,44].

NOTE : The Signal-to-Slope Overload Noise ratio can be determined from the SNG using the relationship :

$SNO = -10 \log(10^{-SQNR/10} - 10^{-SNG/10})$.

7.2.3 Cross-correlation Functions.

The use of the auto- and cross-correlation functions as a measure of performance is related to the use of the PSD function for determining the quality of a waveform coder via the Fourier Transform (cf. chapter 2 section 1.2.5).

As the waveform coder attempts to preserve the spectral characteristics of the input signal by producing a facsimile copy thereof, so the cross-correlation function gives a measure of the degree to which this has been achieved. The auto-correlation function and the PSD (of the input signal) form a Fourier Transform pair (ie. the PSD function of the input signal can be determined from the auto-correlation function via Fourier Transform techniques). Similarly, the auto-correlation function of the output signal contains information pertaining to the PSD of the output.

The cross-correlation function on the other hand is the correlation between the input and output signals. If the auto-

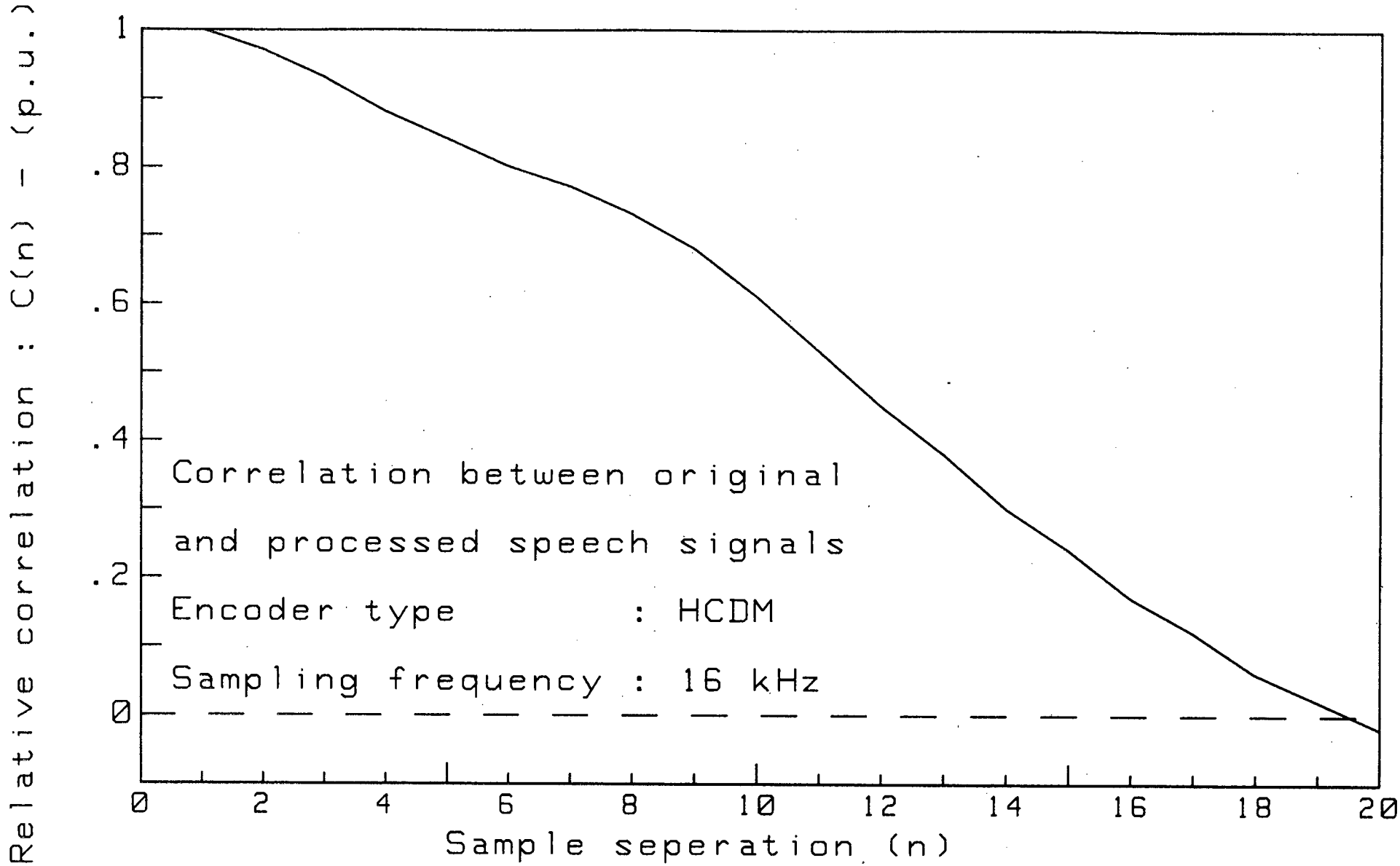


Fig.7.12a : Cross-correlation Function of sampled speech.

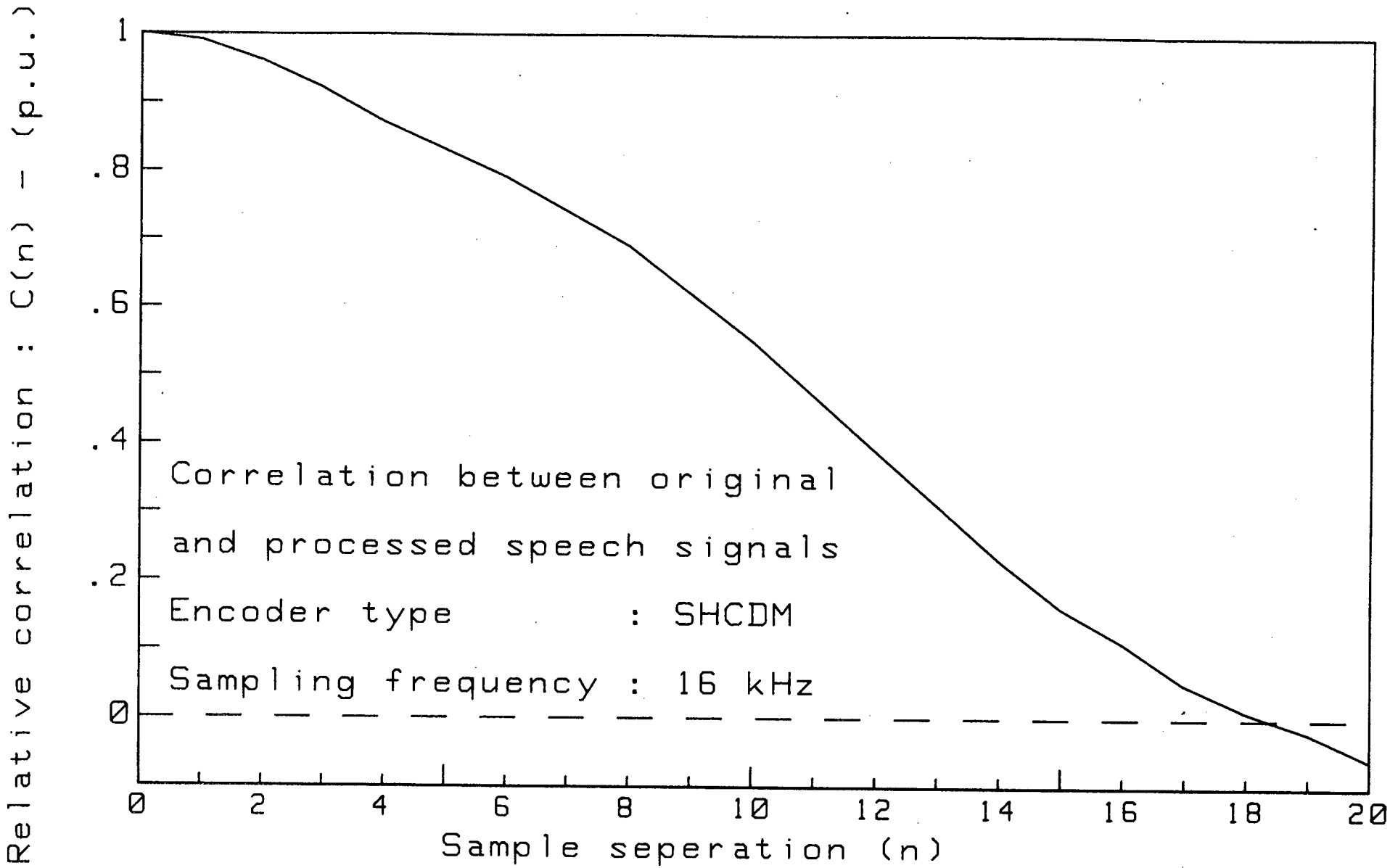


Fig.7.12b : Cross-correlation Function of sampled speech.

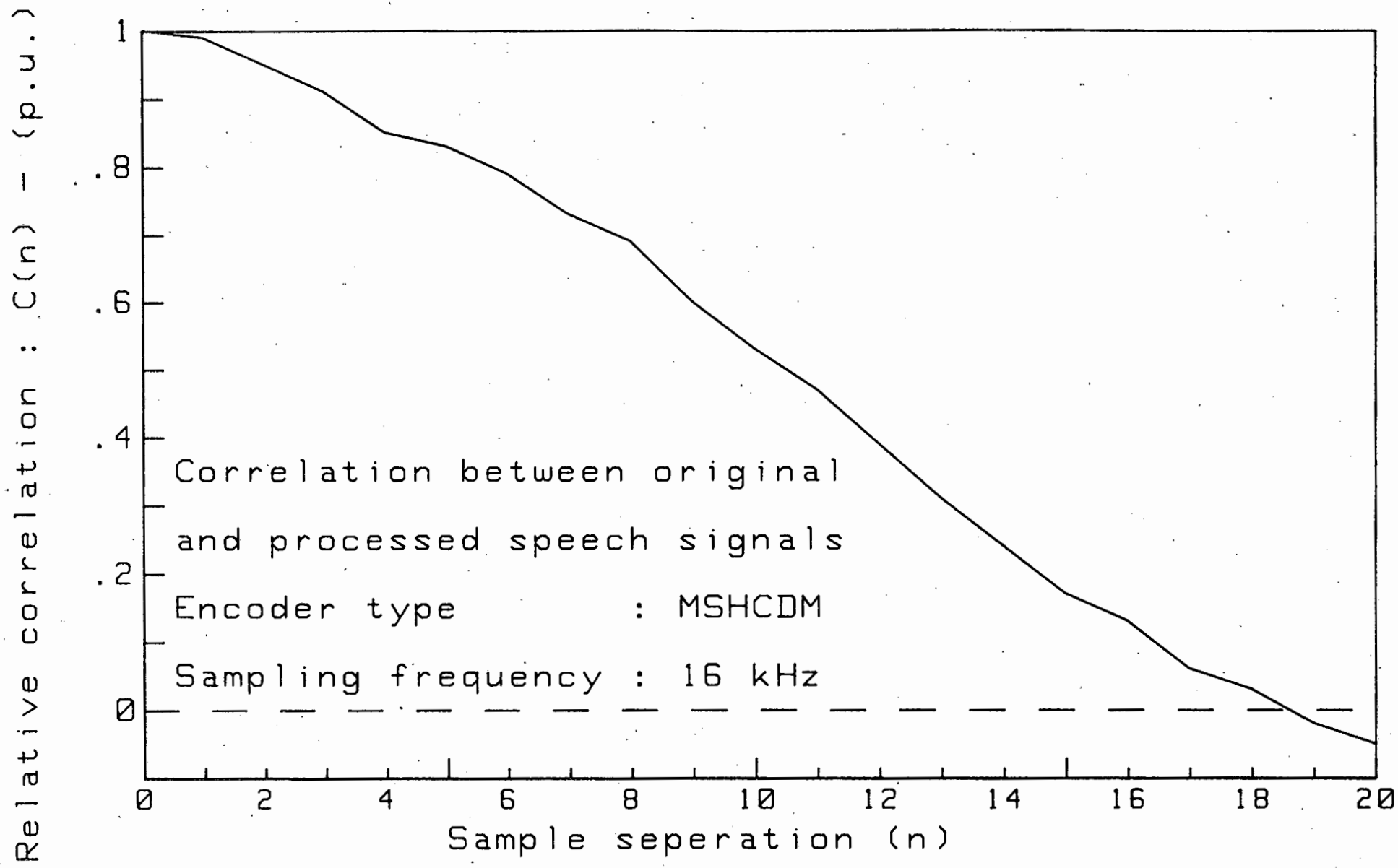


Fig.7.12c : Cross-correlation Function of sampled speech.

and cross-correlation functions are identical then the output signal represents an exact replica of the input signal. So, by comparing the two correlation functions it is possible to determine the degree to which the PSD of the input has been preserved.

The cross-correlation functions are plotted in fig. 7.12a, b, and c. The use of the cross-correlation functions is not intended to provide a rigorous method of comparing the relative performances of the different codecs, but rather an approximate indication of the degree to which the PSD of the input has been preserved (an alternative method would be to actually calculate the PSD function for the input and output speech signals).

7.2.4 Using the Feedforward Mode for Syllabic Companding.

In previous discussion it was mentioned that in order to preserve channel bandwidth the syllabic companding factor was calculated using the decoded signal instead of the actual input signal. Fig. 7.13 shows the effect of using the feedforward mode for calculating the syllabic factor (ie. using the actual input speech signal). This has been done for the MSHCDM codec alone because the effect is likely to be similar for the other two hybrid codecs.

From fig. 7.13 it is clear that there is no particular advantage to be gained from using the feedforward mode, especially when weighed against the additional bandwidth required to transmit the syllabic information.

7.2.5 Extending the Constant Factor Logic for HCDM.

In the hybrid compandor of Magill et al[24] the constant factor logic utilised five quantizer error signals (ie. b_n, . . . , b_{n-4}) as opposed to the three (b_n, . . . , b_{n-2}) used in the HCDM encoder.

The effect of this extended companding logic has been investigated with the dynamic range curves (SQNR vs RMS signal level) given in fig. 7.14. The extended logic is given in Table 7.1.

Table 7.1 : Instantaneous Companding Logic Rule.

b^n	b^{n-1}	b^{n-2}	b^{n-3}	b^{n-4}	Multiplication factor (k_n)
+	+	+			1.50
-	-	-			1.50
+	+	-			1.00
-	-	+			1.00
+	-	-			0.66
+	-	-	-	-	1.00
-	+	+			0.66
-	+	+	+	+	1.00
-	+	-			0.66
+	-	+			0.66

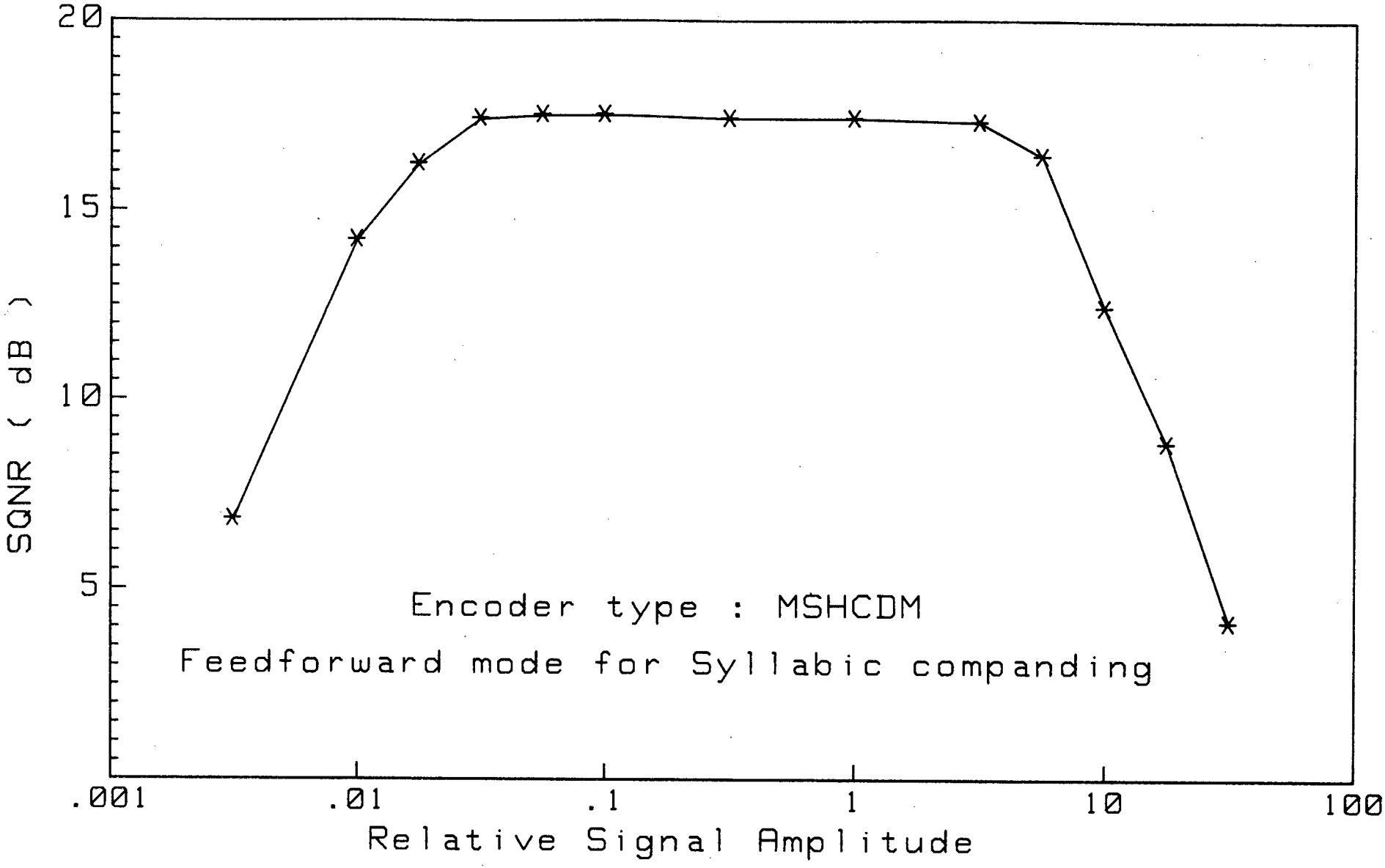


Fig.7.13 : SQNR as a function of RMS signal amplitude.

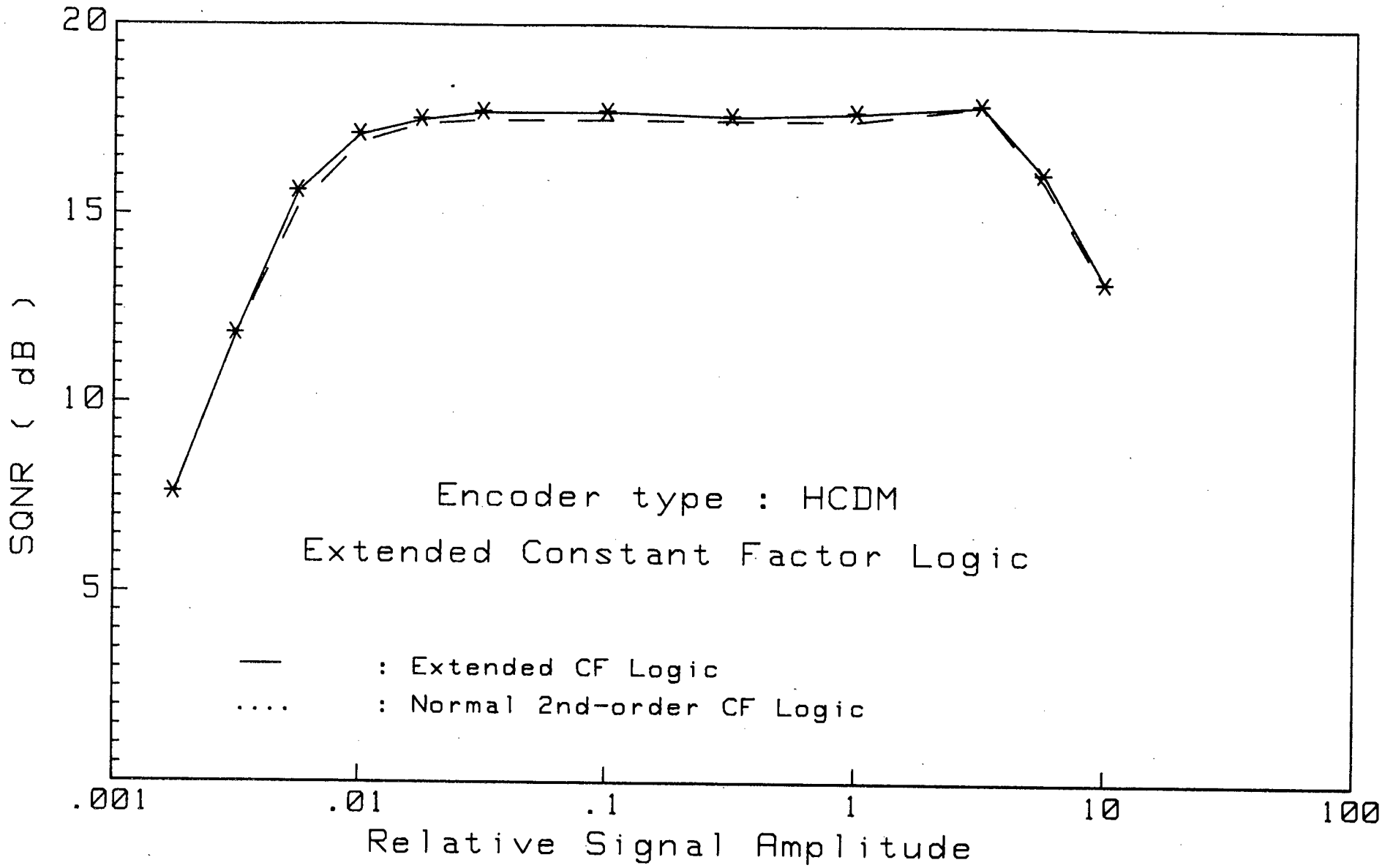


Fig.7.14 : SQNR as a function of RMS signal level.

From the results plotted in fig. 7.14 it is clear that there is very little advantage to be gained from extending the logic required to generate the instantaneous companding factor. Hence, in order to minimize the circuitry that would be required to implement this module the second-order constant factor compandor (using the present binary output and the previous two) is used.

7.2.6 High Quality Microphone vs Telephone Handset.

For all of the objective tests a high quality microphone has been used whereas for the subjective tests the speech was recorded using the microphone contained in a standard telephone handset. The frequency characteristics of the two are significantly different (cf. fig. 5.2).

The performance of the three hybrid codecs has been measured using the telephone handset microphone, and the results are shown in fig. 7.15. A reduction in the peak value of the SQNR is the result of the removal, by the telephone microphone (which exhibits band-pass characteristics), of the lower frequency components of the speech signal. It is these low frequency components which the delta modulators code the easiest and hence, when they are removed the 3 - 4 dB reduction in SQNR is experienced.

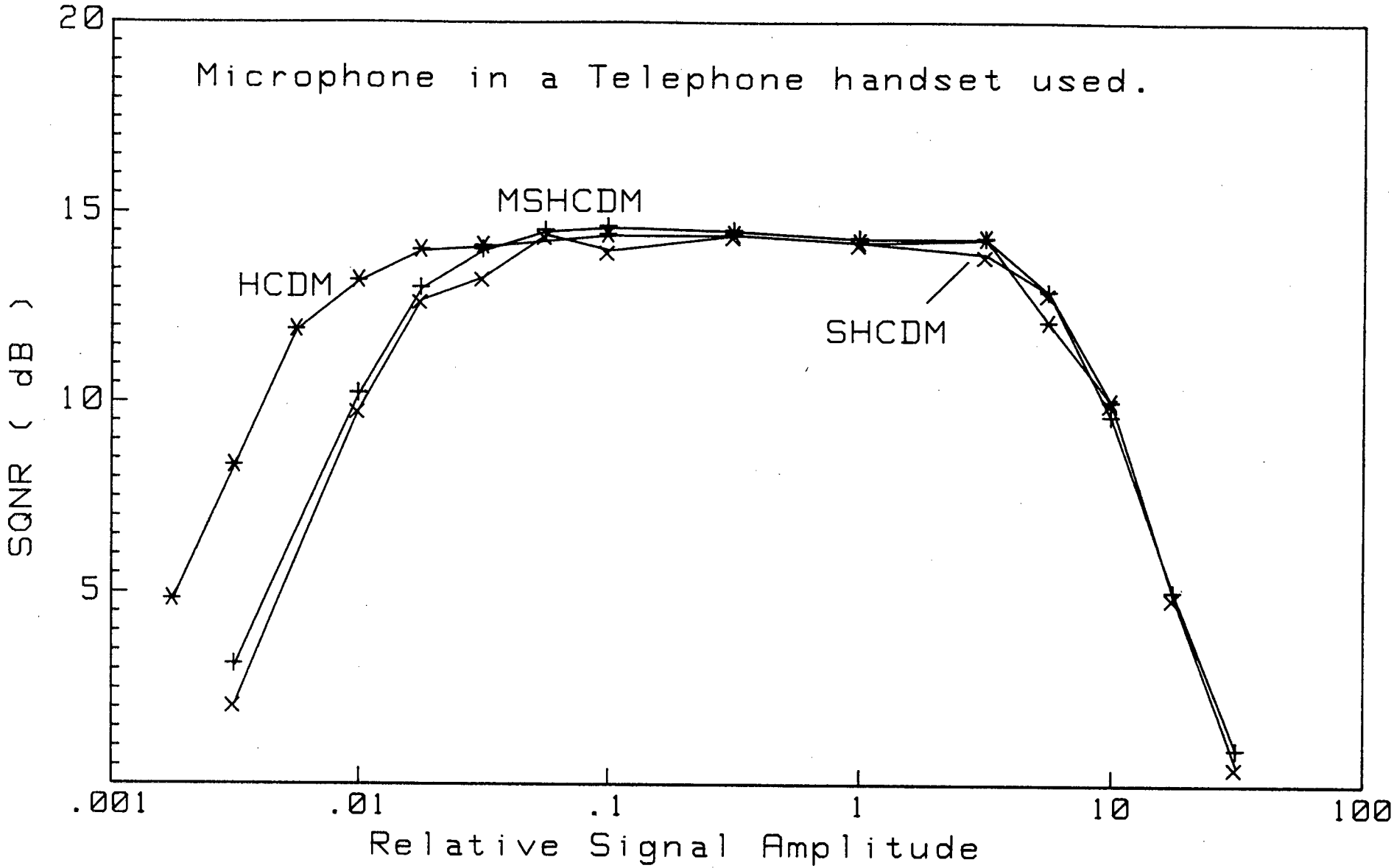


Fig.7.15 : Dynamic Range of the hybrid encoders.

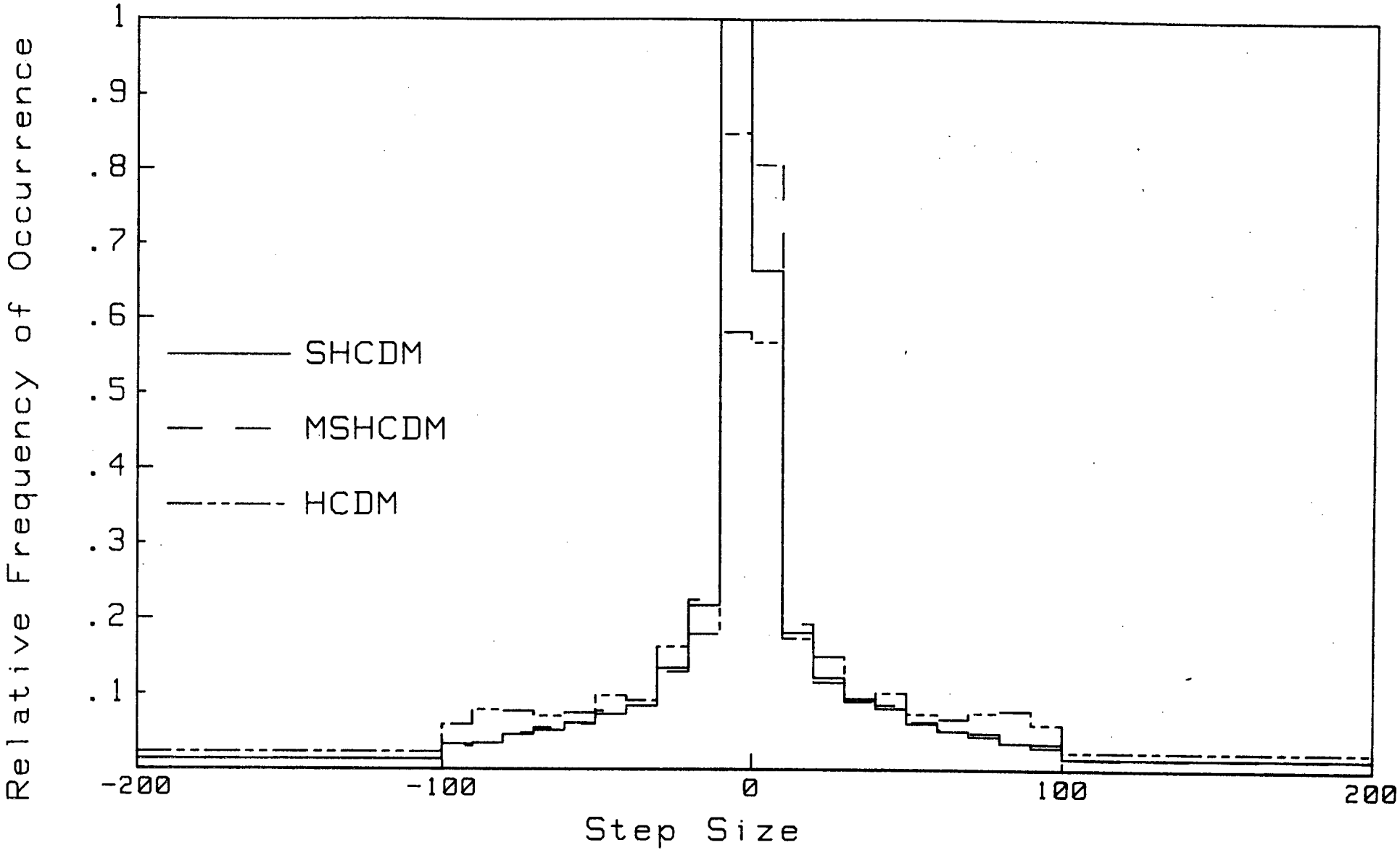


Fig.7.16 : Step Size Histograms

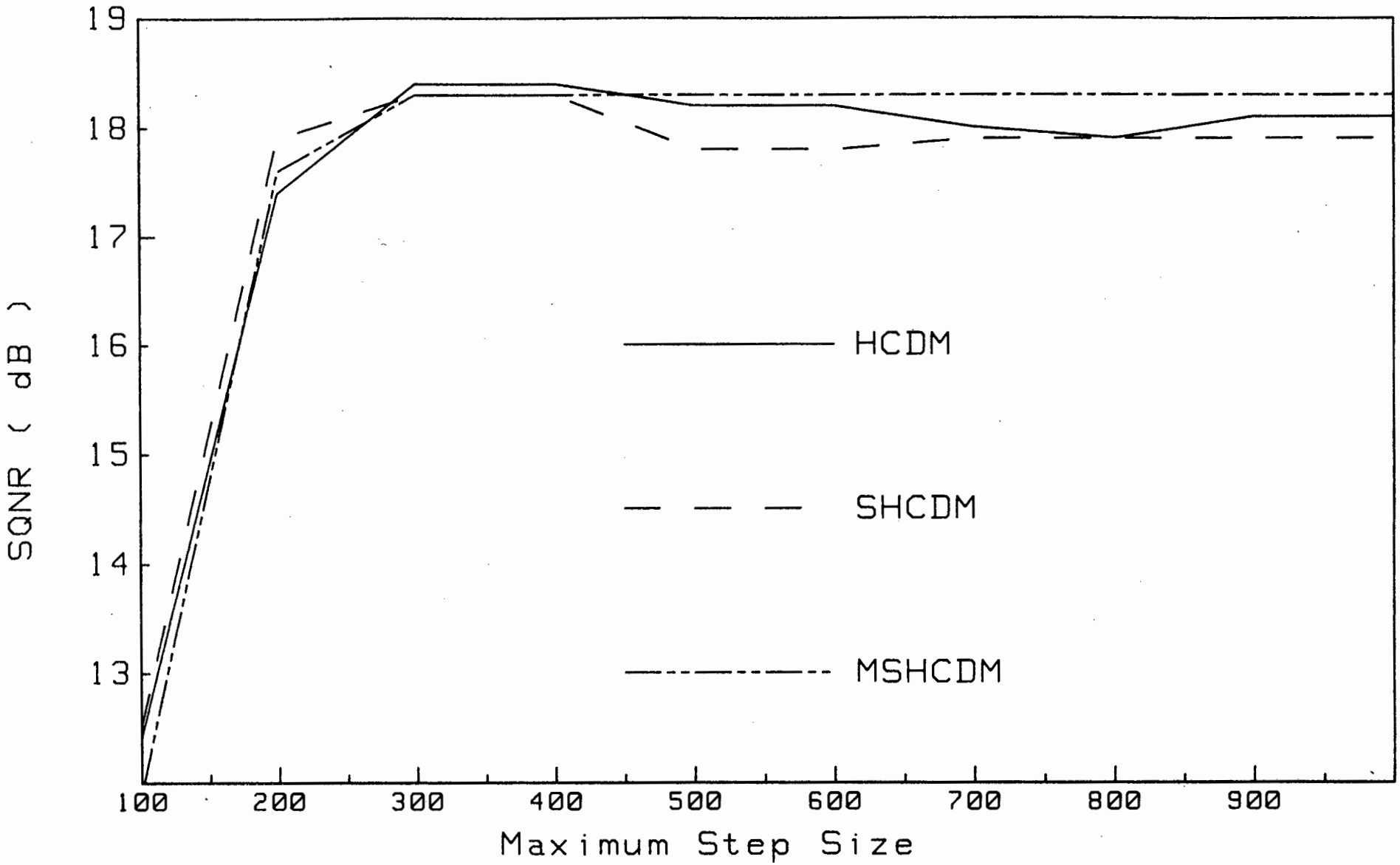


Fig.7.17 : SQNR as a function of Maximum Step Size.

7.2.7 Step Size Histograms.

Fig. 7.16 shows the step size histograms for the three codecs. The interesting feature is that whereas the SHCDM and MSHCDM encoders show a more-or-less uniform decrease (in number of occurences) as the step size increases, the HCDM has a much wider spread of step sizes in the -100 to +100 range.

Referring to fig. 7.17 (the variation of the SQNR as a function of the maximum permissible step size) it is significant that for the MSHCDM encoder no increase in SQNR is achieved by increasing the permissible maximum step size beyond the -300 to +300 range. Both the HCDM and SHCDM encoders appear to offer an improvement in performance if the maximum step sizes are limited to the -400 to +400 range. However, taking into account the fact that the peaks in the curves for these two encoders are likely to vary with different speakers and speech segments it is probably better to consider only the general trend exhibited by these curves (ie. for the SHCDM encoder a maximum step size of 700 could be used, while for the HCDM encoder a value of 900 could be used).

The significance of this is that for the MSHCDM encoder an 8-bit step size word (giving a maximum step size of $|\gamma_{max}| = 256$) could be used without significantly reducing the quality of the decoded speech. The HCDM and SHCDM encoders would require a

10-bit step size word length (giving a maximum step size of $|\gamma_{max}| = 1024$).

7.3 Subjective Results.

7.3.1 Significance of Results.

Once the subjective tests have been performed and the various scores for the three codecs have been determined it is necessary to be able to evaluate the significance of the different statistical results. In other words, it is desirable to be able to determine whether the differences in the statistics for the three codecs are significant or not.

The test for significance is performed using the Z-score test[38]. The Z-score is defined as :

$$Z = (P_1 - P_2) / \sigma_p$$

where

P_1 is the proportion of listeners who (a) heard the sentence correctly, (b) recognised the speaker, or (c) judged the test sentence to be better than the reference for hybrid codec 1,

P_2 is the proportion for hybrid codec 2.

$$\sigma_p = \sqrt{pq(1/N_1 + 1/N_2)}$$

where

N_1 is the number of tests (ie. listeners x sentences) for hybrid codec 1,

N_2 is the number of tests for hybrid codec 2,

and

$$p = (N_1 . P_1 + N_2 . P_2) / (N_1 + N_2)$$

$$q = 1 - p.$$

In all cases the table listed in Appendix G is used and a value of 5% is used for the degree of significance. In other words, we can be sure that, with a 5% margin of confidence, our assumptions regarding the significance of the statistical results are correct.

