AUTOMATIC CONTROL OF ATTENUATION IN A DIGITAL COMMUNICATION NETWORK

by

JUAN MARCIAL FONSECA GUERRA

A thesis submitted to the
University of Aston in Birmingham
for the degree of
Doctor of Philosophy

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Electrical and Electronic Engineering

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To my wife and children,

whose love and patience were

beside me during these three years.

To my beloved parents.

Perhaps the most valuable result of all education is the ability to make yourself do the thing you have to do, when it ought to be done, whether you like it or not; it is the first lesson that ought to be learned; and however early a man's training begins, it is probably the last lesson that he learns thoroughly.

Thomas Henry Huxley

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Summary

This thesis describes the investigation of an adaptive method of attenuation control for digital speech signals in an analogue-digital environment and its effects on the transmission performance of a national telecommunication network. The first part gives the design of a digital automatic gain control, able to operate upon a P.C.M. signal in its companded form and whose operation is based upon the counting of peaks of the digital speech signal above certain threshold levels.

A study was made of a digital automatic gain control (d.a.g.c.) in open-loop configuration and closed-loop configuration. The former was adopted as the means for carrying out the automatic control of attenuation. It was simulated and tested, both objectively and subjectively.

The final part is the assessment of the effects on telephone connections of a d.a.g.c. that introduces gains of 6 dB or 12 dB. This work used a Telephone Connection Assessment Model developed at The University of Aston in Birmingham.

The subjective tests showed that the d.a.g.c. gives advantage for listeners when the speech level is very low. The benefit is not great when speech is only a little quieter than preferred. The assessment showed that,when a standard British Telecom earphone is used,insertion of gain is desirable if speech voltage across the earphone terminals is below an upper limit of -38 dBV. People commented upon the presence of an adaptive-like effect during the tests. This could be the reason why they voted against the insertion of gain at level only little quieter than preferred, when they may otherwise have judged it to be desirable.

A telephone connection with a d.a.g.c. in has a degree of difficulty less than half of that without it. The score Excellent plus Good is 10-30% greater.

Key Words

TELEPHONY, DIGITAL TRANSMISSION, PULSE-CODE MODULATION.

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CHAPTER 1

INTRODUCTION

As progress is made towards a wholly digital telephone network[1,2,3,4,5] different stages will present a variety of problems and improvements. Among the problems, may be mentioned interfacing and matching difficulties such as the interface between two different PCM systems[6,7] and differences in circuit losses[8] between the talker and point of measurement.

During the many years before a fully-digital network is established, many connections will be made partly over analogue circuits and partly over digital circuits[3]. There will be wide variations in the loudness loss of the different combinations of analogue circuits which may be between digital and analogue parts of the network[4]. An analogue connection with a high loss will produce at the output of the decoder a signal having both a low level and a poor signal-to-quantisation noise ratio. Where local exchanges are digital but not the subscriber lines, to achieve improvements (such as enabling more lossy subscribers' lines to be used[3]) if transmission loss between local exchanges is generally reduced in the whole telephone network, or improvement of the return loss between telephone and local circuits, etc.), it could be necessary to introduce loss automatically among other arrangements.

There is thus a need to adjust gain or attenuation at exchanges forming interfaces between analogue and digital portions of the network. This adjustment cannot be preset because the analogue signal will arrive at the interface over different connections with different loudness loss. A gain control is therefore required which will automatically adapt its setting to suit the loudness of the analogue input signals. If such an automatic gain control operates upon the analogue signals, it needs to be provided individually for each circuit. However, equipment which processes the multiplexed output of a PCM encoder can be shared between a group of circuits.

To provide automatic gain control, a suitable parameter of the signal, representing the speech level, must be measured. When speech is transmitted by periodic sampling, a convenient parameter to use is the proportion of samples whose magnitudes exceed a given threshold value[9].

Previous work[10,11,12], involved the design and construction of a device able to introduce gain automatically which operated in closed-loop configuration and processed 30 speech channels. It inserted gains in steps of six dB, avoided putting maximum gain whenever there was no speech, and processed the PCM word in its companded form, using for that a programmable read-only memory as look-up table. However, it oscillated in the presence of a sine-wave.

This thesis describes subsequent work aimed to improve the device by eliminating the oscillation. In Chapter 2, the background of the previously reported d.a.g.c. is given, together with a mathematical model to explain the oscillation. Three different solutions are discussed.

Chapter 3 gives the design and the simulation of the d.a.g.c. in open-loop configuration selected in Chapter 2, able to introduce gain as well as attenuation with 6 dB and 3 dB steps. The simulation was carried out just for one channel, because of the slow speed of operation of the computer available. Its behaviour was studied in the light of several time constants for the onset of gain. Sine-wave, Gaussian noise and speech signals were used as inputs.

Chapter 4 presents the subjective tests made to a telephone network with d.a.g.c. in, able to introduce just gain, to find out the acceptance of it among listeners and the improvements and impairments they detect when it is in use. The 6 dB loss insertion was left out because it represents the solution to other kinds of problems[5,6,7,13,14].

Chapter 5 studies the introduction of noise due to the d.a.g.c.

Chapter 6 describes the assessment of the effects on the transmission performance of a national telephone network which is mixed (analogue and digital), and has a d.a.g.c. incorporated, introducing gain because the level reaching the listener at the digital side has both low level and poor signal-to-noise ratio. The operation of it will be in the direction analogue-digital only. The assessment was carried out using a computer simulation of a telephone connection devised at the Electrical and Electronic Engineering Department of this University[31,32,33] called Telephone Connection Assessment Model (TCAM).

Chapter 7 presents the conclusions and suggests further work.

CHAPTER 2

DIGITAL AUTOMATIC GAIN CONTROL

2.1 PULSE CODE MODULATION (PCM) SYSTEMS

A PCM system, as shown in Fig. 2.1, involves the following basic processes[15,16,17]:

1. Sampling
2. Quantizing
3. Coding
4. Transmission
5. Conversion of the group of binary digits into a set of amplitudes in time sequence.
6. Reconstruction of the analogue signal in time produced at step 5.

As Nyquist's sampling theorem lays down[17,18,19,20] a continuous waveform, whose maximum frequency is f_{max} can be represented by and reconstructed from 2 x f_{max} samples. Even though the speech signals, for telephony, are limited to a 3.4 KHz, because of lack of ideal filters, the sampling rate is 8000 Hz.

The second process allows binary encoding. The analogue signal is divided into a number of discrete levels, each of them is described by a unique binary code. If a sample lies within a range it will be allocated the particular code assigned to that region.

This process introduces an error, called quantization error (nq)[11,19,20], whose maximum value is equal to one half

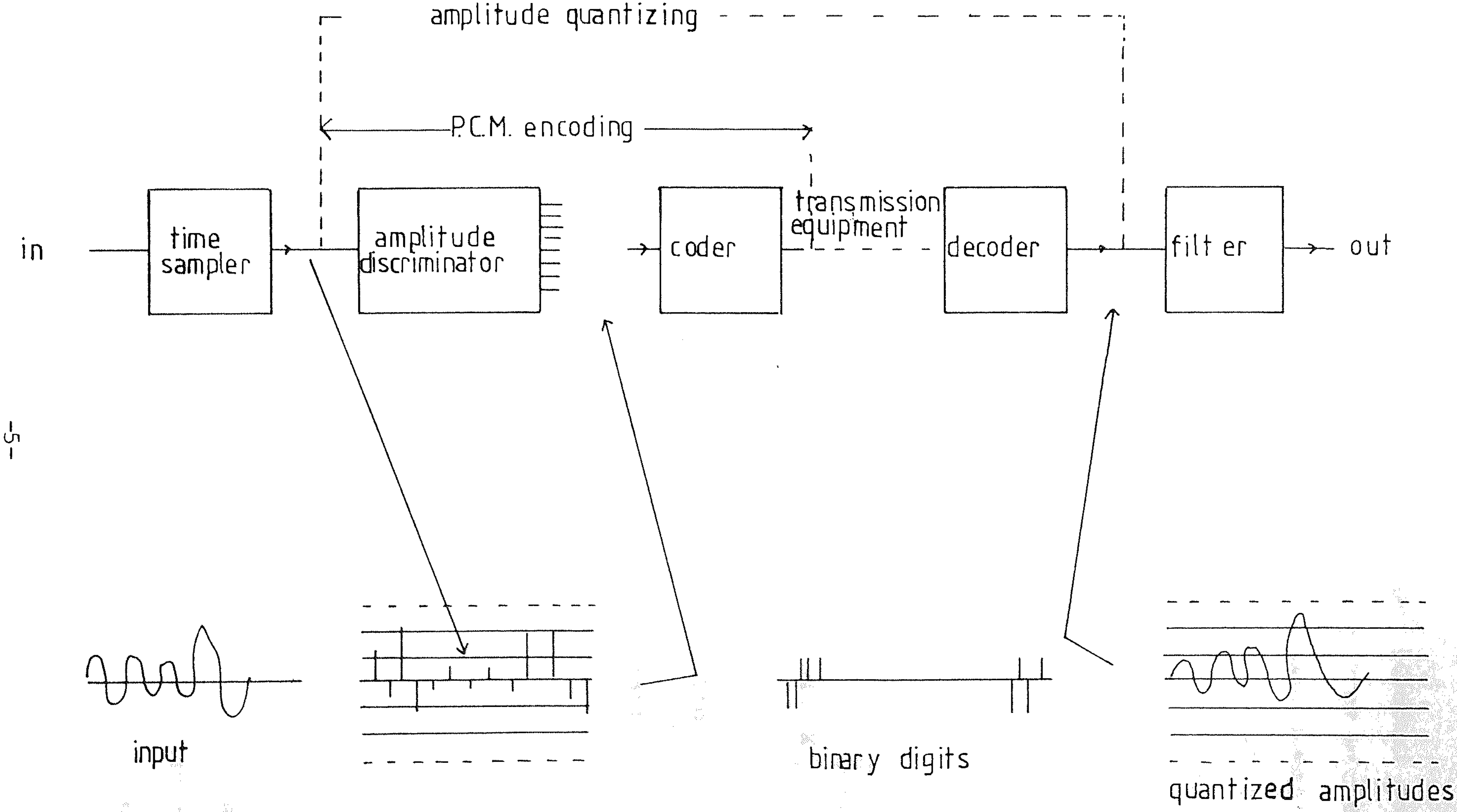


Fig. 2.1 PCM transmission Systems

of the amplitude of the appropriate range. It may be reduced increasing the number of increments.

The encoding will be linear if the interval between decision levels are uniform. As this type of encoding has signal-to-nq ratio for low amplitude signals lower than for high amplitude signals, a non-linear encoding which guarantees a signal-to-nq ratio constant over the whole voltage range, is preferred. This is achieved using the A/87.6/13 Companding Law, 8-bit system(17,21). This is recommended by both CCITT[16] and CEPT(15). It is expressed by(17,21):

$$\frac{y}{Y} = \frac{A}{1+\log A} \cdot \frac{x}{X} \qquad |x| \leq X/A$$

$$\frac{x}{X} = \frac{1}{1+\log A}\{1 + \log A(x/X)\} \qquad \frac{X}{A} \leq |x| \leq X$$

where $\frac{x}{X}$ = input value-to-limiting value ratio

$\frac{y}{Y}$ = output value-to-limiting value ratio.

$A = 87.6$.

Fig. 2.2. shows the A-law with 13 segments. 7 bits are used to encode each sample, the first three bits specify the segment and the 4 other bits specify the magnitude of the quantised sample. The 8th bit is the sign digit.

2.1.1 <u>32-channel PCM System</u>

British Telecom has introduced(23) the second generation of PCM system which is the 32-channel system, recommended by CEPT and CCITT. As shown in Fig. 2.3, the 32 channels are

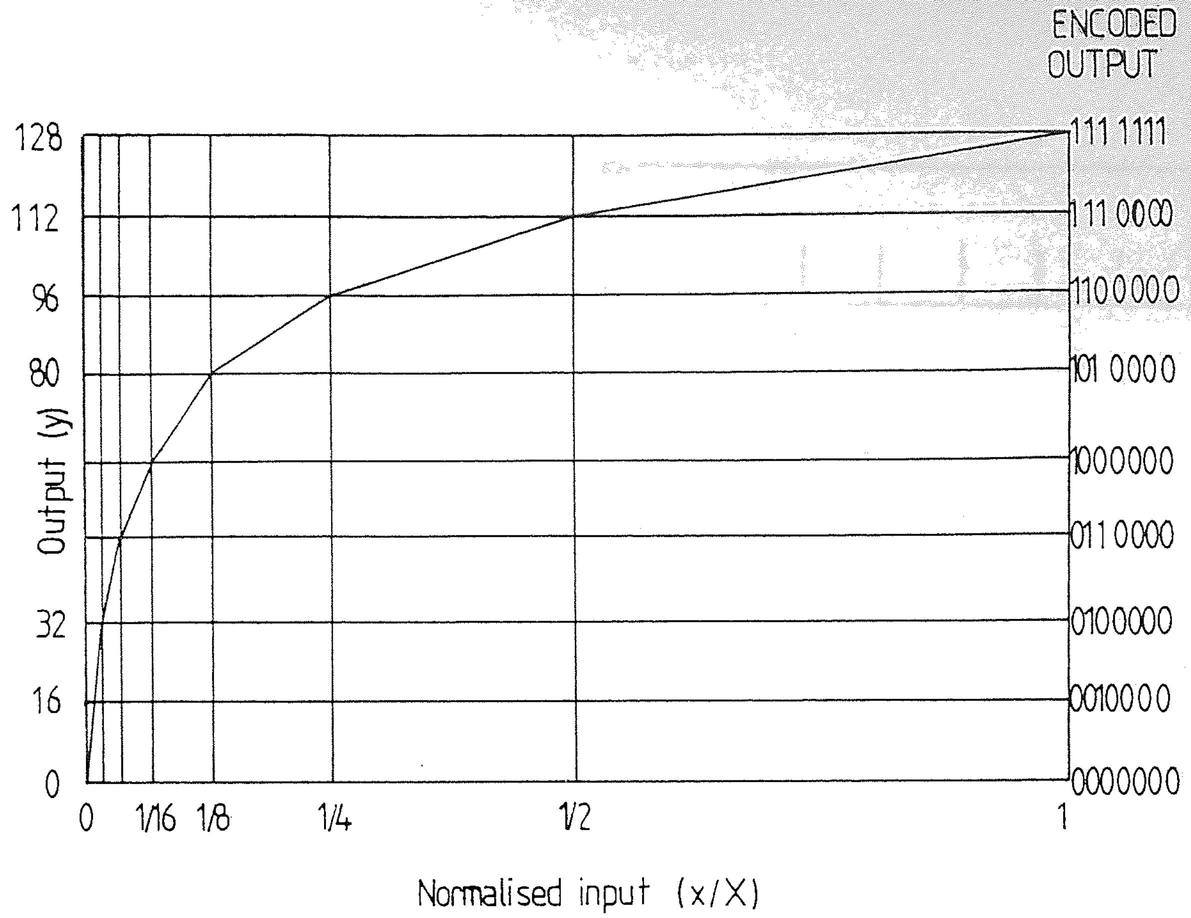


Fig. 2.2 Compression characteristic
(Positive segments only)

names TSO, TS1, TS2, ...TS31. TSO (frame alignment and TS16 (Signalling information) convey information about signalling for the remaining channels and frame and multi-frame alignment. The rest of the channels are speech channels.

2.2 PRINCIPLE OF OPERATION OF DIGITAL AUTOMATIC GAIN CONTROL

The d.a.g.c. is designed to work with the 30 speech channels and to avoid the unit introducing maximum gain when there is no speech.

The principle is based upon the counting of peaks which occur in the speech signal; the number of peaks counted is compared with a model speech signal defined by the negative

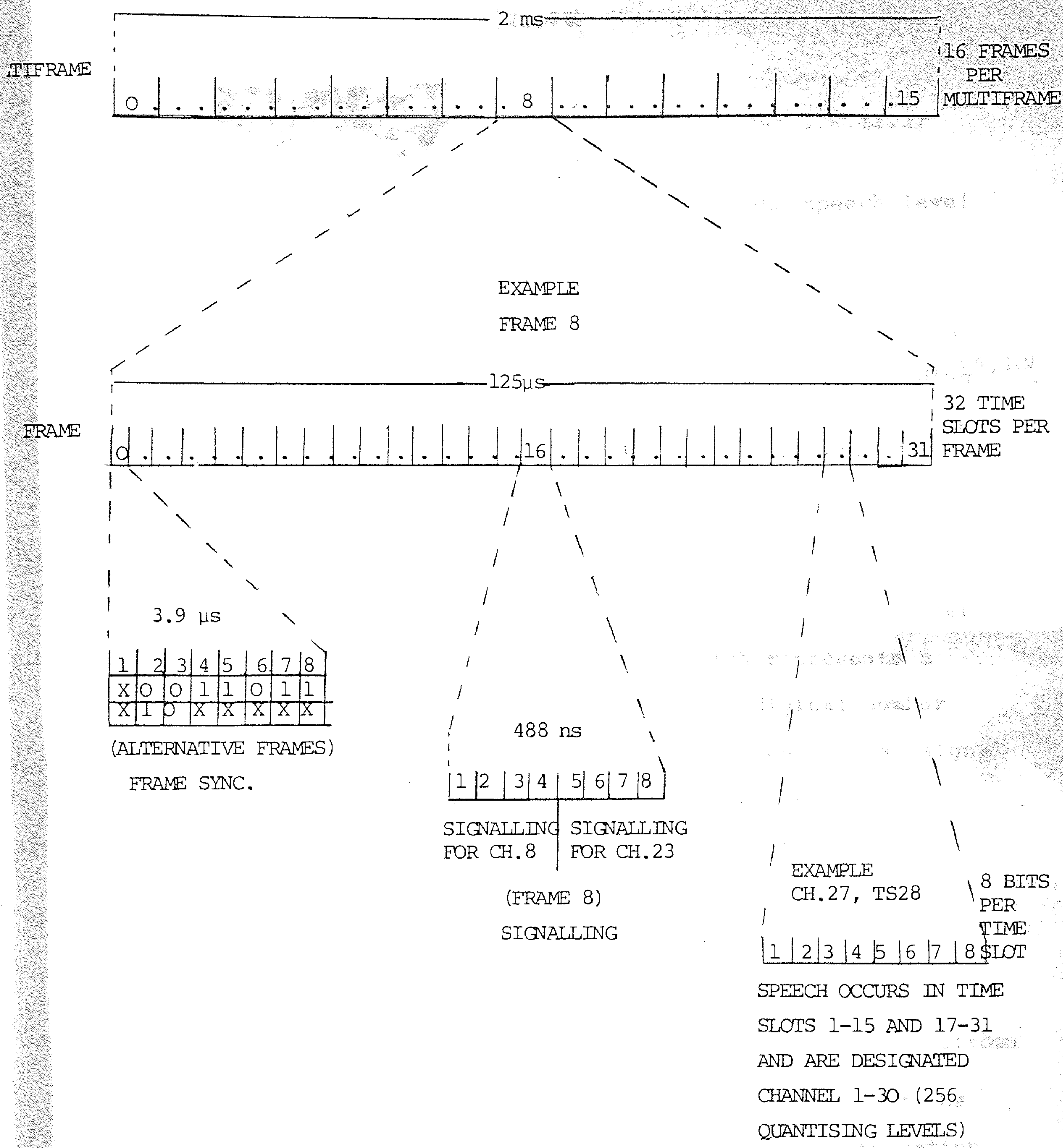


Fig. 2.3 Relationship between digit channel and frame for 32-channel PCM system

exponential distribution[21,24]:

$$P(X > x_r) = e^{-\sqrt{2}\,x_r} \qquad (2.1)$$

where x_r = | Instantaneous speech voltage/RMS speech level |

2.3 DESIGN PRINCIPLE

As the insertion of gain or attenuation is made at a digital stage, the design is based upon complete recoding[9,10]. Let U be the digital representation of an analogue sample V. If this code number U is recoded to a new digital number X_d, which represents an analogue sample KV, then a loss of -20 logK has been introduced. If K > 1, gain is introduced. For simplicity in the circuitry the different values for K may be 1/2; 1;2 and 4, which represents a shift to the right or to the left of the digital number if it was in its linear form. If K = 1, the digital signal is unchanged. In all the d.a.g.c. studied the recoding was achieved by using a programmable read-only memory, p.r.o.m., and in this way the digital signal may be used in its companded form[10,13].