7.3.2 Intelligibility Test.

The results of the intelligibility tests are given in Table 7.2.

Table 7.2 : Codec intelligibility scores.

CODEC	CORRECT	PARTLY CORRECT	INCORRECT
HCDM	239	10	1
SHCDM	197	45	8
MSHCDM	203	44	3

The values listed above reflect the product of the number of listeners and the number of sentences heard correctly, partly correct or incorrectly for each listener. Hence, for 50 listeners and 5 sentences per codec the maximum possible value is 250.

From these results it would appear at first that the HCDM codec is significantly better than the other two. However, in the majority

of cases for the sentences which were scored 'PARTLY CORRECT' the listeners heard one word incorrectly, and this was invariably the first word of the sentence. In all such cases the listeners interpretted the sentence correctly (ie. they understood the semantic content). In Table 7.3 the 'CORRECT' and 'PARTLY CORRECT' scores have been combined.

Table. 7.3 : Modified intelligibility scores.

CODEC	CORRECT & PARTLY CORRECT	INCORRECT
HCDM	249	1
SHCDM	242	8
MSHCDM	247	3

Table 7.4 : Z-scores for codec comparisons.

COMPARISON	P_1	P_2	N_1	N_2	Z-score
HCDM & SHCDM	0.996	0.968	250	250	2.355
HCDM & MSHCDM	0.996	0.988	250	250	1.004
SHCDM & MSHCDM	0.968	0.988	250	250	1.524

Then, using the Z-score method, and referring to Table 7.4, it can be seen that there is a significant difference between the scores of the SHCDM and HCDM codecs. The main purpose of the intelligibility test was however to establish that this measure was at a satisfactory level for the isopreference tests, and not so much

for the purpose of comparing the three codecs. From the results shown in Tables 7.2. and 7.3 it can be seen that the three codecs do have a sufficiently high level of intelligibility to facilitate the isopreference tests.

7.3.3 Speaker Recognition.

As with the intelligibility tests the main purpose of this test was to establish that each of the codecs in question had a satisfactory level of speaker-identity retention. It should be noted that none of the speakers were known to the listeners before the tests and hence the results are likely to be 'coloured' by a certain learning factor (ie. as the listeners become more familiar with the voices so it becomes easier for them to identify the speaker).

The results presented in Table 7.5 reflect the number of times the speakers were identified correctly or incorrectly for each codec. The 'NOT SURE' score was used to represent the cases where, had the listener been more familiar with the speakers, a 'CORRECT' result would possibly have been recorded.

Using the Z-score as a measure of the significance (Table 7.6) it can be seen that there is no significant difference between the three codecs as far as 'CORRECT' and 'INCORRECT' sentences are

Table 7.5 : Speaker recognition scores.

	CORRECT	NOT SURE	INCORRECT
HCDM	148	55	47
SHCDM	153	55	42
MSHCDM	139	76	35

concerned and that only in the case of the MSHCDM is there any significant difference with the other two, for the 'NOT SURE' score. However, considering that it is quite common for listeners to have difficulty recognising even familiar voices (using telephone handsets), and taking into account the fact that, as

Table 7.6 : Z-scores for speaker recognition tests.

COMPARISON	P_1	P_2	N_1	N_2	Z-score
CORRECT					
HCDM & SHCDM	0.592	0.612	250	250	0.457
HCDM & MSHCDM	0.592	0.556	250	250	0.814
SHCDM & MSHCDM	0.612	0.556	250	250	1.270
NOT SURE					
HCDM & SHCDM	0.220	0.220	250	250	0.000
HCDM & MSHCDM	0.220	0.304	250	250	2.136
SHCDM & MSHCDM	0.220	0.304	250	250	2.136
INCORRECT					
HCDM & SHCDM	0.188	0.168	250	250	0.584
HCDM & MSHCDM	0.188	0.140	250	250	1.449
SHCDM & MSHCDM	0.168	0.140	250	250	0.867

mentioned previously, the speakers were totally unfamiliar to the listeners, the results of these tests indicate a satisfactory reproduction of the speakers' voices by the three codecs.

Having now established that the retention of the speaker's identity and the intelligibility for the three codecs are at a satisfactory level the results of the isopreference tests are presented.

7.3.4 Isopreference Tests.

The results for these tests are presented in both graphical and tabular form (Fig. 7.18 & Table 7.7). In Table 7.7 the differences between the three hybrid codecs have been tested for significance for each level of the SNR of the reference system. This was done because in the isopreference tests themselves the three codecs were not compared directly with each other, but to the reference system. From the method outlined in chapter 6 it should be clear that the most important region of the curves of fig. 7.18 is the area in which the codecs are isopreferent with the reference system (this area has been shaded).

Referring to fig. 7.18 it can be seen that the three curves have the characteristics that one would expect - ie. when the SNR of the reference system is very low the majority of the listeners

Table 7.7 : Results of Isopreference tests.

COMPARISON	SNR (dB)	P_1	P_2	N_1	N_2	|Z-score|
HCDM & SHCDM	-6	1.00	0.96	50	50	1.429
	-3	0.88	1.00	50	50	2.526
	0	0.50	0.80	50	50	3.145
	+2	0.24	0.50	50	50	2.693
	+4	0.32	0.28	50	50	0.436
	+6	0.22	0.06	50	50	2.306
	+8	0.20	0.10	50	50	1.400
	+10	0.08	0.04	50	50	0.842
HCDM & MSHCDM	-6	1.00	0.98	50	50	1.005
	-3	0.88	0.98	50	50	1.960
	0	0.50	0.76	50	50	2.693
	+2	0.24	0.84	50	50	6.019
	+4	0.32	0.56	50	50	2.417
	+6	0.22	0.38	50	50	1.746
	+8	0.20	0.32	50	50	1.368
	+10	0.08	0.32	50	50	3.000
SHCDM & MSHCDM	-6	0.96	0.98	50	50	0.586
	-3	1.00	0.98	50	50	1.005
	0	0.80	0.76	50	50	0.483
	+2	0.50	0.84	50	50	3.615
	+4	0.28	0.56	50	50	2.836
	+6	0.06	0.38	50	50	3.862
	+8	0.10	0.32	50	50	2.701
	+10	0.04	0.32	50	50	3.644

prefer the test system (eg. for a SNR of -6 dB P_2 for the MSHCDM encoder is 0.98 - ie. 49 listeners preferred the test system), while on the other hand, when the SNR of the reference system is increased sufficiently the majority of the listeners prefer the reference system.

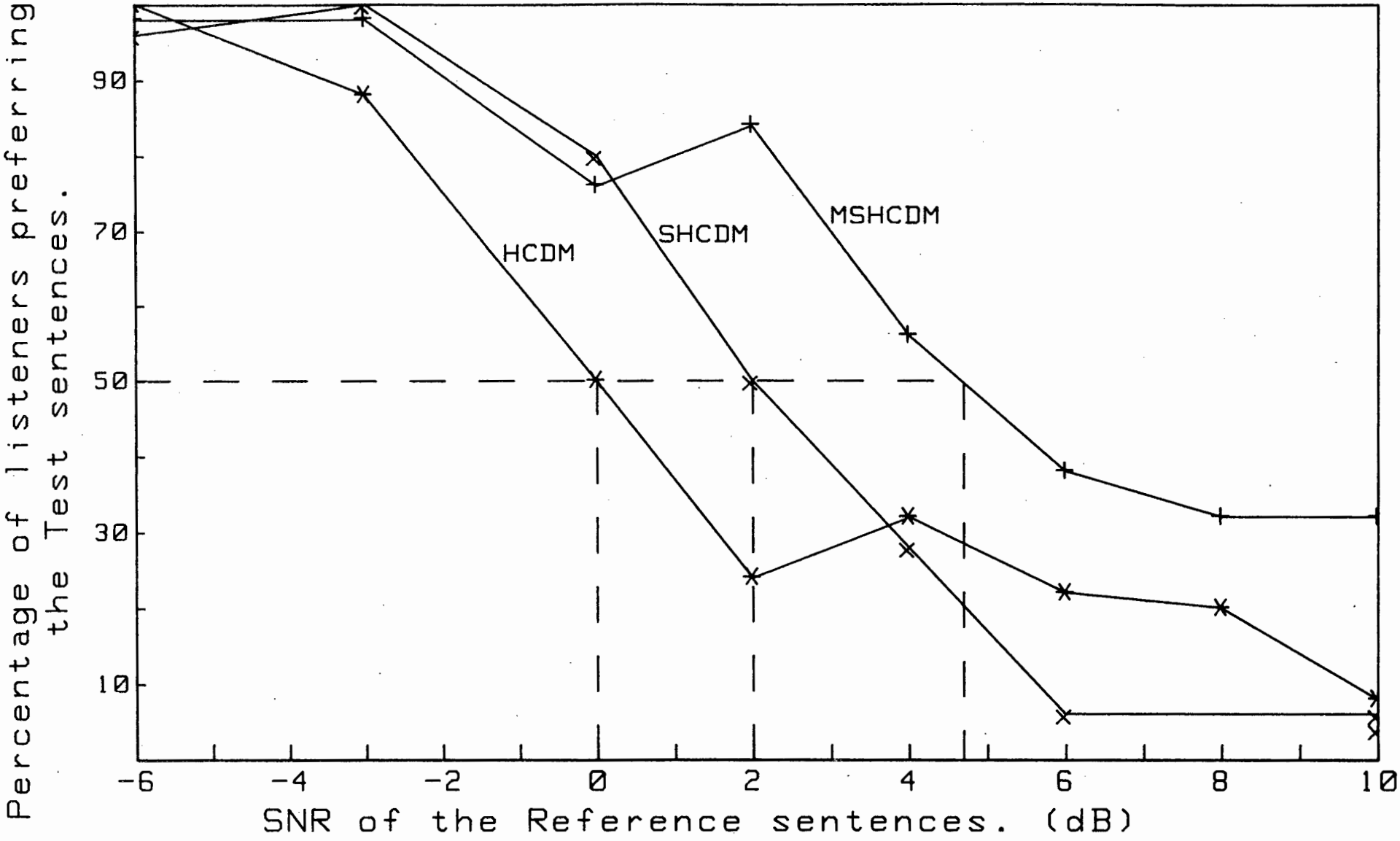


Fig.7.18 : Psychometric curves for the Isopreference test.

From the curves shown in fig. 7.18 and from the results shown in Table 7.7 it should be clear that the MSHCDM encoder's performance is significantly better than the other two. The MSHCDM codec has a subjective SNR of approximately 4.8 dB compared to the SNR's of the HCDM and SHCDM codecs of 0 dB and 2 dB respectively. Hence, both the hybrid codecs using Song Voice instantaneous adaptation show an improvement in quality over the HCDM codec of Un et al.

7.4 Summary.

The objective results show that the performance of the MSHCDM codec is, at worst, equivalent to that of the HCDM codec. The Granular Signal-to-Noise ratios indicate that a moderate improvement is to be had using the MSHCDM codec. If used to predict the outcome of the subjective tests it is significant that the SNG and SNGSEG measures indicate that the best of the three codecs is the MSHCDM codec, followed by the SHCDM codec, and then the HCDM codec. As has been discussed, this has in fact been born out by the subjective results. Comparing the SNG and SNGSEG curves of the three codecs it is noticable that for low RMS levels the HCDM codec is better. Essentially this means that the 'idle channel' performance of this codec is better. However, despite this fact, subjectively the SHCDM and MSHCDM codecs rated better. This would appear to indicate that the improvement in 'idle channel' performance offered by the HCDM codec is not perceptually

significant.

Overall, the most significant results are those obtained from the subjective tests. The results of these indicate a clear advantage offered by the MSHCDM codec. This also correlates with the results of the SNG tests in that the codec producing the least amount of granular noise is likely to be preferred to the others (assuming of course that all the codecs have been previously optimized).

Basically the difference in granular noise powers is caused by the relatively slower change in step size of the Song Voice compandor as compared to the Constant Factor compandor (the Song Voice step size varies linearly whereas the Constant Factor step size varies exponentially). The important point arising out of the difference between the natures of the step size adaptations is that :

(i) During silent periods, or very low amplitude signal periods, the Song Voice compandor produces a binary output signal 110011001100 . . . 110011. . .[21] (resulting in a step size sequence +1 +2 -1 -2 +1 +2 -1 -2 . . .) whereas the Constant Factor compandor produces a binary output sequence 1010101010 (resulting in a step size sequence of +1 -1 +1 -1 +1 -1 . .). Hence, during these periods the MSHCDM codec produces more granular noise than the HCDM codec.

(ii) On the other hand, during large signal amplitude periods the exponential step size adaptation of the Constant Factor compandor produces considerably more noise than the linearly adapting Song Voice compandor.

Considering the fact that voiced speech is subjectively more susceptible to the addition of granular noise than unvoiced speech (cf. chapter 2), and the fact that voiced speech has a higher probability of occurence than unvoiced speech (chapter 2), the performance of the codecs during speech portions with high RMS levels is probably more significant.

7.4.1 Implementation Considerations.

The implementation of the three algorithms is another significant factor in deciding between them. The stated aim of the research project is to produce an integrated circuit version of the delta modulation speech codec. It is important therfore that, in order to reduce the cost of a system using the codec, the codec algorithm be efficient in terms of hardware as well. In this regard the MSHCDM codec offers a significant advantage over the HCDM codec in that one less multiplication operation is required. This is due to the fact that the instantaneaous companding factor of the MSHCDM codec is generated by addition as opposed to multiplication - the multiplication of Λ by γ_0 and b_{n-1} is very

simple as γ_0 is 1 and b_{n-1} is either +1 or -1 (the instantaneous companding is given by : $\gamma_n = |\gamma_{n-1}|.b_n + \gamma_0.\Lambda.b_{n-1})$. Also, the multiplication by the factor α (ie. $\Lambda = \alpha.E$) is much simpler to achieve in the MSHCDM case than the HCDM case. This is because to implement a multiplication by 0.25 requires a simple two-place arithmetic shift whereas to implement a multiplication by 0.8 is far more complicated.

Based on the results of the objective and subjective tests, and considering that the MSHCDM codec involves a significantly less complex hardware implementation, we consider this codec to offer an improved method for implementing a hybrid companding delta modulator in digital hardware.

CHAPTER 8

PROSPECTIVE FUTURE DEVELOPMENT

Having established the superiority of the MSHCDM codec it is necessary, in conclusion, to outline the direction in which future research developments could take place.

8.1 Extending to a Variable-Rate System.

There are basically two possible methods for extending the improvements in performance offered by the hybrid companding schemes discussed in this work. Overall system improvement could be achieved at the expense of considerable complexity, either to achieve an improvement in the output speech quality, or in order to reduce the transmission bandwidth while maintaining the speech quality. An alternative technique, which can be implemented with a moderate increase in complexity, involves using variable-rate sampling.

It has been found that by combining the hybrid companding technique with a variable-rate sampling scheme an improvement of between 3 and 4 dB can be realised[45]. The sampling rate of the encoder is adjusted according to the input speech activity. By using a buffer the encoder binary output signal can be transmitted at a fixed rate.

By extending this variable-rate system to a multisubscriber system an improvement in SQNR of approximately 10 dB, over the conven-

tional single-subscriber HCDM system, can be achieved[46]. This improvement is realised by employing a dynamic buffer control algorithm which utilises statistical multiplexing; and a variable-rate sampling scheme which adjusts the sampling rate according to both the speech activity and the buffer occupancy.

An important element of the variable-rate system design is the buffer size. If the buffer is too small overflow will occur frequently during high speech activity periods. However, if the buffer is too large, the resultant delay experienced at the decoder gives rise to unnatural output speech. The use of statistical multiplexing (which is similar in concept to TASI) necessitates the design of a buffer control scheme that reduces the probability of overflow.

Based on the findings of Un and Kim et al[45,46], as regards the multisubscriber variable-rate system, it was envisaged that a similar degree of improvement for the MSHCDM system could be attained if the fixed-rate version, developed to date, were extended to a variable-rate one. Preliminary work on this has already been carried out (by S.C.Hall, as part of a postgraduate research project) with the initial results confirming the findings of Un et al[45]. Further work still needs to be done on developing a suitable efficient buffer control algorithm.

8.2 Optimization of the Algorithm Implementation.

All investigations to date have involved implementing the codec algorithms as detailed previously in this work. However, because the eventual aim of this research is to implement an Application Specific Integrated Circuit (ASIC) version of the codec it will be necessary to optimize the design in terms of, for instance, the number of gates required to realise a certain function.

It is envisaged that the syllabic compandor used in the three hybrid codecs could be replaced by a simpler rectifier-low-pass filter combination.

ie. the syllabic companding function

$$\Lambda = \alpha E = \alpha \sum_{i=1}^{M-1} (X_i - X_{i-1})^2 / (M - 1)$$

could be replaced by a simple low-pass function

$\Lambda = \alpha E_i$ which is updated at the syllabic rate.

where

$E_i = L . E_{i-1} + A_i$ which is updated at the sampling rate.

A_i is the absolute value of X_i (the approximation signal), and

$L = \exp(-\beta_s T)$,

β_s being the inverse of the syllabic low-pass filter time constant.

Work would need to be done on optimising and evaluating the design of the module to perform this function.

8.3 Correlating the Results of Objective and Subjective Tests.

Perhaps the single most important development in the field of speech codec performance analysis is the development of a single test procedure that would enable different codecs to be rated on a single quality scale according to their relative performances. Ideally it would be desirable to have a single objective measure from which the subjective analyses results could be accurately predicted.

The subjective test procedure used here followed the recommendations of the IEEE closely and was, we believe, adequate enough to illustrate the superiority of the MSHCDM codec over the HCDM codec. However, in the course of the tests a few points arose which should be highlighted to prevent possible confusion.

Firstly, it is common practice for the speech signals used for the purpose of both subjective and objective tests to be recorded using high quality microphones. This practice would be acceptable if the eventual system were to be incorporated in a high quality dedicated communications link (where it is not unlikely that high quality microphones might be used). However, if the system were

to be incorporated into an extensive commercial communications network it is more meaningful to conduct these tests using a microphone, and playback equipment, characteristic of the devices used in the network. In this way more meaningful objective results would be obtained - ie. the results would be indicative of the quality of a realistic implementation of the system under discussion.

8.3.1 Inclusion of Network Parameters.

In this work the performance of the codecs alone have been considered. However, when the codecs are included in a network a number of factors are likely to affect their performance.

Firstly, if a packet switched network is used, the effect of packet losses on the codecs' performance would need to be ascertained. Secondly, if the variable-rate system mentioned previously is implemented then the effect of buffer overflows would need to be investigated and the buffer control algorithm modified accordingly. It is common for simple bit error performance ratings to be quoted. However, in the network systems mentioned above the channel bit error rate is not likely to be the most significant influence on performance. In any event, it is quite common for digital channels with bit error rates of 10^{-7}, or better, to be used, and most waveform codecs perform adequately

under these conditions.

Another factor that could be investigated is the design of a suitable protocol for voice packet switching networks.

In conclusion it should be stated that the primary objective at this stage is to characterise the performance of the hybrid codecs using a variable-rate system. Once that has been concluded the ASIC implementation should be finalised. This, in turn, will enable the scope of the research project to be extended to network considerations, such as variable-rate multisubscriber systems, packet switched protocols, etc.

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APPENDIX A

In this appendix the derivation of the equations for the Song Voice Adaptive Delta Modulation (SVADM) encoder is given[16].

Referring to fig. 3.8 it should be noted that the predictor (in the encoder) can only use the past output sequence

$b_{k-1}, b_{k-2}, \ldots, b_1$ (which shall be represented as b^{k-1})

and the past predicted sequence

$X_{k-1}, X_{k-2}, \ldots, X_1$ (which shall be represented as X^{k-1})

to predict the sample of the source Y_k. The estimator (in the decoder) on the other hand can use b^k and X^k to estimate Y_k.

To optimise the predictor Song et al[16] minimised the mean square of the difference between Y_k and X_k. Noting that X_k is a function of b^{k-1} and X^{k-1} (ie. $X_k = X_k(b^{k-1}, X^{k-1})$) the mean square of the difference can be represented by :

$$E\{ Y_k - X_k(b^{k-1}, X^{k-1}) \}^2. \qquad (1)$$

If the output of the estimator is represented by R_k then the optimum estimator is obtained by minimising

$$E\{ Y_k - R_k(b^k, X^k) \}^2. \qquad (2)$$

In the system of Song et al the two past samples were used and hence b^{k-1} and X^{k-1} can be given by 'b_{k-1}, b_{k-2}' and 'X_{k-1}, X_{k-2}' respectively.

The minimisation of (1) and (2) leads to the equations :

$$X_k = E\{ Y_k \mid X_{k-1}, X_{k-2}, b_{k-1}, b_{k-2} \} \qquad (3)$$

$$R_k = E\{ Y_k \mid X_k, X_{k-1}, b_k, b_{k-1} \} \qquad (4)$$

In the derivation of Song et al it was assumed that a Markov source was used, with samples given by :

$$Y_k = \rho Y_{k-1} + \lambda_{k-1}. \qquad (5)$$

If each λ_{k-1} is uncorrelated, normal with a zero mean and standard deviation σ, then it can be shown that

$$X_{k+1} = \rho R_k. \qquad (6)$$

The result in (6) can be derived by substituting (5) into (3).

Using the two past samples, the estimator equation is given by :

$$R_k = \int_{-\infty}^{\infty} Y_k P(Y_k \mid X_{k-1}^{\,k}, b_{k-1}^{\,k}) dY_k$$

which has solutions[16] :

(7a) $b_k = +1, \; b_{k-1} = -1$:

$$R_k = \frac{\int_{-\infty}^{X_{k-1}} [\rho q'(z_{k-1}) Y_{k-1} + (\sigma/\sqrt{2\pi}) \exp(-\tfrac{1}{2} z_{k-1}^{\,2})] P(Y_{k-1}) dY_{k-1}}{\int_{-\infty}^{X_{k-1}} q'(z_{k-1}) P(Y_{k-1}) dY_{k-1}}$$

(7b) $b_k = +1, \; b_{k-1} = +1$:

$$R_k = \frac{\int_{X_{k-1}}^{\infty} [\rho q'(z_{k-1}) Y_{k-1} + (\sigma/\sqrt{2\pi}) \exp(-\tfrac{1}{2} z_{k-1}^{\,2})] P(Y_{k-1}) dY_{k-1}}{\int_{-\infty}^{X_{k-1}} q'(z_{k-1}) P(Y_{k-1}) dY_{k-1}}$$

(7c) $b_k = -1, \; b_{k-1} = +1$:

$$R_k = \frac{\int_{X_{k-1}}^{\infty} [\rho q'(z_{k-1}) Y_{k-1} + (\sigma/\sqrt{2\pi}) \exp(-\tfrac{1}{2} z_{k-1}^{\,2})] P(Y_{k-1}) dY_{k-1}}{\int_{-\infty}^{X_{k-1}} q'(z_{k-1}) P(Y_{k-1}) dY_{k-1}}$$

(7d) $b_k = -1, \; b_{k-1} = -1$:

$$R_k = \frac{\int_{-\infty}^{X_{k-1}} [\rho q'(z_{k-1}) Y_{k-1} + (\sigma/\sqrt{2\pi}) \exp(-\tfrac{1}{2} z_{k-1}^{\,2})] P(Y_{k-1}) dY_{k-1}}{\int_{-\infty}^{X_{k-1}} q'(z_{k-1}) P(Y_{k-1}) dY_{k-1}}$$

where

$$q(y) = 1 - q'(y) \quad \text{and} \quad q'(y) = (1/\sqrt{2\pi}) \int_z \exp(-u^2/2)\,du \qquad (7e)$$

$$z_{k-1} = (X_k - \rho Y_{k-1})/\sigma \qquad (7f)$$

To simplify the results $q'(y)$ is approximated by :

$$q'(y) \approx \begin{cases} \frac{1}{2}\exp(-ay^2) & \text{for } y > 0 \\ 1 - \frac{1}{2}\exp(-ay^2) & \text{for } y < 0 \end{cases} \qquad (8)$$

It has been found that the value of 'a' which results in the greatest simplification of (7) is $a = 0.5$[16].

If the input samples are highly correlated ($\rho > 0.9$) then (7) can be reduced to :

(9a) $bk = +1,\ bk-1 = -1$:

$$R_k = (2/\sqrt{2\pi})\sigma + \{ X_k - (\sigma\exp(-\tfrac{1}{2}y_k^2))/(\sqrt{2\pi}\, q(y_k))$$

(9b) $bk = +1,\ bk-1 = +1$:

$$R_k = (2/\sqrt{2\pi})\sigma + \{ X_k - (\sigma\exp(-\tfrac{1}{2}y_k^2))/(\sqrt{2\pi}\, q(y_k))$$

(9c) $bk = -1,\ bk-1 = +1$:

$$R_k = (2/\sqrt{2\pi})\sigma + \{ X_k - (\sigma\exp(-\tfrac{1}{2}y_k^2))/(\sqrt{2\pi}\, q(y_k))$$

(9d) $bk = -1,\ bk-1 = -1$:

$$R_k = (2/\sqrt{2\pi})\sigma + \{ X_k - (\sigma\exp(-\tfrac{1}{2}y_k^2))/(\sqrt{2\pi}\, q(y_k))$$

where

$$y_k = (\rho X_{k-1} - X_k)/\sigma \qquad (9e)$$

Since ρ is approximately equal to 1,

$$R_k = \rho X_{k+1} \approx X_{k+1}$$

and

$$y_k \approx (X_{k-1} - X_k)/\sigma,$$

which allows (9) to be re-written as :

$$-y_{k+1} = \pm\sqrt{2/\pi} + (\exp(-\tfrac{1}{2}y_k^2))/(q'(y_k)\sqrt{2\pi}) \qquad b_{k-1} = +1 \qquad (10a)$$

$$-y_{k+1} = \pm\sqrt{2/\pi} - (\exp(-\tfrac{1}{2}y_k^2))/(q'(y_k)\sqrt{2\pi}) \qquad b_{k-1} = -1 \qquad (10b)$$

The system described by (9) is highly non-linear and not easily implementable using digital integrated circuits.

From equation (5), using the approximation $\rho \approx 1$:

$$\sigma^2 = E(Y_k - \rho Y_{k-1})^2 \approx E(Y_k - Y_{k-1})^2$$

which, considering that X_k is the estimate of Y_k, modifies to

$$\sigma^2 \approx E(X_k - X_{k-1})^2.$$

If a further approximation is made, namely

$$\sigma^2 \equiv (X_k - X_{k-1})^2$$

then

$$y_k \approx \mathrm{sgn}(X_{k-1} - X_k). \qquad (11)$$

Using equations (7e) and (11) the equations (9) simplify to :

for $X_k - X_{k-1} > 0$

$$R_k = X_{k+1} = \begin{cases} X_k + 0.51(X_k - X_{k-1}) & b_k = +1,\ b_{k-1} = -1 \qquad (12a) \\ X_k + 1.15(X_k - X_{k-1}) & b_k = +1,\ b_{k-1} = +1 \qquad (12b) \\ X_k - 0.51(X_k - X_{k-1}) & b_k = -1,\ b_{k-1} = +1 \qquad (12c) \\ X_k - 1.15(X_k - X_{k-1}) & b_k = -1,\ b_{k-1} = -1 \qquad (12d) \end{cases}$$

and for $X_k - X_{k-1} < 0$

$$R_k = X_{k+1} = \begin{cases} X_k - 0.51(X_k - X_{k-1}) & b_k = +1,\ b_{k-1} = -1 \quad (13a) \\ X_k - 1.15(X_k - X_{k-1}) & b_k = +1,\ b_{k-1} = +1 \quad (13b) \\ X_k + 0.51(X_k - X_{k-1}) & b_k = -1,\ b_{k-1} = +1 \quad (13c) \\ X_k + 1.15(X_k - X_{k-1}) & b_k = -1,\ b_{k-1} = -1 \quad (13d) \end{cases}$$

From equations (13a) and (13b), for $b_k = +1$ $(X_{k+1} - X_k) > 0$. If we let $k = n - 1$, then when $b_{n-1} = +1$ $(X_n - X_{n-1}) > 0$. However, this contradicts the condition for equations (13a) - (13d). Hence, equations (13b) and (13c) cannot exist. Similarly, equations (12a) and (12d) can be ignored, and so the system simplifies to :

$$R_k = X_{k+1} = \begin{cases} X_k + 0.815| X_k - X_{k-1} | - 0.3| X_k - X_{k-1} | & \\ \qquad b_k = +1,\ b_{k-1} = -1 \quad (14a) \\ X_k + 0.815| X_k - X_{k-1} | + 0.3| X_k - X_{k-1} | & \\ \qquad b_k = +1,\ b_{k-1} = +1 \quad (14b) \\ X_k - 0.815| X_k - X_{k-1} | + 0.3| X_k - X_{k-1} | & \\ \qquad b_k = -1,\ b_{k-1} = +1 \quad (14c) \\ X_k - 0.815| X_k - X_{k-1} | - 0.3| X_k - X_{k-1} | & \\ \qquad b_k = -1,\ b_{k-1} = -1 \quad (14d) \end{cases}$$

R_k can then be written as :

$$R_k = X_{k+1} = X_k + g + h.$$

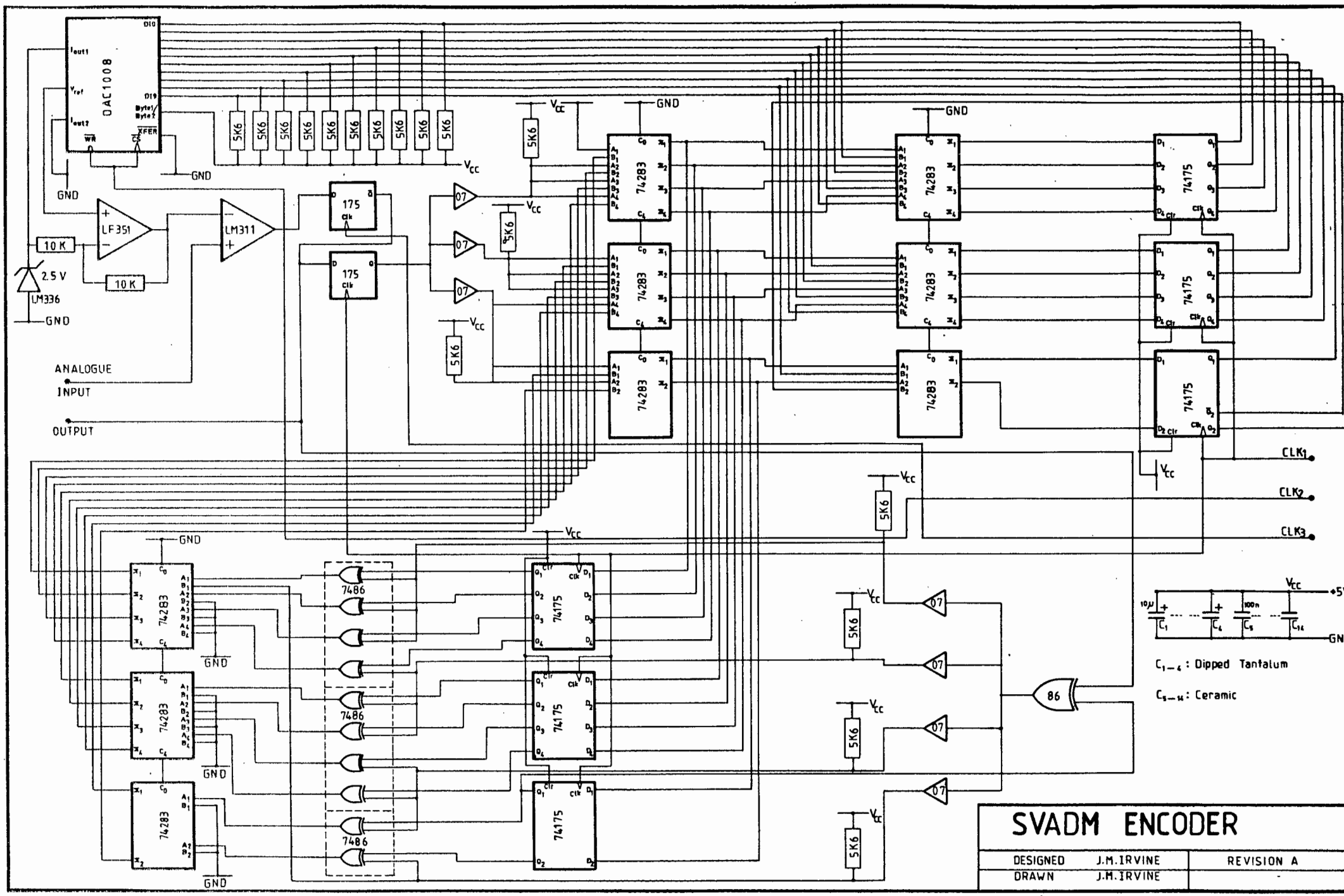
where

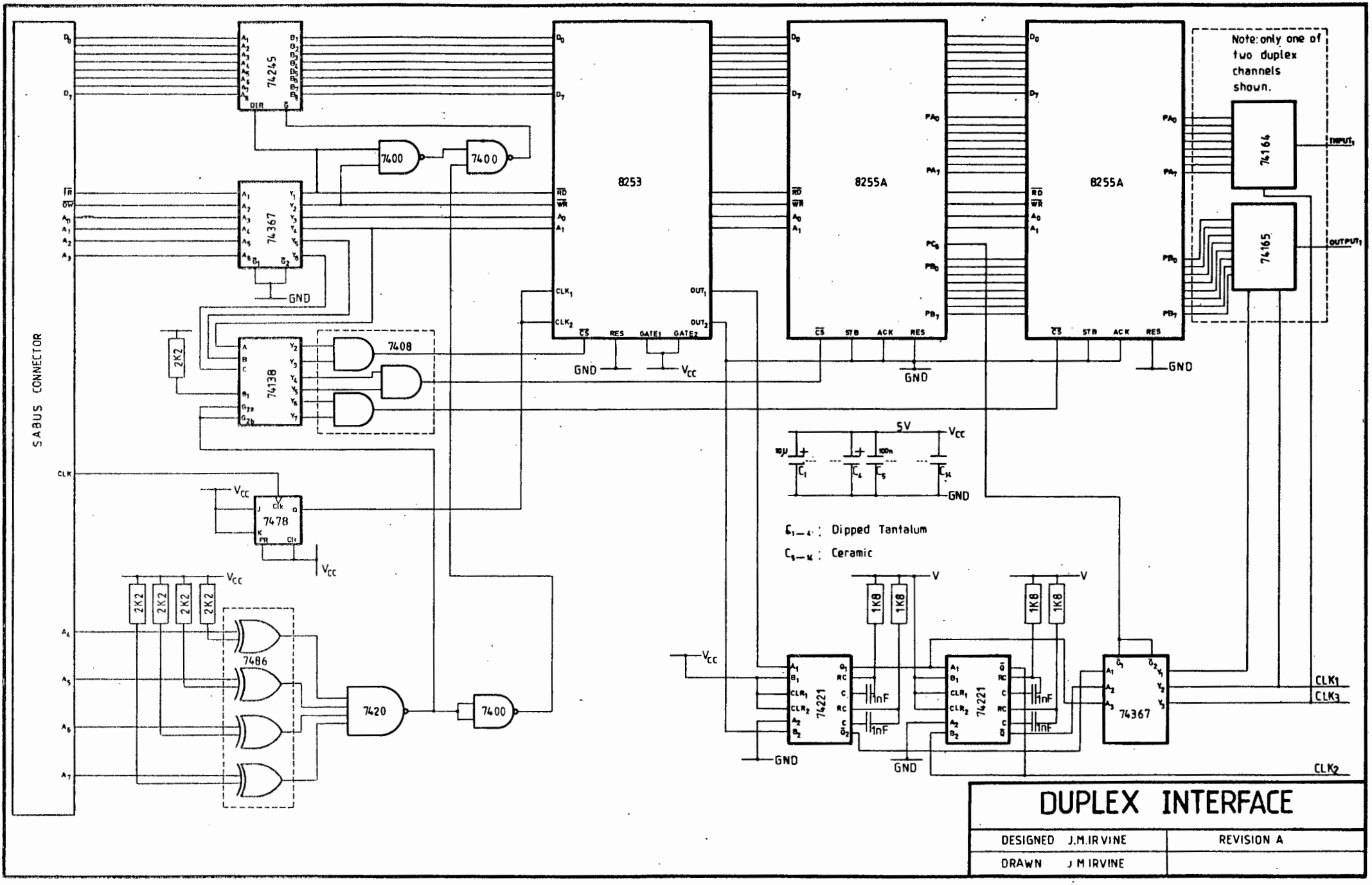
$g = g(b_k, X_k - X_{k-1})$, and

$h = h(b_{k-1}, X_k - X_{k-1})$.