Using the p.r.o.m. to introduce gain or attenuation, we are performing the equivalent to the following algorithm:

If the digital signal is in the first segment of the A-companding law, where it is linear, gain or attenuation is achieved by shifting to the left or to the right[7]. If the PCM word is outside the first segment, 6 dB the attenuation is achieved by substracting 00010000 from the digital signal; to achieve gain we just add 00010000 to the PCM word[7]. To achieve 12 dB gain (K = 4), we may add 00010000 twice or 00100000 once to the digital signal.

2.4 OPERATION OF THE p.r.o.m. AS A LOOK-UP TABLE

As the PCM law used has 256 intervals (128 positives, 128 negatives) and 3 gains are provided, a 1024-word, 8-bit p.r.o.m. is sufficient. This has a 10-bit address word which may be split in two: the last 8 bits are the PCM itself and the other two represent the different setting values, as follows:

01	6 dB	of Attenuation
00	0 dB	of Gain
10	6 dB	of Gain
11	12 dB	of Gain

These two bits are formed by a circuit called the controller, which takes its information from the output if in closed-loop configuration or from the input if in open-loop configuration. The controller is capable of working manually or automatically.

2.5 THE CONTROLLER IN CLOSED-LOOP CONFIGURATION

2.5.1 Principle

Fig. 2.4 shows a diagram of the closed-loop configuration.

The controller works in the following way: during a time interval T, it counts how many occasions the output is away from certain threshold level T_L. The occurrence of

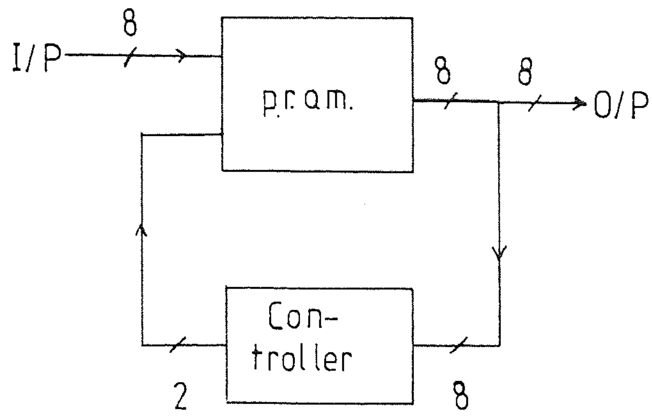


Fig. 2.4 Closed-loop configuration

instantaneous speech samples above T_L may be weighted W_1 and those below, W_2. As low values are more probable than high ones[17,18], the weight W_2 is assigned the value of -1. The number of samples is restricted by the store capacity used in the hardware and by the size of W_1.

Each time the sample is above T_L, W_1 is added to a mid-value, M_o. If it is below T_L, W_2 is added. To avoid setting of maximum gain because of silence periods[12], $W_3=0$ is added each time there is no speech. The presence of the speech is detected using a speech detector, which is formed by a simple digital comparator.

The new value, $N_v = M_o + W_i$, where i = 1, 2 or 3, is compared with two different values, L_1 and L_2, where $L_1 < L_2$. When $N_v \leq L_1$, T_L has seldom been reached and, therefore, gain is introduced. If $N_v \geq L_2$, T_L has frequently been reached and, therefore, attenuation is required.

The figures M_o, W_1, L_1 and L_2 may be chosen in such a way that when N_V is between L_1 and L_2 neither attenuation nor gain is necessary; that is, the existing setting is maintained.

Before choosing T_L, it is necessary to establish the normal talker level. This is the speaker whose clipping level could be situated [17,21,10] 12-18 dB above its r.m.s. level. Taking account of the companding law used, the threshold level can be at 6 dB below the clipping level of the A/87.6/13 PCM law encoder; that is where the seventh segment starts[17].

Using equation (2.1), for the normal speaker, whose clipping level is, say, 12 dB above its r.m.s. level, the probability of reaching T_L is:

$$p_o = 6.00\%$$

Considering the talker range as $\pm$ 3 dB, typical values for talkers who are entering the weak and strong levels, are:

$$p_2 = 1.86\% \quad \text{weak ones}$$
$$p_3 = 13.57\% \quad \text{strong ones}$$

Therefore, if T_L has been reached with a probability p which is:

$$1.86 \leqslant p \leqslant 13.57\% \qquad (2.2)$$

the level is considered normal and the gain is not adjusted, because we are in closed loop-configuration.

The following inequalities may be written:

$$L_2 > M_o + p_3 W_1 S_n + (1 - p_3) W_2 S_n \qquad (2.3a)$$

$$L_1 < M_o + p_2 W_1 S_n + (1 - p_2) W_2 S_n \qquad (2.3b)$$

where S_n is number of samples in time interval T, that is $S_n = 8000T$ if T in seconds.

W_1 and M_o (W_2 is already - 1), must be chosen to convert (2.3a) and (2.3b) into equalities when p is not satisfying relation (2.2).

The period T is chosen to avoid rapid fluctuation in the gain. It is an important parameter and its selection will be dealt with in detail in Chapter 4.

If T is, say, 100 ms, then $S_n = 800$ samples. L_2 is determined by the store capacity. To use the circuit efficiently, L_1 is chosen equal to zero and L_2 maximum capacity of the store device used; thus if a z-bit device is being used, then

$$L_2 = 2^z - 1 \qquad (2.4)$$

For z, say, 10 and solving (2.3) and (2.4):

$M_o = 754$

$W_1 = 31$

$L_2 = 1023$

As this confirmation is closed-loop, the output will become normal (that is, in equilibrium) after the corresponding setting values have been chosen; unless the signal is too weak and requires more than 12 dB of gain or too strong and needs more than 6 dB of attenuation.

A d.a.g.c. to the above design was built and tested[10,12] by connecting it between the send and the receive terminals of a standard Conference of European Postal and Telegraph Administration (CEPT) 30-channel PCM system. Under manual control, the system was found to operate correctly on every channel. Tests were carried out under automatic control, using input signals consisting of pure tones, Gaussian noise and speech. Satisfactory operation was obtained when noise and speech signals were transmitted, and the measured results agreed with the calculated characteristics. Fig.2.5 shows the results with Gaussian noise input. When sinusoidal

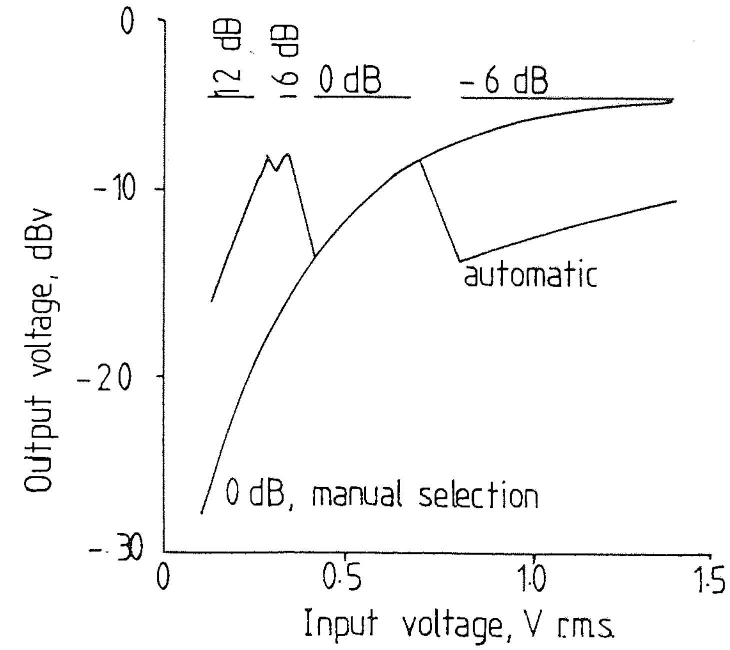


Fig. 2.5 Characteristic i/p/o/p for the closed loop configuration d.a.g.c. with Gaussian noise as i/p

signals were transmitted, it was found that the gain setting of the system sometimes oscillated.

In order to investigate the reason for this oscillation, a mathematical model of the d.a.g.c. in closed-loop configuration was made and will be discussed in the next section.

2.5.2 Mathematical Model of the d.a.g.c. in Closed-loop Configuration

The d.a.g.c. in closed-loop configuration may be visualised as a 'percentage analyser', that is a device that calculates and analyses the percentage of time the output is above certain threshold level, T_L. Fig. 2.6 shows the model.

$p(nT)$: percentage of time the signal $\tilde{y}$ is above T_L

$K_p(nT)$: setting value depending on $p(nT)$

$$K_p(nT) = 1 \text{ if } p_2 < p(nT) \leqslant p_3$$
$$K_p(nT) = \tfrac{1}{2} \text{ if } p(nT) > p_3$$
$$K_p(nT) = 2 \text{ if } p(nT) \leqslant p_2$$

$K_o(nT)$: real setting, which multiply the input, therefore its value depends upon $K_p((n-1)T)$ and $K_o((n-1)T)$.

In general, $p(nT)$ is a function of the r.m.s. level of the signal, the output $\tilde{y}$,

$$p(nT) = f(\tilde{y})$$

where $\tilde{y} = T_L/(\text{r.m.s. level of the signal})$.

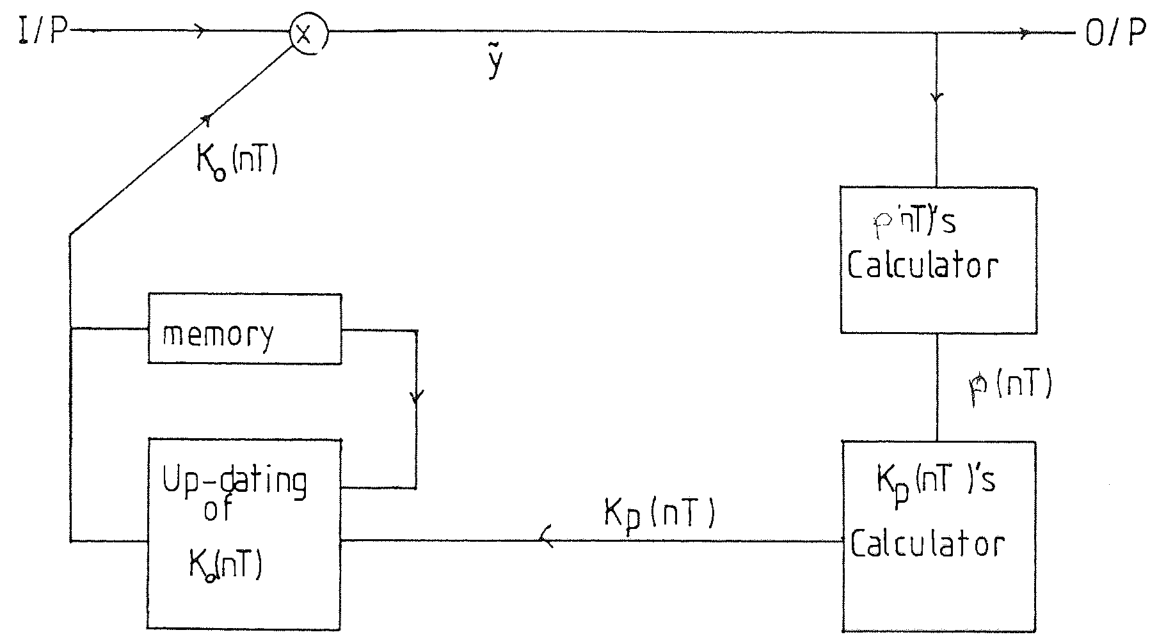


Fig. 2.6 Mathematical model of the d.a.g.c. in closed-loop configuration

p(nT), percentage of time the signal is above certain threshold level

$K_p(nT)$, setting value depending on p(nT)

$$K_p(nT) = 1 \text{ if } p_2 \leq p(nT) \leq p_3$$

$$K_p(nT) = 2 \text{ if } p(nT) > p_2$$

$$K_p(nT) = \tfrac{1}{2} \text{ if } p(nT) < p_3$$

$K_o(nT)$, real setting, which multiply the I/P, therefore its value depends upon $K_o|(n-1)T|$ and $K_p|(n-1)T|$

$K_p(nT)$ is determined from the definition of normal talker; if $p_2 \leqslant p(nT) \leqslant p_3$ then $K_p(nT) = 1$, otherwise $K_p(nT) = 1/2$ or 2.

$$Ko(nT) = g(K_p((n-1)T), Ko((n-1)T))$$

As $g(\ )$ is not a proper function, it will not be used further.

It is necessary to calculate the value of $p((n+1)T)$ as a function of $p(nT)$ and $K_p(nT)$.

Then,

$$p((n+1)T) = f\{\tilde{y}/K_p(nT)\}$$

Rearranging :

$$p((n+1)T) = f\{\frac{f^{-1}(p(nT))}{K_p(nT)}\}$$

Now, if $K_p(nT) = 1$, that is the level $\tilde{y}/K_p(nT)$ is in the range of the normal talker defined, gain should be kept,

$$p((n+1)T) = p(nT)$$

Let y_1 and y_2 be the extremes of the range of level for no change of gain is necessary and p_4 and p_5 the percentages of time that y_1 and y_2 respectively, will be above the threshold level T_L. Figure 2.7 shows this arrangement for r.m.s. values of y_1 greater than y_2.

Let Δy be the ratio between y_1 and y_2 expressed in dB.

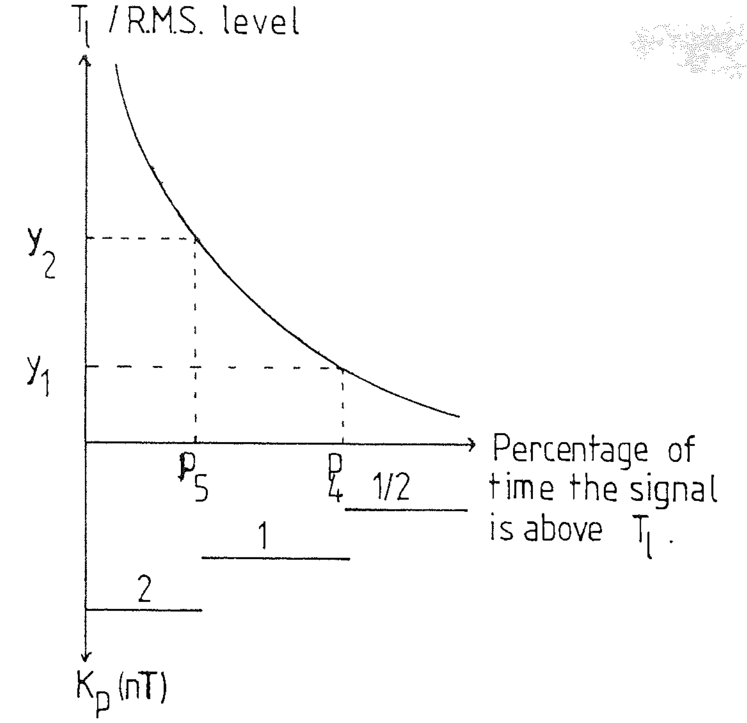


Fig. 2.7 R.m.s. speaker level in function of percentage of time above T_L

$$\lim_{p(nT)\to p_4} p((n+1)T) = \lim_{p(nT)\to p_4}\{f\left|\frac{f^{-1}(p(nT))}{K_p(nT)_a}\right|\} = f\{\frac{y_1}{K_p(nT)_a}\}$$

and

$$\lim_{p(nT)\to p_5} p((n+1)T) = \lim_{p(nT)\to P_5} f\{\frac{f^{-1}(p(nT))}{K_p(nT)_g}\} = f\{\frac{y_2}{K_p(nT)_g}\}$$

$K_p(nT)_g$ is $K_p(nT)$ correspondent to gain (greater than 1)

$K_p(nT)_a$ is $K_p(nT)$ correspondent to attenuation (less than 1)

To avoid oscillation, it is necessary that:

$$f\{\frac{y_1}{K_p(nT)_a}\} = p_5 \quad \text{and}$$

$$f\{\frac{y_2}{K_p(nT)_g}\} = p_4$$

which gives:

$$y_1/K_p(nT)_a = y_2 \tag{2.5}$$

and

$$y_2/K_p(nT)_g = y_1 \tag{2.6}$$

Combining (2.5) and (2.6) :

$$K_p(nT)_g . K_p(nT)_g = 1$$

Calling $K_p(nT)_g$ and $K_p(nT)_a$, expressed in dB, as K_g and K_a respectively; the following table is formed:

1. $Kg + Ka = 0$

1-1 : $\Delta y < Kg$	Oscillation
1-2 : $\Delta y > Kg$	No oscillation

2. $Kg + ka \neq O$

2-1 : $\Delta y > Kg$ & $\Delta y > |K_a|$ No oscillation

2-2 : $\Delta y > Kg$ & $\Delta y < |K_a|$ Damping possible

2-3 : $\Delta y < Kg$ & $\Delta y > |K_a|$ Damping possible

2-4 : $\Delta y < Kg$ & $\Delta y < |K_a|$ Oscillation possible.

If, for example, $y_1/y_2 = 1$ and $K_a + K_g = O$[9], then oscillation will occur even for speech signals.

Now, it is appropriate to study the design detailed at Section 2.5.1.

2.5.3 Speech Signals

To calculate p(nT), eq. (2.1) is used:

$$p(nT) = e^{-\sqrt{2}\tilde{y}}$$

$$f^{-1}\left|p(nT)\right| = \frac{-\log p(nT)}{\sqrt{2}}$$

$$p((n+1)T) = e^{\frac{+\log p(nT)}{K_p(nT)}}$$

If $K_p(nT) = 1$, then $p((n+1)T) = p(nT)$, the gain should be kept, in other words, equilibrium has been reached.

In the present case, the normal talker is defined as that being 9 dB and 15 dB below clipping level and T_L is 6 dB below it. Then $p_4 = 13.6\%$ and $p_5 = 1.86\%$.

$K_g = 2$ and $K_a = 1/2$

Therefore:

$$\lim_{p(nT) \to p_4} p(n + 1)T = f\{\frac{y_1}{K_p(nT)_a}\} = e^{+(\log 0.136)/\frac{1}{2}} = 0.0186$$

$$\lim_{p(nT) \to p_5} p(n + 1)T = f\{\frac{y_2}{K_p(nT)_g}\} = e^{+(\log 0.0186)/2} = 0.136$$

Therefore, with speech signals and the parameters chosen there will not be oscillation at all.

2.5.4 Sine-wave Signal

This is analysed taking the parameters already given.

$P(X \geq x_r)$ has to be chosen for a sine-wave. Fig. 2.8 illustrates this:

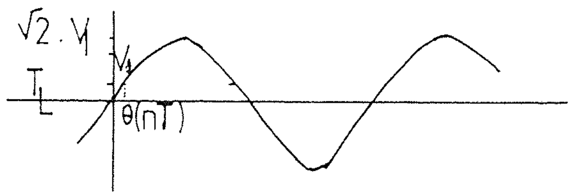


Fig. 2.8 Sine-wave used to calculate $P(X > x_r)$

From the above figure, the percentage of time the signal is above T_L is:

$$p(nT) = \frac{180 - 2\theta(nT)}{180}$$

$$\theta(nT) = 90(1 - p(nT)) \qquad (2.7)$$

We will have:

$$\sqrt{2}V_I\sin\theta(nT) = T_L \tag{2.8}$$

$\theta(nT)$ yields to $p(nT)$ and this to $K_p(nT)$

$$\sqrt{2}K_p(nT)V_I\sin\theta\{(n+1)T\} = T_L \tag{2.9}$$

Using (2.8) and (2.9)

$$\text{Sin}\theta\{(n+1)T\} = \frac{\text{Sin}\theta(nT)}{K_p(nT)}$$

$$\text{Sin}\theta\{(n+1)T\} = \frac{\text{Sin}90(1-p(nT))}{K_p(nT)}$$

$$\text{Sin}90\{1-p((n+1)T)\} = \frac{1}{K_p(nT)}\cdot\text{Cos}90p(nT)$$

$$\text{Cos}90p((n+1)T) = \frac{1}{K_p(nT)}\text{Cos}(90\,p(nT))$$

$$p(n+1)T) = \frac{1}{90}\text{Cos}^{-1}\{\frac{\text{Cos}90p(nT)}{K_p(nT)}\}$$

If $K_p(nT) = 1$

$$p((n+1)T) = p(nT)$$

and the equilibrium has been reached.

Checking for oscillation with $p_4 = 13.6\%$ and $p_5 = 1.86\%$:

$$\frac{1}{90}\text{Cos}^{-1}\{\tfrac{1}{2}\text{Cos}90(0.186)\} = 67.7\%$$

$$\frac{1}{90}\text{Cos}^{-1}\{\frac{1}{\frac{1}{2}}\text{Cos}90(0.136)\} = 0^*$$

* Actually, the equation can not be satisfied, which means argument of Cos^{-1} is > 1

Therefore, the d.a.g.c. is not reaching equilibrium.