APPENDIX B

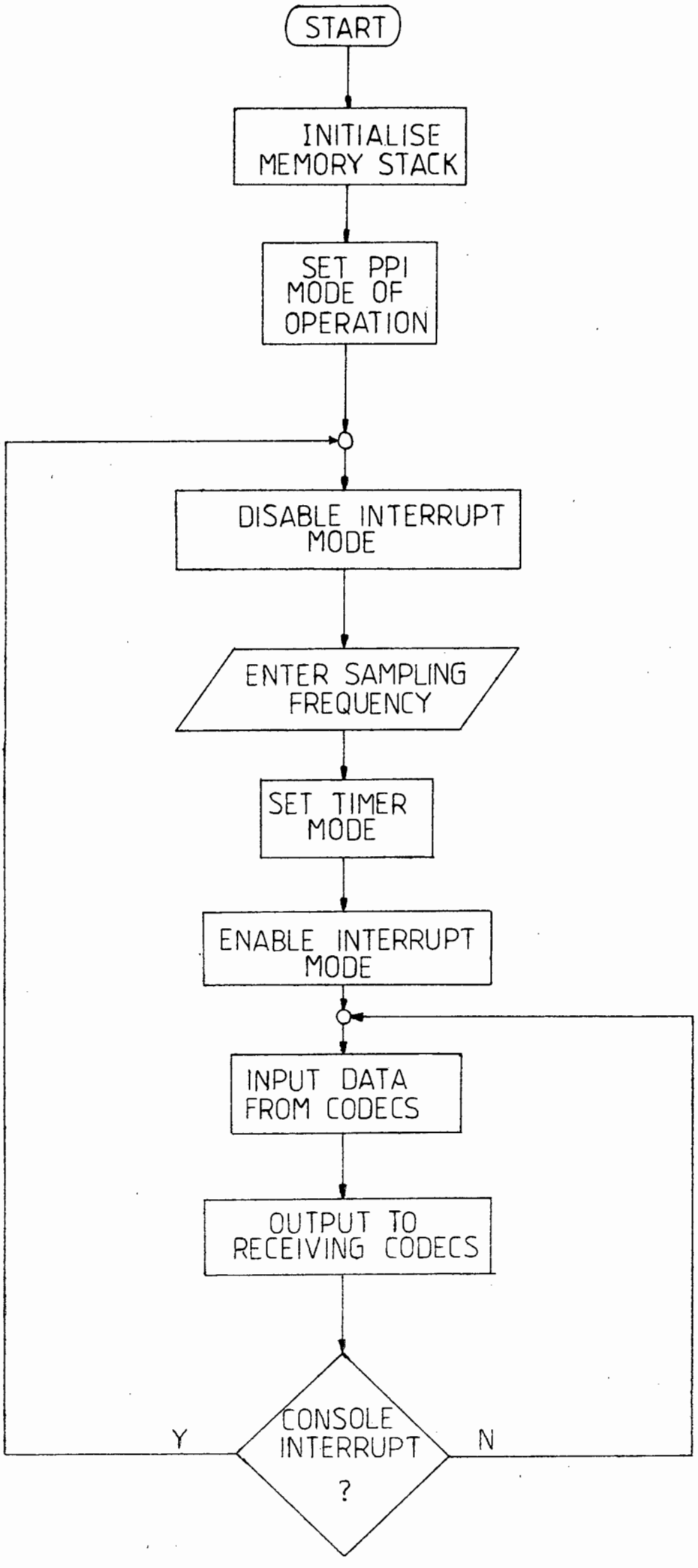
This appendix contains the circuit diagrams for the hardware implementation of the Song Voice Adaptive Delta Modulation (SVADM) system described in Chapter 4.


DAC1008
LF351
LM311
LM336
2.5 V
10 K
10 K
ANALOGUE
INPUT
OUTPUT
GND
V_{CC}
5K6
175
74283
74175
7486
07
86
CLK1
CLK2
CLK3
+5V
10µ
100n
C_{1-4} : Dipped Tantalum
C_{5-14} : Ceramic
SVADM ENCODER
DESIGNED J.M.IRVINE
DRAWN J.M.IRVINE
REVISION A


SABUS CONNECTOR
74245
74367
7400
7400
8253
8255A
8255A
Note: only one of two duplex channels shown.
74164
74165
INPUT1
OUTPUT1
GND
2K2
74138
7408
GND
Vcc
5V
Vcc
10µ
100n
C1
C4
C5
C14
GND
7478
Vcc
Vcc
C1-4 : Dipped Tantalum
C5-14 : Ceramic
2K2
2K2
2K2
2K2
7486
7420
7400
1K8
1K8
1K8
1K8
V
V
Vcc
74221
74221
74367
1nF
1nF
1nF
1nF
GND
GND
CLK1
CLK3
CLK2
DUPLEX INTERFACE
DESIGNED J.M.IRVINE
REVISION A
DRAWN J M IRVINE

APPENDIX C

This appendix contains the software for implementing the full duplex communications link described in Chapter 4.


START
INITIALISE MEMORY STACK
SET PPI MODE OF OPERATION
DISABLE INTERRUPT MODE
ENTER SAMPLING FREQUENCY
SET TIMER MODE
ENABLE INTERRUPT MODE
INPUT DATA FROM CODECS
OUTPUT TO RECEIVING CODECS
CONSOLE INTERRUPT ?
Y
N

```
LOC  OBJ       LINE       SOURCE STATEMENT

                    1 $PAGELENGTH(40)
                    2 $TITLE('Appendix C - Duplex Communication Program')
                    3 ;*****************************************************************;
                    4 ;                                                                 ;
                    5 ;                       VOICE PACKET SWITCHING                    ;
                    6 ;                       ----------------------                    ;
                    7 ;                                                                 ;
                    8 ;                       J.M.IRVINE                                ;
                    9 ;                       3rd JUNE 1983                             ;
                   10 ;                                                                 ;
                   11 ;                                                                 ;
                   12 ;*****************************************************************;
                   13 ;                                                                 ;
                   14 ;       TELPHN.A85: A Program to give direct Full Duplex communication ;
                   15 ;       ----------                                                ;
                   16 ;                                                                 ;
                   17 ;*****************************************************************;
1800               18         ORG 1800H
                   19                         ;
003F               20 UOTEST  EQU 003FH       ; Test for char in USART 0.
0072               21 OUTMSG  EQU 0072H       ; Prints a string on the console.
0060               22 GETECH  EQU 0060H       ; Input from the console.
1C00               23 BUFFER  EQU 1C00H       ; Temporary storage.
1C01               24 TOP     EQU 1C01H       ; Start of Data Stack.
000D               25 CR      EQU 0DH         ; ASCII <CR>.
000A               26 LF      EQU 0AH         ; ASCII <LF>.
003B               27 MASK    EQU 3BH         ; Mask programming byte.
                   28 $EJECT
```

```
LOC   OBJ        LINE        SOURCE STATEMENT

                   29 ;*************************************************************************;
                   30 ;                                                                         ;
                   31 ; MAINLINE OF THE PROGRAM.                                               ;
                   32 ; ========================                                               ;
                   33 ; INPUTS          : BYTES OF DATA FROM THE DELTA MODULATOR, CONTROL      ;
                   34 ;                 : CHARACTERS, AND CHARACTERS FROM THE CONSOLE.         ;
                   35 ; OUTPUTS         : BYTES OF DATA TO THE DEMODULATOR.                    ;
                   36 ; CALLS           : RATE, SETUP, UOTEST.                                 ;
                   37 ; DESTROYS        : NONE.                                                ;
                   38 ; DESCRIPTION     : The outputs from the modulators of the two telephones ;
                   39 ;                 : must be relayed on to the respective demodulators.   ;
                   40 ;                                                                        ;
                   41 ;*************************************************************************;
                   42                              ;
                   43                              ;
1800  F3           44 BEGIN:  DI                   ; Disable all interrupts.
1801  31001F       45         LXI     SP,STAK      ; Initialise an independent stack.
1804  CD2A18       46         CALL    SETUP        ; Initialise the interface card of the SABUS kit.
1807  00           47 LOOP:   NOP                  ;
1808  C30718       48         JMP     LOOP         ;
                   49                              ;
                   50 ;.........................................................................;
                   51                              ;
                   52                              ; The next section of program is called by an
                   53                              ; RST 7.5 interrupt.  The strobing of the PPI
                   54                              ; input ports and the interrupts are simultaneous.
                   55                              ; When the Strobe goes high it causes data from the
                   56                              ; input (serial-parallel) register to be latched, and
                   57                              ; at the same time the RST 7.5 latch is set.
                   58                              ;
180B  F5           59 STROBE: PUSH    PSW          ; Save the relevant registers.
180C  E5           60         PUSH    H            ;
180D  FB           61         EI                   ; Enable the interrupts again.
                   62                              ;
180E  DBFC         63         IN      0FCH         ; Input the most recent byte from Tel. 2.
```

```
 LOC   OBJ        LINE        SOURCE STATEMENT

 1810  32001C       64          STA     BUFFER  ; Store for later output.
 1813  DBF8         65          IN      0F8H    ; Input the most recent 8 bits from the DM.
                    66                          ;
 1815  00           67          NOP             ; Because the RST 7.5 interrupt and
 1816  00           68          NOP             ; the output (parallel-serial) register
 1817  00           69          NOP             ; "load" are simultaneous it is necessary
 1818  00           70          NOP             ; to ensure that the load is complete before
 1819  00           71          NOP             ; outputting new information to this register.
                    72                          ;
 181A  D3FD         73          OUT     0FDH    ; Output to Tel. 2.
 181C  3A001C       74          LDA     BUFFER  ; Recover the input from Tel. 2.
 181F  D3F9         75          OUT     0F9H    ; Output to Tel. 1.
                    76                          ;
 1821  CD3F00       77          CALL    UOTEST  ; Test for interrupt from the console.
 1824  C43218       78          CNZ     RATE    ;
                    79                          ;
 1827  E1           80          POP     H       ; Retore the registers.
 1828  F1           81          POP     PSW     ;
 1829  C9           82          RET             ;
                    83 $EJECT
```

```
 LOC   OBJ        LINE       SOURCE STATEMENT

                    84 ;*****************************************************************************;
                    85 ;                                                                             ;
                    86 ; PROCEDURE     : SETUP.                                                      ;
                    87 ; INPUTS        : NONE.                                                       ;
                    88 ; OUTPUTS       : CONTROL WORDS TO INTERFACE CARDS.                           ;
                    89 ; CALLS         : CLOCKS.                                                     ;
                    90 ; DESTROYS      : A, F/F's.                                                   ;
                    91 ; DESCRIPTION   : Control words are sent to the interface card in order       ;
                    92 ;               : to set it in the desired mode.  The 8255 Parallel I/O       ;
                    93 ;               : device is set to have two input channels, two output        ;
                    94 ;               : channels, and two channels for handshaking.                 ;
                    95 ;                                                                             ;
                    96 ;*****************************************************************************;
                    97                                  ;
182A  3EB4          98 SETUP:  MVI     A,0B4H  ; Configure both PPI Interfaces (8255's) for Mode 1
182C  D3FB          99         OUT     0FBH    ; i.e. Port A Input, Port B Output, Port C Handshake.
182E  3EBC         100         MVI     A,0BCH  ; Port C0 PC6/7 = output, Port C1 PC6/7 = input.
1830  D3FF         101         OUT     0FFH    ;
                   102                         ;
1832  F3           103 RATE:  DI              ; Disable interrupts while changes are made.
1833  3E0D         104         MVI     A,0DH   ; Disable the CODEC clocks.
1835  D3FB         105         OUT     0FBH    ;
1837  3E0E         106         MVI     A,0EH   ; Reset the CODEC F/F's.
1839  D3FB         107         OUT     0FBH    ;
                   108                         ;
183B  3ECC         109         MVI     A,0CCH  ; Eqivalent to zero signal.
183D  D3F9         110         OUT     0F9H    ; Initialise the DM decoder.
                   111                         ;
183F  CD5818       112         CALL    CLOCKS  ; Set the programmable interval timer.
                   113                         ;
1842  3E0F         114         MVI     A,0FH   ; Set the F/F Clear lines.
1844  D3FB         115         OUT     0FBH    ;
1846  3E0C         116         MVI     A,0CH   ; Enable the CODEC clocks.
1848  D3FB         117         OUT     0FBH    ;
```

```
 LOC   OBJ        LINE        SOURCE STATEMENT

 184A  DBFA        119 STRB:   IN      ØFAH    ; Because of indeterminate starting conditions
 184C  E62Ø        12Ø         ANI     2ØH     ; the interface registers are loaded
 184E  CA4A18      121         JZ      STRB    ; and read before enabling the Codec.
 1851  DBF8        122         IN      ØF8H    ; Reset the IBF line.
                   123                         ;
 1853  3E3B        124         MVI     A,MASK  ; Interrupt mask enabling the RST 7.5 interrupt.
 1855  3Ø          125         SIM             ; Set the mask.
 1856  FB          126         EI              ; Future input/output is interrupt controlled.
 1857  C9          127         RET             ;
                   128 $EJECT
```

```
 LOC   OBJ        LINE        SOURCE STATEMENT

                   129 ;*************************************************************************;
                   130 ;                                                                         ;
                   131 ; PROCEDURE     : CLOCKS.                                                 ;
                   132 ; INPUTS        : CHARACTERS FROM THE CONSOLE.                            ;
                   133 ; OUTPUTS       : USER PROMPTS TO CONSOLE SCREEN PLUS CONTROL             ;
                   134 ;               : INFORMATION TO THE PROGRAMMABLE INTERVAL TIMER.          ;
                   135 ; CALLS         : GETECH, OUTMSG.                                         ;
                   136 ; DESTROYS      : A,F/F's.                                                ;
                   137 ; DESCRIPTION   : The 8253 Programmable Timer can be set to operate       ;
                   138 ;               : at the desired bit rate with a second clock being       ;
                   139 ;               : used for strobing. This second timer runs at one        ;
                   140 ;               : eigth the bit rate.                                     ;
                   141 ;                                                                         ;
                   142 ;*************************************************************************;
                   143                             ;
 1858  216819      144 CLOCKS: LXI     H,SCREEN; String to clear the screen.
 185B  CD7200      145         CALL    OUTMSG  ;
 185E  21C018      146         LXI     H,MSG   ; Start of ASCII string.
 1861  CD7200      147         CALL    OUTMSG  ; Send the string to the console screen.
                   148                         ;
 1864  CD6000      149 RESP:   CALL    GETECH  ; Get the user's response.
 1867  21B618      150         LXI     H,TOPA  ;
 186A  FE41        151         CPI     41H     ; Check for ASCII 'A'.
 186C  CA9818      152         JZ      INIT    ; User chose first option.
 186F  21B818      153         LXI     H,TOPB  ;
 1872  FE42        154         CPI     42H     ; Check for ASCII 'B'.
 1874  CA9818      155         JZ      INIT    ; Second option chosen.
 1877  21BA18      156         LXI     H,TOPC  ;
 187A  FE43        157         CPI     43H     ; Check for ASCII 'C'.
 187C  CA9818      158         JZ      INIT    ; Third option chosen.
 187F  21BC18      159         LXI     H,TOPD  ;
 1882  FE44        160         CPI     44H     ; Check for ASCII 'D'.
 1884  CA9818      161         JZ      INIT    ; Fourth option chosen.
 1887  21BE18      162         LXI     H,TOPE  ;
 188A  FE45        163         CPI     45H     ; Check for ASCII 'E'.
```

```
 LOC   OBJ        LINE        SOURCE STATEMENT

 188C  CA9818      164          JZ      INIT    ; Last option chosen.
 188F  213819      165          LXI     H,ERROR ; Invalid option has been selected.
 1892  CD7200      166          CALL    OUTMSG  ;
 1895  C36418      167          JMP     RESP    ; Keep looping until valid option chosen.
                   168                          ;
                   169                          ;
 1898  3E76        170 INIT:    MVI     A,76H   ; Mode Word for Counter #1.
 189A  D3F7        171          OUT     0F7H    ; Mode 3, 16-bit binary counter.
 189C  7E          172          MOV     A,M     ; Set the output clock rate for Counter #1.
 189D  D3F5        173          OUT     0F5H    ;
 189F  23          174          INX     H       ; Get the MSB.
 18A0  7E          175          MOV     A,M     ;
 18A1  D3F5        176          OUT     0F5H    ; MSB sent last.
 18A3  3EB6        177          MVI     A,0B6H  ; Mode Word for Counter #0.
 18A5  D3F7        178          OUT     0F7H    ; Same mode as Counter #1.
 18A7  3E08        179          MVI     A,08H   ; Set the output clock rate for Counter #0.
 18A9  D3F6        180          OUT     0F6H    ; It is 1/8th the rate for Counter #1.
 18AB  3E00        181          MVI     A,00H   ;
 18AD  D3F6        182          OUT     0F6H    ; Send the MSB last.
 18AF  216D19      183          LXI     H,MESG  ; Send a message to the console screen.
 18B2  CD7200      184          CALL    OUTMSG  ;
 18B5  C9          185          RET             ;
                   186 $EJECT
```

```
 LOC   OBJ        LINE       SOURCE STATEMENT

                   187 ;*************************************************************************;
                   188 ;                                                                         ;
                   189 ;                          THE DATA AREA.                                 ;
                   190 ;                                                                         ;
                   191 ;*************************************************************************;
                   192                          ;
 18B6  A0          193 TOPA:   DB      0A0H     ; Corresponds to  9600 baud, DM sample clock.
 18B7  00          194         DB      00H      ;
 18B8  60          195 TOPB:   DB      60H      ; Corresponds to 16000 baud.
 18B9  00          196         DB      00H      ;
 18BA  30          197 TOPC:   DB      30H      ; Corresponds to 32000 baud.
 18BB  00          198         DB      00H      ;
 18BC  1C          199 TOPD:   DB      1CH      ; Corresponds to 56000 baud.
 18BD  00          200         DB      00H      ;
 18BE  18          201 TOPE:   DB      18H      ; Corresponds to 64000 baud.
 18BF  00          202         DB      00H      ;
                   203                          ;
 18C0  54484520    204 MSG:    DB      'THE POSIIBLE CODEC BAUD RATES ARE :',CR,LF
       504F5349
       49424C45
       20434F44
       45432042
       41554420
       52415445
       53204152
       45203A0D
       0A
 18E5  28612920    205         DB      '(a)  9600',CR,LF
       20393630
       300D0A
 18F0  28622920    206         DB      '(b) 16000',CR,LF
       31363030
       300D0A
 18FB  28632920    207         DB      '(c) 32000',CR,LF
       33323030
```

```
 LOC   OBJ        LINE        SOURCE STATEMENT

       300D0A
 1906  28642920    208         DB      '(d) 56000',CR,LF
       35363030
       300D0A
 1911  28652920    209         DB      '(e) 64000',CR,LF
       36343030
       300D0A
 191C  454E5445    210         DB      'ENTER THE REQUIRED OPTION >'
       52205448
       45205245
       51554952
       4544204F
       5054494F
       4E203E
 1937  00          211         DB      00H     ;
                   212                         ;
 1938  0D494E56    213 ERROR:  DB      CR,'INVALID OPTION. ENTER "a","b","c","d" or "e" >'
       414C4944
       204F5054
       494F4E2E
       20454E54
       45522022
       61222C22
       62222C22
       63222C22
       6422206F
       72202265
       22203E
 1967  00          214         DB      00H     ;
                   215                         ;
 1968  1B          216 SCREEN: DB      1BH     ; ASCII <ESC>.
 1969  48          217         DB      48H     ; Cursor to top of screen.
 196A  1B          218         DB      1BH     ;
 196B  4A          219         DB      4AH     ; Clear to end of screen.
 196C  00          220         DB      00H     ;
```

```
 LOC   OBJ        LINE        SOURCE STATEMENT

                   221                        ;
 196D  0D0A544F    222 MESG:   DB      CR,LF,'TO UPDATE THE SAMPLING RATE TYPE ANY CHARACTER >'
       20555044
       41544520
       54484520
       53414D50
       4C494E47
       20524154
       45205459
       50452041
       4E592043
       48415241
       43544552
       203E
 199F  00          223         DB      00H     ;
                   224                         ;
 1F00              225         ORG     1F00H
 1F00              226 STAK:   DS      1       ;
                   227                         ;
 1FFD              228         ORG     1FFDH
 1FFD  C30B18      229         JMP     STROBE  ; 1FFD is the trap address for RST 7.5.
                   230         END

0 error(s) detected


User symbols

BEGIN  1800 A   BUFFER 1C00 A   CLOCKS 1858 A   CR     000D A   ERROR  1938 A   GETECH 0060 A   INIT   1898 A
LF     000A A   LOOP   1807 A   MASK   003B A   MESG   196D A   MSG    18C0 A   OUTMSG 0072 A   RATE   1832 A
RESP   1864 A   SCREEN 1968 A   SETUP  182A A   STAK   1F00 A   STRB   184A A   STROBE 180B A   TOP    1C01 A
TOPA   18B6 A   TOPB   18B8 A   TOPC   18BA A   TOPD   18BC A   TOPE   18BE A   UOTEST 003F A
```

APPENDIX D

The simulation package described in chapter 5 is contained on a floppy disc (5 1/4 inch) labelled "CODSIM : Codec Simulation Package". It is possible for the system to operate using only the two built-in disc drives of the HP9836C desk-top computer, but if a hard disc (such as the HP9134XV) is available then it is advisable to copy the system on to this disc as the access times are considerably shorter and generally the available storage space is greater.

D.1 Using a hard disc.

If a hard disc is used then it will be necessary for the user to alter the software accordingly. The changes necessary are very simple to make.

In each of the programs in the package there are statements specifying which disc drives are used. These statements occur in the initialisation section of the software (in the initialisation routine - in the subroutine section) and are :

```
Drive_0$ = ":INTERNAL,4,0"
Drive_1$ = ":INTERNAL,4,1"
```

If the two built-in drives are to be used then these need not be changed. However, if a hard disc is to be used then these statements should be changed to :

```
Drive_0$ = ":HP9134,700"
Drive_1$ = ":HP9134,700",
```

or a similar appropriate designation depending on which disc

drive is used (refer to the operating/user manual for the drive concerned). The "700" in the designation indicates that interface "7" is to be used and the drive is at address "00".

D.2 Getting started.

The computer is switched off and you wish to commence working. In order to boot the appropriate system the following procedure should be followed.

(i) Insert the system disc, labelled "CODSIM : System Disc", into the right hand disc drive and close the drive's door.

(ii) Switch on the computer, the Multiprogrammer (HP6942A) and any other relevant peripheral devices.

(iii) When the computer has completed booting remove the disc from the drive and replace it with the disc labelled "CODSIM : System Extensions".

(iv) Now type :

LOAD BIN"AP2_1" EXECUTE (the underlined word indicates the actual key to be pressed).

When this has completed type :

LOAD BIN"GRAPH2_1" EXECUTE

If the HP9134XV hard disc (or any other disc with the 'XV' appendage) is to be used then type :

LOAD BIN"XV2_1" EXECUTE

(v) If necessary the changes mentioned in D.1 should be made at

this stage. The programs in the simulation library that need to be adjusted are :

RECORD

PLAYBACK

CODECS

MEASURE

POLATE

and, as mentioned before, are kept on the "CODSIM : Codec Simulation Package" disc. This disc should be inserted in the right hand disc drive. If a hard disc is to be used the programs should be loaded into memory, edited and then stored on the hard disc.

eg. LOAD "MASTER" EXECUTE

edit the program as described insection D.1,

STORE "MASTER:HP9134,700" EXECUTE

Once this has been done the default mass storage device should be changed :

MSI ":HP9134,700" EXECUTE

(Only to be done if the built-in disc drives are not to be used.)

(vi) The system should now be in a usable form. To commence using the simulation package type :

LOAD "MASTER",1 EXECUTE

This will start the system by displaying a menu.

D.3 Using the simulation package.

The system has been designed to interact with the user via a set of menus. The main menu contains a list of the possible sub-systems that can be accessed. The menu is shown below :

Select a function from the following :

1) Aquire speech data at 8 kHz

2) Interpolate 8 kHz data

3) Process interpolated data

4) Determine objective quality measures

5) Replay speech data

6) Terminate

Enter the required option >

The user selects the desired option by typing the corresponding number. When the selected function has completed the main menu will again be displayed and the user can enter a new option.

D.3.1 Aquire speech data at 8 kHz.

Before options 2-5 can be successfully chosen it is necessary to record some speech and store it. This is done by selecting option 1. The program will chain to the input interface program. When the system has completed the interface configuration the memory configuration is carried out. The length of the speech segment that will be recorded is 10 seconds (above this value lack of memory space becomes a problem. Once all configurations have been

completed the system will be ready to commence recording speech segments and the message

Press CONTINUE when ready to record

will be displayed. When the CONTINUE key is pressed the system commences the sampling, quantizing and digitizing process, stopping when the 10 second segment has been recorded.

The RMS level of the signal will be calculated and displayed, along with the peak signal values. The user is then asked whether this segment should be saved or discarded :

Do you wish to save this segment (Y or N) ?

If the 'Y' option is selected the computer then asks for the name of the file in which the data must be stored.

Enter the desired file name >

The file name can be entered using either upper or lower case alphanumeric characters (lower case characters are converted to upper case by the program). A maximum of eight characters can be used for the file name. If the file name already exists the user is asked if that file should be overwritten and, if not, a new name will be prompted for.

Once the data has been stored or discarded the user is asked if any more data is to be recorded. If the response to this is in the affirmative then the process begins again. If not, the user

is returned to the main menu.

D.3.2 Interpolate 8 kHz data.

If option two is selected the computer needs to know three things

- the name of the source file,
  the required effective sampling rate,
  the name of the destination file.

All files containing 8 kHz data have the suffix "_0" appended to the name chosen by the user (to indicate that the file contains 'original' data - ie. unprocessed). This suffix need not be specified by the user as the computer automatically affixes the correct suffix. In response to the prompt

Enter the name of the source file >

a name, containing at most eight characters, must be entered.

Next the new effective sampling rate is entered.

Enter the sampling rate (Possible values : 16, 24, 32, 64) >

These values are in kHz. Any values other than those shown will be rejected.

The third piece of information is the name of the file in which the interpolated data is to be stored. As in previous cases the user responds to the prompt

Enter the destination file name >

by entering a name (maximum of eight characters). If no name is specified the computer uses the name of the source file, the only difference being the suffix appended. All processed files are given the "_P" appendage to indicate 'processed' data.

eg. If no destination file name is specified and the source file name was "TEST_O", then the destination file will be given the name "TEST_P"

Once the interpolation is complete the user is returned to the main menu.

D.3.3 Process interpolated data.

In describing this section it is assumed that the user is familiar with the designs of the various codecs. There are seven codecs that have been implemented in this system (once the user is thoroughly familiar with the simulation system's design details it will be possible to implement other codec designs).

The first thing the user is asked is which codec should be used to process the speech data. It has been made possible to select a range of codecs - ie. one segment of speech could be processed by all or just some of the seven codecs.

Codec Type

1) LDM

2) CVSD

3) CFDM

4) SVADM

5) SHCDM

6) MSHCDM

7) HCDM

Start	Stop	Increment
1	1	1

A codec type '0' returns to main menu.

Enter the desired value >

The value that is to be entered by the user is shown on the screen in inverse video. If no value is enterd then the default value displayed on the screen will be assumed. Once a value is entered the relevant value in the table is updated and the inverse video mode is turned off with the next value now being displayed in inverse video. Once all the values have been entered the user is asked :

Do you wish to keep these values (Y or N) ?

The default answer is 'Y'. If the 'N' option is selected then the 'Start' value is shown in inverse video and the values can be

altered as before.

eg. If values for 'Start', 'Stop' and 'Increment' of 2, 6 and 2 respectively are chosen then the speech will be processed by the CVSD, SVADM and MSHCDM codecs.

There are various design parameters affecting the codecs' performance. These include the leakage time constants (for the leaky integrators), maximum and minimum step sizes, maximum signal range, etc. Depending on the codecs selected, these parameters are presented to the user in a similar fashion to that described above. If a design parameter is not associated with the selected codec it will not be presented to the user for modification. Initially the default parameter values are displayed and it is up to the user to change them.

As in previous cases the user must specify the source file that is to be processed. It should be noted that the codecs implemented here have been designed to allow the user to decide whether 16, 24, 32 or 64 kHz data is to be processed (the choice is reflected in the parameter default values).

Lastly, the user enters the name of the files in which the processed data is to be stored.

Enter the destination file name >

If a range of codecs, or parameter values, had been selected then the file name is given a numerical appendage to indicate the seperate files.

eg. If the name 'TEST' had been selected then the processed data will be stored in 'TEST1_P', 'TEST2_P', 'TEST3_P', etc.

D.3.5 Determine objective quality measures.

This section is virtually identical to the previous one except that the various signal-to-noise ratios are calculated and the processed data is not stored simply because the routines to calculate the noise components of the signal destroy the original signal.

Before asking for the source file name the user is aked if the granular and overload SNR's must be calculated.

Do you require SNG (Y or N) ?

and

Do you require SNO (Y or N) ?

The user is also asked

Do you require the segmental values (Y or N) ?

The relevant results will subsequently be sent to the printer.

D.3.6 Replay speech data.

The first thing the user must tell the program is the sampling rate to be used. Because of the interface hardware the maximum sampling rate is limited to 16 kHz. The user will be asked :

Possible output data rates are 8 kHz and 16 kHz.

Enter the required data rate (kHz) >

Next the user will be asked for the source file name :

Enter the name of the file from which the data is to be read >

As in previous cases only the name need be specified and not the "_P" suffix (the computer will do this). The data is divided into 14 sections and the user may specify the start and finish sections. The default values are 1 and 14 respectively. If a value for the finish section is specified to be less than the start then the default value is assumed.

Start	Finish	Continuous
1	14	YES

Enter the desired value or toggle value (ie. Y/N).

As before the value that must be entered is highlighted on the screen by being shown in inverse video. The 'Continuous' parameter specifies whether or not the selected sections are output repeatedly or just once.

Once the relevant speech segment commences outputting it can be

interrupted by pressing any key on the keyboard.

To terminate the output hit any key on the keyboard. If this is done the following prompt will be displayed.

Do you wish to output new sections of present segment (Y/N) ?

If 'Y' is selected then the values shown in the table above can be updated and the process continues as described. Otherwise, the user is asked :

Do you wish to output a new speech segment (Y/N) ?

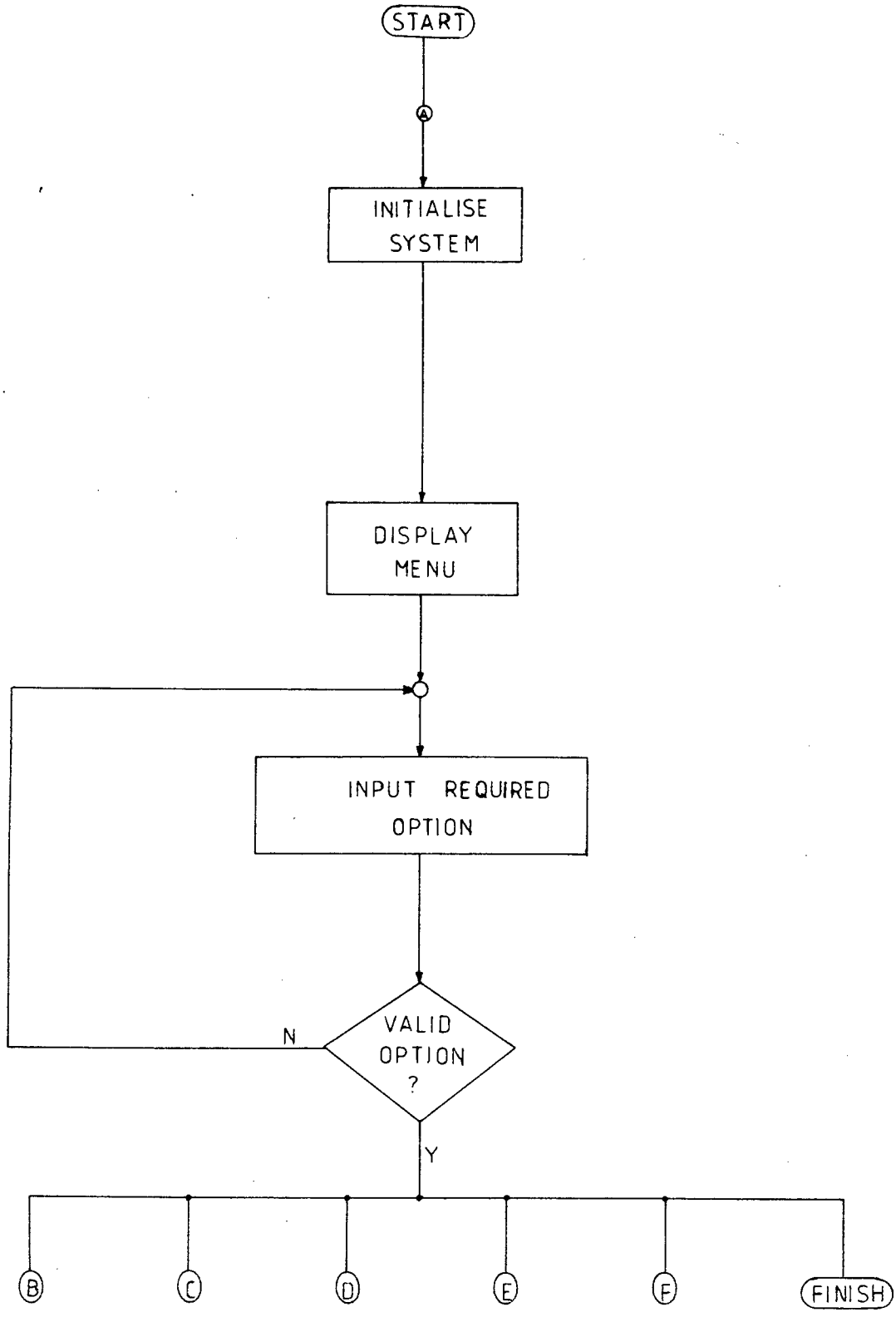
In this case, if 'Y' is selected the entire process described in this section repeats itself. If 'N' is selected then the user is returned to the main menu.

D.3.7 Terminate.

This option gets the user out of the simulation system and back to the normal operating system.

APPENDIX E

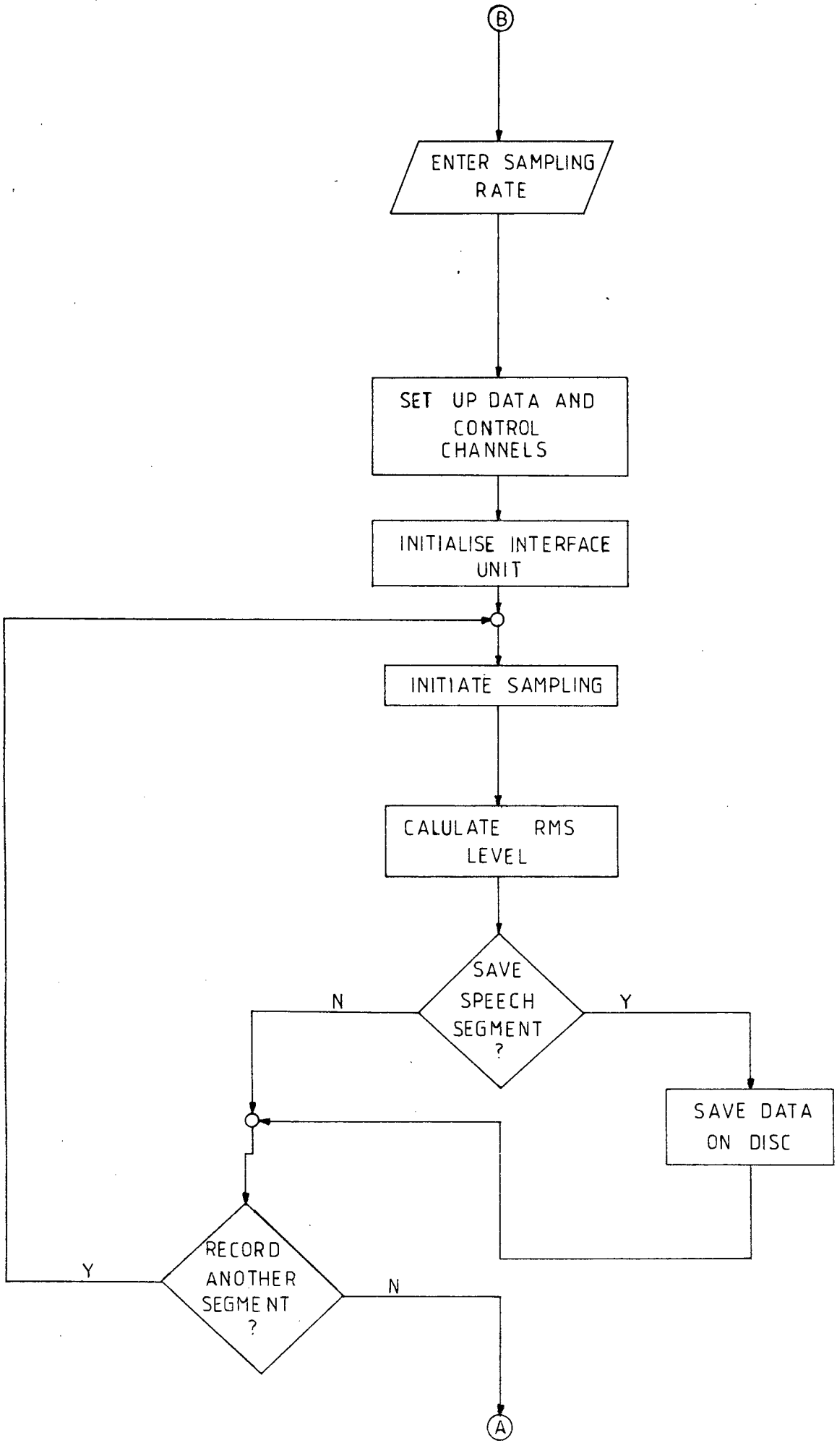
This appendix contains the logic flowcharts and the software listings of the simulation system described in Chapter 5 and Appendix D.