Let us look for $\Delta y = y_1 - y_2$

Using (2.9), and calling $K_p(nT)V_1 = \sigma_1$ and

$K_p|(n+1)T|V_1 = \sigma_2$,

$$\sqrt{2}\ \sigma_1 \mathrm{Cos}90\ p_4 = T_L$$

$$\sqrt{2}\ \sigma_2 \mathrm{Cos}90\ p_5 = T_L$$

We want $\dfrac{\sigma_1}{\sigma_2} = y_2/y_1$

$(\sigma_1 - \sigma_2)$ in dB is approximately 0.2 dB.

This is case 1-1 of oscillation.

In order to avoid oscillation with the sinewave we must achieve:

$$\frac{\mathrm{Cos}90p_4}{\mathrm{Cos}90p_5} = 1/2$$

$$\mathrm{Cos}90p_4 = 0.5\mathrm{Cos}90p_5$$

The maximum argument for left-hand side cosine is 1/2, then $p_4 = 0.66667$ is minimum value and $p_5 = 0$.

For speech signals, these figures correspond to a range of 27 dB.

Concluding, the oscillation in the presence of a sine-wave signal occurs because a sinusoidal signal has a different distribution of voltage with time from speech signals. If we assign a normal-operation range of 6 dB for speech signal, this will correspond to about 0.2 dB for a sinewave. Consequently the system never will reach equilibrium.

2.6 METHODS OF AVOIDING OSCILLATION

2.6.1 General

Although the closed-loop system described in Section 2.5 operated successfully with speech signals, the gain setting oscillated when sinusoidal signals were transmitted. In practice, pure tones are used as test signals for lining up circuits and sinusoidal carriers are used for data transmission over telephone connections. It is therefore necessary that a d.a.g.c. system shall operate satisfactorily for any channel which is transmitting a sinusoidal signal.

Two possible modifications to the previous system were considered:

(1) Associating with the closed-loop d.a.g.c. an ancilliary circuit to detect the presence of a sinusoidal input and alter the operation of the system accordingly.

(2) Replacing the closed-loop d.a.g.c. with an open-loop system.

2.6.2 Closed-loop with Predictor

If it is desired to develop the first idea, it could be done if the presence of the sinewave is detected initially. When a sine wave is present, the contribution to the value N_V ($=M_O + W_i$; $i = 1$; 2 or 3) is always the same. In consequence an easy method of detection is to observe if the increment or decrement is the same (constant) for two or more periods t; if this happens, the minimum gain of those that are in the oscillation would be set. Now, to set the minimum gain is the same as to take an average between the values which are occurring during the oscillation, because in order to attenuate, N_V must go up and this is much quicker than to go down.

A possible method is to observe the new value N_V every t seconds, $t < T$. If N_V is continuously increasing or decreasing then oscillation is present.

In oscillation, the following relationship will exist:

$$N_V(3t) > N_V(2t) > N_V(t) > N_V(0)$$

or

$$N_V(3t) < N_V(2t) < N_V(t) < N_V(0)$$

where $N_V(nt)$ is the N_V measured every t seconds.

Fig. 2.9 shows the predictor circuit designed with comparators and memories.

2.6.3 Controller in Open Loop-configuration

Another way of solving the oscillation problem is opening the loop of Fig. 2.4; this is illustrated in Fig. 2.10.

An open-loop system must monitor the level of the input signal to determine which of several ranges it lies in and

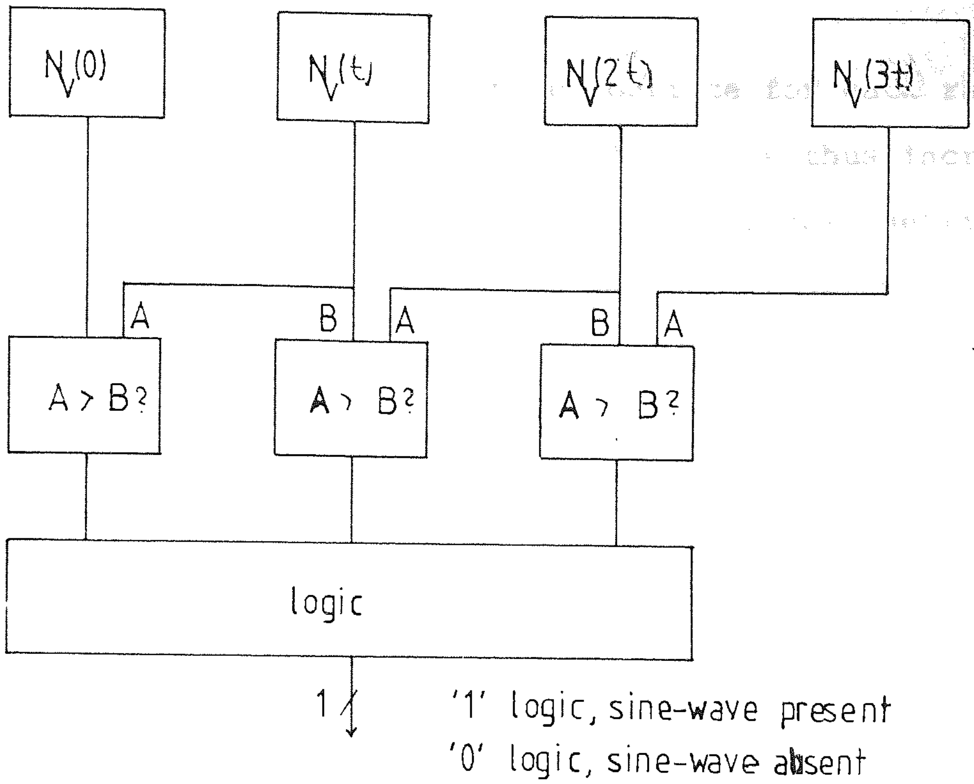


Fig. 2.9 Ancilliary circuit for a d.a.g.c. in closed-loop configuration.

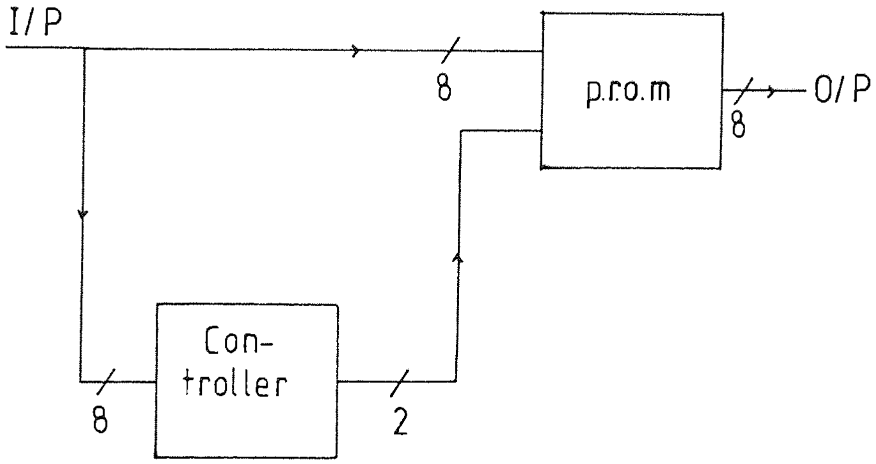


Fig. 2.10 Open-loop configuration

insert the gain or attenuation appropriate for each range of levels. The number of decision levels is thus increased from two or three (for a system having four gain settings). As before, the system operates by counting the number of input samples, N which are above threshold level T_L during a time period T. Two different approaches are considered: Open-loop Controller number 1[12] and Open-loop Controller number 2.

2.6.4 Open-loop Controller Number 1

Let N be the number of samples above T_L, during T seconds of speech signal, then may be written:

If $N \leqslant N_1$ insert 12 dB gain

If $N_1 < N \leqslant N_2$ insert 6 dB gain

If $N_2 < N \leqslant N_3$ insert 0 dB gain

If $N_3 < N$ insert 6 dB loss

where $N_1 < N_2 < N_3$

If T is, say, 1000 ms and T_L is 9 dB below top level and zero dB is provided for speakers whose r.m.s. speech levels are between 9 dB and 15 dB below the limiting level of the PCM coder, typical values for N_1, N_2 and N_3 are:

$N_1 = 29$; $N_2 = 476$; $N_3 = 1945$

These figures will require a 13-bit shift register (11 bits plus two more to store the remaining bits to address the p.r.o.m.). As before, a speech detector is incorporated to avoid setting maximum gain when silent periods are present. Provisions should be taken to ensure that the design averages speech signals only, this is made taking the decisions when 8000T samples of speech have been counted. During silent periods, the last setting is kept.

2.6.5 Open-loop Controller Number 2

In this configuration three threshold levels are used, instead of one, and the controller counts how many times, N, the speech signal is above each one.

Let T_{L1}, T_{L2} and T_{L3} be the threshold levels and

$$T_{L1} > T_{L2} > T_{L3}$$

If it is required to span $\pm$ 3 dB, then

$$T_{L1} = T_{L2} + 6$$

$$T_{L1} = T_{L3} + 12$$

For instance, if T_{L2} is 6 dB above the r.m.s. level of a speaker needing 0 dB, then T_{L3} and T_{L1} are at:

T_{L3} 0 dB above the r.m.s. level

T_{L1} 12 dB above the r.m.s. level

Therefore, a normal talker sees the threshold levels where they are. But a talker needing 6 dB loss will see T_{L1} as less than 9 dB above; T_{L2} as less than 3 dB above and T_{L3} as more than 3 dB below. If the speech level is weak, T_{L1} is more than 15 dB above, T_{L2} more than 9 dB above and T_{L3} more than 3 dB above.

If T = 100 ms the following relationship may be drawn:

If $N > T_{L1}$ more than 3 times 6 dB loss

If $N > T_{L2}$ more than 3 times 0 dB gain

$N > T_{L1}$ less than 3 times

If $N > T_{L2}$ less than 3 times 6 dB gain

$N > T_{L3}$ more than 3 times

If $N > T_{L3}$ less than 3 times 12 dB gain

During silent periods, the d.a.g.c. will maintain the existing setting.

Of the solutions to the oscillation problem of the d.a.g.c. in close-loop configuration studied here, the open-loop system is simpler than the closed-loop with predictor. Between the two alternatives for the open-loop configuration, the one with one single threshold level was simpler to implement than the one having a single counter and three different threshold levels.

As the open-loop system was adopted, it will be dealt with in the next chapter; Fig. 2.11 shows a possible hardware implementation for the open-loop controller number 2. The incoming signal is converted from serial to parallel; it is compared with the threshold levels (T_{L1}, T_{L2}, T_{L3}) and if the signal is above them, the adders 1, 2 and 3 will increase the counts stored in the shift registers 1, 2 and 3. To prevent the insertion of gain when there is no speech, the magnitude of the incoming signal is compared with a fourth threshold level; if it is above, this is counted by adder 4 and stored in the shift register 4. The shift registers operate in synchronism with the PCM system and present their contents for each channel in turn to the comparators 1, 2, 3 and 4 to be compared with N. When the number of active samples, those above the fourth threshold level, which has been stored reaches 8000T the logic circuit in Fig. 2.11 enables the output from comparators 1, 2 and 3 to modify the control bits, used to form the address to the p.r.o.m., accordingly with the relationships given in 2.6.5.

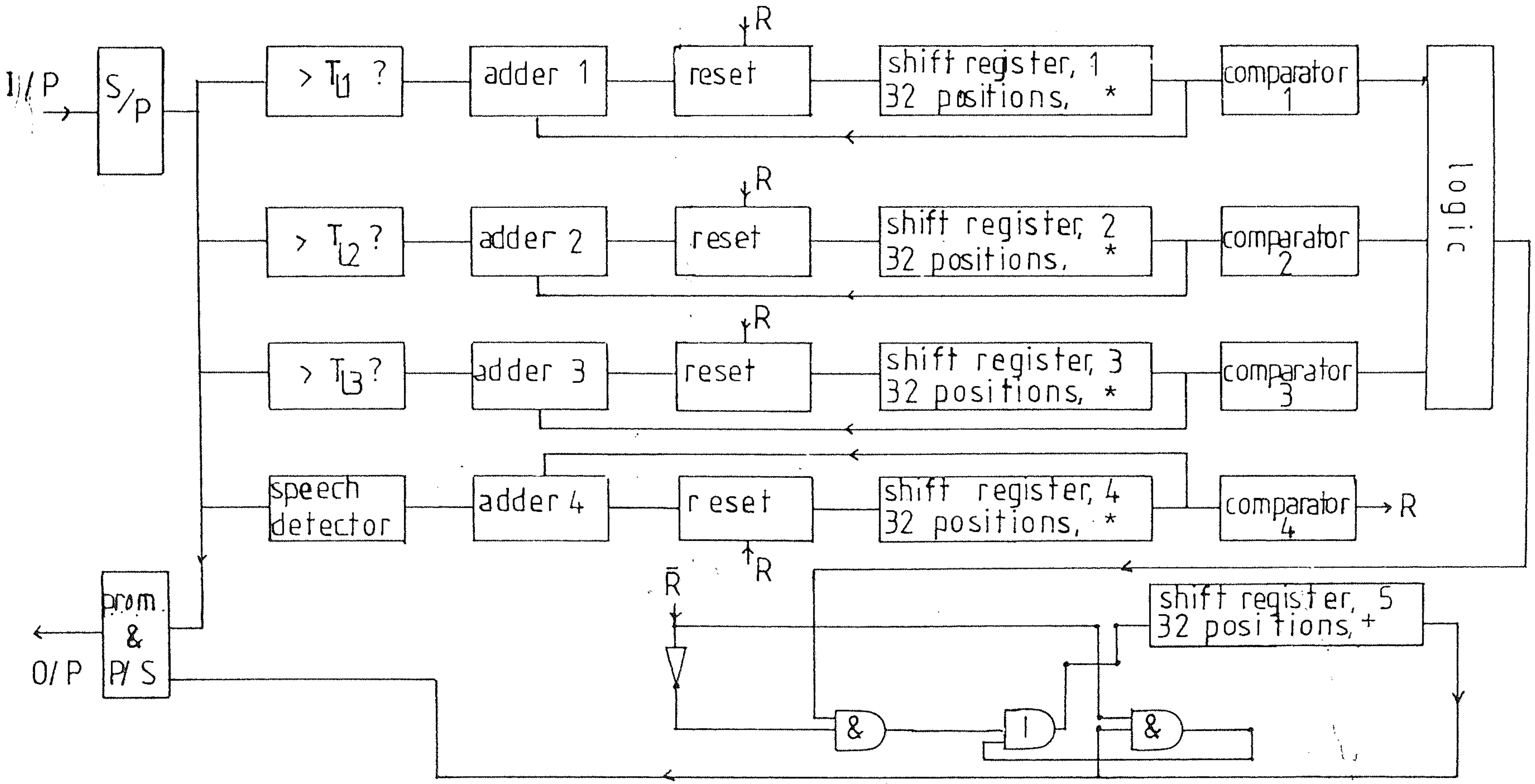


Fig. 2.11 Open-Loop Controller No. 2.

*Number of bits depends on precision chosen (or hardware)

+two bits

CHAPTER 3

DESIGN AND SIMULATION OF A D.A.G.C. IN OPEN-LOOP CONFIGURATION

3.1 DESIGN

This section describes the design of the d.a.g.c. in open-loop configuration, being the one developed that has 3 counters and one threshold level as described in section 2.6.4.

As stated in section 2.6.4, let N be the number of samples above certain threshold level T_{L_1}. This d.a.g.c. operates using the following relationships:

If	$N \leq N_1$	insert 12 dB gain
If	$N_1 < N \leq N_2$	insert 6 dB gain
If	$N_2 < N \leq N_3$	insert 0 dB gain
If	$N_3 < N$	insert 6 dB loss

N_1, N_2 and N_3 are figures to be calculated and they satisfy the relationships:

$$N_1 < N_2 < N_3$$

It is necessary to determine the appropriate values for them in accordance with the threshold level chosen and the speech level taken as reference.

In principle, the threshold level can be anywhere. Its value is restricted only by the number of samples which may be required to define a speaker requiring no change of gain. For instance, if the threshold level is too high (within one of the upper segments of the A-law) the number of samples required may be zero if the time constant is not big enough and the level of the signal is too far from the threshold level. If the threshold level is too low (second or third segment, for example), the number of samples necessary to take a decision could be high and not possible to achieve because of hardware limitations.

To calculate N_1, N_2 and N_3, Eq (2-1) is used. For 6 dB steps, we have the gain setting values of -6 dB, 0 dB, 6 dB and 12 dB. A talker who receives 0 dB is named reference talker (RT). In order to generalise, Fig. 3.1 may be used.

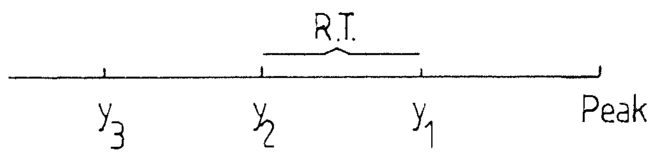


Fig. 3.1

Reference Talker Definition

For 6 dB-step device, from Fig. 3.1 it is defined:

$$\Delta y = y_1 - y_2 = y_2 - y_3 = 6 \text{ dB}$$

The threshold level could be at one of the limits of the talker taken as reference. If it is at y_1, then a speaker who produces an r.m.s. of y_1 reaches the threshold level in N_3 occasions, where

$$N_3 = 8000Te^{-\sqrt{2}} \tag{3.1}$$

If y_2 r.m.s. level is produced T_{L_1} is reached N_2 times, where:

$$N_2 = 8000Te^{-\sqrt{2}.10^{\Delta y/20}} \tag{3.2}$$

Also, if y_3 r.m.s. is produced, the threshold level is attained as many as:

$$N_1 = 8000Te^{-\sqrt{2}.10^{\Delta y/10}} \tag{3.3}$$

To avoid setting maximum gain because of silence, a speech detector is incorporated. To do this a second threshold level, T_{L_2}, is used to define when a sample is considered to be speech or just noise. Whenever the P.C.M. word is below T_{L_2}, nothing is added to the counter. Each sample above T_{L_2} is termed an active sample. In fact, when the number of active samples (NAS) is equal to the number of samples that there are in the time constant T, a decision is made. This makes the actual time to take a decision greater than the time constant chosen. To see how much larger it is, this can be worked out for a speaker needing 0 dB. Let T_{L_2} be y_t dB below the clipping level. The y_1 dB-talker is (y_1-y_t) dB above T_{L_2}; the time constant for this speaker will be:

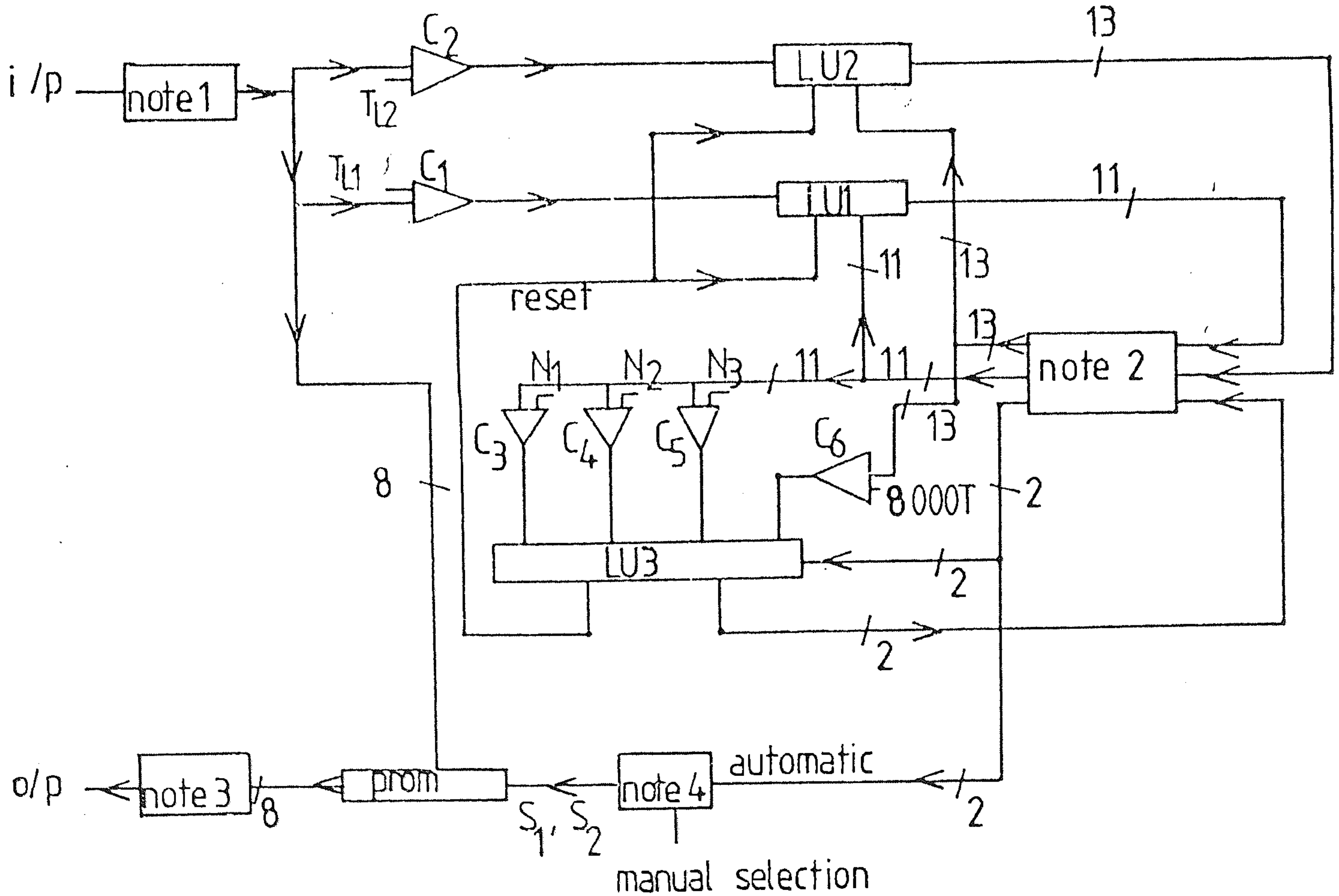


Fig. 3.2 d.a.g.c. in open-loop configuration No. 1
Note 1: serial-to-parallel converter; Note 2: shift register, 32-positions, 26-bits; Note 3: parallel-to-serial converter; Note 4: data selector.

$$T_r = T/(e^{-\sqrt{2}.10^{-(y_1-y_t)/20}}) \tag{3.4a}$$

for the speech level which is at upper limit. If he is at the lower limit:

$$T_r = T/(e^{-\sqrt{2}.10^{-(y_2-y_t)/20}}) \tag{3.4b}$$

If T=0.1 seconds y_1 and y_2 equal to -9 dBr and -15 dBr relative to the clipping level, and y_t = -40 dBr, then the reference talkers have a time constant which varies from 104 ms to 108 ms. For a talker 6 dB below the RT, the time constant limits are from 108 ms to 117 ms. Therefore to set 12 dB gain it is necessary to have more than 117 ms. For setting 6 dB loss, it is necessary to have less than 104 ms. Actually, to achieve 100 ms at the time constant chosen, the signal must be very high.

Fig. 3.2 shows the design.

The p.r.o.m. contains a look-up table for the appropriate output word for every input word for each of the gain settings provided. Each address is selected by a 10-bit word which comprises 8 bits from an incoming P.C.M. sample and two, S_1 and S_2, corresponding to the gain control for that channel. The incoming P.C.M. signal is converted from serial to parallel mode before processing and the output signal is converted back to serial mode, as shown in Fig. 3.2. It is also necessary to arrange for digits contained in time slots 0 and 16, used normally for frame alignment and signalling respectively, to bypass the signal processing system lest they be processed by it. This is not shown in Fig. 3.2.

The speech detector, as has been stated, is to avoid gaps in the speech being treated as low-level signals and thus causing maximum gain to be inserted.

The samples whose magnitude exceed T_{L_1} are detected by digital comparator C_1, and those below T_{L_2} are detected by C_2. The comparators are connected to logic units LU1 and LU2 respectively. Each sample exceeding the threshold levels causes the LU to add 1 to the count stored for that channel in a shift register. This operates in synchronism with the P.C.M. system and presents its contents for each channel in turn to the logic units LU1, 2 and 3. For each channel it stores: the number of samples exceeding T_{L_1} (processed by LU1), the number of samples exceeding T_{L_2} (processed by LU2) and the gain control digits S_1 and S_2 (processed by LU3). When the number of active samples exceeding T_{L_2} (determined by comparator C_6) which has been stored reaches 8000T, the number of samples exceeding T_{L_1} which has been stored is read out. This number, N, is compared with N_1, N_2 and N_3 by digital comparators C_3, C_4 and C_5 whose outputs cause logic unit LU3 to select appropriate values of S_1 and S_2 to be used in addressing the p.r.o.m. The counts of samples exceeding T_{L_1} and T_{L_2} are then reset to zero and values S_1 and S_2 are inserted by LU1, 2 and 3 respectively. S_1 and S_2 are thus reassessed every T seconds.

The above design is slightly modified for 3 dB-step operation in the following way. It is necessary to use three more comparators connected to LU3. The p.r.o.m. must be addressed using 11 bits instead of 10, as one more is necessary for the third control bit, S_3. The shift register must also be increased in size.

3.2 COMPUTER SIMULATION OF THE D.A.G.C. IN OPEN-LOOP CONFIGURATION

The simulation was made in real-time and due to the low speed of the computer, only one channel was processed. The language employed was machine language.

The computer played two roles. First, as a memory, storing the look-up table and, second, as the digital automatic gain control itself. There as an interface circuit which converted the P.C.M. word from serial to parallel and vice-versa. Also, from this interface the necessary clocks to drive the PDP-11 were extracted. Thus, for instance, a clock was generated to enable the computer to read the P.C.M. word already in parallel form; a slightly different version of the former, to do the conversion the other way around.

Fig. 3.3 shows the pulses generated by the interface itself and the pulses given by the P.C.M. equipment.

About the allocations of the T_{L_1}, it was placed at one of the limits of the talker taken as reference. For simulation purposes, the reference talker chosen was that whose r.m.s. level is between 9 dB and 15 dB below the

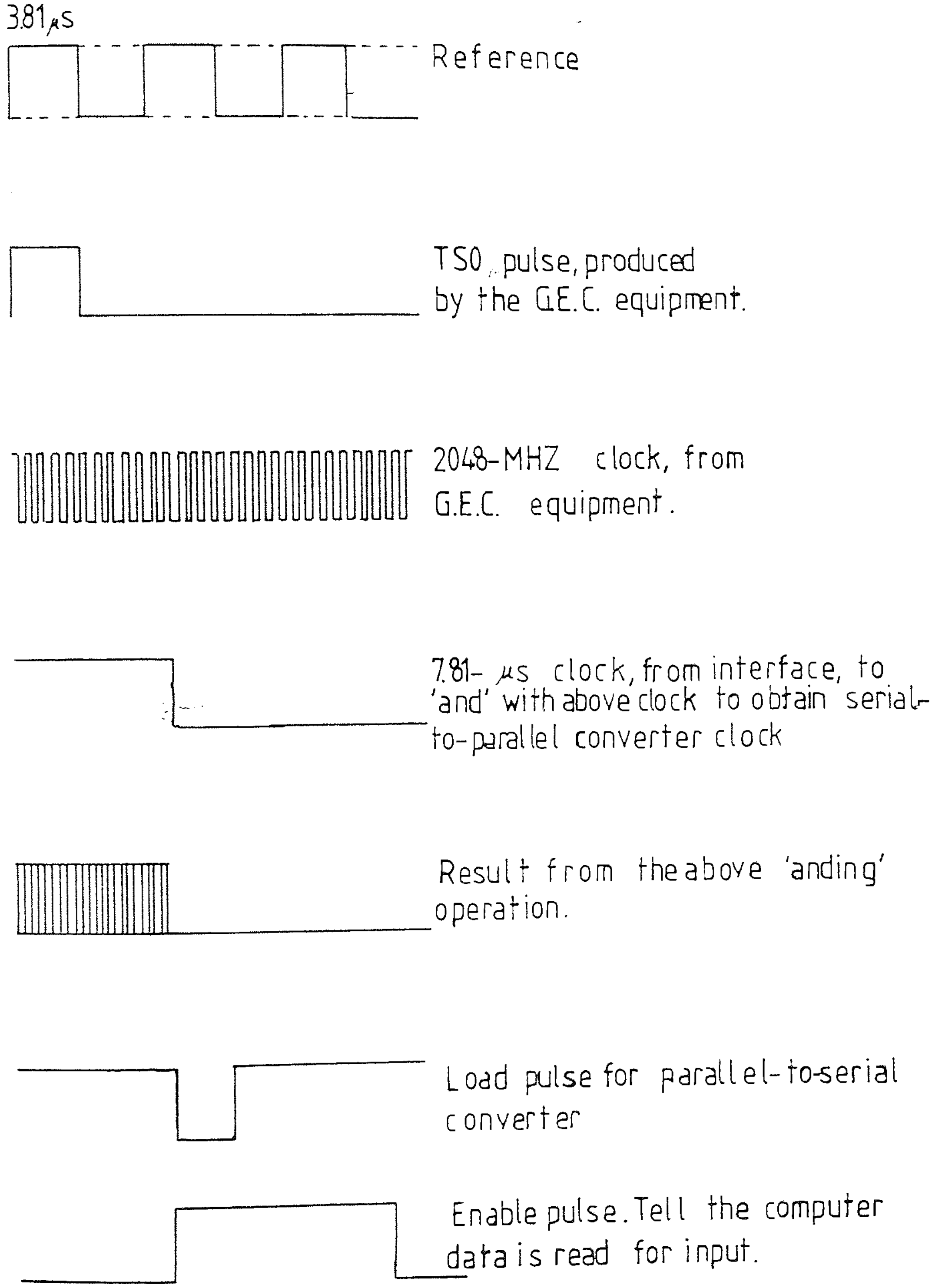


Fig. 3.3 Interface Signals

clipping level. This talker may be considered a loud one[21]. The reference talker will receive 0 dB gain; talkers whose r.m.s. levels are between 15 dB and 21 dB below clipping will receive 6 dB gain. Those 21 dB or more below top receive 12 dB gain and those less than 9 dB below it receive 6 dB loss.

Fig. 3.4 shows the different talkers, for the 6 dB step-scale.

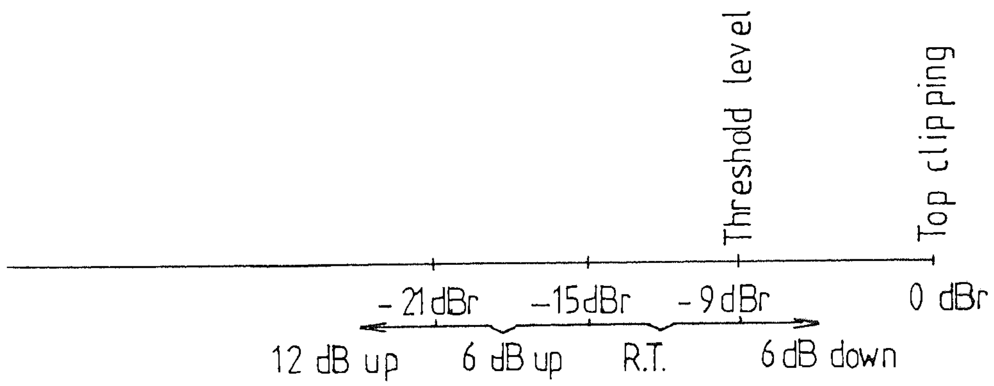


Fig. 3.4

Zone of operation of a 6 dB step d.a.g.c.

The r.m.s. level of a median talker is at -26 dBr relative to the clipping level[21]; thus to get attenuation the speaker must have a very loud voice. At this stage, our interest is to have the d.a.g.c. in open-loop configuration working and to find out its performance with different parameters.

The threshold level, T_{L_1} was situated at -9 dB, relative to clipping, for a system using 6 dB steps and at -10.5 dB for a system using 3 dB steps. Fig. 3.5 shows the zone levels for 3 dB steps.

Using equations (3.1), (3.2) and (3.3) and a time constant of T=100 ms, 0 dB will be introduced if the signal lies within y_1 and y_2 between 48 and 195 occasions. If the T_{L_1} is reached more than 195 times, 6 dB loss is inserted. If the threshold level is achieved between 48 and 3 times (N_2 and N_1), a gain of 6 dB is introduced and if it is reached less than 3 times 12 dB gain is switched in.

Doing similar calculations for 3 dB-step d.a.g.c. and T=0.1 seconds and the zones as defined in Fig. 3.5, we have the following relationship:

$N>294$	6 dB	loss
$194<N\leq 294$	3 dB	loss
$112<N\leq 194$	0 dB	gain
$45<N\leq 112$	3 dB	gain
$16<N\leq 45$	6 dB	gain
$3<N\leq 16$	9 dB	gain
$N\leq 3$	12 dB	gain

The second threshold level, T_{L_2}, was, as this stage, 40 dB below clipping level.

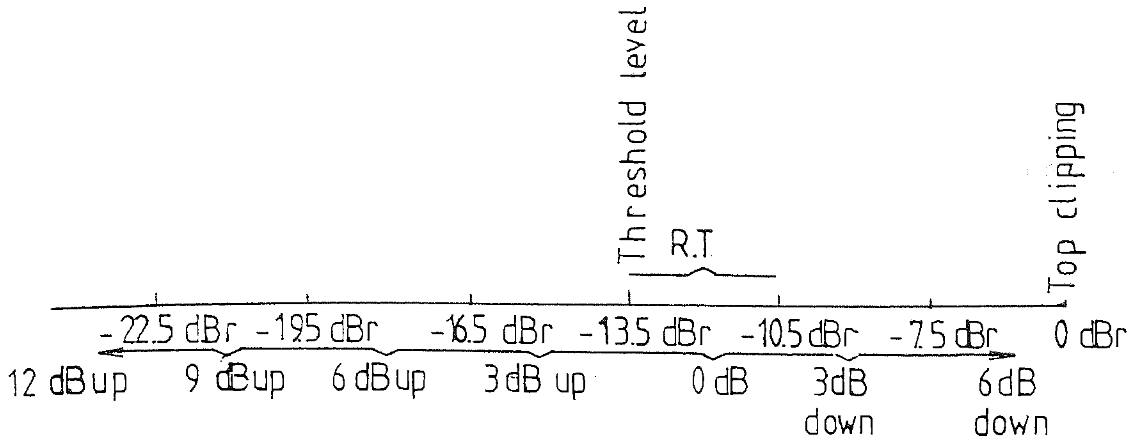


Fig. 3.5 Zone of operation for a 3 dB step d.a.g.c.

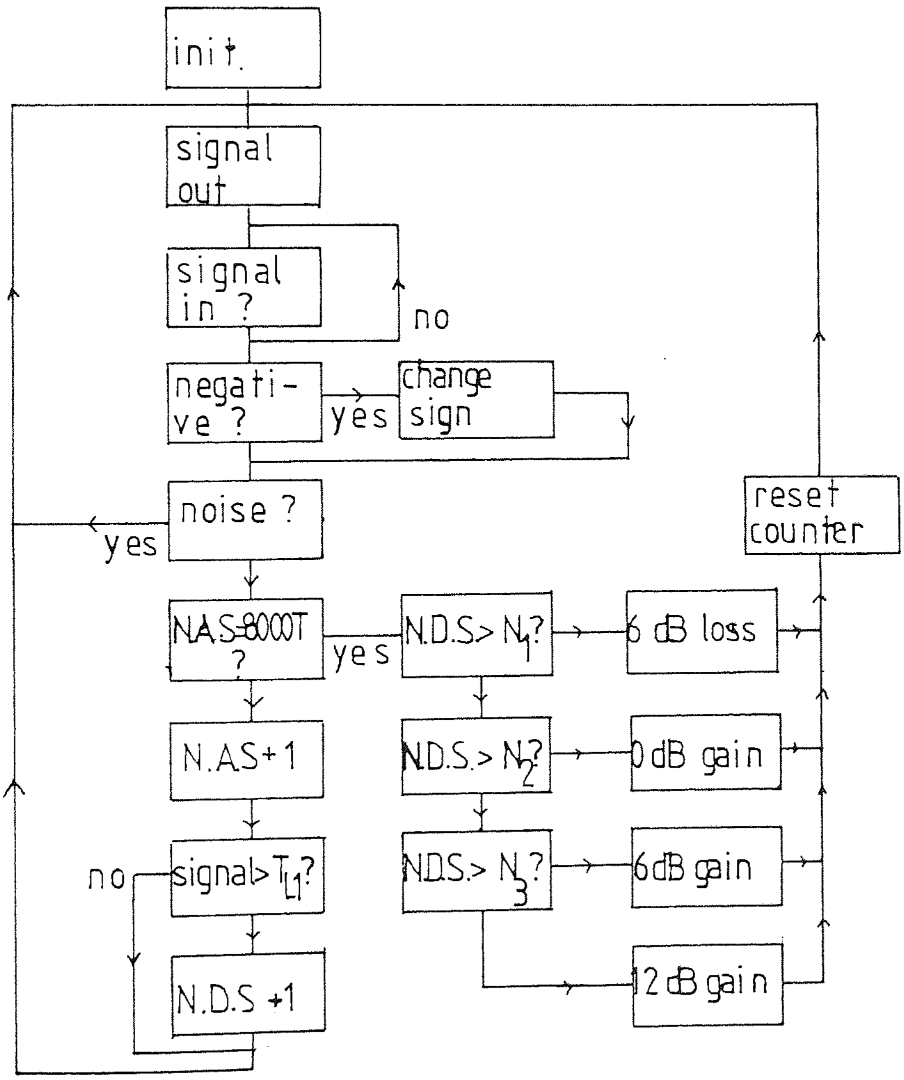


Fig. 3.6 Flow Diagram

The software itself works as follows: the first step is initialisation, then the computer reads the P.C.M. word, when it is available; the software is checking continuously a pulse which comes from the standard 32-channel P.C.M. equipment in order to ensure that the PDP-11 reads the word when it is present. It tests whether the word is positive or negative in order to choose the appropriate first and second threshold levels, T_{L_1} and T_{L_2}. It checks if the word is an active sample; if not, the computer immediately sends the word out, with the previous setting, to the P.C.M. equipment and waits for the arrival of the next sample. This is the method chosen to avoid inserting maximum gain when no speech signals are present. If so, it is counted as an active one and the word is sent out, with the previous setting, to the P.C.M. system.

Then it is necessary to test that the time constant has been reached, that is if NAS is equal to a preset value which represents the number of samples in the time constant chosen. If the time constant has not been reached, the word is compared with T_{L_1}, if it is above, the sample is counted as a decision sample, DS. The number of decision samples, NDS, required to decide the setting values to be used is in accordance with the rules established in section 3.1. Then the computer waits for the next sample to arrive. If the time constant has been reached, the PDP-11 determines in which region the number of decision samples lies in, in order to choose the new setting, which will be used for at least

T_r seconds starting with the next sample. The programme has facilities for modifying the time constant, also the values of the threshold levels. Fig. 3.6 shows the flow diagram.

The d.a.g.c. was connected between the send and the receive terminals of a standard CEPT 32-channel P.C.M. system[12,22]; however only one channel was processed. The design was extended to a d.a.g.c. able to introduce gain or attenuation in steps of 3 dB, as follows: 6 dB loss, 3 dB loss, 0 dB gain, 6 dB gain, 9 dB gain and 12 dB gain. The software was modified accordingly.

3.3. TEST OF THE D.A.G.C. IN OPEN-LOOP CONFIGURATION

3.3.1 General

The purpose of these tests was to study the performance of the d.a.g.c. when it is operated manually or automatically and to see its behaviour using different types of inputs (sine-wave, Gaussian noise and real speech) and different time constants. The tests were made using the two versions of d.a.g.c. were developed, namely, 3 dB step d.a.g.c. and the 6 dB one.

3.3.2 6 dB Step D.A.G.C.

3.3.2.1 Static Test

An 8-bit binary number was used as input. Table 3.1 shows the results obtained. The outputs were as expected under both manual and automatic operations; the settings were, of course, 12 dB up or 6 dB down. This is because the word was 100% above or 0% above T_{L_1}; therefore gain or loss was required.

I/P	MANUAL (0 dB)	AUTOMATIC	GAIN (dB)
10101101	10101101	10001101	12
10101100	10101100	10001100	12
10101110	10101110	10001110	12
10001111	10001111	10011111	-6
11110000	11110000	11010000	-6
10110100	10110100	10010100	12
00110100	00110100	00010100	12
01110000	01110000	01010000	12
01110001	01110001	01010010	12
11100000	11100000	11000000	12

TABLE 3.1

Static Test of the d.a.g.c.

3.3.2.2. Sine-Wave as Input

The tests were made using different time constants, which means different number of active samples and different number of decision samples. This is taken into account with the Eqns. (3.1), (3.2) and (3.3). For each time constant, the computer was given the values and worked out the appropriate figures.