START
A
INITIALISE SYSTEM
DISPLAY MENU
INPUT REQUIRED OPTION
VALID OPTION ?
N
Y
B
C
D
E
F
FINISH

```
!-----------------------------------------------------------------------
!        MASTER LIBRARY FOR SPEECH DATA AQUISITION AND PROCESSING
!        ========================================================
!
! CODSIM          : Codec Simulation Package.
! M.Sc Thesis     : The use of Delta Modulation for Low Bit-Rate Digital
!                   Speech Coding.
! Author          : J.M. Irvine.
!                   Department of Electrical and Electronic Engineering
!                   University of Cape Town.
! Unit Number     : 0.
! Latest Update   : 1st December, 1984.
!-----------------------------------------------------------------------
! Programs developed for the Hewlett Packard 2000 Series desk-top
! computers.  All development work done on an HP9836C computer with the
! interfacing between the analogue and digital environments done using
! the HP6942A Multiprogrammer.
!-----------------------------------------------------------------------
! This program provides the key to the data aquistion and processing
! system.  Access is provided through a main menu listing the possible
! options available.  Program control is then transferred to the soft-
! ware unit corresponding to the selected option.
!-----------------------------------------------------------------------
!<<<<<<<<<<<<<<<<< INITIALISE THE STATE OF THE CONTROLLER >>>>>>>>>>>>>>>>
!                  ======================================
!
etup:   !
PRINTER IS CRT                  ! The default printer device is the monitor.
CONTROL 1,12;1                  ! Turn the key label display off.
CONTROL 1,5;140                 ! Set the default alpha colour to Cyan.
PRINT USING "@"                 ! Clear the alpha display.
GRAPHICS OFF                    ! Turn the graphics display off.
!
DIM Function$[40]               ! Allocate space for the selected command.
!
Stop$=" !"                      ! Control character to stop program execution.
Run$=" R"                       ! Control character to start program execution.
!
```

```
!<<<<<<<<<<<<<<<<<<<<<<<< DISPLAY THE MASTER MENU >>>>>>>>>>>>>>>>>>>>>>>>
!                         ======================
!
PRINT TABXY(10,5),CHR$(132);
PRINT "Select a function from the following :";CHR$(128)
PRINT
PRINT TAB(10),"(1) Acquire speech data at 8 kHz"
PRINT
PRINT TAB(10),"(2) Interpolate 8 kHz data"
PRINT
PRINT TAB(10),"(3) Process interpolated data"
PRINT
PRINT TAB(10),"(4) Determine objective quality measures"
PRINT
PRINT TAB(10),"(5) Replay 8 kHz data"
PRINT
PRINT TAB(10),"(6) Terminate"
!
DISP "Enter the required option >"
!
ON ERROR GOTO Check_error
!
Wait_answer:   !
ON KBD GOTO Get_answer                    ! Poll the controller keyboard
GOTO Wait_answer                          ! for a response from the user.
!
Get_answer:   !
!
Answer=VAL(KBD$)                          ! Input the user's response from
OFF KBD                                   ! the keyboard buffer.
!
SELECT Answer
CASE 1
  DISP "Function 1 selected"
  Function$="LOAD ""RECORD:INTERNAL,4,0"" X"
  OUTPUT KBD;Stop$;Function$;Run$;
  !
CASE 2
  DISP "Function 2 selected"
  Function$="LOAD ""POLATE:INTERNAL,4,0"" X"
  OUTPUT KBD;Stop$;Function$;Run$;
  !
CASE 3
  DISP "Function 3 selected"
  Function$="LOAD ""CODECS:INTERNAL,4,0"" X"
  OUTPUT KBD;Stop$;Function$;Run$;
  !
CASE 4
  DISP "Function 4 selected"
  Function$="LOAD ""MEASURE:INTERNAL,4,0"" X"
  OUTPUT KBD;Stop$;Function$;Run$;
  !
CASE 5
  DISP "Function 5 selected"
  Function$="LOAD ""PLAYBACK:INTERNAL,4,0"" X"
  OUTPUT KBD;Stop$;Function$;Run$;
  !
CASE 6
  GOTO Finish
```

```
  !
CASE ELSE
  BEEP 1000,.1
  DISP CHR$(11);CHR$(129);"INVALID OPTION !!!";CHR$(128)
  WAIT 1.5
  DISP CHR$(11);"Please re-enter the required option >"
  GOTO Wait_answer
END SELECT
!-------------------------------------------------------------------------
!
inish: !
PRINTER IS CRT
CONTROL 1,5;139                         ! Set colour to green.
CONTROL 1,12;0                          ! Turn softkey labels back on.
MASS STORAGE IS ":INTERNAL,4,0"         ! Reset default drive to right drive.
STOP
!
!-------------------------------------------------------------------------
!
heck_error: !
IF ERRN=32 THEN                         ! If an error has occured then check
  GOTO 1030                             ! for an invalid choice by the user -
END IF                                  ! ie. a non-numeric choice (ERRN=32).
!
!-------------------------------------------------------------------------
!
END
!
```


B
ENTER SAMPLING RATE
SET UP DATA AND CONTROL CHANNELS
INITIALISE INTERFACE UNIT
INITIATE SAMPLING
CALULATE RMS LEVEL
SAVE SPEECH SEGMENT ?
N
Y
SAVE DATA ON DISC
RECORD ANOTHER SEGMENT ?
Y
N
A

```
!-------------------------------------------------------------------------
!            BUFFERED A/D PROGRAM FOR SPEECH DATA AQUISITION
!            ==============================================
!
! CODSIM          : Codec Simulation Package.
! M.Sc Thesis     : The use of Delta Modulation for Low Bit-Rate Digital
!                   Speech Coding.
! Author          : J.M. Irvine.
!                   Department of Electrical and Electronic Engineering
!                   University of Cape Town.
! Unit Number     : 1.
! Latest Update   : 1st December, 1984.
!-------------------------------------------------------------------------
! Programs developed for the Hewlett Packard 2000 Series desk-top
! computers.  All development work done on an HP9836C computer with the
! interfacing between the analogue and digital environments done using
! the HP6942A Multiprogrammer.
!-------------------------------------------------------------------------
! This program controls the buffered A/D process.  The multiprogrammer
! is initialised to perform continuous data aquisition.  The input
! analogue signal is sampled and converted to digital form at a rate of
! 8000 samples per second.  This data is transferred to the controller
! via memory cards on the multiprogrammer and when sampling is complete
! the data is transferred to one of the mass storage devices (the
! default device is the HP9134 hard disc).  Once the storage is
! complete the user may then listen to the stored data.  The length of
! the sampled speech segment is 3 seconds which is sufficiently long
! for phonetically balanced sentences such as the Harvard sentences.
!-------------------------------------------------------------------------
!
!<<<<<<<<<<<<<<< DEFINE THE VARIABLES USED BY THE PROGRAM >>>>>>>>>>>>>>>
!                ----------------------------------------
!
OPTION BASE 1
INTEGER Card,Mistake,A_d_slot,D_a_slot                  ! Multiprogrammer card
INTEGER Mem_in_slot,Tb_slot,Slot_no                     ! variables.
INTEGER Complete,No_of_errors,No_of_cards
INTEGER Card_variable,Intr_card,Next_in_buf
INTEGER Loop,Loop1,Differential,Underflow
!
REAL Intr_svc_delay,Transfer_oh,Total_delay             ! System performance
REAL Transfer_rate,Reinit_time,Head_room                ! parameters.
REAL Block_size,Reference_word,Time_available
REAL Transfer_time,Additional_rdgs,Trigger_rate
!
DIM Function$[40],Error$[40],Mr_command$[30],File_name$[10]
ALLOCATE INTEGER Configuration(0:16,5)
!
Stop$=" !"                                  ! Control characters to start &
Run$=" R"                                   ! stop program execution.
!*************************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<<<<<<<<<< M A I N L I N E >>>>>>>>>>>>>>>>>>>>>>>>>>>
!                             ===============
!
GOSUB Initialise                ! Reset the mode of the controller.
GOSUB Sample_rate               ! Set up the variables for 8 kHz sampling.
GOSUB Io_paths                  ! Set up data and control paths.
Xfer(Op$)                       ! Load BINARY subprogram.
!
! Determine the multiprogrammer card configuration.
!
Multi_set_up(Configuration(*),Tb_slot,Mem_in_slot,A_d_slot)
!
GOSUB Re_init                   ! Set the multiprogrammer cards in the
                                ! correct operating state.
!
!<<<<<<<<<<<< NOW THE SPEECH SEGMENTS ARE SAMPLED AND STORED >>>>>>>>>>>>
!             ============================================
!
Input_speech:  !
GOSUB Clear_disp                ! Re-initialise the VDU displays.
GOSUB Speech_adc                ! Sample and store the speech using the
                                ! ADC (Analogue-to-Digital Conversion)
                                ! module.
GOTO Input_speech
!*****************************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<<<<<< S U B R O U T I N E S >>>>>>>>>>>>>>>>>>>>>>>>
!                        ====================
!
!<<<<<<<<<<<<<<<< SPECIFY THE DEFAULT MASS STORAGE DEVICES >>>>>>>>>>>>>>>>
!                ----------------------------------------
!
Initialise: !
  Drive_0$=":INTERNAL,4,0"          ! Set the default mass storage devices.
  Drive_1$=":INTERNAL,4,1"
  PRINTER IS CRT                    ! Use the VDU for displays.
  CONTROL 1,12;1                    ! Turn the key label display off.
  No_of_buffers=8                   ! The number of arrays used to store the
                                    ! sampled speech data.
!
!<<<<<<<<<<<<<< CLEAR THE ALPHANUMERIC AND GRAPHICS DISPLAYS >>>>>>>>>>>>>>
!              ---------------------------------------------
!
Clear_disp: !
  PRINTER IS CRT                              ! Use the video monitor.
  GCLEAR                                      ! Clear the graphics display.
  ALPHA OFF
  PRINT USING "@"                             ! Clear the Alpha display.
  PRINT CHR$(128);CHR$(140);                  ! Select normal video with
  ALPHA ON                                    ! Colour = Cyan.
  DISP " "
  RETURN
!*****************************************************************************
!
!<<<<<<<<<<<<<<< CALCULATE MEMORY AND TIMING REQUIREMENTS >>>>>>>>>>>>>>>
!               ----------------------------------------
!
Sample_rate:!
  Op$="MR"
  Trigger_rate=1/8000                         ! The sampling period.
  Intr_svc_delay=.0229                        ! Interrupt servicing time.
  Transfer_oh=.0013                           ! All values in SECONDS.
  Total_delay=Intr_svc_delay+Transfer_oh
  Transfer_rate=7.5355E-5                     ! Data transfer speed.
  Reinit_time=.00091
!
!<<<<<<<< CALCULATE REFERENCE WORD,TIME AVAILABLE AND BLOCK SIZE >>>>>>>>
!         ------------------------------------------------------
!
  Head_room=Total_delay/Trigger_rate
  Reference_word=INT(4085-Head_room)
  Time_available=(Reference_word*Trigger_rate-Reinit_time)
  Block_size=4085                             ! Minimum memory required.
  Transfer_time=Block_size*Transfer_rate
Calculate:!
  Additional_rdgs=Transfer_time/Trigger_rate
  IF Additional_rdgs>=1 THEN
    Block_size=Block_size+Additional_rdgs
    Transfer_time=Additional_rdgs*Transfer_rate
    GOTO Calculate
  END IF
  Block_size=INT(Block_size)
!
```

```
0 !<<<<<<<<<<<<<<<< PRINT OPERATIONAL PARAMETERS ON SCREEN >>>>>>>>>>>>>>>>
0 !                 ------------------------------------
0 !
0  GOSUB Clear_disp
0  PRINT TABXY(5,3);"CONTINUOUS MODE OPERATIONAL PARAMETERS"
0  PRINT TABXY(5,4);"------------------------------------"
0  PRINT TABXY(8,5);"TRIGGER RATE (microsecs):";Trigger_rate*1.E+6
0  PRINT TABXY(8,6);"BLOCK SIZE       (words):";Block_size
0  PRINT TABXY(8,7);"REFERENCE WORD          :";Reference_word
0  IF Time_available>1 THEN
0    PRINT TABXY(8,8);"TIME AVAILABLE    (secs):";
0    PRINT USING "D.DDDD";Time_available
0  ELSE
0    PRINT TABXY(8,8);"TIME AVAILABLE   (msecs):";
0    PRINT USING "DDD.DD";Time_available
0  END IF
0 !
0 !<<<<<<<<<<<<<<<<<<<<<< MEMORY SPACE IS NOW RESERVED >>>>>>>>>>>>>>>>>>>>
0 !                       ----------------------------
0 !
0 ! A total of 8 buffers, each representing approximately 1.3 seconds worth
0 ! of speech, are used to store the sampled speech signal.
0 !
0  ALLOCATE INTEGER Buf1(Block_size),Buf2(Block_size)
0  ALLOCATE INTEGER Buf3(Block_size),Buf4(Block_size)
0  ALLOCATE INTEGER Buf5(Block_size),Buf6(Block_size)
0  ALLOCATE INTEGER Buf7(Block_size),Buf8(Block_size)
0  !
0  ! Another 4 buffers are used for calculating the RMS level of the input
0  ! speech signal.
0  !
0  ALLOCATE REAL Buffer1(Block_size),Buffer2(Block_size)
0  ALLOCATE REAL Buffer3(Block_size),Buffer4(Block_size)
0  !
0  RETURN
0 !*****************************************************************************
0 !
0 !<<<<<<<<<<<<<<<<<<<< SET UP THE DATA TRANSFER PATHS >>>>>>>>>>>>>>>>>>>>
0 !                     ------------------------------
0 !
0 Io_paths:!
0  ASSIGN @Multi TO 823                              ! Multiprogrammer address.
0  ASSIGN @Mr_readback TO 82305;FORMAT OFF           ! Buffer to Memory path.
0  ASSIGN @Intr_list TO 82312                        ! Interrupt List.
0  ASSIGN @Intr_cause TO 82310                       ! Interrupt Cause.
0  RETURN
0 !*****************************************************************************
```

```
0 !<<<<<<<<<<<<<<<<<<<<<<< INITIALIZE THE SYSTEM CARDS >>>>>>>>>>>>>>>>>>>>>>>
0 !                         ---------------------------
0 !
0 Re_init:!
0 !
0  OUTPUT @Multi;"CC";Tb_slot;"T"                    ! Clear Timer/Pacer card.
0  OUTPUT @Multi;"WC";Tb_slot;0;"T"                  ! Turn off Timer/Pacer.
0  OUTPUT @Multi;"WF";Tb_slot+.2;1;"T"               ! Continuous mode Timer.
0  OUTPUT @Multi;"CC";A_d_slot;"T"                   ! Clear the A/D card.
0  OUTPUT @Multi;"SF";A_d_slot;"1,0.005T"            ! Set A/D input range.
0  OUTPUT @Multi;"CC";Mem_in_slot;"T"                ! Clear Memory card #1.
0  OUTPUT @Multi;"CC";Mem_in_slot+1;"T"              ! Clear Memory card #2.
0  OUTPUT @Multi;"WF";Mem_in_slot+1;0;"T"            ! Clear the REF REGISTER.
0  OUTPUT @Multi;"WF";Mem_in_slot+.1;21;"T"          ! Disable Memory card #1.
0  !
0  ! SET THE TIME BASE CARD'S CLOCK FREQUENCY
0  ! ----------------------------------------
0  OUTPUT @Multi;"WF";Tb_slot;Trigger_rate*1.E+6/2;"UT"
0  !
0  ! PROGRAM THE DIFFERENTIAL COUNTER OF THE MEMORY CARDS
0  ! ---------------------------------------------------
0  OUTPUT @Multi;"WF";Mem_in_slot+1;Reference_word;"T"
0  !
0  ! INSTRUCT MEMORY ON NUMBER OF READINGS TO BE TAKEN
0  ! ------------------------------------------------
0  OUTPUT Mr_command$;"MR";Mem_in_slot;Block_size;"T"
0  !
0  ! ENABLE THE BUFFERED ANALOGUE TO DIGITAL CONVERSION
0  ! --------------------------------------------------
0  OUTPUT @Multi;"WF";Mem_in_slot+.1;1;"T"
0  !
0  ! ARM THE MEMORY CARD
0  ! ------------------
0  OUTPUT @Multi;"AC";Mem_in_slot+1;"T"
0  RETURN
0 !*****************************************************************************
```

```
0 !<<<<<<<<<<<<<< THE ANALOGUE-TO-DIGITAL CONVERSION ROUTINE >>>>>>>>>>>>>>
0 !               ----------------------------------------
0 !
0 Speech_adc:!
0  PRINT TABXY(0,20);"When ready hit the ";
0  PRINT CHR$(129);" CONTINUE ";CHR$(128);" key"
0  PAUSE
0  Next_in_buf=1
0  !
0  ON INTR 8 GOTO Interrupt                    ! The accumulation and storage
0  ENABLE INTR 8;2                             ! of data is interrupt driven.
0  !
0  GOSUB Clear_disp                            ! Blank the screen.
0  !
0  OUTPUT @Multi;"CY";Tb_slot;"T"              ! Cycle the time base card.
0  !
0  DISP "Stop speaking at the next tone..."
0 Sampling:!
0  GOTO Sampling
0 !
```

```
!<<<<<<<<<<<<<<<<<<<<<<< INTERRUPT SERVICE ROUTINE >>>>>>>>>>>>>>>>>>>>>>>
!                        ------------------------
!
Interrupt:!
 IF SPOLL(@Multi)=64 THEN                ! Multiprogrammer has interrupted.
   !
   ENTER @Intr_cause;Complete,No_of_errors,No_of_cards
   ENTER @Intr_list;Error$
   IF No_of_cards>0 THEN
     OUTPUT @Multi;Mr_command$
     ON Next_in_buf GOTO L1,L2,L3,L4,L5,L6,L7,L8
     !
L1:  ENTER @Mr_readback;Buf1(*),Differential,Underflow
     GOTO Data_is_in
     !
L2:  ENTER @Mr_readback;Buf2(*),Differential,Underflow
     GOTO Data_is_in
     !
L3:  ENTER @Mr_readback;Buf3(*),Differential,Underflow
     GOTO Data_is_in
     !
L4:  ENTER @Mr_readback;Buf4(*),Differential,Underflow
     GOTO Data_is_in
     !
L5:  ENTER @Mr_readback;Buf5(*),Differential,Underflow
     GOTO Data_is_in
     !
L6:  ENTER @Mr_readback;Buf6(*),Differential,Underflow
     GOTO Data_is_in
     !
L7:  ENTER @Mr_readback;Buf7(*),Differential,Underflow
     GOTO Data_is_in
     !
L8:  ENTER @Mr_readback;Buf8(*),Differential,Underflow
     BEEP 3000,.1
     GOSUB Re_init
     GOSUB Store_on_disc
     DISP "Do you wish to record more speech ? >"
Input_answer:!
     ON KBD GOTO Response
     GOTO Input_answer
Response:!
     R$=UPC$(KBD$)
     OFF KBD
     IF R$="N" THEN
       Function$="LOAD ""MASTER:INTERNAL,4,0"" X"
       OUTPUT KBD;Stop$;Function$;Run$;
     ELSE
       RETURN                            ! Return to the program mainline.
     END IF
   !
   ELSE
     DISP "SOME OTHER MULTI INTERRUPT HAS OCCURED"
     BEEP 500,1
     STOP
   END IF
 !
 ELSE
   DISP "SOME OTHER HPIB DEVICE HAS INTERRUPTED"
```

```
   BEEP 500,1
   STOP
 END IF
 !
Data_is_in:!
 IF (Differential=4095) OR (Underflow<>0) THEN Problems
 Next_in_buf=(Next_in_buf MOD No_of_buffers)+1
 ENABLE INTR 8;2
 GOTO Sampling
!*************************************************************************
```

```
0 !<<<<<<<<<<<<<<<<<<<< STORE SPEECH SEGMENT ON DISC >>>>>>>>>>>>>>>>>>>>
0 !                      ----------------------------
0 !
0 Store_on_disc:!
0   Drive$=Drive_1$                  ! Use the default disc drive.
0   !
0   PRINT "Maximum sampled value  = ";MAX(Buf1(*),Buf2(*),Buf3(*),Buf4(*),Buf5
,Buf6(*),Buf7(*),Buf8(*))
0   PRINT "Minimum sampled value  = ";MIN(Buf1(*),Buf2(*),Buf3(*),Buf4(*),Buf5
,Buf6(*),Buf7(*),Buf8(*))
0   GOSUB Calc_rms                   ! Evaluate the RMS signal level.
0   !
0   DISP "Do you wish to store that Speech Segment ? >"
0 Get_answer:ON KBD GOTO Answer
0   GOTO Get_answer
0 Answer:R$=UPC$(KBD$)
0   IF R$="Y" THEN
0     ON ERROR GOSUB Wrong_disc
0     MASS STORAGE IS Drive$
0     !
0     INPUT "Enter file name under which data is to be stored >",File_name$
0     File_name$=File_name$&"_0"
0     !
0     !<<<<<<<<<<<<<<<< SPECIFY FILE NAME AND SIZE >>>>>>>>>>>>>>>>
0     !
0 Create:    !
0     ON ERROR GOTO Duplicate
0     CREATE BDAT File_name$,8*Block_size,2
0     OFF ERROR
0     !
0     !<<<<<<<<<<<<<<<<<< SPECIFY I/O PATH NAME >>>>>>>>>>>>>>>>>>
0     !
0     DISP "Transferring data to disc."
0     ASSIGN @Data TO File_name$
0     !
0     FOR Next_in_buf=1 TO No_of_buffers
0     !
0     !<<<<<<<<<<<<<<< OUTPUT DATA TO DISC >>>>>>>>>>>>>>>>>>>
0     !
0       SELECT Next_in_buf
0       CASE 1
0         OUTPUT @Data;Buf1(*)
0       CASE 2
0         OUTPUT @Data;Buf2(*)
0       CASE 3
0         OUTPUT @Data;Buf3(*)
0       CASE 4
0         OUTPUT @Data;Buf4(*)
0       CASE 5
0         OUTPUT @Data;Buf5(*)
0       CASE 6
0         OUTPUT @Data;Buf6(*)
0       CASE 7
0         OUTPUT @Data;Buf7(*)
0       CASE 8
0         OUTPUT @Data;Buf8(*)
0       END SELECT
0       !
0     NEXT Next_in_buf
```

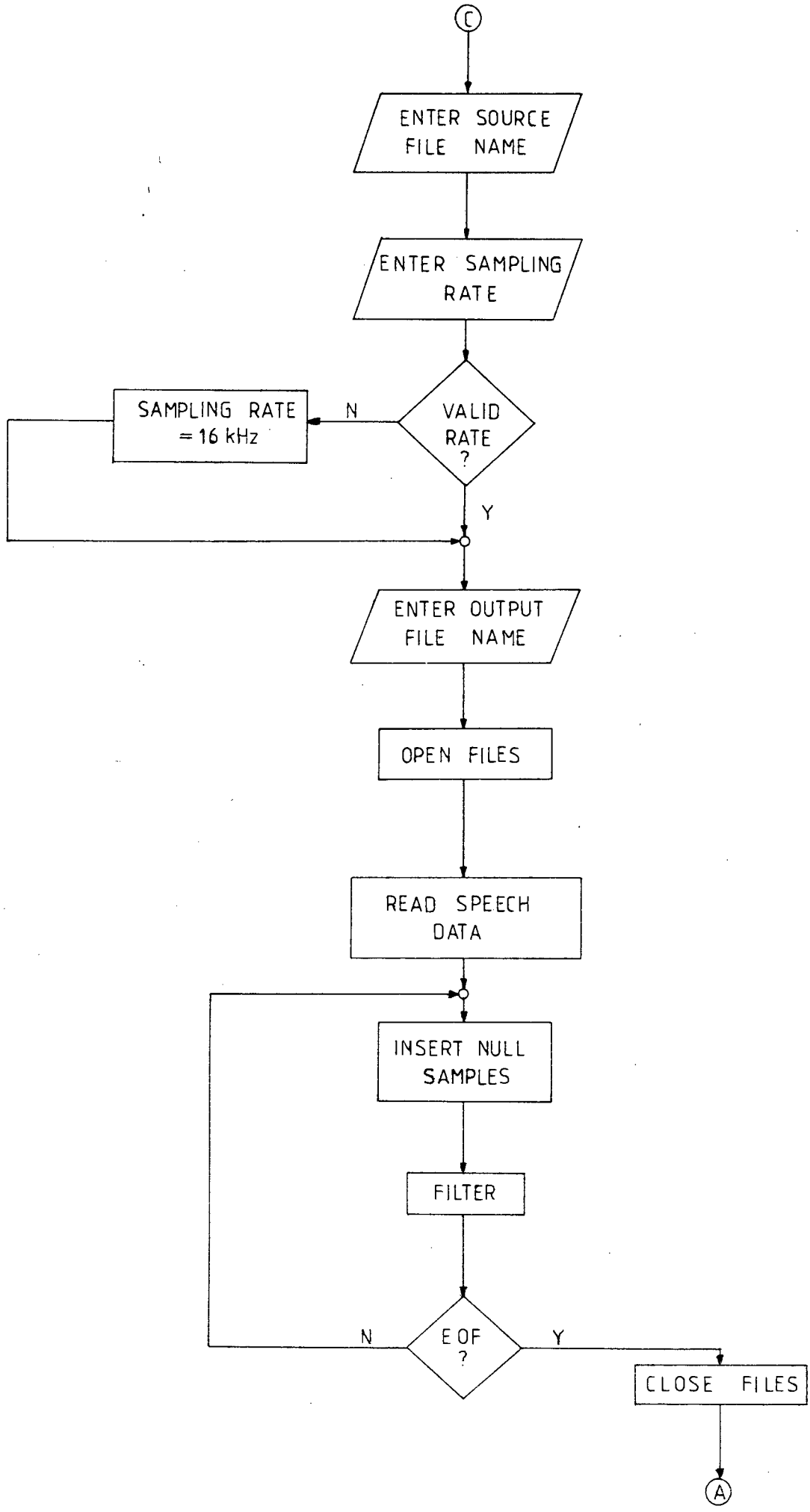
```
660     ASSIGN @Data TO *                                   ! Close the I/O path.
670   END IF
680   MASS STORAGE IS Drive_0$
690   RETURN
700 !*****************************************************************************
```

```
710 !<<<<<<<<<<<<<<<<<<<<<  RECOVER FROM A SYSTEM ERROR >>>>>>>>>>>>>>>>>>>>>>
720 !                        ---------------------------
730 !
740 Wrong_disc:!
750  IF ERRN=72 THEN           ! Attempted to store data on non-existant disc.
760    BEEP 500,.2
770    PRINT "Hard disc is not present, so use default devices."
780    PRINT "Place an initialised disc in the LEFT drive."
790    PRINT "Hit ";CHR$(129);" CONTINUE ";CHR$(128);" when ready."
800    Drive_1$=":INTERNAL,4,1"
810    PAUSE
820  ELSE
830    DISP ERRN,ERRM$
840    STOP
850  END IF
860  RETURN
870 !
880 Duplicate:              !A duplicate file name has been specified.
890  BEEP 1000,.1
900  DISP "A file of that name already exists.  Must it be overwritten (Y/N) ?"
910  ON KBD GOTO Overwrite
920  !
930 Spin: !
940  GOTO Spin
950  !
960 Overwrite: !
970  R$=UPC$(KBD$)
980  OFF KBD
990  !
000  IF R$="Y" THEN
010    DISP "Purging old version of the file."
020    PURGE File_name$
030  ELSE
040    INPUT "Enter a new file name for the data >",File_name$
050    File_name$=File_name$&"_0"
060  END IF
070  GOTO Create
080 !*****************************************************************************
```

```
090 !<<<<<<<<<<<<<<<<<<  CALCULATE THE RMS SIGNAL VALUE  >>>>>>>>>>>>>>>>>>>
100 !                    ------------------------------
110 !
120 Calc_rms:!
130  DISP "Calculating the RMS signal value."
140  MAT Buffer1= Buf1                          ! Make a copy of the first
150  MAT Buffer2= Buf2                          ! half of the speech segment.
160  MAT Buffer3= Buf3
170  MAT Buffer4= Buf4
180  MAT Buffer1= Buffer1 . Buffer1             ! Square each sample in the
190  MAT Buffer2= Buffer2 . Buffer2             ! first half.
200  MAT Buffer3= Buffer3 . Buffer3
210  MAT Buffer4= Buffer4 . Buffer4
220  MAT Buffer2= Buffer1+Buffer2               ! Determine the sum of the
230  MAT Buffer4= Buffer3+Buffer4               ! squared sample values.
240  MAT Buffer4= Buffer2+Buffer4
250  Signal_pwr=SUM(Buffer4)
260  MAT Buffer1= Buf5                          ! Copy each of the samples in
270  MAT Buffer2= Buf6                          ! the second half of the
280  MAT Buffer3= Buf7                          ! speech segment.
290  MAT Buffer4= Buf8
300  MAT Buffer1= Buffer1 . Buffer1             ! Square each of these
310  MAT Buffer2= Buffer2 . Buffer2             ! samples.
320  MAT Buffer3= Buffer3 . Buffer3
330  MAT Buffer4= Buffer4 . Buffer4
340  MAT Buffer2= Buffer1+Buffer2               ! Determine the sum of the
350  MAT Buffer4= Buffer3+Buffer4               ! squared samples in the
360  MAT Buffer4= Buffer2+Buffer4               ! second half of the segment.
370  Signal_rms=SQR(Signal_pwr+SUM(Buffer4)/(8*Block_size))
380  PRINT USING """Input signal RMS level = "",4D.DD";Signal_rms
390  DISP
400  RETURN
410 !*************************************************************************
```

```
120 !<<<<<<<<<<<<<<<<<<<< ALTERNATE INTERRUPT HANDLING >>>>>>>>>>>>>>>>>>>>
130 !                      ---------------------------
140 !
150 Problems:!
160  IF Differential=4095 THEN
170    BEEP 1500,.1
180    PRINT TABXY(8,9);CHR$(129);"MEMORY OVERFLOW HAS OCCURRED"
190    PRINT TABXY(11,10);"PROGRAM HAS BEEN HALTED";CHR$(128)
200  END IF
210  IF Underflow<>0 THEN
220    BEEP 1500,.1
230    PRINT TABXY(8,9);CHR$(129);"MEMORY UNDERFLOW HAS OCCURRED"
240    PRINT TABXY(11,10);"PROGRAM HAS BEEN HALTED";CHR$(128)
250  END IF
260  DISP " "
270  STOP
280  END
290 !*************************************************************************
```

```
500 !<<<<<<<<<<<<<<<<<<<<<<< BINARY LOADER SUB-PROGRAM >>>>>>>>>>>>>>>>>>>>>>>
510 !                        -------------------------
520 !
530  SUB Xfer(Op$)                              ! Transfer the special instruction
540    CLEAR 823                                ! from the supplemental instruction
550    WAIT 4                                   ! set kept on disc.
560    ENTER 82310;Dummy,Dummy,Dummy
570    ALLOCATE Ascii$[80]
580          !
590    Name$=Op$&"Z:INTERNAL,4,0"
700    ASSIGN @Disc_file TO Name$
710    ON END @Disc_file GOTO Eof
720          !
730 Rd_file:ENTER @Disc_file;Ascii$
740    OUTPUT 823;Ascii$
750    GOTO Rd_file
760          !
770 Eof:OFF END @Disc_file
780    ASSIGN @Disc_file TO *
790    DEALLOCATE Ascii$
800    PRINT TABXY(5,11);"<MR> SUPPLEMENTAL INSTRUCTION LOADED INTO MULTI"
810       ! CLEAR SELF TEST SRQ
820    ENTER 82310;Complete,No_of_errors,No_of_cards
830    WAIT 1
840    DISP " "
850  SUBEND
860 !*****************************************************************************
870 !
880 !<<<<<<<<<<<<< DETERMINE MULTIPROGRAMMER CARD CONFIGURATION >>>>>>>>>>>>>>
890 !               ------------------------------------------
900 !
910  SUB Multi_set_up(INTEGER Configuration(*),Tb_slot,Mem_in_slot,A_d_slot)
920    OPTION BASE 1
930    PRINT TABXY(5,13);"DETERMINING MULTI CONFIGURATION"
940    !
950    FOR Slot_no=15 TO 0 STEP -1              ! Read data format of each card
960      OUTPUT 823;"RF";Slot_no;"T"            ! starting at SLOT #0.
970 !
980      FOR Card_variable=1 TO 5               ! Five variables returned per
990                                             ! card - CARD ID, DATA TYPE,
000                                             !        LSB, SIZE, LIMIT
010                                             !
020        ENTER 82304 USING "%,K";Configuration(Slot_no,Card_variable)
030      NEXT Card_variable
040      SELECT Configuration(Slot_no,1)
050      CASE 1
060        Tb_slot=Slot_no
070      CASE 6
080        Mem_in_slot=Slot_no
090      CASE 52
100        A_d_slot=Slot_no
110      CASE ELSE
120      END SELECT
130    NEXT Slot_no
140  SUBEND
150 !*****************************************************************************
```