The time constants were varied from 50 ms to 1000 ms. Fig. 3.7 shows the results for a 1000 Hz sine-wave and a time constant of 100 ms. It can be seen that 0 dB was not achieved. This is logical because even though there is no oscillation, the sine-wave is virtually a fixed level. The results were the same for each time constant.

3.3.2.3 Gaussian Noise Test

To simulate speech, Gaussian noise was used as input. The output was measured with a British Post Office speech volt meter[30], whose readings were dB relative 1V. Table 3.2 shows the results obtained for different time constants and for manual operation. Fig. 3.8 shows the input/output characteristics. It was observed that the settings oscillated, which was due to the random nature of the signal[9]. It can be seen from Table 3.2 that the changes always occurred at the same point, independent of the time constant used.

3.3.2.4 Speech Test

At this stage, the speech test was limited to use real speech as input and measured the output, both manual and automatic, and using different time constants. Table 3.3 shows the measurements taken. The author did not notice appreciable degradation of the speech quality.

3.3.3 TEST TO THE 3 dB STEP D.A.G.C.

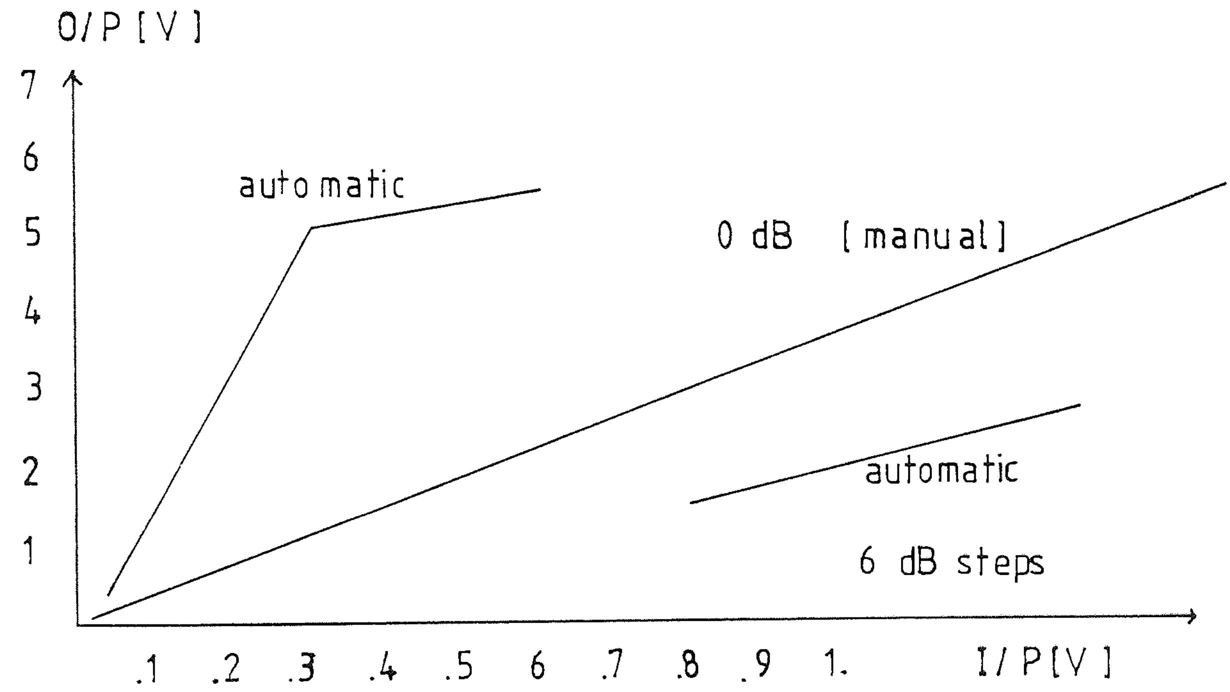


Fig. 3.7 Input/Output characteristic with sine-wave
$f = 1000$ Hz, $T = 0.1$ sec.

TABLE 3.2

Gaussian Noise Test

Time Constant (ms)		50	100	200	300	400	500
I/P	MAN	AUTOMATIC			AUTOMATIC		
mV	dBV	dBV	dBV	dBV	dBV	dBV	dBV
5	-53.80	-43.0	-43.2	-42.9	-42.5	-42.8	-43
10	-49.16	-37.2	-37.2	-37.0	-37.0	-36.8	-37.2
15	-46.00	-34.0	-33.8	-34.0	-33.5	-33.6	-34.0
20	-43.60	-31.5	-31.8	-31.5	-31.5	-31.5	-31.5
25	-41.60	-29.5	-29.8	-29.8	-29.5	-29.5	-29.8
50	-35.80	-23.8	-23.5	-23.5	-23.2	-23.2	-23.2
60	-34.20	-22.0	-22.0	-22.0	-21.8	-21.5	-22.6
70	-32.60	-20.5	-20.6	-20.5	-20.2	-20.2	-20.8
80	-31.2	-19.8	-19.4	-19.6	-19.5	-19.3	-19.6
90	-30.5	-18.6	-18.6	-18.5	-18.5	-18.5	-18.5
100	-29.65	-17.5	-17.6	-17.4	-17.5	-17.2	-17.6
150	-26.0	-14.1	-14.0	-14.0	-14.0	-13.3	-14.0
200	-23.60	-11.8	-11.7	-11.7	-11.5	-11.4	-11.6
300	-16.50	- 4.0	- 4.5	- 4.5	- 4.2	- 4.0	- 4.4
400	-14.0	- 4.0	- 8.0	- 8.0	- 8.0	- 8.0	- 8.0
500	-12.0	- 6.5	- 6.5	- 6.2	- 6.0	- 6.0	- 6.0
600	-10.70	- 5.2	- 5.2	- 5.0	- 5.0	- 6.0	- 5.0
700	- 9.60	- 9.5	- 9.5	- 9.5	- 9.5	- 9.5	- 9.5
800	- 8.6	- 8.5	- 8.5	- 8.5	- 8.8	- 8.5	- 8.6
900	- 8.0	- 8.5	- 8.0	- 8.0	- 8.0	- 7.5	- 8.0
1000	- 7.30	- 7.5	- 7.5	- 7.5	- 7.6	- 7.2	- 7.5
1100	- 6.80	- 7.5	- 7.0	- 6.8	- 7.0	- 6.8	- 7.0
1200	- 6.5	- 8.0	- 7.7	- 7.5	- 8.0	- 7.0	- 7.0
1300	- 6.2	- 8.0	- 8.0	-12.0	-12.0	-11.6	-12.0
1400	- 6.0	-11.0	-12.0	-12.0	-12.0	-12.0	-12.0
1500	- 5.6	-12.0	-11.8	-11.6	-11.5	-11.5	-11.6
1600	- 5.6	-11.5	-11.8	-11.6	-11.5	-11.5	-11.6
1800	- 5.2	-11.2	-11.2	-11.2	-11.2	-11.2	-11.3
1900	- 5.1	-11.2	-11.2	-11.2	-11.2	-11.2	-11.2

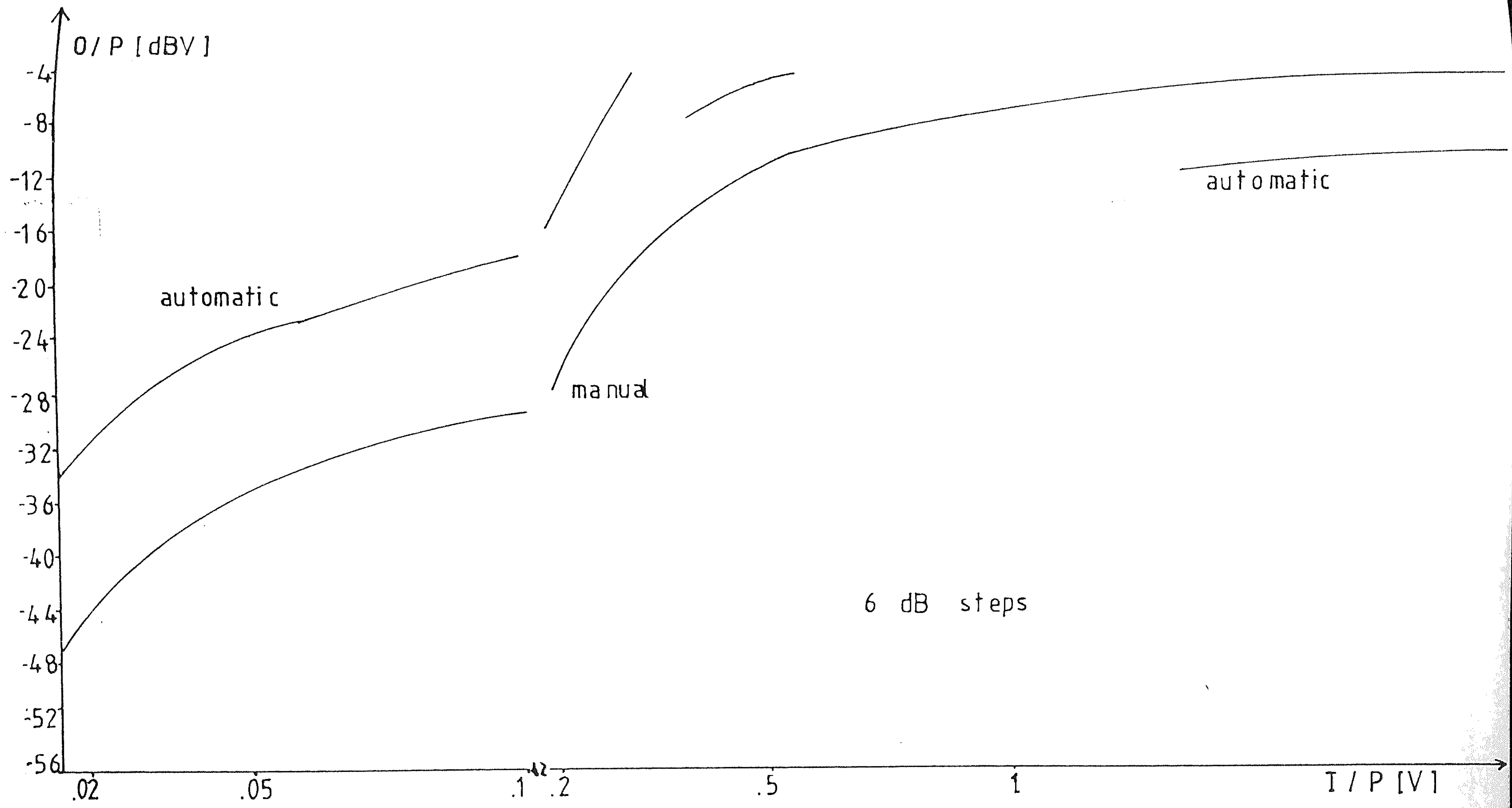


Fig. 3.8 Input/Output characteristic with Gaussian noise.

TABLE 3.3

Speech Test

	I/P	O/P (M)	O/P (A)
1	4.0	-48.00	-36.00
2	7.0	-43.00	-31.00
3	9.6	-40.00	-28.00
4	14.2	-37.00	-24.00
5	25.2	-32.00	-21.00
6	40.0	-28.00	-17.50
7	70.0	-23.00	-12.00
8	125.0	-18.00	-11.50
9	223.0	-13.00	- 7.00
10	281.0	-11.00	-11.00
11	316.0	-10.00	-10.00
12	446.0	- 7.00	- 7.00
13	630.0	- 4.00	-10.00
	(mv)	(dBv)	(dBv)

3.3.3.1 Sine-Wave Test

The results were similar to that of the 6 dB d.a.g.c. sine-wave test; that is independent of the time constant, and the changes occurred always at the same point.

3.3.3.2 Gaussian Noise Test

Table 3.4 shows the results and Fig. 3.9 the input/output characteristics. The results were independent of the time constant. As in the 6 dB step tests, the change always occurred at the same point and the settings oscillated[(9)].

3.3.3.3 Speech Test

The test was carried out in a similar fashion as in 6 dB step test. Table 3.5 presents the results. The author did not observe any degradation of the speech quality.

TABLE 3.4

3 dB steps, 1000 ms, 200 ms

		1000 ms	200 ms	200 ms
I/P (mV)	O/P(M) (dBV)	O/P(A) (dBV)	O/P(A) (dBV)	6dB steps (dBV)
5	-53.80	-55.8	- 5.1	-53.2
10	-49.16	-51.0	-46.0	-37.6
15	-46.0	-48.5	- 4.0	-33.8
20	-43.60	-46.0	-31.6	-32.1
25	-41.60	-30.0	-29.8	-30.2
50	-35.80	-23.6	-23.4	-24.7
60	-34.20	-21.6	-22.0	-22.1
70	-32.60	-20.5	-20.6	-21.0
80	-31.20	-19.4	-19.6	-19.2
90	-30.50	-18.5	-18.5	-18.5
100	-29.65	-17.5	-17.4	-17.4
200	-26.00	-11.6	-11.6	-11.6
300	-16.50	- 4.4	- 4.6	- 4.6
350	-14.60	- 5.6	- 5.6	- 3.2
400	-14.00	- 8.0	- 8.0	- 8.0
450	-13.00	- 8.0	- 7.0	- 7.2
500	-17.0	- 6.0	- 7.0	- 6.0
525	-11.50	- 8.5	- 8.5	- 5.5
600	-10.60	- 7.4	- 7.5	- 4.8
650	-10.00	- 7.0	- 7.0	-10.0
700	- 9.60	- 9.6	- 9.6	- 9.8
750	- 9.20	- 9.2	- 9.2	- 9.2
800	- 8.60	- 8.5	- 8.8	- 8.6
850	- 8.20	- 8.2	- 8.5	- 8.4
900	- 8.00	- 8.0	- 8.0	- 8.0
950	- 7.40	-10.2	-10.2	- 9.6
1000	- 7.30	-10.1	-10.2	-11.6
1100	- 6.80	- 9.8	- 9.8	- 7.0
1200	- 6.50	- 9.2	- 9.5	- 6.8
1300	- 6.20	- 9.0	- 9.0	-12.2
1400	- 6.00	- 8.5	- 8.8	-12.0
1500	- 5.60	- 9.0	- 9.0	-11.8
1600	- 5.60	-11.6	-11.6	-11.6
1800	- 5.2	-11.2	-11.4	-11.1

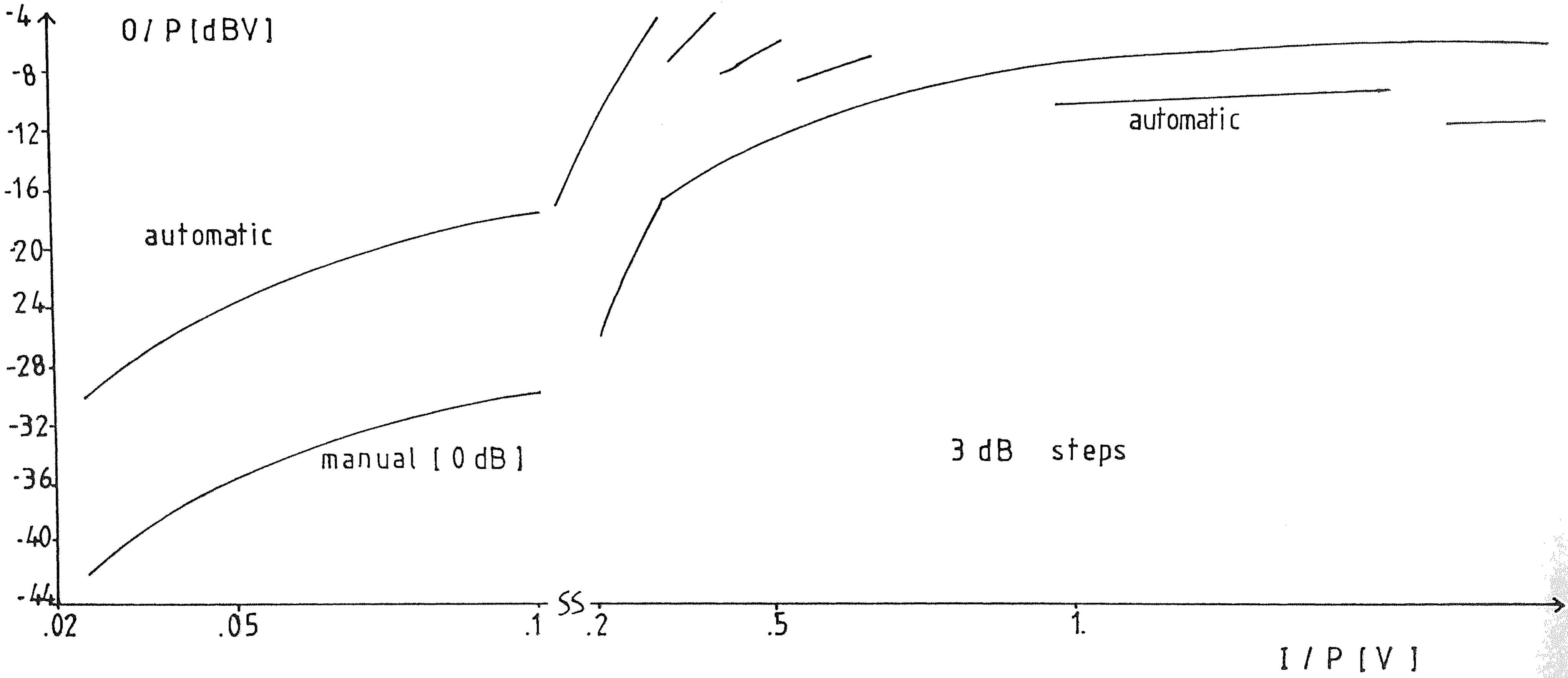


Fig. 3.9 Input/Output characteristics with Guassian noise

TABLE 3.5

Speech Test

3 dB Step

	O/P	I/P	GAIN
1	-37	-48	12
2	-31	-43	12
3	-27	-40	12
4	-24	-37	12
5	-21	-32	12
6	-15	-28	12
7	-10	-23	12
8	-10	-18	9
9	- 9	-15	6
10	-11	-13	3
11	-11	-11	0
12	-13	-10	-3
13	-13	- 9	-4
14	-13	- 7	-6
15	-10	- 4	-6
	(dBV)	(dBV)	(dB)

CHAPTER 4

SUBJECTIVE TESTS

4.1 GENERAL

This chapter describes the subjective tests carried out on a telephone local connection with a d.a.g.c. incorporated which was able to introduce 0 dB, 6 dB or 12 dB gains. The objective was to find out the response of listeners and the improvements and impairments detected when the d.a.g.c. is in use. The 6 dB loss insertion feature was left out because it represents the solution to a rather different problem [5,6,7,13,14]. However, in the previous chapter, the 6 dB loss insertion property was studied objectively.

The purpose of the test was threefold: firstly to determine whether a connection is better with or without d.a.g.c.; secondly to choose levels of speech at the listener's ear at which it is appropriate to introduce the d.a.g.c. and thirdly to determine an acceptable time constant. The tests were carried out in three stages, the first preliminary stage, had the aim of achieving a general idea of how people react to the main parameters involved: i.e. time constant and speech levels. Once conclusions had been drawn from this stage, the next was carried out, which was a larger and more thorough test. The speech levels were reduced to just two; however, the time constant was kept as a parameter. Finally, the last stage was to find out the differences between time constants.

4.2 TEST EQUIPMENT ARRANGEMENT

The subjective tests themselves were made on a connection as shown in Fig. 4.1[24].

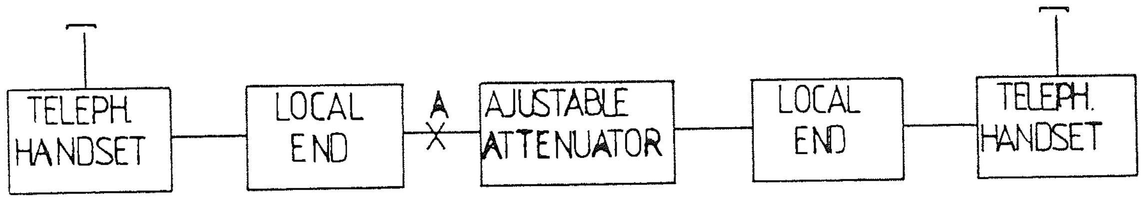


Fig. 4.1

Basic Model for Tests

A tape recorder was placed at A; it had previously recorded speech samples according to the rules stated elsewhere[24]; that is the speech material is recorded at a previously chosen level, together with a calibrate tone, Fig. 4.2(a). Then, the recorded material is replayed to measure its level, Fig. 4.2(b) and, finally, the sentences are re-recorded into recorder R-2 with uniform level, Fig. 4.2(c).