C
ENTER SOURCE FILE NAME
ENTER SAMPLING RATE
VALID RATE ?
N
SAMPLING RATE = 16 kHz
Y
ENTER OUTPUT FILE NAME
OPEN FILES
READ SPEECH DATA
INSERT NULL SAMPLES
FILTER
EOF ?
N
Y
CLOSE FILES
A

```
!-----------------------------------------------------------------------
!       PROGRAM FOR INTERPOLATION FROM 8 kHz DATA TO A HIGHER RATE
!       ==========================================================
!
! CODSIM          : Codec Simulation Package.
! M.Sc Thesis     : The use of Delta Modulation for Low Bit-Rate Digital
!                   Speech Coding.
! Author          : J.M.Irvine.
!                   Department of Electrical and Electronic Engineering
!                   University of Cape Town.
! Unit Number     : 2.
! Latest Update   : 3rd December, 1984.
!-----------------------------------------------------------------------
! Programs developed for the Hewlett Packard 200 Series desk-top
! computers.  All development work done on an HP9836C computer with the
! interfacing between the analogue and digital environments done using
! the HP6942A Multiprogrammer.
!-----------------------------------------------------------------------
! This program interpolates the sampled speech data from 8 kHz to an
! effective sampling rate which is selected by the user.  Only integer
! multiples of the base sampling rate are realisable.  The possible
! sampling rates available with this package are 16, 24, 32, 48 and 64
! kHz.  For a 10 second speech segment it is advisable to have a hard
! disc available as the mass storage device (a 5 1/4 inch discs cannot
! store sufficient data).
!-----------------------------------------------------------------------
!
!<<<<<<<<<<<<<<<< DEFINE THE VARIABLES USED BY THE PROGRAM >>>>>>>>>>>>>>>
!                 ----------------------------------------
!
OPTION BASE 1
!
INTEGER I,J,K,Rate,Position,Last_flag,Rate_blocks,Block_size
!
DIM In_file$[8],Out_file$[8],File_name$[12]
DIM Function$[40],Error$[40]
!
ALLOCATE INTEGER Speech(1:10283)     ! The buffer for the input speech.
ALLOCATE INTEGER Voice(1:8,1:10283) ! The buffer for the output speech.
ALLOCATE REAL Vbuff(1:10283)         ! The filter operates on real values.
ALLOCATE REAL Window(1:121)          ! The window length for filtering.
!
Run$=" R"                    ! Control character to start program execution.
Stop$=" !"                   ! Control character to stop program execution.
!***********************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<<<<<<<<<< M A I N L I N E >>>>>>>>>>>>>>>>>>>>>>>>>>>>
!                            ===============
!
GOSUB Initialise             ! Reset the mode of the controller.
!
GOSUB General_param          ! Input the necessary file information, etc.
!
Rate_blocks=Rate/8           ! There are 8 blocks of data for 8 kHz
                             ! sampled data.
GOSUB Open_file              ! Open the speech data files.
!
FOR I=1 TO 8                                 ! Interpolate the 8 blocks of 8 kHz
                                             ! data.
  GOSUB Get_input                            ! Read a block of input data.
  Stretch(Rate,Speech(*),Voice(*))           ! Insert null samples.
  !
  FOR J=1 TO Rate_blocks
    !
    DISP "Processing 8 X ";Rate_blocks;" blocks of data.  Block : ";I,J
                                             !
                                             ! If the data is interpolated to 16
    FOR K=1 TO Block_size                    ! kHz then only twice the number of
      Vbuff(K)=Voice(J,K)                    ! samples are produced.  If to 24,
    NEXT K                                   ! then three times, etc.
                                             !
    Filter(PI,Position,Last_flag,Rate,Window(*),Vbuff(*))
    MAT Speech= Vbuff
    GOSUB Save_output
    !
  NEXT J
!
NEXT I
!
ASSIGN @Source TO *                  ! Close the source file.
ASSIGN @Destination TO *             ! Close the output file.
!
GOSUB Clear_disp                     ! Clear the dsplay screen.
!
! Return the user to the main menu of the simulation system.
! ---------------------------------------------------------
!
Finish:  !
  Function$="LOAD ""MASTER:INTERNAL,4,0"" X"
  OUTPUT KBD;Stop$;Function$;Run$;
  !*****************************************************************************
```

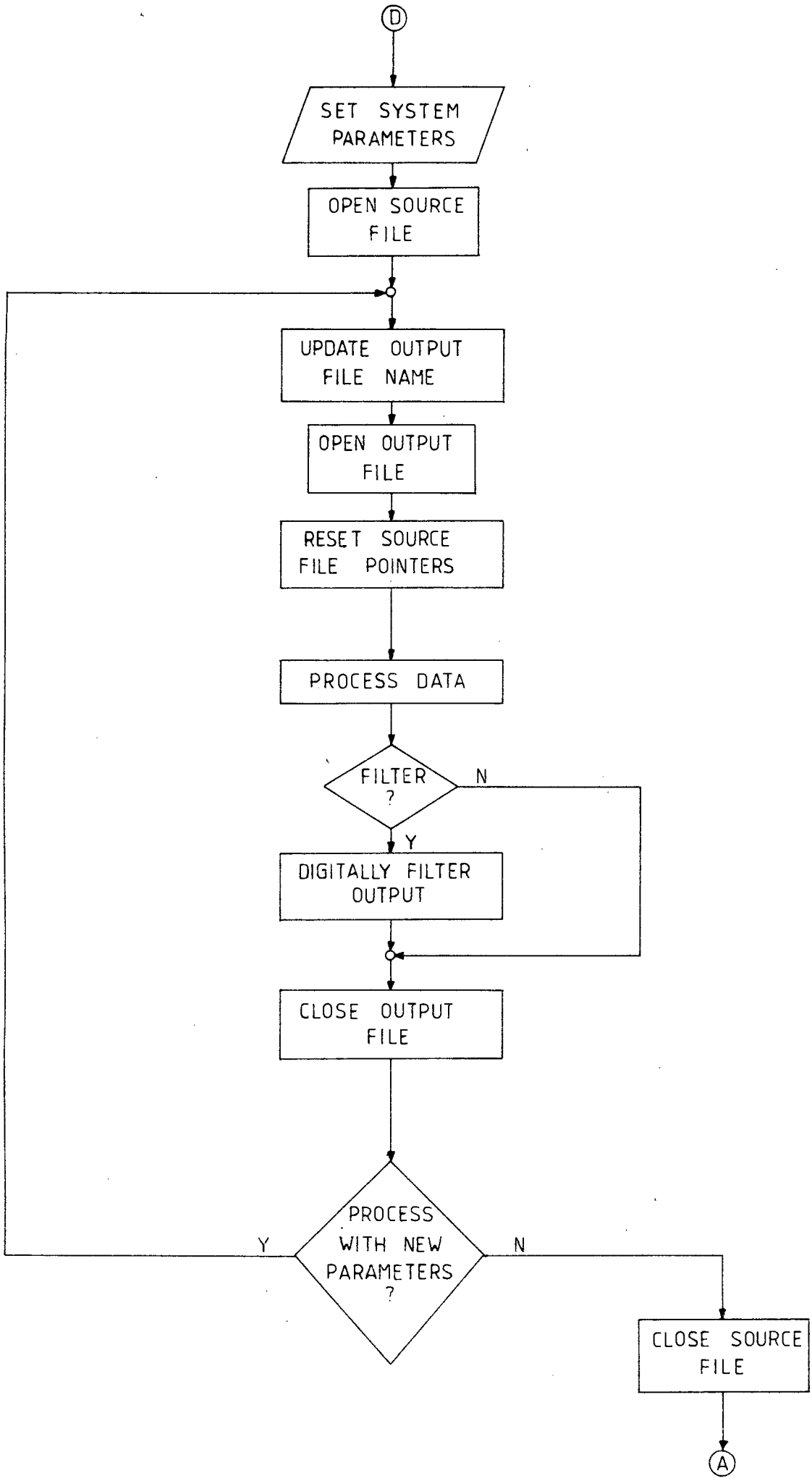
```
!
!<<<<<<<<<<<<<<<<<<<<<<<< S U B R O U T I N E S >>>>>>>>>>>>>>>>>>>>>>>
!                        ====================
!
!<<<<<<<<<<<<<<<< SPECIFY THE DEFAULT MASS STORAGE DEVICES >>>>>>>>>>>>>>>
!                 ----------------------------------------
!
Initialise: !
 Drive_0$=":INTERNAL,4,0"                    ! Set the default mass storage
 Drive_1$=":INTERNAL,4,1"                    ! devices.
 PRINTER IS CRT                              ! Use the VDU for displays.
 CONTROL 1,12;1                              ! Turn the key label display off.
 !
 ! Initialise the values of the simulation system control parameters.
 !-------------------------------------------------------------------
 !
 Block_size=10283
 In_file$="TEST2"
 Out_file$="TEST2"
 Rate=16
 Rate_blocks=8
 Position=60
 Last_flag=0
 !
 MAT Voice= (0)                              ! Clear the arrays used for
 MAT Vbuff= (0)                              ! processing the speech data.
 MAT Speech= (0)
 MAT Window= (0)
 !
 !<<<<<<<<<<<< CLEAR THE ALPHANUMERIC AND GRAPHICS DISPLAYS >>>>>>>>>>>>>
 !             ---------------------------------------------
 !
Clear_disp:!
 GCLEAR                                      ! Clear the graphics display.
 CONTROL 1,12;1                              ! Turn the key label display off.
 ALPHA OFF
 PRINT USING "@"                             ! Clear the Alpha display.
 PRINT CHR$(128);CHR$(140);                  ! Select normal video with
 ALPHA ON                                    ! Colour = Cyan.
 DISP " "
 RETURN
!*****************************************************************************
```

```
50 !<<<<<<<<<<<<<<< OPEN THE FILE CONTAINING THE SOURCE DATA >>>>>>>>>>>>>>>
60 !                 ---------------------------------------
70 !
80 Open_file:!
90  Drive$=Drive_0$
00  MASS STORAGE IS Drive$
10  File_name$=In_file$&"_0"
20  ASSIGN @Source TO File_name$
30  !
40  !
50  ! Create and open a file into which the interpolated data can be stored.
60  !
70  File_name$=Out_file$&"_P"
80  CREATE BDAT File_name$,Rate_blocks*Block_size,2
90  ASSIGN @Destination TO File_name$
00  !
10  RETURN
20 !*************************************************************************
30 !
40 !<<<<<<<<<<<<<<< READ THE DATA FROM THE FILES STORED ON DISC >>>>>>>>>>>>>>
50 !                 ------------------------------------------
60 !
70 Get_input:!
80  DISP "Reading speech data off disc."
90  ENTER @Source;Speech(*)
00  DISP
10  RETURN
20 !*************************************************************************
30 !
40 !<<<<<<<<<<<<<<<<<< STORE THE INTERPOLATED SPEECH DATA >>>>>>>>>>>>>>>>>>
50 !                   ----------------------------------
60 !
70 Save_output:!
80  DISP "Writing interpolated data to disc."
90  OUTPUT @Destination;Speech(*)
00  DISP
10  RETURN
20 !*************************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<< DETERMINE THE SPEECH FILE NAMES >>>>>>>>>>>>>>>>>>>>
!                     ------------------------------
!
General_param: !
 !
 GOSUB Clear_disp
 !
 PRINT TAB(22);CHR$(132);"Speech file information :";CHR$(128);CHR$(10)
 PRINT TAB(20);"Source file name        : ";CHR$(129);In_file$;CHR$(128);CH
(10)
 PRINT TAB(20);"Effective sampling rate :";Rate;" kHz";CHR$(10)
 PRINT TAB(20);"Destination file name   : ";Out_file$;CHR$(10)
 PRINT TAB(10);CHR$(129);" The default setting is selected by pressing 'ENT
";CHR$(128)
 INPUT "Enter the source file name",In_file$
 In_file$=UPC$(In_file$)
 PRINT TABXY(20,4);"Source file name        : ";In_file$;"          ";CHR$(10

 !
 PRINT TAB(20);"Effective sampling rate :";CHR$(129);Rate;" kHz";CHR$(128)
 INPUT "Enter the required rate (16, 24, 32, 48, 64 kHz)",Value
 IF (Value=16) OR (Value=24) OR (Value=32) OR (Value=48) OR (Value=64) THEN
   Rate=Value
 END IF
 PRINT TABXY(20,6);"Effective sampling rate :";Rate;" kHz";CHR$(10)
 !
 PRINT TAB(20);"Destination file name   : ";CHR$(129);Out_file$;CHR$(128)
 INPUT "Enter the destination file name",Out_file$
 Out_file$=UPC$(Out_file$)
 PRINT TABXY(20,8);"Destination file name   : ";Out_file$;"          ";CHR$(1

 RETURN
 !*****************************************************************************
 END
```


D
SET SYSTEM PARAMETERS
OPEN SOURCE FILE
UPDATE OUTPUT FILE NAME
OPEN OUTPUT FILE
RESET SOURCE FILE POINTERS
PROCESS DATA
FILTER ?
N
Y
DIGITALLY FILTER OUTPUT
CLOSE OUTPUT FILE
PROCESS WITH NEW PARAMETERS ?
Y
N
CLOSE SOURCE FILE
A

```
!-----------------------------------------------------------------------
!           CODEC SIMULATION PROGRAMS FOR SUBJECTIVE ANALYSIS
!           =================================================
!
! CODSIM         : Codec Simulation Package.
! M.Sc Thesis    : The use of Delta Modulation for Low Bit-Rate Digital
!                  Speech Coding.
! Author         : J.M.Irvine.
!                  Department of Electrical and Electronic Engineering
!                  University of Cape Town.
! Unit Number    : 3.
! Latest Update : 3rd December, 1984.
!-----------------------------------------------------------------------
!
! Programs developed for the Hewlett Packard 200 Series desk-top
! computers.  All development work done on an HP9836C computer with the
! interfacing between the analogue and digital environments done using
! the HP6942A Multiprogrammer.
!-----------------------------------------------------------------------
!
! This program processes the speech data, stored on disc, using the
! simulated delta modulators (ie. the LDM, CVSD, CFDM, SVADM, SHCDM,
! MSHCDM, and HCDM coders).  Once the data has been procesed by the
! relevant codec it is stored on disc, thus enabling subjective tests
! to be performed using this processed data.
!-----------------------------------------------------------------------
!
!
!<<<<<<<<<<<<<<<< DEFINE THE VARIABLES USED BY THE PROGRAM >>>>>>>>>>>>>>>
!                  ---------------------------------------
!
OPTION BASE 1
INTEGER Syllable,Err1,Err2,Err3,So                  ! The codec parameters -
REAL Alpha,Beta,Step,New_step                       ! step sizes, leakage
REAL Big_alpha,Signal,New_signal                    ! control variables, etc.
REAL Max_step,Max_cf,Max_sig,Min_sig
!
INTEGER Position,Processor_no,So_start,So_fin       ! Simulation control
INTEGER Syll_start,Syll_fin,Proc_start,Proc_fin     ! variables - sampling
INTEGER So_inc,Syll_inc,Proc_inc                    ! rate, codec type, etc.
REAL Rate,Alpha_start,Alpha_fin,Beta_start
REAL Beta_fin,Alpha_inc,Beta_inc
!
DIM In_file$[8],Out_file$[8],File_name$[12]
DIM Function$[40],Error$[40]
!
ALLOCATE INTEGER Speech(1:10283)          ! The buffer for the input speech.
ALLOCATE INTEGER P_speech(1:10283)        ! The buffer for processed speech.
ALLOCATE REAL Voice(1:10283)              ! Codecs operate on real values.
ALLOCATE REAL Syllab(1:1000)              ! Syllabic companding filter.
ALLOCATE REAL Window(1:60)                ! The window length for filtering.
!
Run$=" R"                    ! Control character to start program execution.
Stop$=" !"                   ! Control character to stop program execution.
!***************************************************************************
```

```
0   !<<<<<<<<<<<<<<<<<<<<<<<<<<<< M A I N L I N E >>>>>>>>>>>>>>>>>>>>>>>>>>>>
0   !                            ===============
0   !
0   GOSUB Initialise             ! Reset the mode of the controller.
0   GOSUB Set_parameters         ! Set the parameters controlling the codec
0                                ! type and the parameter ranges.
0   File_count=0                 ! Initialise output file count parameter.
0   GOSUB Open_file              ! Open the file containing the speech data.
0   !
0   FOR Processor_no=Proc_start TO Proc_fin STEP Proc_inc
0     !
0     ! It is possible to process the speech data using a range of different
0     ! parameter settings.  This enables the effects of the parameters to be
0     ! evaluated subjectively.
0     !-------------------------------------------------------------------------
0     !
0     FOR Alpha=Alpha_start TO Alpha_fin STEP Alpha_inc
0       FOR Beta=Beta_start TO Beta_fin STEP Beta_inc
0         FOR So=So_start TO So_fin STEP So_inc
0           FOR Syllable=Syll_start TO Syll_fin STEP Syll_inc
0             !
0             RESET @Source            ! Reset the source file to start.
0             File_count=File_count+1  ! The number of output files.
0             GOSUB Open_outfile       ! Open a file for the output data.
0             !
0             MAT Syllab= (0)          ! Clear the syllabic filter.
0             MAT Window= (0)          ! Clear the digital output filter.
0             !
0             GOSUB Print_message      ! Output a status message to VDU.
0             FOR Blocks=1 TO Rate_blocks
0               !
0               DISP "Processing ";Blocks;" out of ";Rate_blocks;" of ";Rate;"
Hz data."
0               GOSUB Get_input        ! Read the input speech data off disc.
0               MAT Voice= Speech      ! Copy from Integer to Real.
0               !
0               ! The procedure "Codecs", which performs the simulation,
0               ! consists of a compiled PASCAL module.  The listing of this
0               ! module is contained in Unit no. 6.
0               !-----------------------------------------------------------
0               !
0*              Codecs(So,Err1,Pos,Err2,Err3,Processor_no,Syllable,Beta,Rate,A
ha,Signal,Max_cf,Max_sig,Min_sig,New_step,Max_step,Big_alpha,New_signal,Syllab
0               !
0               IF Filter_flag=1 THEN
0                 Filter(Rate,Window(*),Voice(*))
00              END IF
10              !
20              MAT P_speech= Voice    ! Make an integer copy of the speech.
30              GOSUB Save_output      ! Copy the processed data into a file.
40              !
50            NEXT Blocks
60            ASSIGN @Destination TO * ! Close the file containing output.
70            !
80          NEXT Syllable
90        NEXT So
00      NEXT Beta
10    NEXT Alpha
20  NEXT Processor_no
```

```
30  !
40  ASSIGN @Source TO *                    ! Close the source file.
50  GOSUB Clear_disp                       ! Clear the dsplay screen.
60  !
70  ! Return the user to the main menu of the simulation system.
80  ! ------------------------------------------------------------
90  !
00 Finish: !
10  Function$="LOAD ""MASTER:INTERNAL,4,0"" X"
20  OUTPUT KBD;Stop$;Function$;Run$;
30  !*************************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<<<<<< S U B R O U T I N E S >>>>>>>>>>>>>>>>>>>>>>>
!                        ====================
!
!<<<<<<<<<<<<<<<< SPECIFY THE DEFAULT MASS STORAGE DEVICES >>>>>>>>>>>>>>>
!                ---------------------------------------
!
Initialise: !
 Drive_0$=":INTERNAL,4,0"                    ! Set the default mass storage
 Drive_1$=":INTERNAL,4,1"                    ! devices.
 PRINTER IS CRT                              ! Use the VDU for displays.
 CONTROL 1,12;1                              ! Turn the key label display off.
 !
 ! Initialise the values of the simulation system control parameters.
 !-------------------------------------------------------------------
 !
 In_file$="TEST2"
 Out_file$="TEST2"
 !
 Rate=16
 Rate_blocks=8
 Proc_start=1
 Proc_fin=1
 Proc_inc=1
 So_start=1
 So_fin=1
 So_inc=1
 Alpha_start=.25
 Alpha_fin=.25
 Alpha_inc=.01
 Beta_start=1.53
 Beta_fin=1.53
 Beta_inc=.01
 Syll_start=5
 Syll_fin=5
 Syll_inc=1
 !
 Max_step=1000
 Max_cf=1.5^17
 Min_sig=-32768
 Max_sig=+32767
 !
 !<<<<<<<<<<<< CLEAR THE ALPHANUMERIC AND GRAPHICS DISPLAYS >>>>>>>>>>>>>
 !            ---------------------------------------------
 !
 !
Clear_disp:!
 GCLEAR                                      ! Clear the graphics display.
 CONTROL 1,12;1                              ! Turn the key label display off.
 ALPHA OFF
 PRINT USING "@"                             ! Clear the Alpha display.
 PRINT CHR$(128);CHR$(140);                  ! Select normal video with
 ALPHA ON                                    ! Colour = Cyan.
 DISP " "
 RETURN
!*****************************************************************************
```

```
!<<<<<<<<<<<<<<< OPEN THE FILE CONTAINING THE SOURCE DATA >>>>>>>>>>>>>>>
!                ---------------------------------------
!
Open_file:!
 Drive$=Drive_0$
 MASS STORAGE IS Drive$
 File_name$=In_file$&"_P"
 ASSIGN @Source TO File_name$
 RETURN
!*******************************************************************************
!
!<<<<<<<<<<<<<<< READ THE DATA FROM THE FILES STORED ON DISC >>>>>>>>>>>>>>>
!                ---------------------------------------
!
Get_input:!
 DISP "Reading speech data off disc."
 ENTER @Source;Speech(*)
 DISP
 RETURN
!*******************************************************************************
!<<<<<<<<<<<<<<< OPEN A FILE FOR THE PROCESSED SPEECH DATA >>>>>>>>>>>>>>>
!                ---------------------------------------
!
Open_outfile:!
 Drive$=Drive_1$
 MASS STORAGE IS Drive$
 !
 ! Update the name of the output file.
 !
 File_name$=Out_file$&VAL$(File_count)&"_P"
 CREATE BDAT File_name$,Rate_blocks*Block_size,2
 !
 ASSIGN @Destination TO File_name$
 !
 RETURN
!*******************************************************************************
!
!<<<<<<<<<<<<< TRANSFER THE PROCESSED DATA TO THE OUTPUT FILE >>>>>>>>>>>>>
!              ---------------------------------------------
!
Save_output:!
 DISP "Transferring processed speech to disc"
 OUTPUT @Destination;P_speech(*)
 DISP
 RETURN
!*******************************************************************************
```

```
240 !<<<<<<<<<<<<<<<< PRINT A MESSAGE ON THE VDU FOR THE USER >>>>>>>>>>>>>>>>
250 !                 ------------------------------------
260 !
270 Print_message:!
280  !
290  GOSUB Clear_disp
300  !
310  PRINT "Processing the speech data contained in ";In_file$
320  PRINT "Processing using a ";
330  SELECT Processor_no
340  CASE 1
350    PRINT "Linear Delta Modulator";
360  CASE 2
370    PRINT "Continuously Variable Slope Delta Modulator";
380  CASE 3
390    PRINT "Constant Factor Delta Modulator";
400  CASE 4
410    PRINT "Song Voice Adaptive Delta Modulator";
420  CASE 5
430    PRINT "Song Hybrid Companding Delta Modulator";
440  CASE 6
450    PRINT "Modified Song Hybrid Companding Delta Modulator";
460  CASE 7
470    PRINT "Hybrid Companding Delta Modulator";
480  END SELECT
490  PRINT
500  IF Filter_flag=1 THEN
510    PRINT "Digital filtering > YES."
520  ELSE
530    PRINT "Digital filtering > NO."
540  END IF
550  PRINT
560  RETURN
570 !*************************************************************************
```

```
580 !<<<<<<<<<<<<<<< SET THE CONTROL PARAMETERS OF THE SYSTEM >>>>>>>>>>>>>>>
590 !                 ---------------------------------------
600 !
610 Set_parameters:!
620  !
630  ! This subroutine enables all the variable and parameter values to be set
640  ! First the codecs that are to be used are enabled and then the variables
650  ! relating to each are set.  Also, the source and destination files are
660  ! specified, as well as the data rate.
670  !-------------------------------------------------------------------------
680  !
690  !<<<<<<<<<<<<<<<<<<<< DISPLAY THE CODEC-TYPE MENU >>>>>>>>>>>>>>>>>>>>
700  !                      --------------------------
710  !
720  PRINT TAB(18);CHR$(132);"Processor Menu : Select an option";CHR$(128)
730  PRINT TAB(26);CHR$(132);"Codec Type";CHR$(128)
740  PRINT TABXY(26,5);"1)  LDM"
750  PRINT TAB(26);"2)  CVSD"
760  PRINT TAB(26);"3)  CFDM"
770  PRINT TAB(26);"4)  SVADM"
780  PRINT TAB(26);"5)  SHCDM"
790  PRINT TAB(26);"6)  MSHCDM"
800  PRINT TAB(26);"7)  HCDM"
810  PRINT CHR$(10)
820  PRINT TAB(18);CHR$(132);"Start";CHR$(128);TAB(31);CHR$(132);"Stop";CHR$(12
;TAB(43);CHR$(132);"Increment";CHR$(128)
830  !
840  !-------------------------------------------------------------------------
850  !
860 Alter_values:                          ! Enter the required codec types.
870  !
880  PRINT TABXY(19,15);CHR$(129);Proc_start;CHR$(128);TAB(32);Proc_fin;TAB(44)
roc_inc
890  PRINT CHR$(10)
900  PRINT "A codec type '0' returns to main menu."
910  DISP CHR$(140);"Enter the desired value"
920  !
930  GOSUB Keyboard_input
940  IF Value<>9 THEN
950    Proc_start=Value
960  END IF
970  !
980  PRINT TABXY(19,15);Proc_start;TAB(30);CHR$(129);Proc_fin;CHR$(128);TAB(44)
roc_inc
990  GOSUB Keyboard_input
000  IF Value<>9 THEN
010    Proc_fin=Value
020  END IF
030  !
040  PRINT TABXY(19,15);Proc_start;TAB(30);Proc_fin;TAB(42);CHR$(129);Proc_inc;
HR$(128)
050  Increment=0
060  INPUT "Enter the desired value",Increment
070  IF Increment<>0 THEN
080    Proc_inc=Increment
090    Increment=0
100  END IF
110  !
```

```
120   PRINT TABXY(19,15);Proc_start;TAB(30);Proc_fin;TAB(42);Proc_inc
130   DISP "Do you wish to keep these values (Y/N) ?"
140   !
150   ON KBD GOTO Answer
160   !
170 Wait:!
180   GOTO Wait
190   !
200 Answer: !
210   R$=UPC$(KBD$)
220   IF (R$="Y") OR (R$=" E") THEN
230     IF (Proc_start=0) OR (Proc_fin=0) THEN    ! A value of zero for a codec
240       GOTO Finish                             ! type returns the user to
250     END IF                                    ! the main menu.
260   ELSE
270     GOTO Alter_values
280   END IF
290   !
300   !-------------------------------------------------------------------------
310   !
320   ! Update the various design parameters of the different codecs.
330   !
340   GOSUB Clear_disp
350   !
360 Design_param: !
370   !
380   PRINT TABXY(18,2);CHR$(129);"Codec algorithm parameters :";CHR$(128);CHR$(
0)
390   PRINT TAB(21);CHR$(132);"Minimum step size (So)";CHR$(128);CHR$(10)
400   PRINT TAB(18);CHR$(132);"Start";CHR$(128);TAB(30);CHR$(132);"Stop";CHR$(12
8);TAB(41);CHR$(132);"Increment";CHR$(128)
410   PRINT TAB(19);CHR$(129);So_start;CHR$(128);TAB(31);So_fin;TAB(42);So_inc
420   DISP "Enter the desired value"
430   GOSUB Keyboard_input
440   IF Value>0 THEN
450     So_start=Value
460   END IF
470   PRINT TABXY(19,7);So_start;TAB(29);CHR$(129);So_fin;CHR$(128);TAB(42);So_i
nc
480   GOSUB Keyboard_input
490   IF Value>0 THEN
500     So_fin=Value
510   END IF
520   PRINT TABXY(19,7);So_start;TAB(29);So_fin;TAB(40);CHR$(129);So_inc;CHR$(12
8)
530   INPUT "Enter the desired value",Increment
540   IF Increment<>0 THEN
550     So_inc=Increment
560     Increment=0
570   END IF
580   PRINT TABXY(19,7);So_start;TAB(29);So_fin;TAB(40);So_inc
590   !
600   !-------------------------------------------------------------------------
```

```
510  PRINT TABXY(18,12);CHR$(132);"Leakage time constant (Beta)";CHR$(128);CHR$
10)
520  PRINT TAB(18);CHR$(132);"Start";CHR$(128);TAB(30);CHR$(132);"Stop";CHR$(12
);TAB(41);CHR$(132);"Increment";CHR$(128)
530  PRINT TABXY(17,15);CHR$(129);Beta_start;CHR$(128);TAB(29);Beta_fin;TAB(41)
Beta_inc
540  Value=0
650  INPUT "Enter the desired value",Value
660  IF Value<>0 THEN
670    Beta_start=Value
680    Value=0
690  END IF
700  PRINT TABXY(17,15);Beta_start;TAB(27);CHR$(129);Beta_fin;CHR$(128);TAB(41)
Beta_inc
710  INPUT "Enter the desired value",Value
720  IF Value<>0 THEN
730    Beta_fin=Value
740    Value=0
750  END IF
760  PRINT TABXY(17,15);Beta_start;TAB(27);Beta_fin;TAB(39);CHR$(129);Beta_inc;
HR$(128)
770  INPUT "Enter the desired value",Value
780  IF Value<>0 THEN
790    Beta_inc=Value
800    Value=0
810  END IF
820  PRINT TABXY(17,15);Beta_start;TAB(27);Beta_fin;TAB(39);Beta_inc
830  DISP "Do you wish to keep these values (Y/N) ?"
840  !
850  ON KBD GOTO Answer_2
860  !
870 Wait_2:!
880  GOTO Wait_2
890  !
900 Answer_2: !
910  R$=UPC$(KBD$)
920  IF R$="N" THEN
930    GOTO Design_param
940  END IF
950  !
960  !-----------------------------------------------------------------------
```

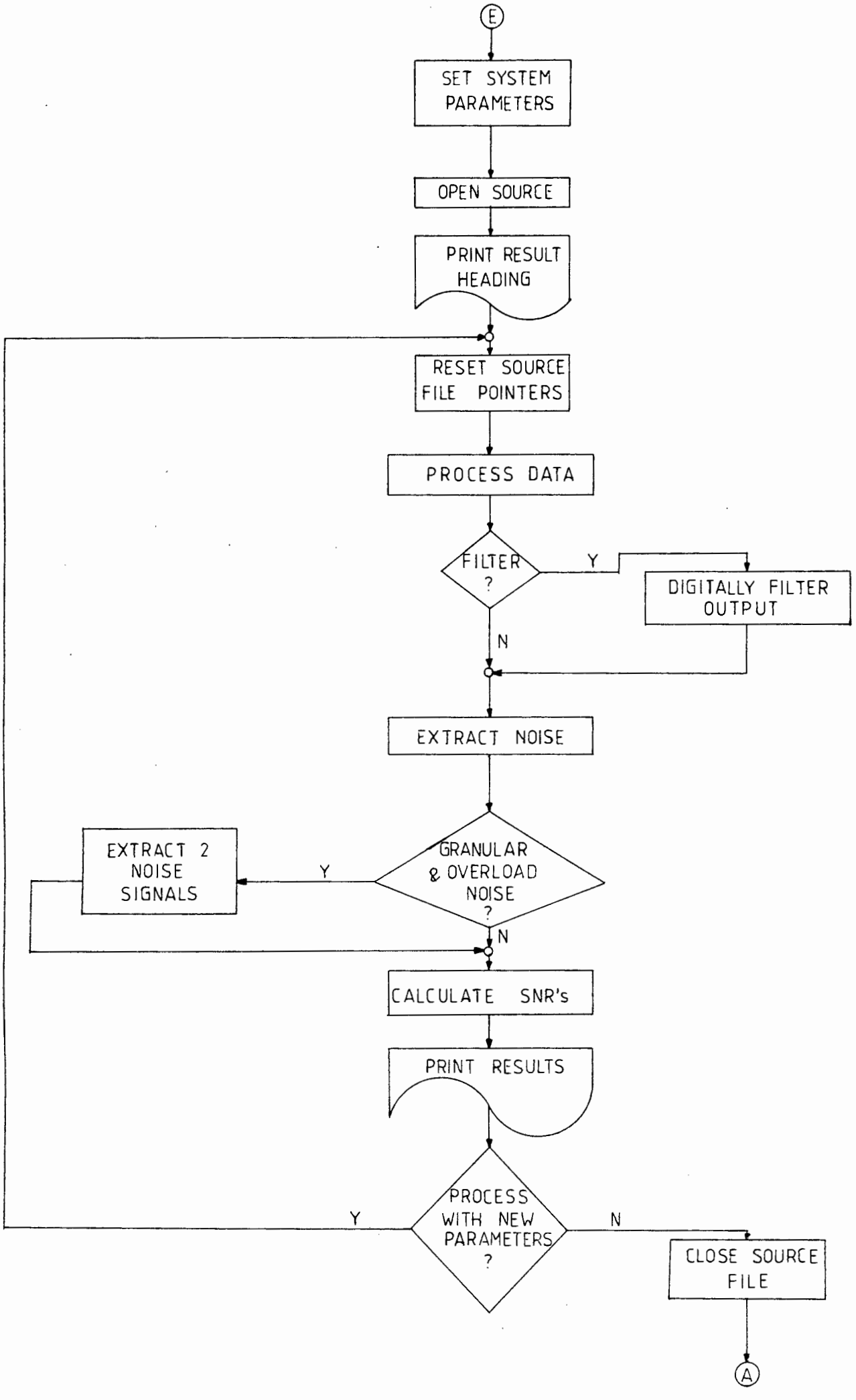
```
970   ! The parameters for the hybrid codecs are only updated if the range of
980   ! of the codec types, chosen previously, includes one or more of the
990   ! hybrid codecs.
000   !
010   IF (Proc_start<=4) AND (Proc_fin<=4) THEN
020     GOTO General_param
030   END IF
040   !
050   GOSUB Clear_disp
060   !
070 Hybrid_param: !
080   !
090   PRINT TABXY(23,2);CHR$(129);"Hybrid Design Parameters :";CHR$(128);CHR$(10

00   PRINT TAB(18);CHR$(132);"Syllabic Companding Constant (Alpha)";CHR$(128);C
R$(10)
10   PRINT TAB(23);CHR$(132);"Start";CHR$(128);TAB(35);CHR$(132);"Stop";CHR$(12
;TAB(46);CHR$(132);"Increment";CHR$(128)
20   PRINT TAB(23);CHR$(129);Alpha_start;CHR$(128);TAB(35);Alpha_fin;TAB(46);Al
a_inc
30   Value=0
40   INPUT "Enter the desired value",Value
50   IF Value<>0 THEN
60     Alpha_start=Value
70     Value=0
80   END IF
90   PRINT TABXY(23,7);Alpha_start;TAB(33);CHR$(129);Alpha_fin;CHR$(128);TAB(46
Alpha_inc
00   INPUT "Enter the desired value",Value
10   IF Value<>0 THEN
20     Alpha_fin=Value
230    Value=0
40   END IF
50   PRINT TABXY(23,7);Alpha_start;TAB(33);Alpha_fin;TAB(44);CHR$(129);Alpha_in
CHR$(128)
60   INPUT "Enter the desired value",Value
70   IF Value<>0 THEN
280    Alpha_inc=Value
290    Value=0
300  END IF
310  PRINT TABXY(23,7);Alpha_start;TAB(33):Alpha_fin;TAB(44);Alpha_inc;CHR$(32)
320  !
330  !-------------------------------------------------------------------------
```

```
340  PRINT TABXY(24,12);CHR$(132);"Syllabic Companding Period";CHR$(128);CHR$(1
)
350  PRINT TAB(23);CHR$(132);"Start";CHR$(128);TAB(35);CHR$(132);"Stop";CHR$(12
);TAB(46);CHR$(132);"Increment";CHR$(128)
360  PRINT TABXY(23,15);CHR$(129);Syll_start;CHR$(128);TAB(35);Syll_fin;TAB(47)
Syll_inc
370  Value=0
380  INPUT "Enter the desired value (milliseconds)",Value
390  IF Value<>0 THEN
400    Syll_start=Value
410    Value=0
420  END IF
430  PRINT TABXY(23,15);Syll_start;TAB(33);CHR$(129);Syll_fin;CHR$(128);TAB(47)
Syll_inc
440  INPUT "Enter the desired value (milliseconds)",Value
450  IF Value<>0 THEN
460    Syll_fin=Value
470    Value=0
480  END IF
490  PRINT TABXY(23,15);Syll_start;TAB(33);Syll_fin;TAB(45);CHR$(129);Syll_inc;
R$(128)
500  INPUT "Enter the desired value (milliseconds)",Value
510  IF Value<>0 THEN
520    Syll_inc=Value
530    Value=0
540  END IF
550  PRINT TABXY(23,15);Syll_start;TAB(33);Syll_fin;TAB(45);Syll_inc
560  DISP "Do you wish to keep these values (Y/N) ?"
570  !
580  ON KBD GOTO Answer_3
590  !
600 Wait_3:!
610  GOTO Wait_3
620  !
630 Answer_3: !
640  R$=UPC$(KBD$)
650  IF R$="N" THEN
660    GOTO Hybrid_param
670  END IF
680  !
690  !-----------------------------------------------------------------------
```

```
700  ! The general parameters are updated at this stage - eg. Maximum step
710  ! sizes, maximum output signal ranges, source file names, destination
720  ! file names, etc.
730  !
740 General_param: !
750  !
760  GOSUB Clear_disp
770  !
780  PRINT TAB(22);CHR$(132);"Speech file information :";CHR$(128);CHR$(10)
790  PRINT TAB(20);"Source file name      : ";CHR$(129);In_file$;CHR$(128);CHR$
10)
800  PRINT TAB(20);"Destination file name : ";Out_file$;CHR$(10)
810  PRINT TAB(20);"Maximum signal value  : ";Max_sig;CHR$(10)
820  PRINT TAB(20);"Minimum signal value  : ";Min_sig;CHR$(10)
830  PRINT TAB(20);"Maximum step size     : ";Max_step;CHR$(10)
840  PRINT TAB(20);"Sampling rate ( kHz ) : ";Rate;CHR$(10)
850  PRINT TAB(10);CHR$(129);" The default setting is selected by pressing 'ENT
'";CHR$(128)
860  INPUT "Enter the source file name",In_file$
870  In_file$=UPC$(In_file$)
880  PRINT TABXY(20,4);"Source file name      : ";In_file$;"          ";CHR$(10)
890  PRINT TAB(20);"Destination file name : ";CHR$(129);Out_file$;CHR$(128)
900  !
910  INPUT "Enter the destination file name",Out_file$
920  Out_file$=UPC$(Out_file$)
930  PRINT TABXY(20,6);"Destination file name : ";Out_file$;"          ";CHR$(10)
940  PRINT TAB(20);"Maximum signal value  : ";CHR$(129);Max_sig;CHR$(128)
950  !
960  Value=0
970  INPUT "Enter the desired value",Value
980  IF Value>0 THEN
990    Max_sig=Value
000    Value=0
10   END IF
020  PRINT TABXY(20,8);"Maximum signal value  : ";Max_sig;"          ";CHR$(10)
30   !
040  PRINT TAB(20);"Minimum signal value  : ";CHR$(129);Min_sig;CHR$(128)
050  !
060  INPUT "Enter the desired value",Value
070  IF Value<>0 THEN
080    Min_sig=Value
090    Value=0
100  END IF
110  PRINT TABXY(20,10);"Minimum signal value  : ";Min_sig;"          ";CHR$(10)
120  !
130  PRINT TAB(20);"Maximum step size     : ";CHR$(129);Max_step;CHR$(128)
140  !
150  INPUT "Enter the desired value",Value
160  IF Value>0 THEN
170    Max_step=Value
180    Value=0
190  END IF
200  PRINT TABXY(20,12);"Maximum step size     : ";Max_step;"          ";CHR$(10)
210  !
```

```
220  PRINT TAB(20);"Sampling rate ( kHz ) : ";CHR$(129);Rate;CHR$(128)
230  INPUT "Enter the sampling rate ( kHz ) : 16, 24, 32, 48, or 64",Value
240  IF (Value=16) OR (Value=24) OR (Value=32) OR (Value=48) OR (Value=64) THEN
250    Rate=Value
260  END IF
270  PRINT TABXY(20,14);"Sampling rate ( kHz ) : ";Rate;CHR$(10)
280  !
290  PRINT TAB(80)
300  !
310  DISP "Do you require the processed speech to be filtered (Y/N) ?"
320  !
330  ON KBD GOTO Answer_4
340  !
350 Wait_4:!
360  GOTO Wait_4
370  !
380 Answer_4: !
390  R$=UPC$(KBD$)
400  IF R$="N" THEN
410    Filter_flag=0
420  END IF
430  !
440  Rate_blocks=Rate_blocks*Rate/8
450  Syll_start=INT(Rate*Syll_start)          ! Convert from a time value to
460  Syll_fin=INT(Rate*Syll_fin)              ! a number of samples (ie. the
470  Syll_inc=INT(Rate*Syll_inc)              ! no. of samples in the syllabic
480                                           ! period).
490  RETURN
500  !*************************************************************************
510  !
520 Keyboard_input: !
530  ON KBD GOTO Get_response
540  !
550 Waiting: !
560  GOTO Waiting
570  !
580 Get_response: !
590  ON ERROR GOTO Waiting
600  Value=VAL(KBD$)
610  OFF KBD
620  OFF ERROR
630  !
640  RETURN
650  !*************************************************************************
660  END
```


E
SET SYSTEM PARAMETERS
OPEN SOURCE
PRINT RESULT HEADING
RESET SOURCE FILE POINTERS
PROCESS DATA
FILTER ?
Y
DIGITALLY FILTER OUTPUT
N
EXTRACT NOISE
GRANULAR & OVERLOAD NOISE ?
Y
EXTRACT 2 NOISE SIGNALS
N
CALCULATE SNR's
PRINT RESULTS
PROCESS WITH NEW PARAMETERS ?
Y
N
CLOSE SOURCE FILE
A

```
!-------------------------------------------------------------------------!
!            CODEC SIMULATION PROGRAMS FOR OBJECTIVE ANALYSIS             !
!            ================================================             !
!                                                                         !
! CODSIM          : Codec Simulation Package.                             !
! M.Sc Thesis     : The use of Delta Modulation for Low Bit-Rate Digital  !
!                   Speech Coding.                                        !
! Author          : J.M.Irvine.                                           !
!                   Department of Electrical and Electronic Engineering   !
!                   University of Cape Town.                              !
! Unit Number     : 4.                                                    !
! Latest Update   : 3rd December, 1984.                                   !
!-------------------------------------------------------------------------!
!                                                                         !
! Programs developed for the Hewlett Packard 200 Series desk-top          !
! computers.  All development work done on an HP9836C computer with the   !
! interfacing between the analogue and digital environments done using    !
! the HP6942A Multiprogrammer.                                            !
!-------------------------------------------------------------------------!
!                                                                         !
! This program processes the speech data, stored on disc, using the       !
! simulated delta modulators (ie. the LDM, CVSD, CFDM, SVADM, SHCDM,      !
! MSHCDM. and HCDM coders).  Once the data has been procesed by the       !
! relevant codec the output is filtered (if required) by a digital        !
! filter and the noise signals extracted.  The signal-to-noise ratio      !
! (SQNR) and the segmental signal-to-noise ratio (SNRSEG) are calcul-     !
! ated and, if specified, the respective granular and segmental values    !
! are evaluated.                                                          !
!-------------------------------------------------------------------------!
```

```
!<<<<<<<<<<<<<<< DEFINE THE VARIABLES USED BY THE PROGRAM >>>>>>>>>>>>>>>
!                ----------------------------------------
!
OPTION BASE 1
INTEGER Syllable,Err1,Err2,Err3,So                    ! The codec parameters -
REAL Alpha,Beta,Step,New_step                         ! step sizes, leakage
REAL Big_alpha,Signal,New_signal                      ! control variables, etc.
REAL Max_step,Max_cf,Max_sig,Min_sig
!
INTEGER Position,Processor_no,So_start,So_fin         ! Simulation control
INTEGER Syll_start,Syll_fin,Proc_start,Proc_fin       ! variables - sampling
INTEGER So_inc,Syll_inc,Proc_inc                      ! rate, codec type, etc.
REAL Rate,Alpha_start,Alpha_fin,Beta_start
REAL Beta_fin,Alpha_inc,Beta_inc
!
REAL Signal_pwr,Seg_signal_pwr,Total_noise            ! Variables for calcul-
REAL Segmental_noise,Total_gran,Segmental_gran        ! ating the various
REAL Total_over,Segmental_over                        ! signal-to-noise ratios.
INTEGER Sig_cnt,Seg_sig_cnt,Noise_cnt
INTEGER Seg_noise_cnt,Gran_cnt,Seg_gran_cnt
INTEGER Over_cnt,Seg_over_cnt
!
DIM In_file$[8],File_name$[12]
DIM Function$[40],Error$[40]
!
ALLOCATE INTEGER Speech(1:10283)                      ! The buffer for the input speech.
ALLOCATE REAL Voice(1:10283)                          ! Codecs operate on real values.
ALLOCATE REAL Voice2(1:10283)
ALLOCATE REAL Syllab(1:1000)                          ! Syllabic companding filter.
ALLOCATE REAL Window(1:60)                            ! The window length for filtering.
!
ALLOCATE Seg_sig(1:515)                               ! Array for segmental calculations.
ALLOCATE Seg_noise(1:515),Seg_gran(1:515),Seg_over(1:515)
!
Run$=" R"                    ! Control character to start program execution.
Stop$=" !"                   ! Control character to stop program execution.
!*************************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<<<<<<<<<< M A I N L I N E >>>>>>>>>>>>>>>>>>>>>>>>>>>
!                            ===============
!
GOSUB Initialise             ! Reset the mode of the controller.
GOSUB Set_parameters         ! Set the parameters controlling the codec
                             ! type and the parameter ranges.
GOSUB Open_file              ! Open the file containing the speech data.
!
GOSUB Print_heading          ! Print a heading for the results.
!
FOR Processor_no=Proc_start TO Proc_fin STEP Proc_inc
  !
  ! It is possible to process the speech data using a range of different
  ! parameter settings.  This enables the effects of the parameters to be
  ! evaluated subjectively.
  !-------------------------------------------------------------------
  !
  FOR Alpha=Alpha_start TO Alpha_fin STEP Alpha_inc
    FOR Beta=Beta_start TO Beta_fin STEP Beta_inc
      FOR So=So_start TO So_fin STEP So_inc
        FOR Syllable=Syll_start TO Syll_fin STEP Syll_inc
          FOR Scale=Scale_start TO Scale_fin STEP Scale_inc
            !
            RESET @Source          ! Reset the source file to start.
            !
            MAT Syllab= (0)        ! Clear the syllabic filter.
            MAT Window= (0)        ! Clear the digital output filter.
            !
            GOSUB Print_message    ! Output a status message to VDU.
            FOR Blocks=1 TO Rate_blocks
              !
              DISP "Processing ";Blocks;" out of ";Rate_blocks;" of ";Rate
 kHz data."
              GOSUB Get_input      ! Read the input speech data off disc.
              MAT Voice= Speech    ! Copy from Integer to Real.
              !
              ! Amplify the speech signal.
              !
              MAT Voice= (10^Scale)*Voice
              !
              !
              ! Update the lumped and segmental signal powers.
              !-----------------------------------------------
              !
              Noises(Signal_sgn,Sig_cnt,0,0,Seg_sig_cnt,Samples_per_seg,Si
al_pwr,Seg_signal_pwr,Seg_sig(*),Voice(*))
              !
              !
              ! The procedure "Codecs", which performs the simulation,
              ! consists of a compiled PASCAL module.  The listing of
              ! this module is contained in Unit no. 6.
              !-------------------------------------------------------
              !
              Codecs(So,Err1,Pos,Err2,Err3,Processor_no,Syllable,Beta,Rate
alpha,Signal,Max_cf,Max_sig,Min_sig,New_step,Max_step,Big_alpha,New_signal,Syll
              !
              IF Filter_flag=1 THEN
                Filter(Rate,Window(*),Voice(*))
              END IF
```

```
1240                ! Extract the noise signal.
1250                !--------------------------
1260                !
1270                MAT Voice= Voice-Speech
1280                MAT Voice2= Voice
1290                !
1300                !
1310                ! Update the lumped and segmental noise powers.
1320                !----------------------------------------------
1330                !
1340                Noises(Noise_sgn,Noise_cnt,0,0,Seg_noise_cnt,Samples_per_seg
,Total_noise,Segmental_noise,Seg_noise(*),Voice(*))
1350                !
1360                !
1370                ! Update the lumped and segmental granular noise powers.
1380                !------------------------------------------------------
1390                !
1400                IF Gran_flag=1 THEN
1410                  Noises(Noise_sgn,Gran_cnt,1,0,Seg_gran_cnt,Samples_per_seg
,Total_gran,Segmental_gran,Seg_gran(*),Voice(*))
1420                !
1430                !
1440                ! Update the lumped and segmental overload noise powers.
1450                !------------------------------------------------------
1460                !
1470                  MAT Voice= Voice2
1480                  Noises(Noise_sgn,Over_cnt,0,1,Seg_over_cnt,Samples_per_seg
,Total_over,Segmental_over,Seg_over(*),Voice(*))
1490                END IF
1500                !
1510                GOSUB Ratios
1520                !
1530              NEXT Blocks
1540                !
1550              PRINT USING "#,D,2X,D.DD,2X,D.DD,2X,DD.D,2X,6D,4X,D.D,DD,3X";Pro
cessor_no,Alpha,Beta,INT(Syllable/Rate),So,Max_step,Scale
1560              PRINT USING "DD.D,3X,DD.D,2X,DD.D,3X,DD.D,2X,DD.D,3X,DD.D";Sgn
r,Snr_seg,Sng,Sng_seg,Sno,Sno_seg
1570                !
1580              ASSIGN @Destination TO *! Close the file containing output.
1590              !
1600            NEXT Scale
1610          NEXT Syllable
1620        NEXT So
1630      NEXT Beta
1640    NEXT Alpha
1650  NEXT Processor_no
1660  !
1670  ASSIGN @Source TO *                 ! Close the source file.
1680  GOSUB Clear_disp                    ! Clear the dsplay screen.
1690  !
1700  ! Return the user to the main menu of the simulation system.
1710  ! ----------------------------------------------------------
1720  !
1730 Finish: !
1740  Function$="LOAD ""MASTER:INTERNAL,4,0"" X"
1750  OUTPUT KBD;Stop$;Function$;Run$;
1760  !*************************************************************************
```

```
770  !<<<<<<<<<<<<<<<<<<<<<<<< S U B R O U T I N E S >>>>>>>>>>>>>>>>>>>>>>>>
780  !                         ====================
790  !
800  !<<<<<<<<<<<<<<<< SPECIFY THE DEFAULT MASS STORAGE DEVICES >>>>>>>>>>>>>>>>
810  !                 ---------------------------------------
820  !
830 Initialise: !
840  Drive_0$=":INTERNAL,4,0"                ! Set the default mass storage
850  Drive_1$=":INTERNAL,4,1"                ! devices.
860  PRINTER IS CRT                          ! Use the VDU for displays.
870  CONTROL 1,12;1                          ! Turn the key label display off.
880  !
890  ! Initialise the values of the simulation system control parameters.
900  !-------------------------------------------------------------------
910  !
920  In_file$="TEST2"
930  Out_file$="TEST2"
940  !
950  Rate=16
960  Rate_blocks=8
970  Proc_start=1
980  Proc_fin=1
990  Proc_inc=1
1000 So_start=1
1010 So_fin=1
1020 So_inc=1
1030 Alpha_start=.25
1040 Alpha_fin=.25
1050 Alpha_inc=.01
1060 Beta_start=1.53
1070 Beta_fin=1.53
1080 Beta_inc=.01
1090 Syll_start=5
1100 Syll_fin=5
1110 Syll_inc=1
1120 !
1130 Scale_start=0
1140 Scale_fin=0
1150 Scale_inc=1
1160 !
1170 Max_step=1000
1180 Max_cf=1.5^17
1190 Min_sig=-32768
1200 Max_sig=+32767
1210 !
1220 Filter_flag=1
1230 Gran_flag=1
1240 !
```

```
2250  !<<<<<<<<<<<< CLEAR THE ALPHANUMERIC AND GRAPHICS DISPLAYS >>>>>>>>>>>>
2260  !              ------------------------------------------
2270  !
2280 Clear_disp:!
2290   GCLEAR                                  ! Clear the graphics display.
2300   CONTROL 1,12;1                          ! Turn the key label display off.
2310   ALPHA OFF
2320   PRINT USING "@"                         ! Clear the Alpha display.
2330   PRINT CHR$(128):CHR$(140);              ! Select normal video with
2340   ALPHA ON                                ! Colour = Cyan.
2350   DISP " "
2360   RETURN
2370 !*************************************************************************
2380 !
2390 !<<<<<<<<<<<<<< OPEN THE FILE CONTAINING THE SOURCE DATA >>>>>>>>>>>>>>
2400 !                ---------------------------------------
2410 !
2420 Open_file:!
2430   Drive$=Drive_0$
2440   MASS STORAGE IS Drive$
2450   File_name$=In_file$&"_P"
2460   ASSIGN @Source TO File_name$
2470   RETURN
2480 !*************************************************************************
2490 !
2500 !<<<<<<<<<<<<<< READ THE DATA FROM THE FILES STORED ON DISC >>>>>>>>>>>>>>
2510 !               ---------------------------------------
2520 !
2530 Get_input:!
2540   DISP "Reading speech data off disc."
2550   ENTER @Source;Speech(*)
2560   DISP
2570   RETURN
2580 !*************************************************************************
```

```
2590 !<<<<<<<<<<<<<<<< PRINT A MESSAGE ON THE VDU FOR THE USER >>>>>>>>>>>>>>>>
2600 !                  ------------------------------------
2610 !
2620 Print_message:!
2630  GOSUB Clear_disp
2640  PRINT "Processing the speech data contained in ":In_file$
2650  PRINT "Processing using a ";
2660  SELECT Processor_no
2670  CASE 1
2680    PRINT "Linear Delta Modulator";
2690  CASE 2
2700    PRINT "Continuously Variable Slope Delta Modulator";
2710  CASE 3
2720    PRINT "Constant Factor Delta Modulator";
2730  CASE 4
2740    PRINT "Song Voice Adaptive Delta Modulator";
2750  CASE 5
2760    PRINT "Song Hybrid Companding Delta Modulator";
2770  CASE 6
2780    PRINT "Modified Song Hybrid Companding Delta Modulator";
2790  CASE 7
2800    PRINT "Hybrid Companding Delta Modulator";
2810  END SELECT
2820  PRINT
2830  IF Filter_flag=1 THEN
2840    PRINT "Digital filtering > YES."
2850  ELSE
2860    PRINT "Digital filtering > NO."
2870  END IF
2880  PRINT
2890  RETURN
2900 !*****************************************************************************
2910 !
2920 !<<<<<<<<<<<<<<<<<<<< PRINT A HEADING FOR THE RESULTS >>>>>>>>>>>>>>>>>>>>
2930 !                     -------------------------------
2940 !
2950 Print_heading:!
2960  PRINTER IS PRT
2970  PRINT "CODSIM : Objective evaluation of codec performance."
2980  PRINT "Filtering : ";
2990  IF Filter_flag=1 THEN
3000    PRINT "Y    ";
3010  ELSE
3020    PRINT "N    ";
3030  END IF
3040  PRINT "  Granular and overload SNR's : ";
3050  IF Gran_flag=1 THEN
3060    PRINT "Y"
3070  ELSE
3080    PRINT "N"
3090  END IF
3100  PRINT "P  Alph  Beta  Syll  So  Maxstp   Scale   SQNR  SNRSEG  SNG  SNGSEG
  SNO  SNOSEG"
3110  FOR I=1 TO 80
3120    PRINT "-";
3130  NEXT I
3140  PRINT
3150  PRINTER IS CRT
3160  RETURN
```

```
!<<<<<<<<<<<<<<< CALCULATE THE RESPECTIVE SEGMENTAL VALUES >>>>>>>>>>>>>>>
!                ----------------------------------------
!
Segmental_snr:!
 !
 Snr_seg=0
 Sng_seg=0
 Sno_seg=0
 !
 FOR I=1 TO 515
   IF Segmental_noise(I)<>0 THEN
     Snr_seg=Snr_seg+10*LGT(Seg_signal_pwr(I)/Segmental_noise(I))
   END IF
   IF Gran_flag=1 THEN
     IF Segmental_gran(I)<>0 THEN
       Sng_seg=Sng_seg+10*LGT(Seg_signal_pwr(I)/Segmental_gran(I))
     END IF
     IF Segmental_over(I)<>0 THEN
       Sno_seg=Sno_seg+10*LGT(Seg_signal_pwr(I)/Segmental_over(I))
     END IF
   END IF
 NEXT I
 !
 Snr_seg=Snr_seg/515
 Sng_seg=Sng_seg/515
 Sno_seg=Sno_seg/515
 !
 MAT Seg_signal_pwr= (0)
 MAT Segmental_noise= (0)
 MAT Segmental_gran= (0)
 MAT Segmental_over= (0)
 !
 RETURN
!*************************************************************************
```

```
4050 !<<<<<<<<<<<<<<< SET THE CONTROL PARAMETERS OF THE SYSTEM >>>>>>>>>>>>>>>
4060 !                 ---------------------------------------
4070 !
4080 Set_parameters:!
4090  !
4100  ! This subroutine enables all the variable and parameter values to be set
4110  ! First the codecs that are to be used are enabled and then the variables
4120  ! relating to each are set.  Also, the source and destination files are
4130  ! specified, as well as the data rate.
4140  !-------------------------------------------------------------------------
4150  !
4160  !<<<<<<<<<<<<<<<<<<<< DISPLAY THE CODEC-TYPE MENU >>>>>>>>>>>>>>>>>>>>
4170  !                      --------------------------
4180  !
4190  PRINT TAB(18);CHR$(132);"Processor Menu : Select an option";CHR$(128)
4200  PRINT TAB(26);CHR$(132);"Codec Type";CHR$(128)
4210  PRINT TABXY(26,5);"1)  LDM"
4220  PRINT TAB(26);"2)  CVSD"
4230  PRINT TAB(26);"3)  CFDM"
4240  PRINT TAB(26);"4)  SVADM"
4250  PRINT TAB(26);"5)  SHCDM"
4260  PRINT TAB(26);"6)  MSHCDM"
4270  PRINT TAB(26);"7)  HCDM"
4280  PRINT CHR$(10)
4290  PRINT TAB(18);CHR$(132);"Start";CHR$(128);TAB(31);CHR$(132);"Stop";CHR$(12
8);TAB(43);CHR$(132);"Increment";CHR$(128)
4300  !
4310  !-------------------------------------------------------------------------
4320  !
4330 Alter_values:                          ! Enter the required codec types.
4340  !
4350  PRINT TABXY(19,15);CHR$(129);Proc_start;CHR$(128);TAB(32);Proc_fin;TAB(44)
;Proc_inc
4360  PRINT CHR$(10)
4370  PRINT "A codec type '0' returns to main menu."
4380  DISP CHR$(140);"Enter the desired value"
4390  !
4400  GOSUB Keyboard_input
4410  IF Value<>9 THEN
4420    Proc_start=Value
4430  END IF
4440  !
4450  PRINT TABXY(19,15);Proc_start;TAB(30);CHR$(129);Proc_fin;CHR$(128);TAB(44)
;Proc_inc
4460  GOSUB Keyboard_input
4470  IF Value<>9 THEN
4480    Proc_fin=Value
4490  END IF
4500  !
4510  PRINT TABXY(19,15);Proc_start;TAB(30);Proc_fin;TAB(42);CHR$(129);Proc_inc;
CHR$(128)
4520  Increment=0
4530  INPUT "Enter the desired value",Increment
4540  IF Increment<>0 THEN
4550    Proc_inc=Increment
4560    Increment=0
4570  END IF
4580  !
```

```
4590  PRINT TABXY(19,15);Proc_start;TAB(30);Proc_fin;TAB(42);Proc_inc
4600  DISP "Do you wish to keep these values (Y/N) ?"
4610  !
4620  ON KBD GOTO Answer
4630  !
4640 Wait:!
4650  GOTO Wait
4660  !
4670 Answer: !
4680  R$=UPC$(KBD$)
4690  IF (R$="Y") OR (R$=" E") THEN
4700    IF (Proc_start=0) OR (Proc_fin=0) THEN    ! A value of zero for a codec
4710      GOTO Finish                             ! type returns the user to
4720    END IF                                    ! the main menu.
4730  ELSE
4740    GOTO Alter_values
4750  END IF
4760  !
4770  !-------------------------------------------------------------------------
4780  !
4790  ! Update the various design parameters of the different codecs.
4800  !
4810  GOSUB Clear_disp
4820  !
4830 Design_param: !
4840  !
4850  PRINT TABXY(18,2);CHR$(129);"Codec algorithm parameters :";CHR$(128);CHR$(
10)
4860  PRINT TAB(21);CHR$(132);"Minimum step size (So)";CHR$(128);CHR$(10)
4870  PRINT TAB(18);CHR$(132);"Start";CHR$(128);TAB(30);CHR$(132);"Stop";CHR$(12
8);TAB(41);CHR$(132);"Increment";CHR$(128)
4880  PRINT TAB(19);CHR$(129);So_start;CHR$(128);TAB(31);So_fin;TAB(42);So_inc
4890  DISP "Enter the desired value"
4900  GOSUB Keyboard_input
4910  IF Value>0 THEN
4920    So_start=Value
4930  END IF
4940  PRINT TABXY(19,7);So_start;TAB(29);CHR$(129);So_fin;CHR$(128);TAB(42);So_i
nc
4950  GOSUB Keyboard_input
4960  IF Value>0 THEN
4970    So_fin=Value
4980  END IF
4990  PRINT TABXY(19,7);So_start;TAB(29);So_fin;TAB(40);CHR$(129);So_inc;CHR$(12
8)
5000  INPUT "Enter the desired value",Increment
5010  IF Increment<>0 THEN
5020    So_inc=Increment
5030    Increment=0
5040  END IF
5050  PRINT TABXY(19,7);So_start;TAB(29);So_fin;TAB(40);So_inc
5060  !
5070  !-------------------------------------------------------------------------
```

```
5080  PRINT TABXY(18,12);CHR$(132);"Leakage time constant (Beta)";CHR$(128);CHR$
(10)
5090  PRINT TAB(18);CHR$(132);"Start";CHR$(128);TAB(30);CHR$(132);"Stop";CHR$(12
8);TAB(41);CHR$(132);"Increment";CHR$(128)
5100  PRINT TABXY(17,15);CHR$(129);Beta_start;CHR$(128);TAB(29);Beta_fin;TAB(41)
;Beta_inc
5110  Value=0
5120  INPUT "Enter the desired value",Value
5130  IF Value<>0 THEN
5140    Beta_start=Value
5150    Value=0
5160  END IF
5170  PRINT TABXY(17,15);Beta_start;TAB(27);CHR$(129);Beta_fin;CHR$(128);TAB(41)
;Beta_inc
5180  INPUT "Enter the desired value",Value
5190  IF Value<>0 THEN
5200    Beta_fin=Value
5210    Value=0
5220  END IF
5230  PRINT TABXY(17,15);Beta_start;TAB(27);Beta_fin;TAB(39);CHR$(129);Beta_inc;
CHR$(128)
5240  INPUT "Enter the desired value",Value
5250  IF Value<>0 THEN
5260    Beta_inc=Value
5270    Value=0
5280  END IF
5290  PRINT TABXY(17,15);Beta_start;TAB(27);Beta_fin;TAB(39);Beta_inc
5300  DISP "Do you wish to keep these values (Y/N) ?"
5310  !
5320  ON KBD GOTO Answer_2
5330  !
5340 Wait_2:!
5350  GOTO Wait_2
5360  !
5370 Answer_2: !
5380  R$=UPC$(KBD$)
5390  IF R$="N" THEN
5400    GOTO Design_param
5410  END IF
5420  !
5430  !-----------------------------------------------------------------------
```

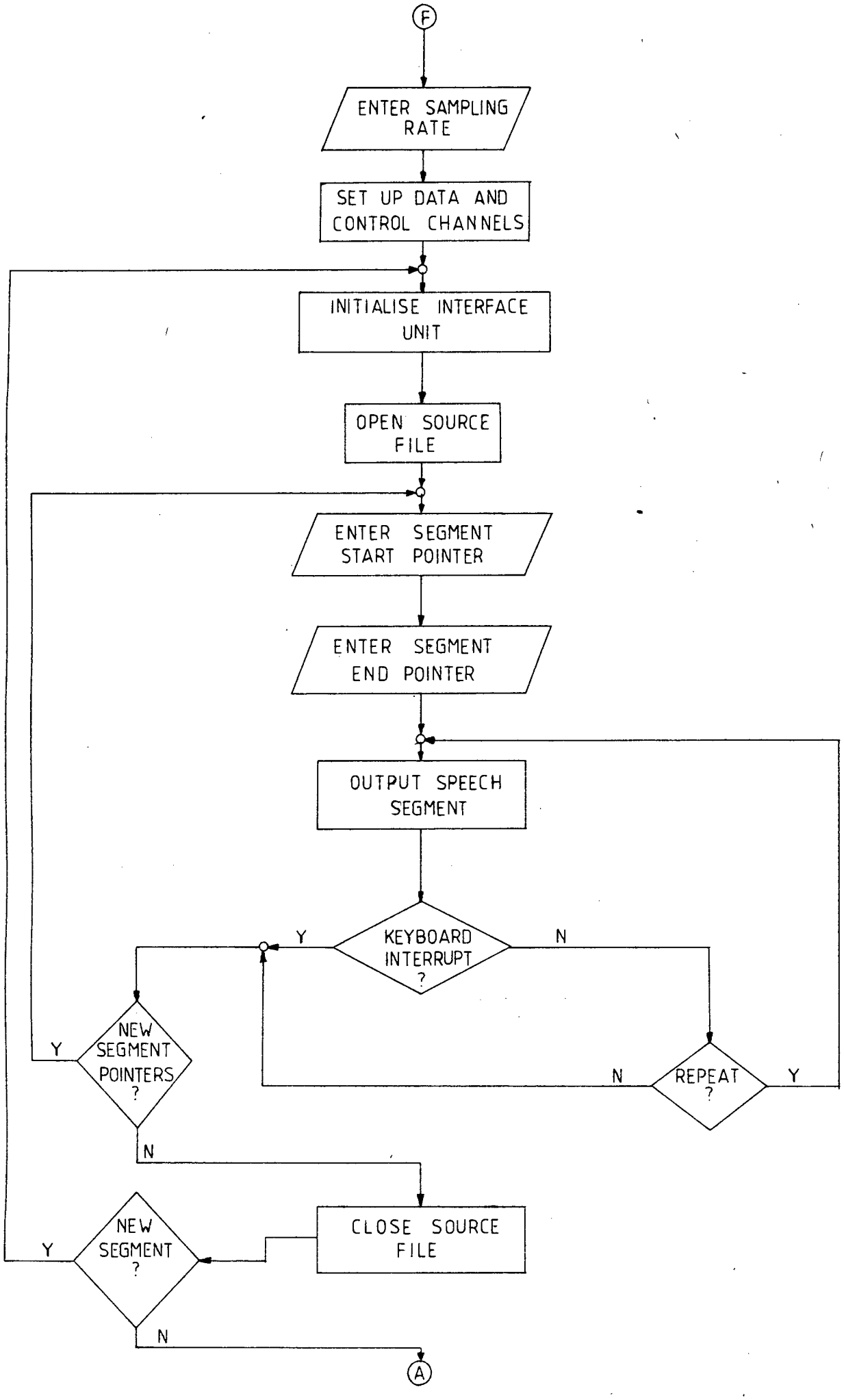
```
5440  ! The parameters for the hybrid codecs are only updated if the range of
5450  ! of the codec types, chosen previously, includes one or more of the
5460  ! hybrid codecs.
5470  !
5480  IF (Proc_start<=4) AND (Proc_fin<=4) THEN
5490    GOTO General_param
5500  END IF
5510  !
5520  GOSUB Clear_disp
5530  !
5540 Hybrid_param: !
5550  !
5560  PRINT TABXY(23,2);CHR$(129):"Hybrid Design Parameters :";CHR$(128);CHR$(10

5570  PRINT TAB(18);CHR$(132);"Syllabic Companding Constant (Alpha)";CHR$(128);C
HR$(10)
5580  PRINT TAB(23);CHR$(132);"Start";CHR$(128);TAB(35);CHR$(132);"Stop";CHR$(12
8);TAB(46);CHR$(132);"Increment";CHR$(128)
5590  PRINT TAB(23);CHR$(129);Alpha_start;CHR$(128);TAB(35);Alpha_fin;TAB(46);Al
pha_inc
5600  Value=0
5610  INPUT "Enter the desired value",Value
5620  IF Value<>0 THEN
5630    Alpha_start=Value
5640    Value=0
5650  END IF
5660  PRINT TABXY(23,7);Alpha_start;TAB(33);CHR$(129);Alpha_fin;CHR$(128);TAB(46
);Alpha_inc
5670  INPUT "Enter the desired value",Value
5680  IF Value<>0 THEN
5690    Alpha_fin=Value
5700    Value=0
5710  END IF
5720  PRINT TABXY(23,7);Alpha_start;TAB(33);Alpha_fin;TAB(44);CHR$(129);Alpha_in
c;CHR$(128)
5730  INPUT "Enter the desired value",Value
5740  IF Value<>0 THEN
5750    Alpha_inc=Value
5760    Value=0
5770  END IF
5780  PRINT TABXY(23,7);Alpha_start;TAB(33);Alpha_fin;TAB(44);Alpha_inc;CHR$(32)
5790  !
5800  !-------------------------------------------------------------------------
```

```
5810  PRINT TABXY(24,12);CHR$(132);"Syllabic Companding Period";CHR$(128);CHR$(1
0)
5820  PRINT TAB(23);CHR$(132);"Start";CHR$(128);TAB(35);CHR$(132);"Stop";CHR$(12
3);TAB(46);CHR$(132);"Increment";CHR$(128)
5830  PRINT TABXY(23,15);CHR$(129);Syll_start;CHR$(128);TAB(35);Syll_fin;TAB(47)
;Syll_inc
5840  Value=0
5850  INPUT "Enter the desired value (milliseconds)",Value
5860  IF Value<>0 THEN
5870    Syll_start=Value
5880    Value=0
5890  END IF
5900  PRINT TABXY(23,15);Syll_start;TAB(33);CHR$(129);Syll_fin;CHR$(128);TAB(47)
Syll_inc
5910  INPUT "Enter the desired value (milliseconds)",Value
5920  IF Value<>0 THEN
5930    Syll_fin=Value
5940    Value=0
5950  END IF
5960  PRINT TABXY(23,15);Syll_start;TAB(33);Syll_fin;TAB(45);CHR$(129);Syll_inc;
CHR$(128)
5970  INPUT "Enter the desired value (milliseconds)",Value
5980  IF Value<>0 THEN
5990    Syll_inc=Value
6000    Value=0
6010  END IF
6020  PRINT TABXY(23,15);Syll_start;TAB(33);Syll_fin;TAB(45);Syll_inc
6030  DISP "Do you wish to keep these values (Y/N) ?"
6040  !
6050  ON KBD GOTO Answer_3
6060  !
6070 Wait_3:!
6080  GOTO Wait_3
6090  !
6100 Answer_3: !
6110  R$=UPC$(KBD$)
6120  IF R$="N" THEN
6130    GOTO Hybrid_param
6140  END IF
6150  !
6160  !-------------------------------------------------------------------------
```

```
6170  ! The general parameters are updated at this stage - eg. Maximum step
6180  ! sizes, maximum output signal ranges, source file names, destination
6190  ! file names, etc.
6200  !
6210 General_param: !
6220  !
6230  GOSUB Clear_disp
6240  !
6250  PRINT TAB(22);CHR$(132);"Speech file information :";CHR$(128);CHR$(10)
6260  PRINT TAB(20);"Source file name      :  ";CHR$(129):In_file$:CHR$(128);CHR
$(10)
6270  PRINT TAB(20);"Destination file name :  ";Out_file$;CHR$(10)
6280  PRINT TAB(20);"Maximum signal value  : ";Max_sig;CHR$(10)
6290  PRINT TAB(20);"Minimum signal value  : ";Min_sig;CHR$(10)
6300  PRINT TAB(20);"Maximum step size     : ";Max_step;CHR$(10)
6310  PRINT TAB(20);"Sampling rate ( kHz ) : ";Rate;CHR$(10)
6320  PRINT TAB(10);CHR$(129);" The default setting is selected by pressing 'ENT
R'";CHR$(128)
6330  INPUT "Enter the source file name",In_file$
6340  In_file$=UPC$(In_file$)
6350  PRINT TABXY(20,4);"Source file name      :  ";In_file$;"          ";CHR$(10)
6360  PRINT TAB(20);"Destination file name :  ";CHR$(129);Out_file$;CHR$(128)
6370  !
6380  INPUT "Enter the destination file name",Out_file$
6390  Out_file$=UPC$(Out_file$)
6400  PRINT TABXY(20,6);"Destination file name :  ";Out_file$;"          ";CHR$(10
)
6410  PRINT TAB(20);"Maximum signal value  : ";CHR$(129);Max_sig;CHR$(128)
6420  !
6430  Value=0
6440  INPUT "Enter the desired value",Value
6450  IF Value>0 THEN
6460    Max_sig=Value
6470    Value=0
6480  END IF
6490  PRINT TABXY(20,8);"Maximum signal value  : ";Max_sig;"          ";CHR$(10)
6500  !
6510  PRINT TAB(20);"Minimum signal value  : ";CHR$(129);Min_sig;CHR$(128)
6520  !
6530  INPUT "Enter the desired value",Value
6540  IF Value<>0 THEN
6550    Min_sig=Value
6560    Value=0
6570  END IF
6580  PRINT TABXY(20,10);"Minimum signal value  : ";Min_sig;"          ";CHR$(10)
6590  !
6600  PRINT TAB(20);"Maximum step size     : ";CHR$(129);Max_step;CHR$(128)
6610  !
6620  INPUT "Enter the desired value",Value
6630  IF Value>0 THEN
6640    Max_step=Value
6650    Value=0
6660  END IF
6670  PRINT TABXY(20,12);"Maximum step size     : ";Max_step;"          ";CHR$(10)
6680  !
```

```
7250  Rate_blocks=Rate_blocks*Rate/8
7260  Syll_start=INT(Rate*Syll_start)          ! Convert from a time value to
7270  Syll_fin=INT(Rate*Syll_fin)              ! a number of samples (ie. the
7280  Syll_inc=INT(Rate*Syll_inc)              ! no. of samples in the syllabic
7290                                           ! period).
7300  Samples_per_seg=Rate*20                  ! The no. of samples in 20 msec.
7310  RETURN
7320  !*****************************************************************************
7330  !
7340 Keyboard_input: !
7350  ON KBD GOTO Get_response
7360  !
7370 Waiting: !
7380  GOTO Waiting
7390  !
7400 Get_response: !
7410  ON ERROR GOTO Waiting
7420  Value=VAL(KBD$)
7430  OFF KBD
7440  OFF ERROR
7450  !
7460  RETURN
7470  !*****************************************************************************
7480  END
```