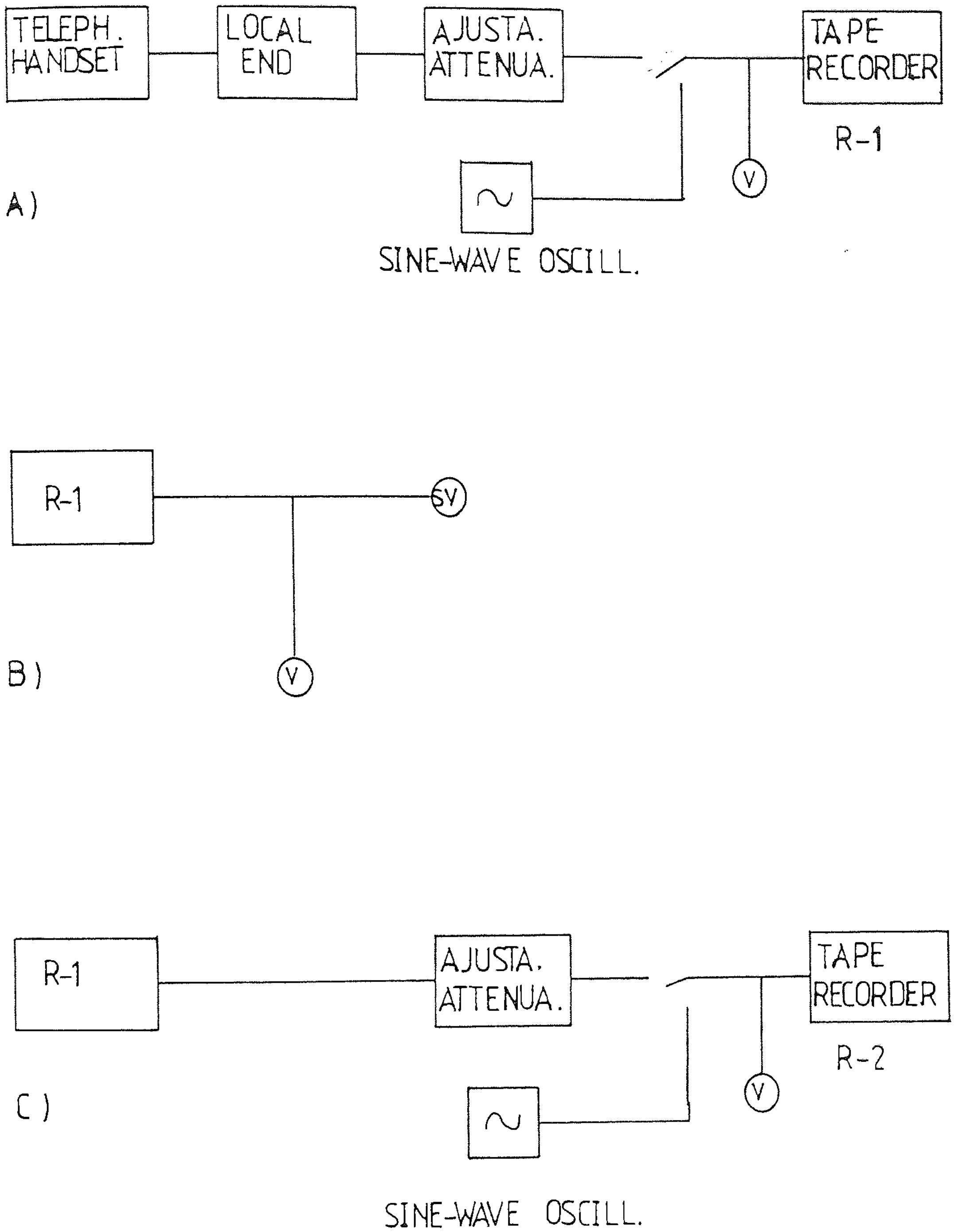


Fig.4.2 Recording of sentences
(a) First time recording of sentences with tones.
(b) Replaying of sentences to measure their levels.
(c) Re-recording of sentences with uniform level.
NOTE: V - volt meter, SV - speech volt meter

The tape recorder R-2 will be connected to A, Fig. 4.3.

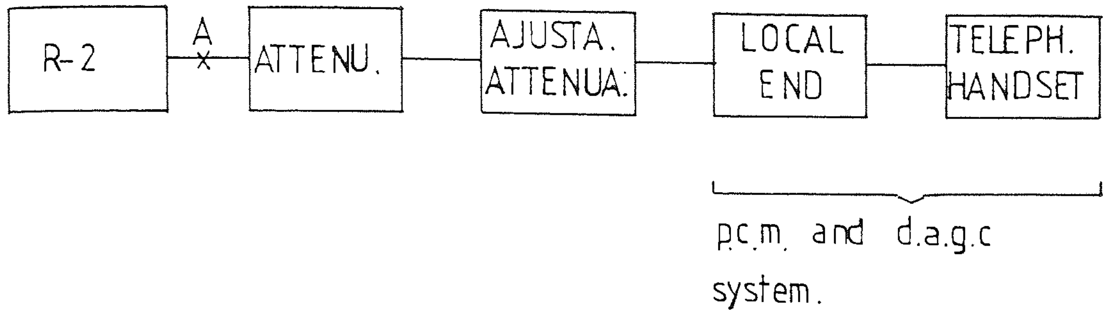


Fig. 4.3

R-2 connected to the basic connection

The digital end has both the P.C.M. system and the d.a.g.c. The final connection, with the PDP-11 computer is shown in Fig. 4.4.

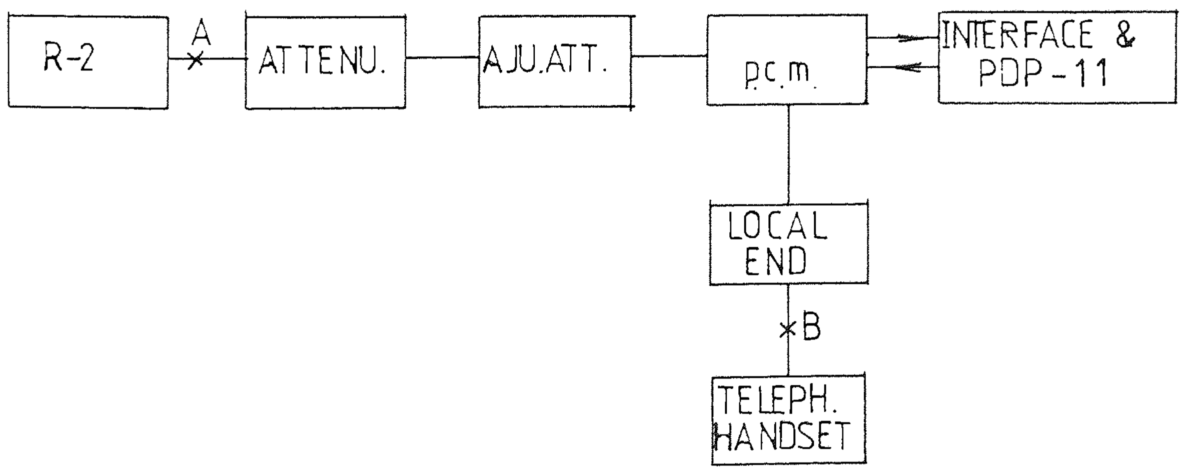


Fig. 4.4

Testing connection with P.C.M. and d.a.g.c. systems.

To avoid doing the above test in real time, a tape recorder was used at B to record the output and play it back in a quiet room at the same level it is heard (or received) at B. A calibration tone was used to achieve this.

4.3 PRELIMINARY TEST

The preliminary test was made with three speech levels, ℓ_1, ℓ_2, ℓ_3, four time constants and one voice. The three levels were levels situated in the region of [24] 1). preferred level (-32 dBV), 2). near quieter than preferred (-38 dBV) and 3). between quieter than preferred and much quieter than preferred (-44 dBV).

The three time constants were 100 ms, 333 ms, 1000 ms and 0 ms. 0 ms is equivalent to the d.a.g.c. not operating. The test consisted of pair comparison, i.e. the listener compares two presentations of speech and decides which one is preferred. The speech samples were 30 seconds in duration, one of which had been processed by the computer and the other not.

With the d.a.g.c. in operation, three time constants were used. The system with d.a.g.c. out is labelled A; d.a.g.c. in and 100 ms as D1; in and 333 ms as D2 and in and 1000 ms as D3.

As this test was fairly simple and short, it was possible to conduct it in real time and therefore it was not necessary to make the recording at point B in Fig. 4.4.

The presentation to the listener of 2 30-second sample was randomised. Fig. 4.5 gives the arrangement for this test.

Fig. 4.5

Randomised arrangement for the preliminary test.

Listeners						
1	A,D2 ℓ_2	A,D1 ℓ_3	D2,A ℓ_1	D2,A ℓ_1	A,D1 ℓ_2	D ,A ℓ_3
2	D1,A ℓ_3	A,D3 ℓ_2	A,D3 ℓ_1	A,D1 ℓ_2	D3,A ℓ_2	D1,A ℓ_1
3	D2,A ℓ_3	D1,A ℓ_1	A,D1 ℓ_2	D2,A ℓ_2	A,D1 ℓ_3	D2,A ℓ_1
4	D3,A ℓ_2	D3,A ℓ_3	A,D2 ℓ_1	D3,A ℓ_1	A,D ℓ_2	D2,A ℓ_3
5	A,D1 ℓ_2	A,D3 ℓ_2	D3,A ℓ_3	D1,A ℓ_1	A,D3 ℓ_1	A,D1 ℓ_3
6	A,D3 ℓ_3	D2,A ℓ_1	A,D2 ℓ_2	D3,A ℓ_2	A,D2 ℓ_3	D3,A ℓ_1

Fig. 4.6

Answers from the 6 subjects to the preliminary test.

1	D2	D1	D1	A	A	D1
2	D1	A	A	A	A	A
3	D2	A	D1	A	D1	A
4	A	D3	A	A	A	A
5	A	A	A	D1	D3	A
6	D3	D2	A	D3	D2	A

Grouping accordingly with the levels, we may present the following results:

Level ℓ_1 (-32 dBV)

A=8

D=4 D1=2 (50%)

D2=1 (25%)

D3=1 (25%)

Level ℓ_2 (-38 dBV)

A=9

D=3 D1=1 (25%)

D2=1 (25%)

D3=1 (25%)

Level ℓ_3 (-44 dBV)

A=5

D=7 D1=3 (75%)

D2=2 (50%)

D3=2 (50%⅛)

The test was not intended to be conclusive[12], but to provide a useful indication of likely performance. The benefit to be derived from the d.a.g.c. is not great when the speech is a little quieter only than preferred but substantial benefit accrues when speech level is too quiet. As previously stated, the d.a.g.c. was designed to adapt to talkers changes in speech levels on a continuous basis, the assumption made is that talkers tend to maintain a roughly constant level appropriate to them during a conversation. It is further assumed that occasional variations in talkers level, leading to changes of gain during conversation, will be acceptable. However, experimental evidence suggests that this assumption may not be valid as listeners, in majority of cases, commented upon the adaptive nature of the experiment. Awareness of this by the listeners may have led them to vote against insertion of gain when they may otherwise have judged it to be desirable. This being the case, assessment of the most suitable duration of the time constant was inconclusive. For this reason, the next stage had the same time constants: 0 ms, 100 ms, 333 ms and 1000 ms. However, the speech levels are confined to where they matter most, i.e. below -38 dBV at listener's ears.

4.4 SECOND STAGE TEST

To have a more conclusive experiment, two voices, α and β, were used, both males; therefore, with two levels and four time constants there were 16 treatments, which are too many to be presented to any individual listener; accordingly the experiment was based upon a balanced incomplete block[26,27,28].

The two levels chosen were l1=-38 dBV and l2=-44 dBV. The former required 6 dB gain and the latter 12 dB gain. The time constants were identified as: t0=0 ms, t1=100 ms, t2=333 ms and t3=1000 ms.

The treatments, v_t, are numbered from 1 to 16. The test was carried out using 16 subjects, or blocks, each one being presented with six, k, 45-second samples of speech; the number of replications, r, was six. The number of times, λ, each pair of treatments appears in the same block, b, was[26, 27, 28]:

$$\lambda = \frac{r(k-1)}{v_t-1} = \frac{6(6-1)}{16-1} = 2$$

TABLE 4.1

Identification of the Treatments

Treat-ments	Levels	Voices	Time Constants
1	11	α	t0
2	11	α	t1
3	11	α	t2
4	11	α	t3
5	11	β	t0
6	11	β	t1
7	11	β	t2
8	11	β	t3
9	12	α	t0
10	12	α	t1
11	12	α	t2
12	12	α	t3
13	12	β	t0
14	12	β	t1
15	12	β	t2
16	12	β	t3

Table 4.1 shows the identification of the treatments. level voices time constants.

Table 4.2 shows the arrangements, randomised in the way they were presented to the subjects.

Subjects	I	II	III	IV	V	VI
A	4	8	1	11	14	15
B	10	11	6	12	2	14
C	14	13	5	10	7	4
D	6	10	15	13	1	16
E	13	6	9	14	3	8
F	12	3	10	15	8	5
G	9	15	11	5	13	2
H	11	4	16	3	9	10
I	1	2	3	4	5	6
J	15	9	4	6	12	7
K	2	7	8	9	10	1
L	3	1	13	7	11	12
M	5	12	14	1	16	9
N	7	14	2	16	15	3
O	16	5	7	8	6	11
P	8	16	12	2	4	13

TABLE 4.2

Randomisation of the Treatments

The listeners gave their judgements of the six sentences accordingly with the following scale [24], Table 4.3:

Score Value	
5	Much louder than preferred
4	Louder than preferred
3	Preferred
2	Quieter than preferred
1	Much quieter than preferred.

TABLE 4.3

Scale 4A

The answers given by the subjects are in Table 4.4.

TREATMENTS	SUBJECTS																TOTAL
	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	
1	3			3					3		3	3	3				18 ↓i
2		4					5		4		3			5		4	25
3					4	3		3	3			4		5			22
4	3		3					3	3	4						4	20
5			3			2	4		3				3		3		18
6					5				4	5					4		26
7			4							4	3	5		4	4		24
8	3				3	4					3				4	3	20
9					3		3	2		3	2		2				15
10			4	4		2		3			2						18
11	3						4	3				4			3		20
12						2				3		4	3			4	19
13			2	1	3		2					3				2	13
14	2		3		4								4	3			19
15	3			3		2	4			4				4			20
16				4				3					4	4	4	3	22
TOTAL	17	20	19	19	22	15	22	17	20	23	16	23	19	25	22	20	$319 = y_t$

j→

TABLE 4.4 Answers from the subjects to the test.
Note: y_{ij} represents the value which is at row i and column j
$N_a = 96$, number of answers.

ANALYSIS OF VARIANCE OF SECOND STAGE TEST

$$SS_T = \sum_i \sum_j Y_{ij}^2 - \frac{Y_{..}^2}{N_a}$$

$N_a = 96$

$a = 16$

$b = 16$

$k = 6$

$\lambda = 2$

$$SS_{blocks} = \sum_{j=1}^{b} \frac{Y_{j}^2}{K} - \frac{Y^2}{N_a}$$

$$SS_{T.adjusted} = k \frac{\sum^{a} Q_i^2}{\lambda a} \qquad Q_i = y_i \cdot \frac{1}{k} \sum_i^{b} x_{ij} \, y_{ij}$$

$$SS_E = SS_T - SS_{T.adjusted} - SS_{blocks}$$

$$SS_T = 1\times(1)^2 + 12\times(2)^2 + \ldots. + 6\times(5)^2 - \frac{319^2}{96} = 62.99$$

$$SS_{blocks} = \sum_{1}^{16} \frac{Y_i^2}{6} - \frac{319^2}{96} = \frac{1}{6}(17^2+20^2+\ldots+20^2) - \frac{319^2}{96} = 19.49$$

$$Q_1 = 18 - \frac{1}{6}(17+19+20+16+23+19) = -1$$

$$Q_2 = 25 - \frac{1}{6}(20+22+20+16+25+20) = 4.5$$

$$Q_3 = 22 - \frac{1}{6}(22+15+17+20+23+25) = 1.67$$

$$Q_4 = 20 - \frac{1}{6}(17+19+17+20+23+20) = 0.67$$

$$Q_5 = 18 - \frac{1}{6}(19+15+22+20+19+22) = -1.50$$

$$Q_6 = 26 - \frac{1}{6}(20+19+22+20+23+22) = 5.00$$

$$Q_7 = 24 - \frac{1}{6}(19+23+16+23+25+22) = 2.67$$

$$Q_8 = 20 - \frac{1}{6}(17+22+15+16+22+20) = 1.37$$

$$Q_9 = 15 - \frac{1}{6}(22+22+17+23+16+19) = -4.83$$

$$Q_{10} = 18 - \frac{1}{6}(20+19+19+15+17+16) = 0.33$$

$$Q_{11} = 20 - \frac{1}{6}(17+20+22+17+23+22) = -0.17$$

$$Q_{12} = 19 - \frac{1}{6}(20+15+23+23+19+20) = -1.00$$

$$Q_{13} = 13 - \frac{1}{6}(19+19+22+22+23+20) = -7.83$$

$$Q_{14} = 19 - \frac{1}{6}(17+20+19+22+19+22+19+25) = -1.33$$

$$Q_{15} = 20 - \frac{1}{6}(17+19+15+12+23+25) = -0.17$$

$$Q_{16} = 22 - \frac{1}{6}(19+17+19+25+22+20) = 1.67$$

$$SS_{T.\text{adjusted}} = \frac{k\sum_{1}^{9} Q_i^2}{\lambda\, a} = \frac{6\ \{(1)^2+(4.5)^2+\ldots+(1.61)^2\}}{2\times 16} = 28.31$$

$$SS_E = 62.99 - 28.31 - 19.49 = 15.19$$

		Degree of Treat.		
Treat:	18.32	15	1.89	8.10
Blocks:	19.49	15	1.30	
Error:	15.19	65	0.24	
Total:	63.00	95		

$F_{.05}$, 15.65 = 2 < 0.10

$F_{.01}$, 15.65 = 2.33 < 0.1 They are very different.

$F_{.001}$, 18.65 = 3.57 < 0.1

The treatments are different at 5%, 1% and 0.1% levels of signficance.

DUNCAN'S MULTIPLE RANGE TEST[22]

$N_a = 96$

$n = 16$

$$\frac{k}{\lambda a} = \frac{6}{2\times 16} = 0.19 \;; \qquad S = \sqrt{0.19\times 0.24} = 0.21$$

The adjusted treatments, Q_i, are multiplied by the factor $\frac{k}{\lambda a}$ in ascending order:

$Q_{13} = -7.83\times 0.19 = -1.47$

$Q_9 = -0.91$

$Q_5 = -0.28$

$Q_{14} = -0.25$

$Q_{12} = -0.19$

$Q_1 = -0.19$

$Q_{15} = -0.03$

$Q_{11} = -0.03$

$Q_{10} = 0.06$

$Q_4 = 0.13$

$Q_8 = 0.25$

$Q_3 = 0.31$

$Q_{16} = 0.31$

$Q_7 = 0.50$

$Q_2 = 0.84$

$Q_6 = 0.94$

Using the table of significant ranges from Duncan's multiple range test with $\alpha = 0.05$ (5%) and 65 degrees of freedom[22,26] we obtained $r_{0.05}(2.65)$ through $r_{0.05}(16.65)$. With these $r_{0.05}(\ )$, we obtained R_2 through R_{16}, where

$R_2 = r_{.05}(2.65)\times 0.21$

$R_3 = r_{.05}(3.65)\times 0.21$

$R_4 = r_{.05}(4.65)\times 0.21$

$R_5 = r_{.05}(5.65)\times 0.21$

$R_6 = r_{.05}(6.65)\times 0.21$

$R_7 = r_{.05}(7.65)\times 0.21$

$R_8 = r_{.05}(8.65)\times 0.21$

$R_9 = r_{.05}(9.65)\times 0.21$

$R_{10} = r_{.05}(10.65)\times 0.21$

$R_{11} = r_{.05}(11.65)\times 0.21$

$R_{12} = r_{.05}(12.65)\times 0.21$

$R_{13} = r_{.05}(13.65)\times 0.21$

$R_{14} = r_{.05}(14.65)\times 0.21$

$R_{15} = r_{.05}(15.65)\times 0.21$

$R_{16} = r_{.05}(16.65)\times 0.21$

Then

$R_2 = 0.6$; $R_3 = 0.63$; $R_4 = 0.65$

$R_5 = 0.67$; $R_6 = 0.68$; $R_7 = 0.69$

$R_8 = 0.70$; $R_9 = 0.70$; $R_{10} = 0.71$

$R_{11} = 0.71$; $R_{12} = 0.71$; $R_{13} = 0.71$

$R_{14} = 0.72$; $R_{15} = 0.72$ and $R_{16} = 0.72$.