F
ENTER SAMPLING RATE
SET UP DATA AND CONTROL CHANNELS
INITIALISE INTERFACE UNIT
OPEN SOURCE FILE
ENTER SEGMENT START POINTER
ENTER SEGMENT END POINTER
OUTPUT SPEECH SEGMENT
KEYBOARD INTERRUPT ?
Y
N
NEW SEGMENT POINTERS ?
Y
N
REPEAT ?
N
Y
NEW SEGMENT ?
Y
N
CLOSE SOURCE FILE
A

```
!-------------------------------------------------------------------------
!          BUFFERED D/A PROGRAM FOR SUBJECTIVE SIGNAL ANALYSIS
!          ===================================================
!
! CODSIM         : Codec Simulation Package.
! M.Sc Thesis    : The use of Delta Modulation for Low Bit-Rate Digital
!                  Speech Coding.
! Author         : J.M. Irvine.
!                  Department of Electrical and Electronic Engineering
!                  University of Cape Town.
! Unit Number    : 5.
! Latest Update  : 2nd December, 1984.
!_________________________________________________________________________
!
! Programs developed for the Hewlett Packard 2000 Series desk-top
! computers.  All development work done on an HP9836C computer with the
! interfacing between the analogue and digital environments done using
! the HP6942A Multiprogrammer.
!_________________________________________________________________________
!
! This program reads the data stored on the discs and outputs it, via a
! memory buffer, to the D/A card in the Multiprogrammer. Once the data
! has been transferred from the mass storage device it is transferred
! to the memory cards in the multiprogrammer and then clocked out to
! the D/A card.  The speech segment can be output once or repeatedly
! and it can be output in shorter segments which can form the whole or
! part of the original segment. Their are two possible output data
! rates : 8 kHz for the original sampled data, and 16 kHz for the pro-
! cessed or interpolated versions of the sampled data.
!_________________________________________________________________________
!
!
!<<<<<<<<<<<<<< DEFINE THE VARIABLES USED BY THE PROGRAM >>>>>>>>>>>>>>
!               ---------------------------------------
!
OPTION BASE 1
INTEGER Card,Mistake,A_d_slot,D_a_slot                ! Multiprogrammer card
INTEGER Mem_in_slot,Tb_slot                           ! variables.
INTEGER Complete,No_of_errors,No_of_cards,Slot_no
INTEGER Card_variable
!
REAL Intr_svc_delay,Transfer_oh,Total_delay           ! System performance
REAL Transfer_rate,Reinit_time,Head_room              ! parameters.
REAL Block_size,Reference_word,Time_available
REAL Transfer_time,Additional_rdgs,Trigger_rate
!
DIM Function$[40],Error$[40],Mw_command$[30],File_name$[10]
ALLOCATE INTEGER Configuration(0:16,5)
!
Run$=" R"                    ! Control character to start program execution.
Stop$=" !"                   ! Control character to stop program execution.
!*************************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<<<<<<<<<< M A I N L I N E >>>>>>>>>>>>>>>>>>>>>>>>>>>
!                              ==============
!
GOSUB Initialise            ! Reset the mode of the controller.
GOSUB Sample_rate           ! Set up the output data rate.
GOSUB Io_paths              ! Set up data and control paths.
Xfer(Op$)                   ! Load BINARY subprogram.
!
! Determine the multiprogrammer card configuration.
!
Multi_set_up(Configuration(*),Tb_slot,Mem_in_slot,D_a_slot)
!
GOSUB Clear_disp
!
GOSUB Re_init               ! Set the multiprogrammer cards in the
                            ! correct operating state.
!
!<<<<<<<<<<<<< THE SPEECH DATA IS READ OFF DISC AND OUTPUT >>>>>>>>>>>>>
!              ==========================================
!
Output_speech:  !
  GOSUB Read_data           ! Transfer the speech data from disc.
New_sections:!
  GOSUB Sections            ! Determine which portions to be listened to.
  ON KBD GOTO Response      ! Check for interrupt from keyboard.
  !
  DISP "To terminate the output hit any key on the keyboard."
  !
  GOSUB Output_data         ! Send the speech data to the multiprogrammer.
  !
Response:  !
  OFF KBD
  GOSUB Re_init
  DISP "Do you wish to output new sections of present segment ? (Y/N)"
  ON KBD GOTO Reply1
  !
Wait1:  !
  GOTO Wait1
  !
Reply1:  !
  R$=UPC$(KBD$)
  OFF KBD
  IF R$="N" THEN
    DISP "Do you wish to output a new speech segment ? ( Y/N )"
    ON KBD GOTO Reply2
    !
```

```
)0 Wait2:!
)00     GOTO Wait2
)10     !
)20 Reply2:!
)30     R$=UPC$(KBD$)
)40     OFF KBD
)50     IF R$="N" THEN
)60       Function$="LOAD ""MASTER:INTERNAL,4.0"" X"
)70       OUTPUT KBD;Stop$;Function$;Run$;
)80     ELSE
)90       GOTO Output_speech
!00     END IF
!10   ELSE
!20     GOTO New_sections
!30   END IF
!40   !*************************************************************************
```

```
150  !<<<<<<<<<<<<<<<<<<<<<<<< S U B R O U T I N E S >>>>>>>>>>>>>>>>>>>>>>>>
160  !                         ====================
170  !
180  !<<<<<<<<<<<<<<<< SPECIFY THE DEFAULT MASS STORAGE DEVICES >>>>>>>>>>>>>>>>
190  !                 ---------------------------------------
200  !
210 Initialise: !
220  Drive_0$=":INTERNAL,4,0"          ! Set the default mass storage devices.
230  Drive_1$=":INTERNAL,4,1"
240  PRINTER IS CRT                    ! Use the VDU for displays.
250  CONTROL 1,12;1                    ! Turn the key label display off.
260  No_of_buffers=14                  ! The number of arrays used to store
270                                    ! the sampled speech data.
280  !
290  !<<<<<<<<<<<<< CLEAR THE ALPHANUMERIC AND GRAPHICS DISPLAYS >>>>>>>>>>>>>>
300  !              ------------------------------------------
310  !
320 Clear_disp:!
330  GCLEAR                                 ! Clear the graphics display.
340  ALPHA OFF
350  PRINT USING "@"                        ! Clear the Alpha display.
360  PRINT CHR$(128);CHR$(140);             ! Select normal video with
370  ALPHA ON                               ! Colour = Cyan.
380  DISP " "
390  RETURN
400 !*****************************************************************************
410 !
420 !<<<<<<<<<<<<<<<<<<<< DETERMINE THE OUTPUT DATA RATE >>>>>>>>>>>>>>>>>>>>
430 !                     ------------------------------
440 !
450 Sample_rate:!
460  Op$="MW"
470  PRINT CHR$(129);TABXY(0,20);
480  PRINT "Possible output data rates are 8 kHz and 16 kHz.";CHR$(128)
490  BEEP 2500,.1
500 Wrong:INPUT "Enter the required data rate ( kHz ) >",Sample_rate
510  IF (Sample_rate<>8) AND (Sample_rate<>16) THEN
520    Error$="INVALID ENTRY -- PLEASE RE-ENTER"
530    GOTO Wrong
540  ELSE
550    PRINT TABXY(0,20);"                                                    "
560  END IF
570 !
```

```
!<<<<<<<<<<<<<<< CALCULATE MEMORY AND TIMING REQUIREMENTS >>>>>>>>>>>>>>>
!                ---------------------------------------
!
Bench_marks:!
 Trigger_rate=1/(Sample_rate*1.E+3)     ! The sampling period.
 Block_size=10283                       ! Size of the original data blocks.
 Blocksize=8*2*Block_size/14            ! Size of sub-segments of speech.
 Total_delay=.0229+.0013
 Head_room=Total_delay/Trigger_rate
 Reference_word=INT(4085-Head_room)
!
!<<<<<<<<<<<<<<<<<<<<< MEMORY SPACE IS NOW RESERVED >>>>>>>>>>>>>>>>>>>>>
!                      ---------------------------
!
! A total of 14 buffers, each representing approximately 0.8 seconds of
! speech, are used to store the sampled output speech signal.
!
 ALLOCATE INTEGER Buf1(Blocksize),Buf2(Blocksize),Buf3(Blocksize)
 ALLOCATE INTEGER Buf4(Blocksize),Buf5(Blocksize),Buf6(Blocksize)
 ALLOCATE INTEGER Buf7(Blocksize),Buf8(Blocksize),Buf9(Blocksize)
 ALLOCATE INTEGER Buf10(Blocksize),Buf11(Blocksize),Buf12(Blocksize)
 ALLOCATE INTEGER Buf13(Blocksize),Buf14(Blocksize)
 RETURN
!*****************************************************************************
!
!<<<<<<<<<<<<<<<<<<< SET UP THE DATA TRANSFER PATHS >>>>>>>>>>>>>>>>>>>
!
Io_paths:!
 ASSIGN @Multi TO 823                           ! Multiprogrammer address.
 ASSIGN @Mw_readout TO 823;FORMAT OFF           ! Buffer to Memory path.
 ASSIGN @Intr_list TO 82312                     ! Interrupt list.
 ASSIGN @Intr_cause TO 82310                    ! Interrupt cause.
 RETURN
!*****************************************************************************
```

```
!<<<<<<<<<<<<<<<<<<<<<< INITIALIZE THE SYSTEM CARDS >>>>>>>>>>>>>>>>>>>>>>
!                       --------------------------
!
Re_init:!
 OUTPUT @Multi;"CC";Tb_slot;"T"                    ! Clear Timer/Pacer card.
 OUTPUT @Multi;"WC";Tb_slot;0;"T"                  ! Turn off Timer/Pacer.
 OUTPUT @Multi;"WF";Tb_slot+.2;1;"T"               ! Continuous mode timer.
 OUTPUT @Multi;"CC";D_a_slot;"T"                   ! Clear the D/A card.
 OUTPUT @Multi;"SF";D_a_slot;"2,1,1T"              ! Program the D/A mode.
 OUTPUT @Multi;"CC";Mem_in_slot;"T"                ! Clear Memory card #1.
 OUTPUT @Multi;"CC";Mem_in_slot+1;"T"              ! Clear Memory card #2.
 OUTPUT @Multi;"WF";Mem_in_slot+1;0;"T"            ! Clear the REF REGISTER.
 OUTPUT @Multi;"WF";Mem_in_slot+.1;21;"T"          ! Disable Memory card #1.
 !
 ! SET THE TIME BASE CARD'S CLOCK FREQUENCY
 ! ----------------------------------------
 OUTPUT @Multi;"WF";Tb_slot;(1.E+3/Sample_rate)/2;"UT"
 !
 ! CYCLE THE TIME/BASE CARD
 ! ------------------------
 OUTPUT @Multi;"CY";Tb_slot;"T"
 !
 ! PROGRAM THE DIFFERENTIAL COUNTER OF THE MEMORY CARDS
 ! ----------------------------------------------------
 OUTPUT @Multi;"WF";Mem_in_slot+1;Reference_word;"T"
 !
 ! INSTRUCT MEMORY ON SIZE OF THE DATA BLOCK
 ! -----------------------------------------
 OUTPUT Mw_command$;"MW";Mem_in_slot;Blocksize;"T"
 !
 ! ENABLE THE BUFFERED DIGITAL TO ANALOGUE CONVERSION
 ! --------------------------------------------------
 OUTPUT @Multi;"WF";Mem_in_slot+.1;1;"T"
 !
 ! ARM THE MEMORY CARD
 ! -------------------
 OUTPUT @Multi;"AC";Mem_in_slot+1;"T"
 RETURN
!*************************************************************************************
```

```
310 !<<<<<<<<<<<<<< DETERMINE SECTION OF SPEECH TO BE OUTPUTTED >>>>>>>>>>>>>>
320 !                ------------------------------------------
330 !
340 Sections:!
350  Start=1
360  Finish=14
370  New_start=0
380  New_finish=0
390  Continuous$="Y"
400  New_continuous$="Y"
410  PRINT TABXY(3,5);CHR$(132);"Speech segments to be listened to:";CHR$(128)
420  PRINT TABXY(5,7);"Start          Finish          Continuous"
430  PRINT TABXY(6,9);CHR$(129);Start;CHR$(128);TAB(24);Finish;TAB(42);
440  IF Continuous$="Y" THEN
450    PRINT " YES "
460  ELSE
470    PRINT " NO  "
480  END IF
490  PRINT TABXY(0,21);"Enter the desired value or toggle value (ie. Y/N)."
500  !
510  INPUT New_start
520  IF (New_start>0) AND (New_start<15) THEN
530    Start=New_start
540  END IF
550  PRINT TABXY(6,9);Start;TAB(22);CHR$(129);Finish;CHR$(128);
560  IF Continuous$="Y" THEN
570    PRINT TAB(42);" YES "
580  ELSE
590    PRINT TAB(42);" NO  "
600  END IF
610  !
620  INPUT New_finish
630  IF (New_finish>=Start) AND (New_finish<15) THEN
640    Finish=New_finish
650  END IF
660  PRINT TABXY(6,9);Start;TAB(22);Finish;TAB(40);CHR$(129);
670  IF Continuous$="Y" THEN
680    PRINT " YES ";CHR$(128)
690  ELSE
700    PRINT " NO  ";CHR$(128)
710  END IF
720  !
```

```
730   INPUT New_continuous$
740   New_continuous$=UPC$(New_continuous$)
750   IF (New_continuous$="Y") OR (New_continuous$="N") THEN
760     Continuous$=New_continuous$
770   END IF
780   IF Continuous$="Y" THEN
790     PRINT TABXY(6,9);Start;TAB(22);Finish;TAB(40);" YES "
800   END IF
810   IF Continuous$="N" THEN
820     PRINT TABXY(6,9);Start;TAB(22);Finish;TAB(40);" NO  "
830   END IF
840   !
850   DISP "Do you wish to keep these values ?"
860   ON KBD GOTO Answer
870 Get_answer: !
880   GOTO Get_answer
890   !
900 Answer: !
910   R$=UPC$(KBD$)
920   OFF KBD
930   IF R$="N" THEN
940     GOTO Sections
950   END IF
960   PRINT TABXY(0,21);"                                                    "
970   RETURN
980 !*******************************************************************************
```

```
990 !<<<<<<<<<<<<<< READ THE DATA FROM THE FILES STORED ON DISC >>>>>>>>>>>>>>
000 !                 ------------------------------------------
010 !
020 Read_data:!
030  Drive$=Drive_1$
040  ON ERROR GOTO Wrong_disc
050 Assign_device: !
060  MASS STORAGE IS Drive$
070 Get_file:!
080  GOSUB File_name
090  ASSIGN @Test_file TO File_name$
100  DISP "Reading speech data off disc."
110  OFF ERROR
120  !
130  ENTER @Test_file;Buf1(*)          ! The speech data has been divided into
140  ENTER @Test_file;Buf2(*)          ! small sections, each about 0.8 sec in
150  ENTER @Test_file;Buf3(*)          ! length.  If the original version of the
160  ENTER @Test_file;Buf4(*)          ! sampled speech is to be listened to
170  ENTER @Test_file;Buf5(*)          ! then only 7 sections are needed,
180  ENTER @Test_file;Buf6(*)          ! whereas if the processed version is
190  ENTER @Test_file;Buf7(*)          ! needed then 14 sections are required.
200  !
210  IF File_type$="P" THEN
220    ENTER @Test_file;Buf8(*)        ! If the file is a processed version of
230    ENTER @Test_file;Buf9(*)        ! the original sampled data (ie. the
240    ENTER @Test_file;Buf10(*)       ! data has been processed by a DM) then
250    ENTER @Test_file;Buf11(*)       ! the extra speech data must be read off
260    ENTER @Test_file;Buf12(*)       ! the disc.
270    ENTER @Test_file;Buf13(*)
280    ENTER @Test_file;Buf14(*)
290    !
```

```
300  ELSE
310    MAT Buf8= Buf1                    ! If the speech segment is the original
320    MAT Buf9= Buf2                    ! sampled version then only half as much
330    MAT Buf10= Buf3                   ! data is available (as compared with
340    MAT Buf11= Buf4                   ! interpolated and processed files) and
350    MAT Buf12= Buf5                   ! so it is repeated.
360    MAT Buf13= Buf6
370    MAT Buf14= Buf7
380  END IF
390  !
400  ASSIGN @Test_file TO *
410  !
420  MASS STORAGE IS Drive_0$
430  RETURN
440 !*************************************************************************
```

```
450 !<<<<<<<< DETERMINE THE NAME OF THE DISC FILE CONTAINING THE DATA >>>>>>>>
460 !          ------------------------------------------------------
470 !
480 File_name:!
490  INPUT "Enter name of file from which the data is to be read >",File_name$
500  File_name$=UPC$(File_name$)
510  DISP "Does this file contain the original or processed data ?"
520  ON KBD GOTO File_type
530 Spin:!
540  GOTO Spin
550 File_type:!
560  R$=UPC$(KBD$)
570  OFF KBD
580  IF R$="P" THEN
590    File_name$=File_name$&"_P"  ! Add the suffix indicating processed data.
600    File_type$="P"
610  ELSE
620    File_name$=File_name$&"_O"  ! Add the suffix indicating original data.
630    File_type$="O"
640  END IF
650  RETURN
660 !*************************************************************************
```

```
!<<<<<<<<<<<<<< SEND THE SPEECH DATA TO THE MULTIPROGRAMMER >>>>>>>>>>>>>>
!               ----------------------------------------
!
Output_data:!
 DISP "Press ";CHR$(129);" CONTINUE ";CHR$(128);" to commence output."
 PAUSE
 !
 DISP
Transfer_data: !
 FOR Buffer_no=Start TO Finish
   OUTPUT @Multi;Mw_command$                    ! Send instruction to the card.
   WAIT .005                                    ! Allow for it to be decoded.
   SELECT Buffer_no
   CASE 1
     OUTPUT @Mw_readout;Buf1(*)
   CASE 2
     OUTPUT @Mw_readout;Buf2(*)
   CASE 3
     OUTPUT @Mw_readout;Buf3(*)
   CASE 4
     OUTPUT @Mw_readout;Buf4(*)
   CASE 5
     OUTPUT @Mw_readout;Buf5(*)
   CASE 6
     OUTPUT @Mw_readout;Buf6(*)
   CASE 7
     OUTPUT @Mw_readout;Buf7(*)
   CASE 8
     OUTPUT @Mw_readout;Buf8(*)
   CASE 9
     OUTPUT @Mw_readout;Buf9(*)
   CASE 10
     OUTPUT @Mw_readout;Buf10(*)
   CASE 11
     OUTPUT @Mw_readout;Buf11(*)
   CASE 12
     OUTPUT @Mw_readout;Buf12(*)
   CASE 13
     OUTPUT @Mw_readout;Buf13(*)
   CASE 14
     OUTPUT @Mw_readout;Buf14(*)
   END SELECT
 NEXT Buffer_no
 !
 IF Continuous$="Y" THEN
   GOTO Transfer_data
 END IF
 !
 RETURN
!*******************************************************************************
```

```
!70 !<<<<<<<<<<<<<< RESOLVE ERRORS ARISING FROM DISC OPERATIONS >>>>>>>>>>>>>>
!80 !                ----------------------------------------
!90 !
200 Wrong_disc:!
210  IF ERRN=72 THEN
220    BEEP 500,.1
230    PRINT "Hard disc is not present !"
240    PRINT "Place the appropriate floppy disc in the LEFT hand disc drive."
250    PRINT "Hit ";CHR$(129);" CONTINUE ";CHR$(128);" when ready."
260    Drive$=":INTERNAL,4,1"
270    PAUSE
280    GOTO Assign_device
290  ELSE
300    IF ERRN=56 THEN
310      BEEP 500,.1
320      DISP "Invalid file name !! Please re-enter >"
330      WAIT 2
340      GOTO Get_file
350    ELSE
360      DISP ERRN,ERRM$
370      STOP
380    END IF
390  END IF
400  STOP
410  END
420 !*****************************************************************************
```

```
430 !<<<<<<<<<<<<<<<<<<<<<<< BINARY LOADER SUB-PROGRAM >>>>>>>>>>>>>>>>>>>>>>>
440 !                        -------------------------
450 !
460  SUB Xfer(Op$)                              ! Transfer the special instruction
470    CLEAR 823                                ! from the supplemental instruction
480    WAIT 4                                   ! set.
490    ENTER 82310;Dummy,Dummy,Dummy
500    ALLOCATE Ascii$[80]
510          !
520    Name$=Op$&"Z:INTERNAL,4,0"
530    ASSIGN @Disc_file TO Name$
540    ON END @Disc_file GOTO Eof
550          !
560 Rd_file:ENTER @Disc_file;Ascii$
570    OUTPUT 823;Ascii$
580    GOTO Rd_file
590          !
600 Eof:OFF END @Disc_file
610    ASSIGN @Disc_file TO *
620    DEALLOCATE Ascii$
630    PRINT TABXY(5,8);"<MR> SUPPLEMENTAL INSTRUCTION LOADED INTO MULTI"
640       ! CLEAR SELF TEST SRQ
650    ENTER 82310:Complete,No_of_errors,No_of_cards
660    WAIT 1
670    DISP " "
680  SUBEND
690 !*************************************************************************
700 !
710 !<<<<<<<<<<<<< DETERMINE MULTIPROGRAMMER CARD CONFIGURATION >>>>>>>>>>>>>
720 !               ----------------------------------------
730 !
740  SUB Multi_set_up(INTEGER Configuration(*),Tb_slot,Mem_in_slot,D_a_slot)
750    OPTION BASE 1
760    PRINT TABXY(5,9);"DETERMINING MULTI CONFIGURATION"
770    FOR Slot_no=15 TO 0 STEP -1              ! Read data format of each card.
780      OUTPUT 823;"RF";Slot_no;"T"            ! Start at slot #15.
790 !
800      FOR Card_variable=1 TO 5               ! Five variables returned per
810                                             ! card - Card ID, Data type,
820                                             !        LSB, Size, Limit.
830                                             !
840        ENTER 82304 USING "%,K";Configuration(Slot_no,Card_variable)
850      NEXT Card_variable
860      SELECT Configuration(Slot_no,1)
870      CASE 1
880        Tb_slot=Slot_no
890      CASE 6
900        Mem_in_slot=Slot_no
910      CASE 48
920        D_a_slot=Slot_no
930      CASE ELSE
940      END SELECT
950    NEXT Slot_no
960  SUBEND
970 !*************************************************************************
```

```
IMPLEMENT

PROCEDURE Codecs ;

CONST
        Block_size     = 10283 ; { The number of samples per array.    }

VAR
        Leakage,                 { The prediction filter constant.     }
        Input_signal   : REAL ;  { The actual sampled input signal.    }
{-------------------------------------------------------------------}



PROCEDURE LDM ;                  { Linear Delta Modulation.            }

VAR
        I              : INTEGER ;

BEGIN

    FOR I := 1 TO Block_size DO
       BEGIN

           Signal       := New_signal ;
           Input_signal := Voice[ I ] ;

           { Update the two-level quantizer binary output signal.     }
           {----------------------------------------------------------}

           IF Input_signal >= Signal
             THEN
               Bn := +1
             ELSE
               Bn := -1 ;


           { Update the coder approximation signal - which is         }
           { equivalent to the decoder output signal under ideal      }
           { channel conditions.                                      }
           {----------------------------------------------------------}

           New_Signal := Signal  +  So * Bn ;
           IF New_signal > Max_sig          { Check that the output of }
             THEN                           { the decoder is within    }
               New_signal := Max_sig        { the permissible range of }
             ELSE                           { of the output signal.    }
               IF New_signal < Min_sig
                 THEN
                   New_signal := Min_sig ;
           Voice[ I ] := New_Signal ;

       END ;

    WRITE( '.' ) ;

END ;
{*******************************************************************}
```

```
PROCEDURE CVSD ;     { Continuously Variable Slope Delta Modulation. }

{ The algorithm used here is based on the algorithms described in    }
{ references [18], [21], and [22].  For further details refer to     }
{ chapter 3 section 2.2.                                             }
{--------------------------------------------------------------------}

CONST
        B               = 0.99 ;  { Step size leakage constant.      }
        H               = 3 ;     { Gain of the prediction integrator. }

VAR
        I,V             : INTEGER ;



BEGIN

    FOR I := 1 TO Block_size DO
       BEGIN

          Bn_2          := Bn_1 ;
          Bn_1          := Bn ;
          Step          := New_step ;
          Signal        := New_signal ;
          Input_signal  := Voice[ I ] ;


          { Update the two-level quantizer binary output signal.     }
          {----------------------------------------------------------}

          IF Input_signal >= Signal
            THEN                                  { Update the two-level }
              Bn := +1                            { quantizer binary     }
            ELSE                                  { output signal.       }
              Bn := -1 ;


          { Update the coder step size according to the syllabic     }
          { companding algorithm.                                    }
          {----------------------------------------------------------}

          IF ( Bn_2 = Bn_1) AND ( Bn_1 = Bn )
            THEN                                  { Apply a pulse to the step }
              V := 150                            { size generation filter.   }
            ELSE
              V := 0 ;

          New_step    := B * Step  +  ( 1-B ) * ( V + So ) ;
```

```
            IF New_step > Max_step                  { Limit the step size }
              THEN                                  { to the specified    }
                New_step := Max_step                { range.              }
              ELSE
                IF New_step < -Max_step
                  THEN
                    New_step := -Max_step
                  ELSE
                    IF ( New_step > 0 ) AND ( New_step < So )
                      THEN
                        New_step := So
                      ELSE
                        IF ( New_step > -So ) AND ( New_step < 0 )
                          THEN
                            New_step := -So ;


            { Update the coder approximation signal - which is          }
            { equivalent to the decoder output signal under ideal       }
            { channel conditions.                                       }
            {-----------------------------------------------------------}

            New_Signal := Leakage * Signal  +
                          Bn * H * ( 1-Leakage ) * New_step ;

            IF New_signal > Max_sig             { Check that the output of }
              THEN                              { the decoder is within    }
                New_signal := Max_sig           { the permissible range of }
              ELSE                              { of the output signal.    }
                IF New_signal < Min_sig
                  THEN
                    New_signal := Min_sig ;

            Voice[ I ] := New_Signal ;

        END ;

    WRITE( ',' ) ;

END ;
{***************************************************************************}
```

```
PROCEDURE CFDM ;                    { Constant Factor Delta Modulation. }

{ The algorithm used here corresponds to Jayant's ADM (cf. reference }
{ [23]).  For further details refer to chapter 2 section 2.3.       }
{-------------------------------------------------------------------}

CONST
        P               = 1.5 ; { The constant factors used by Jayant. }
VAR
        I               : INTEGER ;

BEGIN
    FOR I := 1 TO Block_size DO
       BEGIN
           Bn_1          := Bn ;
           Step          := New_step ;
           Signal        := New_signal ;
           Input_signal  := Voice[ I ] ;

           { Update the two-level quantizer binary output signal.    }
           {---------------------------------------------------------}
           IF Input_signal >= Signal
             THEN                                  { Update the two-level }
               Bn := +1                            { quantizer binary     }
             ELSE                                  { output signal.       }
               Bn := -1 ;

           { Update the coder step size according to the Constant    }
           { Factor companding algorithm.                            }
           {---------------------------------------------------------}
           IF Bn_1 = Bn
             THEN
               New_step := Step * P
             ELSE
               New_step := Step / P ;
           IF New_step > Max_step                { Limit the step size }
             THEN                                { to the specified    }
               New_step := Max_step              { range.              }
             ELSE
               IF New_step < So
                 THEN
                   New_step := So ;

           { Update the coder approximation signal - which is        }
           { equivalent to the decoder output signal under ideal     }
           { channel conditions.                                     }
           {---------------------------------------------------------}
           New_Signal := Leakage * Signal  +  Bn * New_step ;
           IF New_signal > Max_sig          { Check that the output of }
             THEN                           { the decoder is within    }
               New_signal := Max_sig        { the permissible range of }
             ELSE                           { of the output signal.    }
               IF New_signal < Min_sig
                 THEN
                   New_signal := Min_sig ;
           Voice[ I ] := New_Signal ;
       END ;
    WRITE( '.' ) ;
END ;
{*******************************************************************}
```

```
PROCEDURE SVADM ;              { Song Voice Adaptive Delta Modulation. }

VAR
      I                : INTEGER ;

BEGIN

    FOR I := 1 TO Block_size DO
       BEGIN
           Bn_1          := Bn ;
           Step          := New_step :
           Signal        := New_signal ;
           Input_signal := Voice[ I ] ;

           { Update the two-level quantizer binary output signal.     }
           {----------------------------------------------------------}

           IF Input_signal >= Signal
            THEN                                  { Update the two-level }
              Bn := +1                            { quantizer binary     }
            ELSE                                  { output signal.       }
              Bn := -1 ;


           { Update the coder step size according to the Song Voice   }
           { Adaptive algorithm.                                      }
           {----------------------------------------------------------}

           New_step    := ABS( Step ) * Bn   +   So * Bn_1 ;
           IF New_step > Max_step                  { Limit the step size }
             THEN                                  { to the specified    }
               New_step := Max_step                { range.              }
             ELSE
               IF New_step < -Max_step
                 THEN
                   New_step := -Max_step ;


           { Update the coder approximation signal - which is         }
           { equivalent to the decoder output signal under ideal      }
           { channel conditions.                                      }
           {----------------------------------------------------------}

           New_Signal := Leakage * Signal   +   New_step ;
           IF New_signal > Max_sig          { Check that the output of }
             THEN                           { the decoder is within    }
               New_signal := Max_sig        { the permissible range of }
             ELSE                           { of the output signal.    }
               IF New_signal < Min_sig
                 THEN
                   New_signal := Min_sig ;
           Voice[ I ] := New_Signal ;
       END ;

    WRITE( '.' ) ;

END ;
{************************************************************************}
```

```
PROCEDURE SHCDM ;          { Song Hybrid Companding Delta Modulation. }

{ The algorithm used here was developed as part of the research      }
{ project.  It is descibed in detail in chapter 3, section 2.6.      }
{--------------------------------------------------------------------}

VAR
        Sum,
        Last_step,
        Shift_out       : REAL ;

        I               : INTEGER ;



BEGIN

    Sum := 0 ;
    FOR I := 1 TO ( Syllable-1 ) DO    { Update the syllabic energy. }
       Sum := Sum + SQR( Syll_filter[ I+1 ]  -  Syll_filter[ I ] ) ;

    FOR I := 1 TO Block_size DO
       BEGIN

          Bn_1          := Bn ;
          Last_step     := Step ;
          Signal        := New_signal ;
          Input_signal  := Voice[ I ] ;


          { Update the two-level quantizer binary output signal.     }
          {----------------------------------------------------------}

          IF Input_signal >= Signal
            THEN
              Bn := +1
            ELSE
              Bn := -1 ;


          { Update the syllabic step size factor.                    }
          {----------------------------------------------------------}

          Shift_out           := Syll_filter[ Pos ] ;
          Syll_filter[ Pos ] := Signal ;

          IF Pos = 1
            THEN
              Sum := Sum + SQR( Signal - Syll_filter[ Syllable ] )
            ELSE
              Sum := Sum + SQR( Signal - Syll_filter[ Pos-1 ] ) ;

          Pos := Pos + 1 ;
          IF Pos > Syllable
            THEN
              Pos := 1 ;
```

```
            Sum    := Sum - SQR(Shift_out - Syll_filter[ Pos ]) ;

            IF Pos  = Syllable
              THEN
                BEGIN
                    Big_alpha := Alpha * SQRT( Sum/Syllable ) ;
                    IF Big_alpha < So
                      THEN
                        Big_alpha := So ;
                END ;


            { Update the coder instantaneous step factor according to }
            { the Song Voice Adaptive algorithm.                      }
            {---------------------------------------------------------}

            Step := ABS( Last_step ) * Bn  +  Bn_1 * So ;


            {Update the coder step size by combining the instant-     }
            { aneous and syllabic step factors.                       }
            {---------------------------------------------------------}

            New_step := Big_alpha * Step ;
            IF New_step > Max_step                { Limit the step size }
              THEN                                { to the specified    }
                New_step := Max_step              { range.              }
              ELSE
                IF New_step < -Max_step
                  THEN
                    New_step := -Max_step ;


            { Update the coder approximation signal - which is        }
            { equivalent to the decoder output signal under ideal     }
            { channel conditions.                                     }
            {---------------------------------------------------------}

            New_signal := Leakage * Signal + New_step ;

            IF New_signal > Max_sig          { Check that the output of }
              THEN                           { the decoder is within    }
                New_signal := Max_sig        { the permissible range of }
              ELSE                           { of the output signal.    }
                IF New_signal < Min_sig
                  THEN
                    New_signal := Min_sig ;

            Voice[ I ] := New_signal ;

        END ;

    WRITE( '.' ) ;

END ;
{*************************************************************************}
```

```
PROCEDURE MSHCDM ;                { Modified Song Hybrid Companding DM. }

{ The algorithm used here was developed as part of the research       }
{ project.  It is descibed in detail in chapter 3, section 2.6.       }
{---------------------------------------------------------------------}

VAR
        Sum,
        Shift_out       : REAL ;

        I               : INTEGER ;



BEGIN

    Sum := 0 ;
    FOR I := 1 TO ( Syllable-1 ) DO    { Update the syllabic energy. }
       Sum := Sum + SQR( Syll_filter[ I+1 ] -  Syll_filter[ I ] ) ;

    FOR I := 1 TO Block_size DO
       BEGIN

           Bn_1          := Bn ;
           Step          := New_step ;
           Signal        := New_signal ;
           Input_signal  := Voice[ I ] ;


           { Update the two-level quantizer binary output signal.     }
           {----------------------------------------------------------}

           IF Input_signal >= Signal
             THEN
               Bn := +1
             ELSE
               Bn := -1 ;


           { Update the syllabic step size factor.                    }
           {----------------------------------------------------------}

           Shift_out           := Syll_filter[ Pos ] ;
           Syll_filter[ Pos ] := Signal ;

           IF Pos = 1
             THEN
               Sum := Sum + SQR( Signal - Syll_filter[ Syllable ] )
             ELSE
               Sum := Sum + SQR( Signal - Syll_filter[ Pos-1 ] ) ;

           Pos := Pos + 1 ;
           IF Pos > Syllable
             THEN
               Pos := 1 ;

           Sum     := Sum - SQR(Shift_out - Syll_filter[ Pos ]) ;
```

```
            IF Pos  = Syllable
              THEN
                BEGIN
                    Big_alpha := Alpha * SQRT( Sum/Syllable ) ;
                    IF Big_alpha < So
                      THEN
                        Big_alpha := So ;
                END ;


            { The instantaneous companding - Song Voice Adaptive.  In }
            { this application the syllabic factor is included at     }
            { this stage.                                             }
            {---------------------------------------------------------}

            New_step := ABS( Step ) * Bn  +  Bn_1 * Big_alpha * So ;

            IF New_step > Max_step                { Limit the step size }
              THEN                                { to the specified    }
                New_step := Max_step              { range.              }
              ELSE
                IF New_step < -Max_step
                  THEN
                    New_step := -Max_step ;


            { Update the coder approximation signal - which is        }
            { equivalent to the decoder output signal under ideal     }
            { channel conditions.                                     }
            {---------------------------------------------------------}

            New_signal := Leakage * Signal + New_step ;

            IF New_signal > Max_sig           { Check that the output of }
              THEN                            { the decoder is within    }
                New_signal := Max_sig         { the permissible range of }
              ELSE                            { of the output signal.    }
                IF New_signal < Min_sig
                  THEN
                    New_signal := Min_sig ;

            Voice[ I ] := New_signal ;

        END ;

    WRITE( '.' ) ;

END ;
{*************************************************************************}
```

```
PROCEDURE HCDM ;                    { Hybrid Companding Delta Modulation. }

{ The algorithm used in this procedure is the one described by Un et }
{ al [17], [18].  The syllabic companding is combined with Constant  }
{ Factor companding.  For further details refer to chapter 3,        }
{ section 2.4.                                                       }
{--------------------------------------------------------------------}

VAR
        Sum,
        Shift_out       : REAL ;

        I               : INTEGER ;



BEGIN

    Sum := 0 ;
    FOR I := 1 TO ( Syllable-1 ) DO     { Update the syllabic energy. }
       Sum := Sum + SQR( Syll_filter[ I+1 ]  -  Syll_filter[ I ] ) ;

    FOR I := 1 TO Block_size DO
       BEGIN

           Bn_2          := Bn_1 ;
           Bn_1          := Bn ;
           Step          := New_step ;
           Signal        := New_signal ;
           Input_signal  := Voice[ I ] ;


           { Update the two-level quantizer binary output signal.     }
           {----------------------------------------------------------}

           IF Input_signal >= Signal
             THEN
               Bn := +1
             ELSE
               Bn := -1 ;


           { Update the syllabic step size factor.                    }
           {----------------------------------------------------------}

           Shift_out           := Syll_filter[ Pos ] ;
           Syll_filter[ Pos ] := Signal ;

           IF Pos = 1
             THEN
               Sum := Sum + SQR( Signal - Syll_filter[ Syllable ] )
             ELSE
               Sum := Sum + SQR( Signal - Syll_filter[ Pos-1 ] ) ;

           Pos := Pos + 1 ;
           IF Pos > Syllable
             THEN
               Pos := 1 ;
```

```
            Sum     := Sum - SQR(Shift_out - Syll_filter[ Pos ]) ;

            IF Pos  = Syllable
              THEN
                BEGIN
                    Big_alpha := Alpha * SQRT( Sum/Syllable ) ;
                    IF Big_alpha < So
                      THEN
                        Big_alpha := So ;
                END ;


            { The instantaneous companding - Constant Factor.          }
            {----------------------------------------------------------}

            IF ( Bn = Bn_1 ) AND ( Bn_1 = Bn_2 )
              THEN
                Step := Step * 1.5
              ELSE
                IF Bn <> Bn_1
                  THEN
                    Step := Step / 1.5 ;
            IF Step < 1
              THEN
                Step := 1 ;
            IF Step > Max_cf
              THEN
                Step := Max_cf ;

            { Combine syllabic & instantaneous companding factors.     }
            {----------------------------------------------------------}

            New_step     := Big_alpha * Step ;
            IF New_step > Max_step
              THEN
                New_step := Max_step ;

            { Update the coder approximation signal - which is         }
            { equivalent to the decoder output signal under ideal      }
            { channel conditions.                                      }
            {----------------------------------------------------------}

            New_signal := Leakage * Signal  +  Bn * NEW_STEP ;

            IF New_signal > Max_sig          { Check that the output of }
              THEN                           { the decoder is within    }
                New_signal := Max_sig        { the permissible range of }
              ELSE                           { of the output signal.    }
                IF New_signal < Min_sig
                  THEN
                    New_signal := Min_sig ;

            Voice[ I ] := New_signal ;

        END ;

    WRITE( '.' ) ;

END ;
{*****************************************************************}
```

```
{*************************************************************************}
{*********************                                   *********************}
{*********************      M A I N L I N E              *********************}
{*********************                                   *********************}
{*************************************************************************}

BEGIN

    IF Beta > 1000                  { If no leakage is required then a  }
      THEN                          { value for Beta of greater than    }
        Leakage := 1                { 1000 is chosen by the user.       }
      ELSE
        Leakage := EXP( -1 / ( Beta * Rate ) ) ;

    CASE P_no OF

      1 : LDM ;       { Linear Delta Modulation.                          }
      2 : CVSD ;      { Continuously Variable Slope Delta  Modulation.    }
      3 : CFDM ;      { Constant Factor Delta Modulation.                 }
      4 : SVADM ;     { Song Voice Adaptive Delta Modulation.             }

      { In the case of the hybrid delta modulators the code perform-      }
      { ing the syllabic companding has been repeated in each of the      }
      { procedures simulating the hybrids.  Although this is not the      }
      { most efficient use of the code it does help to reduce the         }
      { amount of processing time occupied by subroutine calls.           }
      {-------------------------------------------------------------------}

      5 : SHCDM ;     { Song Hybrid Companding Delta Modulation;          }
      6 : MSHCDM ;    { Modified Song Hybrid Companding DM.               }
      7 : HCDM ;      { Hybrid Companding Delta Modulation.               }

    END ;

END { Of Procedure Codecs. } ;
END { Of Module CODERS. }.
{*************************************************************************}
```

```
MODULE NOISE

{-------------------------------------------------------------------}
{ CODSIM          : Codec Simulation Package.                       }
{ M.Sc Thesis     : The use of Delta Modulation for Low Bit-Rate    }
{                   Digital Speech Coding.                          }
{ Author          : J.M. Irvine.                                    }
{                   Department of Electrical and Electronic         }
{                   Engineering                                     }
{                   University of Cape Town.                        }
{ Unit Number     : 7.                                              }
{ Latest Update   : 1st December, 1984.                             }
{-------------------------------------------------------------------}

$STACKCHECK OFF$     { Disable stack overflow checks.               }
$RANGE OFF$          { Disable array and CASE index range checks.   }
$OVFLCHECK OFF$      { BASIC & PASCAL allow different INTEGER ranges. }
$IOCHECK OFF$        { No error checks after calls on I/O routines. }
$DEBUG OFF$

$SEARCH 'HARD:CSUBDECL'$;    { BASIC-PASCAL conversion code.        }

IMPORT CSUBDECL ;            { Simplifies parameter passing to BASIC. }

EXPORT    { The EXPORT statement includes the TYPE and PROCEDURES to  }
          { be accessed as compiled subroutines by the BASIC program. }
     TYPE
        SEGMENT_NOISE = ARRAY [ 1..2057 ]  OF REAL ;
        VOICE_BUFFER  = ARRAY [ 1..10283 ] OF REAL ;

     PROCEDURE Noises(  VAR Present_sgn,
                            Sample_count,
                            Granular_flag,
                            Overload_flag,
                            Segment_count,
                            Samples_per_segment : BINTVALTYPE ;
                        VAR Total_noise,
                            Segmental_noise     : REAL ;
                            S_Pointer           : DIMENTRYPTR ;
                        VAR Segmental           : SEGMENT_NOISE ;
                            V_Pointer           : DIMENTRYPTR ;
                        VAR Voice               : VOICE_BUFFER ) ;
{-------------------------------------------------------------------}
IMPLEMENT

PROCEDURE Noises ;

CONST
        Block_size    = 10283 ; { The number of samples per array.  }

VAR
        Input_signal  : REAL ;  { The actual sampled input signal.  }

        I,
        Last_sgn      : INTEGER ;
{-------------------------------------------------------------------}
```

```
PROCEDURE Granular_noise ;

VAR
        J               : INTEGER ;

BEGIN

    FOR J := 1 TO Block_size DO
       BEGIN

           Last_sgn     := Present_sgn ;
           IF Voice[ J ] < 0
             THEN
               Present_sgn := -1
             ELSE
               Present_sgn := +1 ;

           IF Last_sgn = Present_sgn
             THEN                    { If two equal signs are detected   }
               Voice[ J ] := 0 ; { then the noise is Overload noise. }

       END ;

END ;
{*************************************************************************}




PROCEDURE Overload_noise ;

VAR
        J               : INTEGER ;

BEGIN

    FOR J := 1 TO Block_size DO
       BEGIN

           Last_sgn     := Present_sgn ;
           IF Voice[ J ] < 0
             THEN
               Present_sgn := -1
             ELSE
               Present_sgn := +1 ;

           IF Last_sgn <> Present_sgn
             THEN                    { If two unequal signs are detected }
               Voice[ J ] := 0 ; { then the noise is Granular noise. }

       END ;

END ;
{*************************************************************************}
```

```
BEGIN

{ This module caculates the total noise energy and the segmental      }
{ noise energy contained in the decoded speech signal.  The noise     }
{ signal, either filtered or unfiltered, is passed to this module by  }
{ the BASIC program.  The segmental noise powers are calculated       }
{ using 20 msec segments.  If the Granular or Overload flags are set  }
{ then the respective noise energies are calculated.  If neither of   }
{ these flags are set then the values caculated correspond to the     }
{ lumped noise and segmental noise energies.                          }
{---------------------------------------------------------------------}

    IF Granular_flag = 1
      THEN
        Granular_noise          { Extract the Granular noise signal. }
      ELSE
        IF Overload_flag = 1
          THEN
            Overload_noise ;    { Extract the Overload noise signal. }

    For I := 1 TO Block_size DO
       BEGIN

           Total_noise := Total_noise  +  SQR( Voice[ I ] ) ;

           IF Sample_count = Samples_per_segment
             THEN
               BEGIN
                   Sample_count                 := 0 ;
                   Segmental[ Segment_count ]   := Segmental_noise ;
                   Segment_count                := Segment_count  +  1 ;
                   Segmental_noise              := 0 ;
               END ;

           Segmental_noise := Segmental_noise  +  SQR( Voice[ I ] ) ;
           Sample_count    := Sample_count  +  1 ;

       END ;


END { Of Procedure Codecs. } ;
END { Of Module CODERS. }.
{*********************************************************************}
```

```
MODULE STRETCHER

{-------------------------------------------------------------------}
{ CODSIM          : Codec Simulation Package.                       }
{ M.Sc Thesis     : The use of Delta Modulation for Low Bit-Rate    }
{                   Digital Speech Coding.                          }
{ Author          : J.M. Irvine.                                    }
{                   Department of Electrical and Electronic         }
{                   Engineering                                     }
{                   University of Cape Town.                        }
{ Unit Number     : 8.                                              }
{ Latest Update   : 4th December, 1984.                             }
{-------------------------------------------------------------------}

$STACKCHECK OFF$      { Disable stack overflow checks.              }
$RANGE OFF$           { Disable array and CASE index range checks.  }
$OVFLCHECK OFF$       { BASIC & PASCAL allow different INTEGER ranges. }
$IOCHECK OFF$         { No error checks after calls on I/O routines. }
$DEBUG OFF$

$SEARCH 'HARD:CSUBDECL'$;    { BASIC-PASCAL conversion code.         }

IMPORT CSUBDECL ;            { Simplifies parameter passing to BASIC. }

EXPORT    { The EXPORT statement includes the TYPE and PROCEDURES to  }
          { be accessed as compiled subroutines by the BASIC program. }
     TYPE
        SPEECH_BUFFER = ARRAY [ 1..10283 ] OF BINTVALTYPE ;
        VOICE_BUFFER  = ARRAY [ 1..8,1..10283 ] OF BINTVALTYPE ;

     PROCEDURE STRETCH( VAR Sample_rate : BINTVALTYPE ;
                            S_pointer   : DIMENTRYPTR ;
                        VAR Speech      : SPEECH_BUFFER ;
                            V_pointer   : DIMENTRYPTR ;
                        VAR Voice       : VOICE_BUFFER ) ;
{-------------------------------------------------------------------}



IMPLEMENT

PROCEDURE STRETCH ;

CONST
        Block_size    = 10283 ; { The number of samples per array.   }

VAR
        I,J,K,
        Rate_factor   : INTEGER ;
{-------------------------------------------------------------------}
```

```
BEGIN

    { The effective sampling rate determines the sample spacing.  If }
    { a sampling rate of 32 kHz is required then the 8 kHz sampled   }
    { data must have two null samples inserted between every sample. }
    {-------------------------------------------------------------------}

    J := 0 ;                          { The subscripts controlling the  }
    K := 1 ;                          { position of the output samples. }

    Rate_factor := Sample_rate DIV 8 ;

    FOR I := 1 TO Block_size DO
       BEGIN

          J := J + Rate_factor ;      { Determine the position of  }
          IF J > Block_size           { the next sample in the new }
            THEN                      { speech segment.            }
              BEGIN

                  K := K + 1 ;
                  J := J - Block_size ;

              END ;

          Voice[ K,J ] := Speech[ I ] ;

       END ;

    WRITE( '.' ) ;

END ;    { Of Procedure Stretch. }
END.     { Of Module Stretcher.  }
{*********************************************************************}
```

```
MODULE FIR

{-------------------------------------------------------------------------}
{ CODSIM          : Codec Simulation Package.                             }
{ M.Sc Thesis     : The use of Delta Modulators for Low Bit-Rate          }
{                   Digital Speech Coding.                                }
{ Author          : J.M. Irvine.                                          }
{                   Department of Electrical and Electronic               }
{                   Engineering                                           }
{                   University of Cape Town.                              }
{ Unit Number     : 9.                                                    }
{ Latest Update   : 9th December, 1984.                                   }
{-------------------------------------------------------------------------}
{                                                                         }
{ The digital filter simulated by this procedure is responsible for       }
{ the removal of the out of band components of the noise produced by      }
{ the delta modulation process.  The purpose of this is to implement      }
{ the digital equivalent of the low-pass filter at the output of the      }
{ decoder so as to enable the correct determination of the signal-        }
{ to-noise ratio of the DM codec.  This filter is also used for the       }
{ interpolation of the speech data to higher effective sampling           }
{ rates.  The filter is designed to operate at 16, 24, 32, 48 and 64      }
{ kHz sampling rates.  A window length of 60 has been used.               }
{-------------------------------------------------------------------------}

$STACKCHECK OFF$     { Disable stack overflow checks.                     }
$RANGE OFF$          { Disable array and CASE index range checks.         }
$OVFLCHECK OFF$      { BASIC & PASCAL allow different INTEGER ranges.     }
$IOCHECK OFF$        { No error checks after calls on I/O routines.       }
$DEBUG OFF$

$SEARCH 'HARD:CSUBDECL'$;    { BASIC-PASCAL conversion code.              }

IMPORT CSUBDECL ;            { Simplifies parameter passing to BASIC.     }

EXPORT    { The EXPORT statement includes the TYPE and PROCEDURES to      }
          { be accessed as compiled subroutines by the BASIC program.     }
     TYPE
        WINDOW_BUFFER = ARRAY [ 1..121 ] OF REAL ;
        VOICE_BUFFER  = ARRAY [ 1..10283 ] OF REAL ;

     PROCEDURE FILTER( VAR PI           : REAL ;
                       VAR Position,
                           Last_flag,
                           Sample_rate : BINTVALTYPE ;
                           W_pointer   : DIMENTRYPTR ;
                       VAR Window      : WINDOW_BUFFER ;
                           V_pointer   : DIMENTRYPTR ;
                       VAR Voice       : VOICE_BUFFER ) ;
{-------------------------------------------------------------------------}
```

```
IMPLEMENT

PROCEDURE FILTER ;

CONST
        Window_length =    60 ; { The no. of elements in the filter. }
        Block_size    = 10283 ; { The number of samples per array.   }
        F_interval    =   250 ; { Frequency spacing for the filter.  }

VAR
        D_filter,
        Filter_in     : ARRAY[ -Window_length..+Window_length ]
                        OF REAL ;

        Filter_out    : REAL ;

        J,K,
        Inp_position  : INTEGER ;
{-------------------------------------------------------------------}

PROCEDURE Initialize_filter ;

VAR
        Theta,
        Hamming       : REAL ;

        I,J,
        Freq_samples  : INTEGER ;

BEGIN
    Freq_samples := ( Sample_rate * 1000 ) DIV ( F_interval ) :
    FOR I := -Window_length to +Window_length DO
      Filter_in[ I ] := Window[ I+61 ] ;

    FOR I := 0 TO Window_length DO
       BEGIN
           { Determine the impulse response weighting sequence.      }
           {---------------------------------------------------------}

           D_filter[ I ] := 1 + COS( 20*PI*I / Freq_samples ) ;
           FOR J := 1 TO 9 DO   { The number of frequency samples in }
                                { the signal pass band.              }
              BEGIN
                  Theta         := ( 2*PI*I*J ) / ( Freq_samples ) ;
                  D_filter[ I ] := D_filter[ I ] + 2 * COS( Theta ) ;
              END ;

           { A Hamming window has been used to reduce the 'Gibson'   }
           { effect - ie. the overshoot caused by the finite length  }
           { of the window of the filter.                            }
           {---------------------------------------------------------}

           Hamming := 0.54 + 0.46 * COS( I*PI / Window_length ) ;
           D_Filter[ I ]  := D_Filter[ I ] * Hamming / Freq_samples ;
           D_Filter[ -I ] := D_Filter[ I ] ;
       END ;
END ;
{*******************************************************************}
```

```
BEGIN

    Initialize_filter ;
    FOR J := 1 TO Block_size DO
       BEGIN
           Filter_in[ Position ] := Voice[ J ] ;

           Filter_out   := 0 ;
           Inp_Position := Position ;
           FOR K := -Window_length TO +Window_length DO
              BEGIN
                  Inp_Position := Inp_Position + 1 ;
                  IF Inp_Position > Window_length
                    THEN
                      Inp_Position := -Window_length ;
                  Filter_out := Filter_out +
                                D_filter[ K ] *
                                Filter_in[ Inp_Position ] ;
              END ;

           Voice[ J ] := Filter_out ;

           Position := Position + 1 ;
           IF Position > Window_length
             THEN
               Position := -Window_length ;

       END ;
    WRITE( '.' ) ;


    { Update the Filter Window buffer before returning to the BASIC  }
    { parent-program.                                                }
    {----------------------------------------------------------------}

    FOR J := -Window_length to +Window_length DO
      Window[ J+61 ] := Filter_in[ J ] ;

END ; { Of Procedure Filter. }
END.  { Of Module FIR.       }
{*****************************************************************}
```