The comparisons yielded:

Treatment 6 ≠ Treatment 5

Treatment 6 ≠ Treatment 3

Treatment 6 ≠ Treatment 4

Treatment 6 ≠ Treatment 8

Treatment 6 = Treatment 2

Treatment 6 = Treatment 7

Treatment 6 = Treatment 3

Treatment 2 = Treatment 3

Treatment 2 = Treatment 7

Treatment 3 = Treatment 7

Treatment 16 = Treatment 12

Treatment 16 = Treatment 14

Treatment 16 = Treatment 15

Treatment 16 ≠ Treatment 13

Treatment 8 = Treatment 4

Treatment 8 = Treatment 5

Treatment 8 = Treatment 1

Treatment 4 = Treatment 5

Treatment 4 = Treatment 1

Treatment 10 ≠ Treatment 9

Treatment 10 = Treatment 11 = Treatment 12

Treatment 10 = Treatment 15 = Treatment 17

Treatment 11 ≠ Treatment 9

Treatment 11 ≠ Treatment 13

Treatment 15 ≠ Treatment 13

Treatment 15 ≠ Treatment 9

Treatment 14 ≠ Treatment 13

Analysing the results, it is concluded that, in general, a telephone network with low levels of speech is preferred with the d.a.g.c. is in use.

At low levels, the preference for the d.a.g.c. is almost independent of the time constant, with a slight preference is for shorter time constant; at high level the preference is for short time constant, even though it was not conclusive whether 100 ms or 333 ms is preferred.

At high level of speech, the mean obtained with the d.a.g.c. in and using a time constant of 1000 ms is equal to that without. It could be said that people prefer some help if the level is too low; if it is in the region of just quieter than preferred they prefer the introduction of gain and a fairly short time constant. Fig. 4.7 gives the results of the test.

The next stage experiment was to find out if there is a conclusive difference or similarity between d.a.g.c. with 100 ms or 333 ms.

4.5 THIRD STAGE TEST

In this experiment, two male voices, α, β, were used; two time constants, $t1 = 100$ ms and $t2 = 333$ ms, and two levels $\ell 1 = -38$ dBV and $\ell 2 = -44$ dBV. Therefore, the design was a 2^3 factorial design [26,27,28].

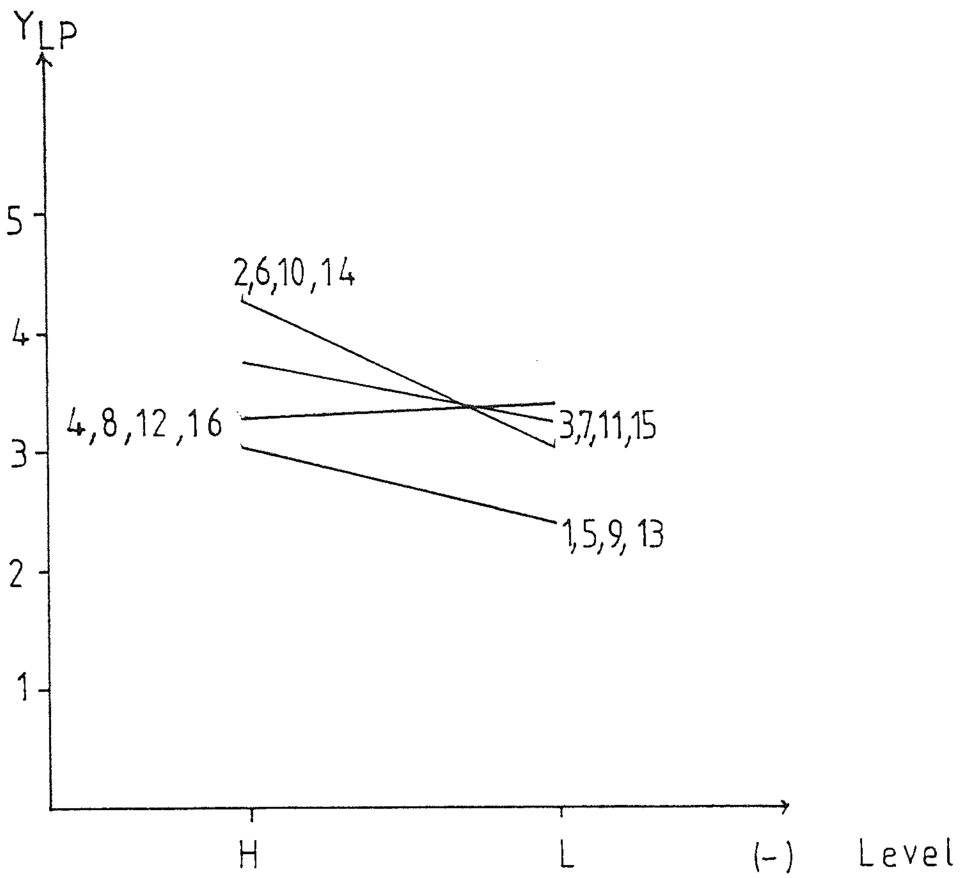


Fig. 4.7 Y_{LP} vs. Level

Notes: 1,5,9,13 No d.a.g.c
2,6,10,14, d.a.g.c. in with 100 ms
3,7,11,15, d.a.g.c. in with 333 ms
9,8,12,16, d.a.g.c. in with 1000 ms

The treatments are defined as follows, Table 4.5:

Treatment	Levels (ℓ)	Voices (v)	Time constants (t)	
1	ℓ_1	α	t_1	(1)
2	ℓ_1	α	t_2	t
3	ℓ_1	β	t_1	v
4	ℓ_2	β	t_2	$v \times t$
5	ℓ_1	α	t_1	ℓ
6	ℓ_2	α	t_2	$\ell \times t$
7	ℓ_1	β	t_1	$\ell \times v$
8	ℓ_2	β	t_2	$\ell \times v \times t$

TABLE 4.5 Identification of Treatments

The randomised arrangement of them is[28,29] in Table 4.6.

5	1	7	2	4	6	3	8
8	3	5	2	4	1	6	7
5	6	2	4	3	8	7	1
6	5	8	2	1	3	7	4
5	3	8	1	4	6	7	2
7	8	2	5	4	1	6	3
7	4	2	8	3	5	1	6
2	1	3	7	5	4	6	8

TABLE 4.6 Randomisation of Treatments

The listeners, 8 altogether, heard 8 45-second speech samples, which were previously processed, and recorded, by the 32-channel P.C.M. equipment and the PDP-11 and gave their opinion about each sample accordingly with the opinion scale 4A[24], Table 4.3. The 8 samples were presented accordingly with the conditions defined in the treatments and in the order given by the randomised arrangement.

The test was carried out in a quiet room in which about 50 dBA of room noise was introduced. The answers the listeners gave are in the Table 4.7.

Treatment	Listeners' Scores								
1	4	4	4	3	4	3	4	2	28
2	3	4	4	5	3	3	4	3	29
3	3	3	3	4	3	3	4	3	26
4	3	5	3	5	3	3	3	4	29
5	3	2	4	4	3	2	3	2	23
6	4	4	4	4	2	2	3	3	26
7	3	3	3	3	3	3	3	2	23
8	3	4	3	3	3	3	3	2	24
$\rightarrow y_{.j}$	26	29	25	31	24	22	27	21	208 $= T$

$\downarrow y_{i}$

TABLE 4.7 Listeners answers to the test.

Using the Yate's method[27], to analyse Table 4.7.

(1)	23	49	96	208	
t_c	26	47	112	8	.25
v_g	23	57	4	4	.13
$t_c \times v_g$	24	55	4	0	0
ℓ	28	3	-2	16	.5
$\ell \times t_c$	29	1	-2	0	0
$\ell \times v_g$	26	1	-2	0	0
$\ell \times t_c \times v_g$	29	3	2	4	.13

t_c represents the time constants; v_g the voices and ℓ the speech levels.

	Sum of Squares	Degree of Freedom	Mean Sum of Squares
Total	34	63	
Row	5.5	7	.7857
Column	10.5	7	
Error	18	49	.3677

$$\sqrt{\frac{4\times 0.3673}{64}} = 0.1515$$

Using t with 49 of freedom[26,27,28].

5% = 2.016 0.1515 = 0.007

1% = 2.682 0.1515 = 0.4062

0.1% = 3.510 0.1515 = 0.5318

As expected the level is a significant factor. The time is not significant at all.

As general conclusion, the d.a.g.c. is preferred and it is more if the level is less than -38 dBV, in relation with the time constant, people prefer a short one (100-333 ms) to a long one.

CHAPTER 5

NOISE INTRODUCED BY THE D.A.G.C.

5.1 QUANTIZING NOISE

A device that introduces attenuation into an A-law P.C.M. system degrades the signal-to-noise ratio at low levels by about 8 dB; at middle and high levels there is no degradation at all[6,7,16,17].

As was discussed in Chapter 2, attenuation is needed at high levels only and therefore the signal-to-noise ratio due to circuit noise is not affected. Also, attention is mainly concentrated here upon the effect of adding gain.

The quantizing noise power, nq, is calculated as follows[7,35,36]:

$$E|(v-Ku)^2|=nq \tag{5.1}$$

where

E is the mean or expected value,

v is the input to the d.a.g.c.,

u is the output of the d.a.g.c.,

K is the coefficient of linear regression of v on u and is expressed as

$$K = E(u.v)/E(u^2) \tag{5.2}$$

K should be 2 or 4, if 6 dB or 12 dB gains were introduced. K represents the value which minimises Eqn. (5.1)[29,31].

In order to calculate Eqn (5.1), it is necessary to express the P.C.M. signal in a mathematical way. The P.C.M. system considered has the following characteristics[6,7,15]

(a) A/87.6/13 law,

(b) Mid-riser, that is there is a decision amplitude at zero,

(c) Decision Level Assignment (D.L.A.), that is the output of the decoder corresponds to the mid-value of the adjacent decision amplitudes.

A digital representation of the compressed signal X_c is expressed in terms of m binary digits[37] which represent the segment number L_s, and n binary digits representing the quantizing step V_s in each segment. Then, the total number of segments, M, in one polarity is 2^m, and the total number of quantizing steps, N_s, is 2^n.

Therefore[7],

$$X_c(L_s,V_s) = V_s+N_s.L_s \tag{5.3}$$

where

$X_c \varepsilon\ (0,1,\ldots N_sM-1)$

$L_s \varepsilon\ (0,1,\ldots M-1)$

$V_s \varepsilon\ (0,1,\ldots N_s-1)$

The digitally linearised expanded signal, y, is[40]:

$$y = 2^{L_s - \eta}\,(V_s + 16\eta + 1/2) \tag{5.4}$$

where

$$\eta \begin{cases} 0 \text{ if } L_s = 0 \\ 1 \text{ if } L_s \neq 0 \end{cases}$$

Using Eqn (5.4), the positive outputs levels for the 8-bit, A-law are given in Table 5.1.

	Segment numbers							
L_s	1	1	2	3	4	5	6	7
OUTPUTS	0.5	16.5	33	66	132	264	528	1056
	1.5	17.5	35	70	140	280	560	1120
	2.5	18.5	37	74	148	296	592	1184
	3.5	19.5	39	78	156	312	624	1248
	4.5	20.5	41	82	164	328	656	1312
	5.5	21.5	43	86	172	344	688	1376
	6.5	22.5	45	90	180	360	720	1440
	7.5	23.5	47	94	188	376	752	1504
	8.5	24.5	49	98	196	392	784	1568
	9.5	25.5	51	102	204	408	816	1632
	10.5	26.5	53	106	212	424	848	1696
	11.5	27.5	55	110	220	440	880	1760
	12.5	28.5	57	114	228	456	912	1824
	13.5	29.5	59	118	236	472	944	1888
	14.5	30.5	61	122	244	488	976	1952
	15.5	31.5	63	126	252	504	1008	2016

TABLE 5.1

Output levels (positive only)

For calculation purposes, these values are defined as u_i, where i is in the range 1 to 128.

The decision amplitudes corresponding to the outputs given in the Table 5.1 are given in the Table 5.2.

Segment numbers

L_s	1	1	2	3	4	5	6	7
V_s	0	16	32	64	128	256	512	1024
1	1	17	34	68	136	271	544	1088
2	2	18	36	72	144	288	576	1152
3	3	19	38	76	152	304	608	1216
4	4	20	40	80	160	320	640	1280
5	5	21	42	84	168	366	672	1344
6	6	22	44	88	176	352	704	1408
7	7	23	46	92	184	368	736	1472
8	8	24	48	96	192	384	768	1536
9	9	25	50	100	200	400	800	1600
10	10	26	52	104	208	416	832	1664
11	11	27	54	108	216	432	864	1728
12	12	28	56	112	224	448	896	1792
13	13	29	58	116	232	464	928	1856
14	14	30	60	120	240	480	960	1920
15	15	31	62	124	248	496	992	1984
16	16	32	64	128	256	512	1024	2048

TABLE 5.2

Decision levels amplitudes (positive only)

They are named v_i, where i ranges from 1 to 128.

The remaining outputs, for set values of 6 dB loss, 6 dB gain and 12 dB gain are shown in the tables 5.3, 5.4 and 5.5. They are indexed u_i, i ranging from 1 to 128.

TABLE 5.3

Output levels (positive only), 6 dB loss

	Segment numbers							
L_s	1	1	2	3	4	5	6	7
OUTPUT	0.5	8.5	16.5	33	66	132	264	528
	0.5	8.5	17.5	35	70	140	280	560
	1.5	9.5	18.5	37	74	148	296	592
	1.5	9.5	19.5	39	78	156	312	624
	2.5	10.5	20.5	41	82	164	328	656
	2.5	10.5	21.5	43	86	172	344	688
	3.5	11.5	22.5	45	90	180	360	720
	3.5	11.5	23.5	47	94	188	376	752
	4.5	12.5	24.5	49	98	196	392	784
	4.5	12.5	25.5	51	102	204	408	816
	5.5	13.5	26.5	53	106	212	424	848
	5.5	13.5	27.5	55	110	220	440	880
	6.5	14.5	28.5	57	114	228	456	912
	6.5	14.5	29.5	59	118	236	472	944
	7.5	15.5	30.5	61	122	224	488	976
	7.5	15.5	31.5	63	126	252	508	1008

TABLE 5.4

Output levels (positive only), 6 dB gain

	Segment numbers							
L_s	1	1	2	3	4	5	6	7
OUTPUT	0.5	33	66	123	264	528	1056	2016
	2.5	35	70	140	280	560	1120	2016
	4.5	37	74	148	296	592	1184	2016
	6.5	39	78	156	312	624	1248	2016
	8.5	41	82	164	328	656	1312	2016
	10.5	43	86	172	344	688	1376	2016
	12.5	45	90	180	360	720	1440	2016
	14.5	47	94	188	376	752	1504	2016
	16.5	49	98	196	392	784	1568	2016
	18.5	51	102	204	408	816	1632	2016
	20.5	53	106	212	424	848	1696	2016
	22.5	55	110	220	440	880	1760	2016
	24.5	57	114	228	456	912	1824	2016
	26.5	59	118	236	472	944	1888	2016
	28.5	61	122	244	488	976	1952	2016
	30.5	63	126	252	504	1008	2016	2016

TABLE 5.5

Output levels (positive only), 12 dB gain

	Segment numbers							
L_s	1	1	2	3	4	5	6	7
OUTPUTS	0.5	66	132	264	528	1056	2016	2016
	6.5	70	140	280	560	1120	2016	2016
	10.5	74	148	296	592	1184	2016	2016
	14.5	78	156	312	624	1248	2016	2016
	18.5	82	164	328	656	1312	2016	2016
	22.5	86	172	344	688	1376	2016	2016
	26.5	90	180	360	720	1440	2016	2016
	30.5	94	188	376	756	1504	2016	2016
	35	98	196	392	784	1568	2016	2016
	39	102	204	400	816	1632	2016	2016
	43	106	212	424	848	1696	2016	2016
	47	110	220	440	880	1760	2016	2016
	51	114	228	456	912	1824	2016	2016
	55	118	236	472	944	1888	2016	2016
	59	122	244	488	976	1952	2016	2016
	63	126	252	504	1008	2016	2016	2016

In order to calculate the quantizing noise power, nq, an exponential distribution is assumed for the input:

$$p(v) = \frac{1}{\sqrt{2}\,\bar{v}} \, e^{-\sqrt{2}\frac{v}{\bar{v}}} \tag{5.5}$$

nq may be expressed as:

$$nq = \int_{o}^{v_1} (v-Ku_i)^2 p(v)\,dv + \sum_{i=1}^{128} \int_{v_i}^{v_{i+1}} (v-Ku_i)^2 p(v)\,dv$$

$$+\int_{v_{128}}^{\infty} (v-Ku_{128})^2 p(v)\,dv \tag{5.6}$$

To calculate nq corresponding to a 6 dB loss, or a gain or 0 dB, 6 dB or 12 dB, it is only necessary to feed into Eqn (5.6) the values given in the Tables 5.3, 5.1, 5.4

and 5.5 in turn, respectively. The decision levels, v_i, are given in Table 5.2.

In Eqn (5.6) K is needed and from Eqn (5.2) it may be expressed:

$$\frac{K}{2} = \frac{u_1 \int_o^{v_1} vp(v)\,dv + \sum_{i=1}^{128} v_i \int_{v_i}^{v_{i+1}} vp(v)\,dv + u_{128} \int_{v_{128}}^{\infty} vp(v)\,dv}{u_1^2 \int_o^{v_1} p(v)\,dv + \sum_{i=1}^{128} v_i^2 \int_{v_i}^{v_{i+1}} p(v)\,dv + v_{128}^2 \int_{v_{128}}^{\infty} p(v)\,dv} \qquad (5.7)$$

For a 6 dB loss, or a gain of 0 dB, 6 dB or 12 dB, K is calculated by feeding the values of the Tables 5.3, 5.1, 5.4 and 5.5 respectively, into Eqn (5.7). v_i values are given in Table 5.2.

The way Tables 5.1, 5.2, 5.3, 5.4 and 5.5 and the Eqns (5.6) and (5.7) has been presented is in accord with the Basic computer program written to calculate nq, and later Q=S/nq, where S is input signal power and S is given by:

$$S = \int v^2 p(v)\,dv$$

Fig. 5.1 shows the S/nq ratio for the gain settings studied. The input is given in dB below full-load sine-wave, dBBFLS, which is the maximum level of an input sinewave not distorted by the coder.

For a 6 dB loss, the curve shape is the same as previously published[6,7,17,34].