APPENDIX F

The digital filter described here was used for the interpolation function as well as for the objective quality measures. A finite impulse response (FIR) filter was used because of its inherent stability[41] and the fact that the phase and amplitude characteristics may be arbitrarily specified.

Having specified the amplitude and phase characteristics the impulse response weighting sequence can be determined. If the amplitude response function is sampled at 1/NT Hz intervals then the frequency samples (G_r) are related to the impulse response g(i)T by the discrete Fourier Transform (DFT) given by :

$$G_r = \sum_{i=0}^{N-1} g(i)T . e^{-j2\pi ir/N} \qquad (F1)$$

Hence, using the inverse DFT the impulse response weighting sequence can be determined.

$$g(i)T = \left(\sum_{r=0}^{N-1} G_r . e^{j2\pi ir/N} \right) / N \qquad (F2)$$

where

$i = 0, 1, 2, \ldots, N-1.$

However, by truncating the Fourier series to N values gives rise to oscillations in the actual amplitude response of the filter. These oscillations, known as the Gibbs phenomenon, can be reduced by applying a window function to the finite series. The three most commonly used window functions for non-recursive digital

filters are the Hamming, Hanning and Blackman windows[41].

(a) Hamming window : $W_r = 0.54 + 0.46 \cos(r\pi/I)$.

(b) Hanning window : $W_r = 0.50 + 0.50 \cos(r\pi/I)$.

(c) Blackman window : $W_r = 0.42 + 0.50 \cos(r\pi/I) + 0.08 \cos(2r\pi/I)$.

where

'I' is the number of terms in the impulse response weighting sequence on either side of g(0)T.

The filter used in the simulation package had the characteristics specified in fig. F1.

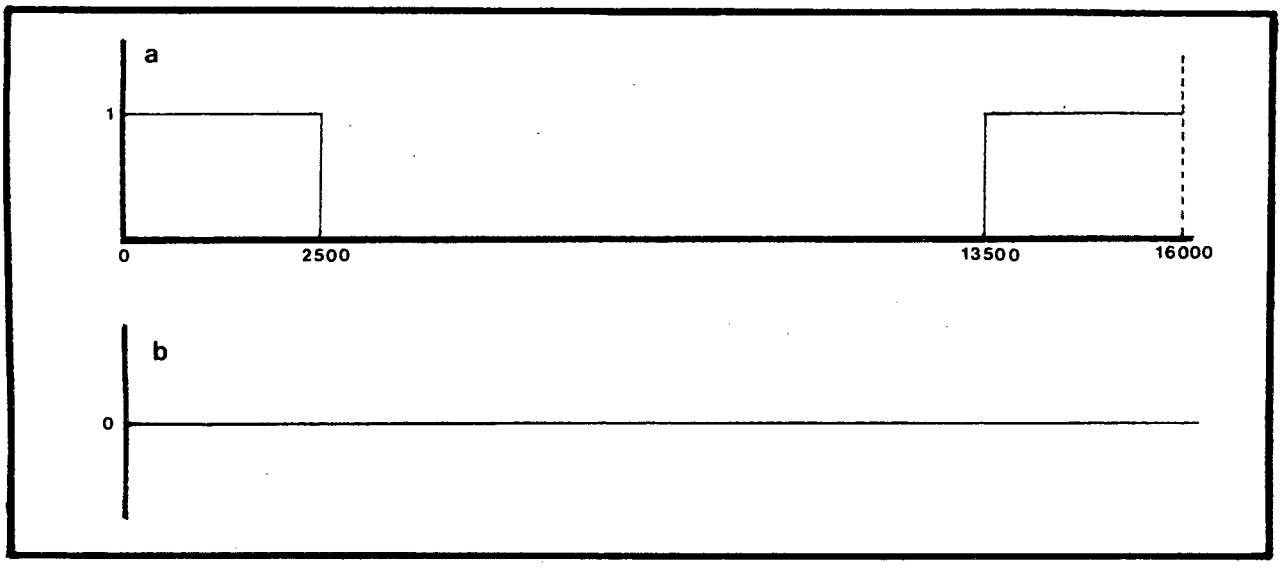


Fig. F1 : (a) Amplitude and (b) Phase characteristics of the FIR filter used in the simulation package.

The filter will first be designed for a sampling frequency, f_s, of 16 kHz and a cut-off frequency, f_c, of 2.5 kHz. If the frequency

samples are seperated by 250 Hz then

$$N = 16.10^3/250 = 64.$$

If ($G_0\ G_1\ G_2\ \ldots\ G_{63}$) is represented by (G), and noting that the value of G_r at a point of discontinuity is given by the average of G_{r-1} and G_{r+1}, then, from fig. F1(a),

(G) = (1, 1, 1, 1, 1, 1, 1, 1, 1, 1, $\frac{1}{2}$, 0, 0, ... , 0, $\frac{1}{2}$, 1, 1, 1, 1, 1, 1, 1, 1, 1).

Representing $e^{-j2\pi/N}$ as W, from equation (F2)

$$g(i)T = \left(\sum_{i=0}^{N-1} G_r . W^{-ir} \right)/N \qquad (F3)$$

Knowing (G), and from equation (F3) :

$$g(i)T = [\ 1 + W^{-i} + W^{-2i} + W^{-3i} + \ldots + W^{-9i} + \tfrac{1}{2}W^{-10i} + \tfrac{1}{2}W^{-54i} + W^{-55i} + \ldots + W^{-62i} + W^{-63i}\ ]/64$$

NOTE : $W^{-63i} = W^{-(64-1)i}$

$= e^{j2\pi(64-1)i/64}$

$= e^{j2\pi i} . e^{-j2\pi i/64}$

$= 1 . e^{-j2\pi i/64}$

$\therefore W^{-63i} = W^{i}$

$\therefore W^{-i} + W^{-63i} = 2\ Re(\ W^{i}\ ) = 2\ \cos(\ 2\pi i/N\ ) = 2\ \cos(\ \varphi\ )$

Therefore,

$$g(i)T = [\ 1 + 2\cos(\varphi) + \ldots + \cos(10\varphi)\ ]/64$$

The values for the impulse response weighting filter are given

below.

Table F1 : Impulse response weighting sequence values - g(i)T.

g(0)T =	0.3125
g(1)T =	0.2645
g(2)T =	0.1466
g(3)T =	0.0205
g(4)T =	-0.0555
g(5)T =	-0.0612
g(6)T =	-0.0917
g(7)T =	0.0243
g(8)T =	0.0377
g(9)T =	0.0184
g(10)T =	-0.0112
g(11)T =	-0.0256
g(12)T =	-0.0165
g(13)T =	0.0041
g(14)T =	0.0176
g(15)T =	0.0143
g(16)T =	0.0000
g(17)T =	-0.0118
g(18)T =	-0.0118
g(19)T =	-0.0023
g(20)T =	0.0074

In this case 20 terms have been used either side of g(0)T and hence the three window functions become :

(a) Hamming : $W_r = 0.54 + 0.46 \cos(r\pi/20)$

(b) Hanning : $W_r = 0.50 + 0.50 \cos(r\pi/20)$

(c) Blackman : $W_r = 0.42 + 0.50 \cos(r\pi/20) + 0.08 \cos(r\pi/10)$.

The values for the impulse response weighting sequence are then modified to the values shown in Table F2 (overleaf).

Table F2 : Modified impulse response weighting sequence.

	Hamming	Hanning	Blackman
g(0)T =	0.3125	0.3125	0.3125
g(1)T =	0.2630	0.2628	0.02618
g(2)T =	0.1433	0.1430	0.1407
g(3)T =	0.0195	0.0194	0.0188
g(4)T =	-0.0507	-0.0502	-0.0472
g(5)T =	-0.0529	-0.0522	-0.0473
g(6)T =	-0.0160	-0.0156	-0.0136
g(7)T =	0.0182	0.0176	0.0146
g(8)T =	0.0257	0.0247	0.0192
g(9)T =	0.0112	0.0106	0.0077
g(10)T =	-0.0060	-0.0056	-0.0038
g(11)T =	-0.0120	-0.0108	-0.0068
g(12)T =	-0.0066	-0.0057	-0.0033
g(13)T =	0.0014	0.0011	0.0006
g(14)T =	0.0047	0.0036	0.0018
g(15)T =	0.0031	0.0021	0.0010
g(16)T =	0.0000	0.0000	0.0000
g(17)T =	-0.0015	-0.0006	-0.0003
g(18)T =	-0.0012	-0.0003	-0.0001
g(19)T =	-0.0002	0.0000	0.0000
g(20)T =	0.0006	0.0000	0.0000

The filter's pulse transfer function is then given by :

$$G(Z) = g(20)T.Z^{-20} + \ldots + g(1)T.Z^{-1} + g(0)T.Z^{0} + g(1)T.Z^{+1} + g(20)T.Z^{+20} \qquad (F4)$$

where

$Z = e^{jwT}$.

The amplitude vs frequency and the phase vs frequency plots are given in fig's. F2 and F3. Fig. F3 also shows the effect of changing the parameter 'I' and the sampling frequency (f_s). Fig. F4 gives an indication of the relative effects of using the

different window functions.

It should be noted that the filter design specified by equation (F4) is not a real-time implementation. In order to make the filter realisable in real time there should be no terms of Z raised to a positive power. However, because this filter was used in the simulation package (which itself is not a real-time system) purely for interpolation and measurement purposes it was preferable to use the non-real time version as in this manner it is possible to achieve a zero phase shift characteristic.

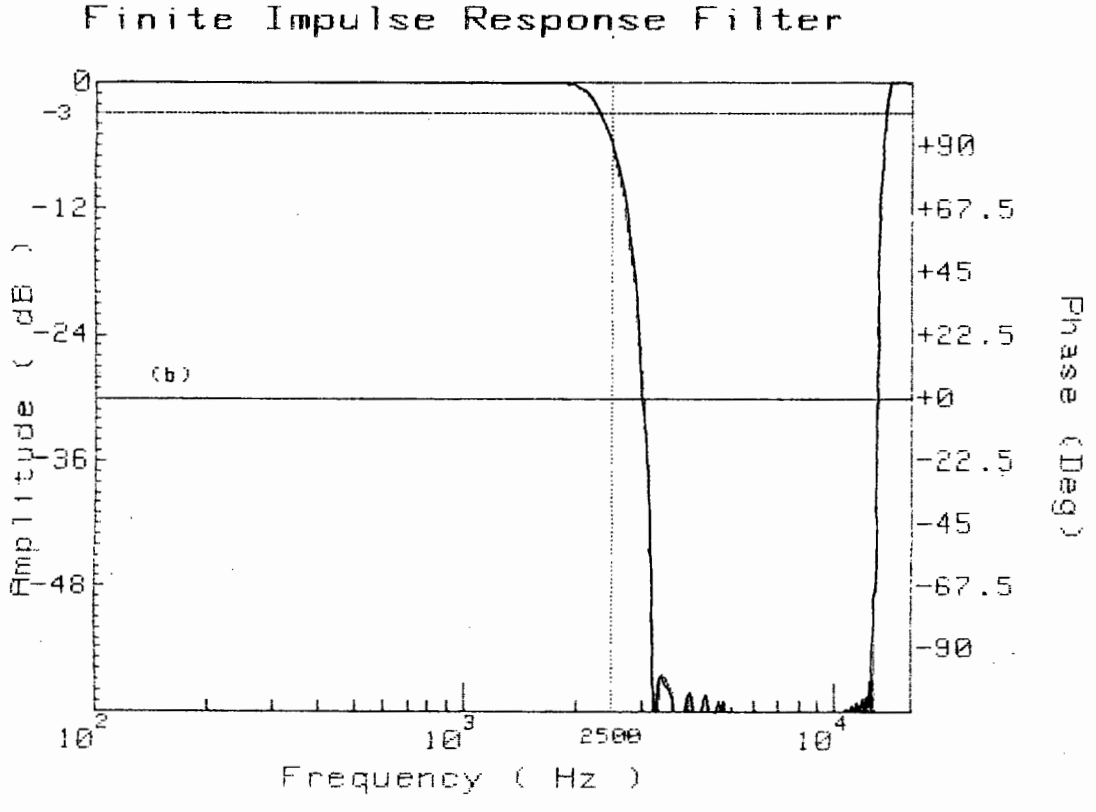
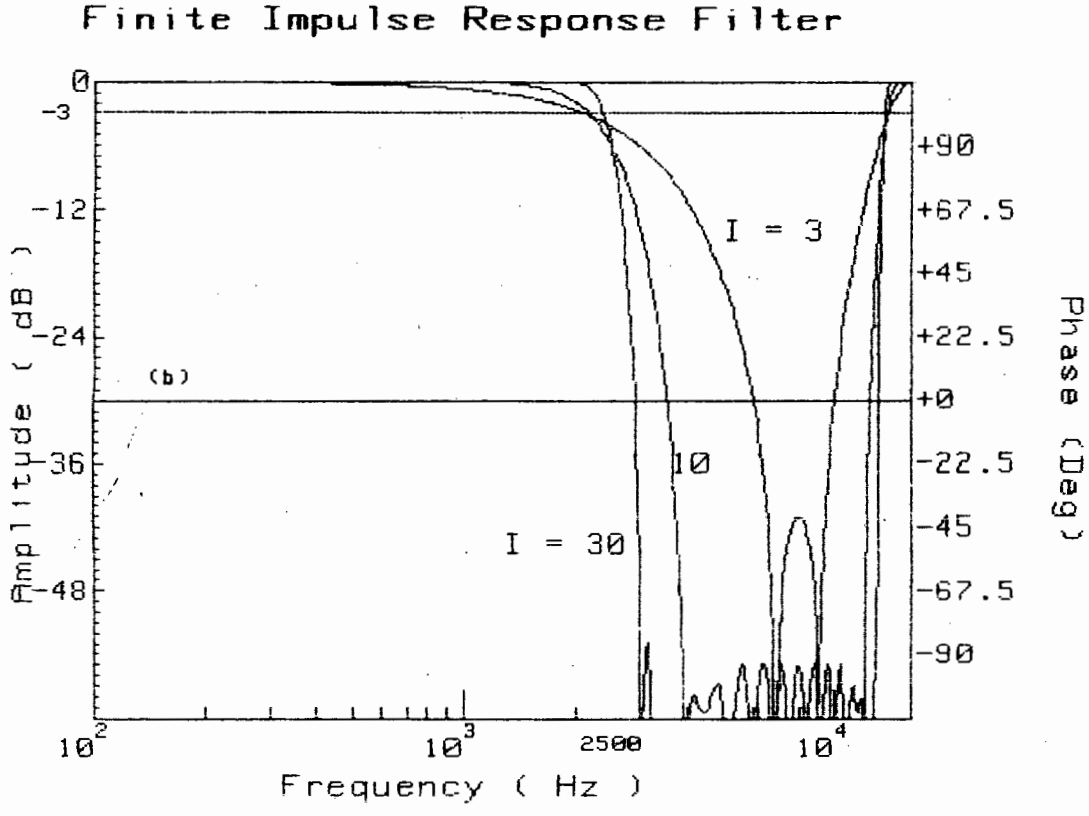


Fig. F2 : (a) Amplitude and (b) Phase response of the FIR filter specified by equation F4.



Sampling frequency	=	16000
Filter cut-off frequency	=	2500
Type of window	=	HAMMING

Fig. F3 : Variation of the amplitude (a) and phase (b) characteristics for different values of 'I'.

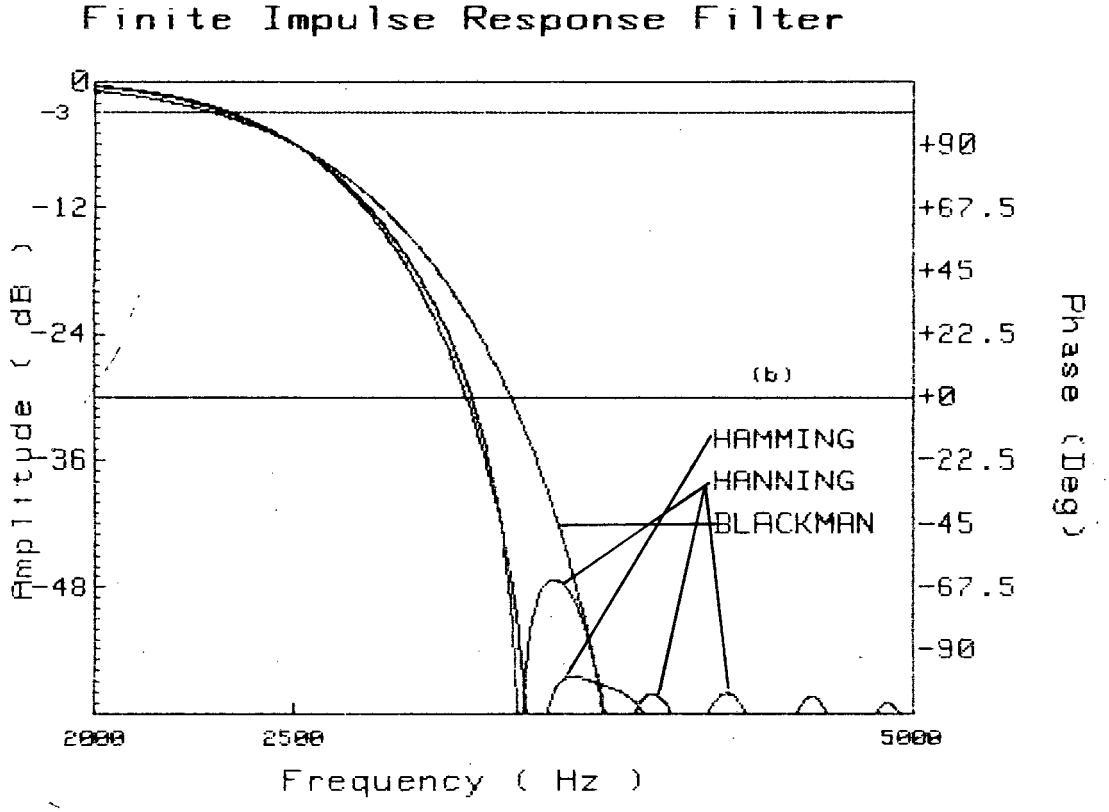


Fig. F4 : Effect of the different window functions on the (a) amplitude and (b) phase characteristics of the filter.

APPENDIX G

This appendix contains the table which gives the critical values of Z for various levels of significance. This table is used in conjunction with the Z-score test to test the significance of results from the subjective tests.

Table G.1.

Level of Significance	0.10	0.05	0.01	0.005
Critical values of Z for One-Tailed Tests	-1.28 1.28	-1.645 1.645	-2.33 2.33	-2.58 2.58
Critical values of Z for Two-Tailed Tests	-1.645 1.645	-1.96 1.96	-2.58 2.58	-3.08 3.08

One-tailed, or one-sided, tests are used when testing values to one side of the mean. The procedure for testing a hypothesis that one system (eg. one codec) is better than another involves the use of one-tailed tests. This is different to testing whether one system is better or worse than another.