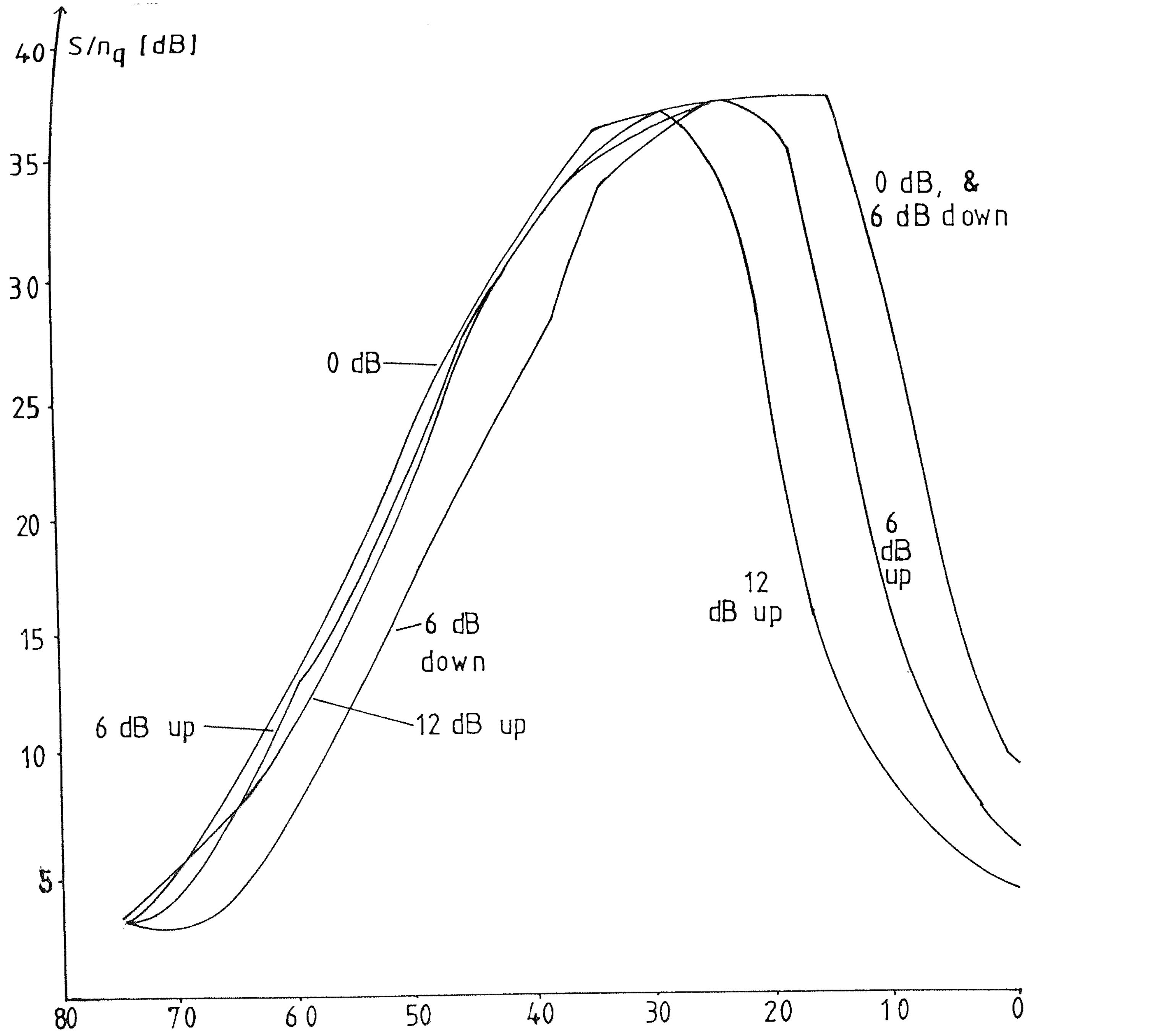


Fig. 5.1 Signal-to-quantizing noise ratio for 6 dB loss, 0 dB and gains of 6 dB and 12 dB.

It can be seen that for gains of 6 dB and 12 dB, the d.a.g.c. does not introduce much noise at low levels, the degradation is of the order of 2 dB only.

At middle levels, the curve is as that of 6 dB loss, but as we approach higher levels, the degradation becomes severe, and is due to the clipping. The clipping level has been lowered to the 6th segment for 6 dB gain and to the 5th for 12 dB gain.

The good performance at low levels, when the signal depends heavily on the first segment, in constrast with the poor performance for 6 dB loss, may be explained by the fact that when gain is introduced there is a one-to-one, or single-valued, correspondence between the P.C.M. words before and after the introduction of gain. When 6 dB loss is introduced, the assignment is multivalued, or two-to-one, that is for each pair of inputs there is just one output.

A different approach to explain the behaviour is as follows:

Table 5.6 shows the decision levels for the half-first segment and the appropriate output for 0 dB and 6 dB loss, and gains of 6 dB and 12 dB. The corresponding outputs are multiplied by the ideal gain of the channel: 1 if 0 dB, 1/2 if 6 dB attenuation, 2 for 6 dB gain and 4 for 12 dB gain.

TABLE 5.6

Decision levels and output level for 0 dB, 6 dB loss, 6 dB gain and 12 dB gain. Half-first segment and positive only.

Decision Levels	0 dB	6dB loss	6 dB up	12 dB up
0				
	1/2	1/2	1/2	1/2
1				
	1.5	1	1.25	1.625
2				
	2.5	3	2.25	2.625
3				
	3.5	3	3.25	3.625
4				
	4.5	5	4.25	4.625
5				
	5.5	5	5.25	5.625
6				
	6.5	7	6.25	6.625
7				
	7.5	7	7.25	7.625
8				
	8.5	9	8.25	8.75
9				
	9.5	9	9.25	9.75
10				
	10.5	11	10.25	10.75
11				
	11.5	11	11.25	11.75
12				
	12.5	13	12.25	12.75
13				
	13.5	13	13.25	13.75
14				
	14.5	15	14.25	14.75
15				
	15.5	15	15.25	15.75

Firstly, the two-to-one assignment mentioned previously may be observed. For 0 dB, the allocation of the quantised amplitudes is the optimum[38] and D.L.A. is in use. Secondly, looking at the outputs for gains of 6 dB and 12 dB on one side and 6 dB loss on the other, the latter presents values which are well away from the optimum allocation; whereas the 6 dB and 12 dB gains are 1/4 of an unit below or above the optimum respectively.

A problem arising here is the output for the first interval of the first segment. If the gain were achieved by shifting the P.C.M. words, the natural output for this interval would be the same output as if we had just 0 dB, because for binary word s0000000 and any shift we get s0000000. Now, as a p.r.o.m. is being used as a look-up table, a more appropriate output could have been chosen. As this factor affects the idle-circuit noise, the output given by the shifting method was chosen for the present work to keep the noise down.

Looking at Table 5.6, it could be said that the 6 dB loss device is, in fact, a much coarser quantiser than those which introduce gain. Eight intervals are being used instead of sixteen.

As the middle levels are reached, where the P.C.M. words lie in the upper segments, there is no degradation at all. Because the decision levels in the A-law system are placed at exact powers of 2 relative to each other[34], the corresponding output will be one that already has the optimum position[17,39,41,42].

When the level of the signal starts to depend heavily on the 7 th segment for a gain of 6 dB and on the 6th segment for 12 dB gain, the clipping factor degrades the signal very badly. As has been pointed out, however, we are not working in this region.

5.2 IDLE CIRCUIT NOISE

The idle circuit noise when gain is introduced is calculated using Gaussian noise as input and calculating just the output.

Fig. 5.2 shows the idle circuit noise for gains of 6 dB and 12 dB, as well as for 0 dB and 6 dB loss. As expected, the noise present when the speech is absent will be increased by the d.a.g.c. when introducing gain. But, in general, the idle circuit noise in a digital connection is very low when only one digital link is in use.

It has been shown(43,44) that when there are both speech-on noise and speech-off noise, a subjective effect of both noises can be expressed as:

$$Ne = (2/3)Noff + (1/3)n.q \tag{5.8}$$

where Ne is the subjective effect of both noises. In this work it is assumed that this relationship will hold.

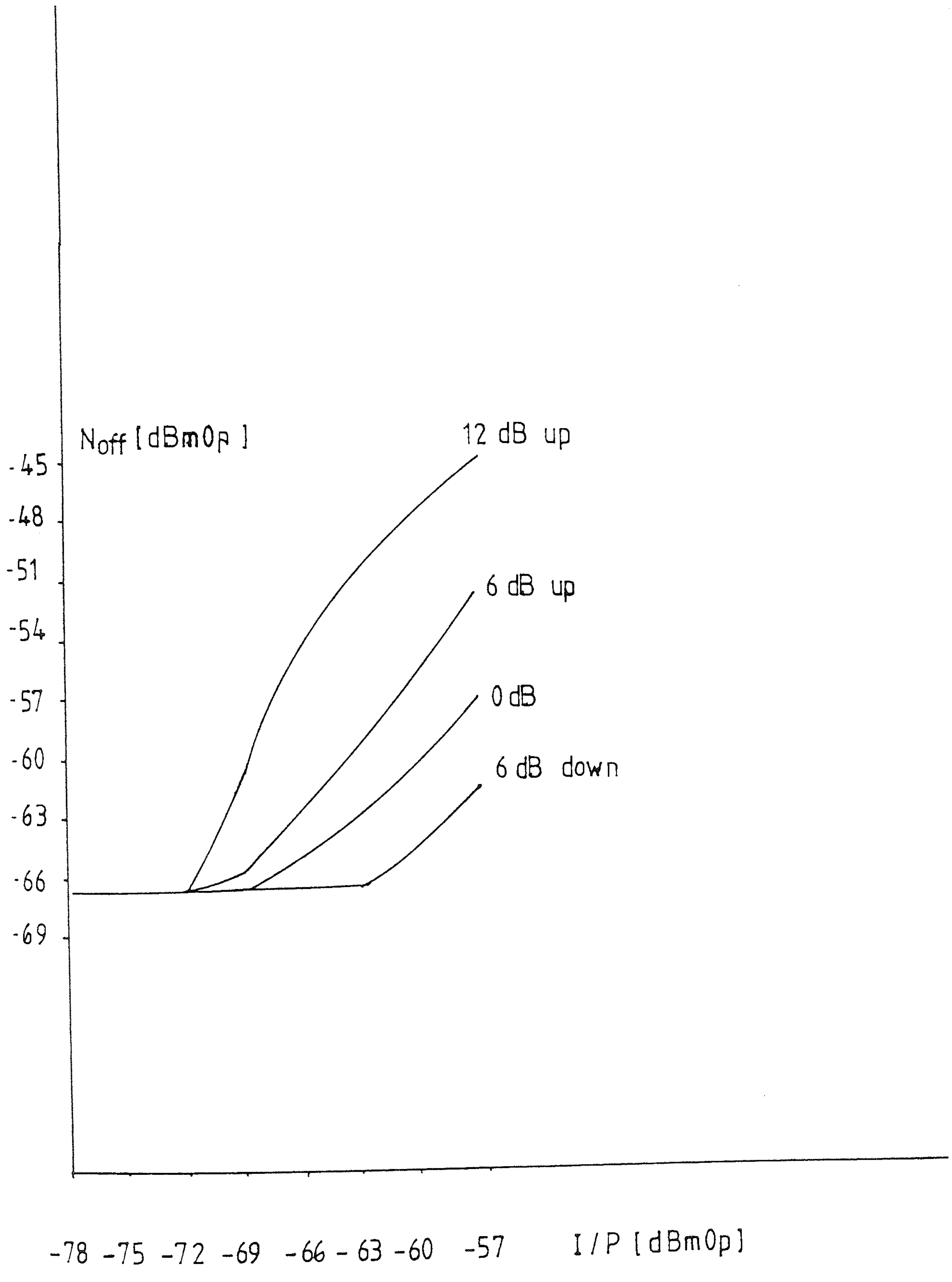


Fig. 5.2 Idle circuit noise for 6 dB loss,
0 dB and gains of 6 dB and 12 dB.

CHAPTER 6

ASSESSMENT OF THE EFFECTS OF A D.A.G.C. ON THE TRANSMISSION PERFORMANCE OF A NATIONAL TELEPHONE NETWORK

6.1 GENERAL

This chapter describes the assessment of the effects on the transmission performance of a national telephone network which uses analogue and digital transmission and has a d.a.g.c. incorporated. The d.a.g.c. is able to introduce gain when the speech level reaching the listener at the digital side is both at a low level and has a poor signal-to-noise ratio. The operation of it is restricted to be in the analogue-to-digital direction only.

The assessment was carried out using a computer model of a telephone connection devised in The Electrical and Electronic Engineering Department of the University[31,32,33] known as Telephone Connection Assessment Model (T.C.A.M.).

6.2 BRIEF DESCRIPTION OF THE T.C.A.M.

It is a 'mechanisitic model' which represents[32] not only the internal structure of a telephone connection but also its external characteristics.

The T.C.A.M. is divided into six cascaded stages. The model's first stage requires a description of the circuit elements comprising the telephone connection, beginning with the handsets. The program stores the sensitivities of the transducer elements which are defined in accordance with the CCITT recommendation P.64[31,32]. During this stage, and during the second, the electro-acoustical sensitivities are combined with chain matrix parameters describing the electrical transmission losses by cascading.

Then stage 3 is run and , using the results from stages 1 and 2, the loudness losses are calculated. Stage 4 determines the noise spectra at the ear from various sources and via various paths; and, finally, stages 5 and 6 estimate the degree of satisfactoriness. This is expressed by the percentage of people likely to suffer some difficulty in conversing (expressed as D%); and by the percentage of Excellent or Good Opinions (E+G%). The model is thus designed to simulate a subjective test.

When all the electrical elements have been specified, there are several other factors which have to be considered, among them the position of the speaker which has been chosen to be the British Model Speaking Position[32], room noise, circuit noise, etc.

In a practical environment there is no control over room noise and this is fixed throughout the assessment at 50 dBA. Circuit noise takes into account electrical noise from all sources within the telephone connection. In the T.C.A.M., this noise, named Ne in Chapter 5, is referred to a common point taken for convenience at the electrical input to the local telephone network and is the figure entered into the computer. nq is constant, nq = - 70 dBmOp and Noff is varied from -57 dBmOp down to -78 dBmOp, at the d.a.g.c. location. nq is altered to take account of the 2 dB degradation due to the d.a.g.c.

6.3 TELEPHONE CONNECTION TO BE ASSESSED

The configurations of telephone connections are quite diverse, but a possible circuit may be represented by a basic block diagram, shown in Fig. 6.1.

The analogue part will generally comprise of a number of different transmission elements. It is assumed that this part will be that which is degrading the signal level. Therefore, from the point of view of loudness loss, the digital part could be considered fairly constant if the digital path is up to the subscribers premises.

The d.a.g.c. is fitted immediately after the analogue signal has been digitised and is in a suitable form for digital manipulation before the signal is transmitted.

6.3.1 British Transmission Plan

Fig. 6.2 shows the existing British Transmission Plan[45]. The maximum loss between local exchanges is 19.5 dB, which appears through LE-GSC-GSC-GSC-LE. The loss allowances for

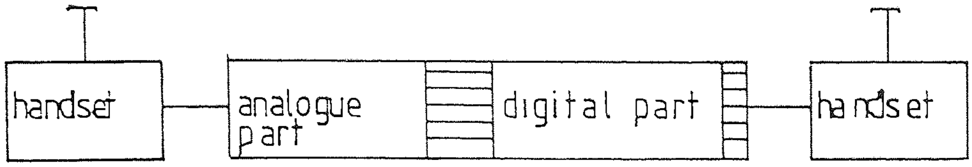


Fig. 6.1   Basic block diagram of a telephone connection.

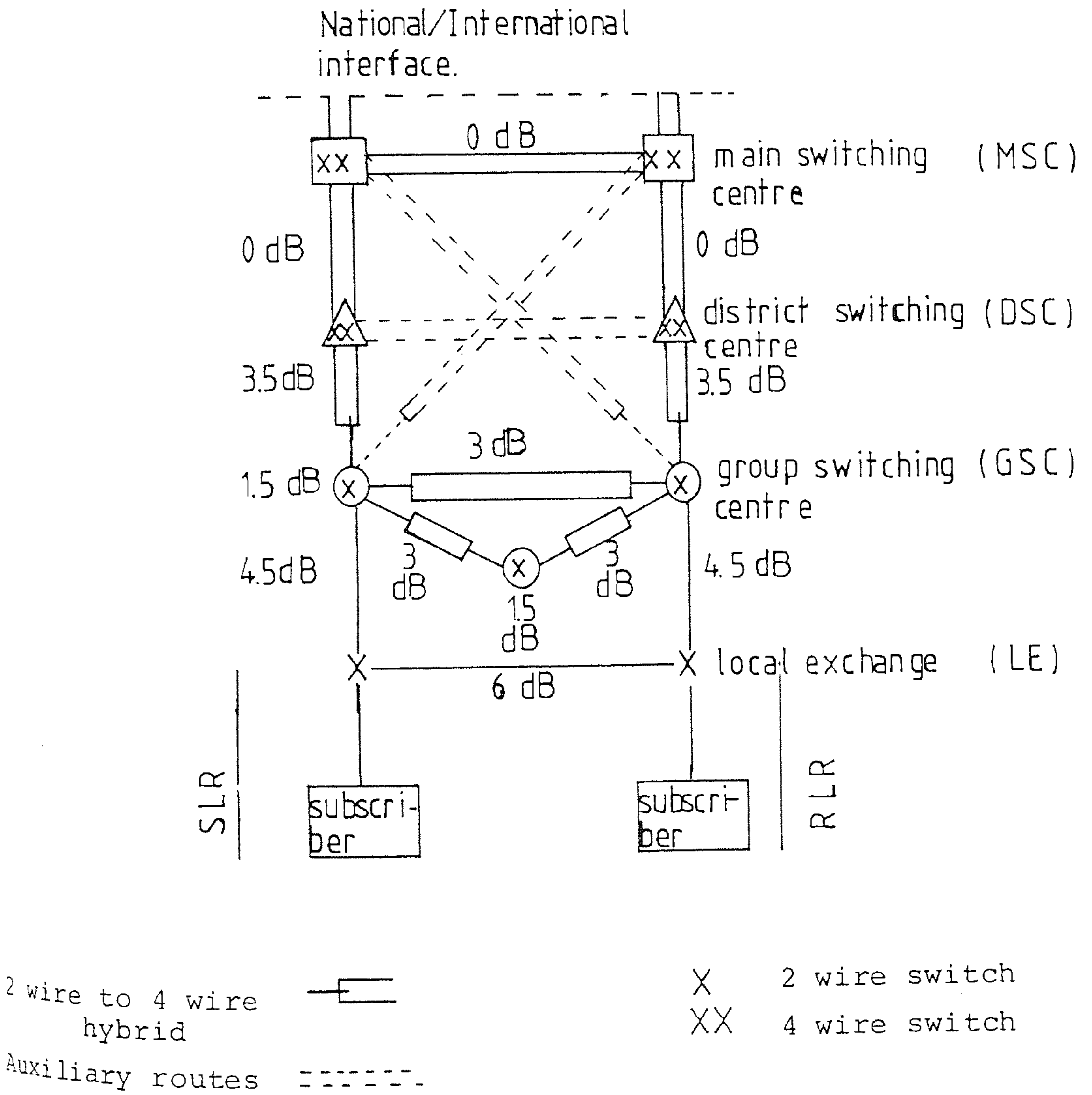


Fig. 6.2 British Transmission Plan.

switching, cabling and mismatching is 1.5 dB.

The P.C.M. junction is considered to have a 3 dB loss in accordance with the present practice, but as British Telecom will adopt, in the future, 6 dB loss junction [45], the assessment is made with this value as well.

6.3.2 Assessment Model

The assessment is made by comparing the performance of any telephone connection without a d.a.g.c. with a similar telephone connection but with d.a.g.c. incorporated.

A telephone connection will be defined by giving all its electrical components (from handset to handset) and numerically they will be specified by loudness ratings [38, 46,47] (Sending Loudness Rating, SLR; Receiving Loudness Rating, RLR; Overall Loudness Rating, OLR). The voltage reaching the listener's earphone, V_L (dBV), is given, referred to 0 dB RLR; that is the level at the input, L, of the local end. L is indicated in all connections used later on.

During the running of the assessment, the factor which determined whether d.a.g.c. was needed for a specific connection was the level of the voltage at L as it was also during the subjective tests. However, as this was a computer simulation for the assessment, the performance of the telephone connection with d.a.g.c. in was studied, even when the speech level reaching the listener was well above the assumed minimum required. This was done to check if people wanted some help with very quiet speech levels rather than from just quiet levels.

For each connection, the idle circuit noise was a parameter, it was varied from high values, although they are rare, to low values. Each specific connection was considered in isolation.

Two types of connection are considered:

(a) Connections in which the listener is connected to a digital exchange via a two-wire analogue local line.

(b) Connections in which the listener is connected to a digital exchange via a digital local line providing the equivalent for four-wire transmission.

For each type of connection the speaker is connected to an exchange in an analogue part of the national network, as shown in Fig. 6.3.

For connection type (b), there is no problem of stability. For connection type (a), it is necessary that the four-wire circuit shall remain stable even when the d.a.g.c. is inserting gain. The conditions to ensure stability are discussed in Appendix A.

It is therefore assumed in this section that the circuit is stable.

6.3.2.1 Connection Number 1

This connection is shown in Fig. 6.3.

The analogue part from handset to the local exchange (LE) is a limiting connection: the subscriber line is 5.9 km long and 0.5 mm diameter copper wire. The telephone handset is the British Telecom 7461. The local exchange is connected to the primary centre (PC) through a loaded cable junction, lcj, of 0.6 mm in diameter and a loss of 4.5 dB at 1 kHz. The PCs have a loss of 5 dB (of which 1.5 dB is allowed for switching, mismatching, etc.).

Foe each connection three types of d.a.g.c. are considered :

(a) the normal one: that which is expected to introduce 0 dB or 6 dB or 12 dB gains;

(b) a second one: introduces 0 dB or 6 dB and,

(c) able to introduce 0 dB or 12 dB.

For each situation, the computer works out results for gains of 0 dB, 6 dB, 9 dB and 12 dB. The purpose is to see the effect of moving the threshold levels at which the introduction of gain starts on the response of the subjects because people preferred the introduction of gain with very quiet levels rather than from quiet levels. The introduction of the 9 dB gain feature is, because when the signal is the limit between the 6 dB zone and the 12 dB

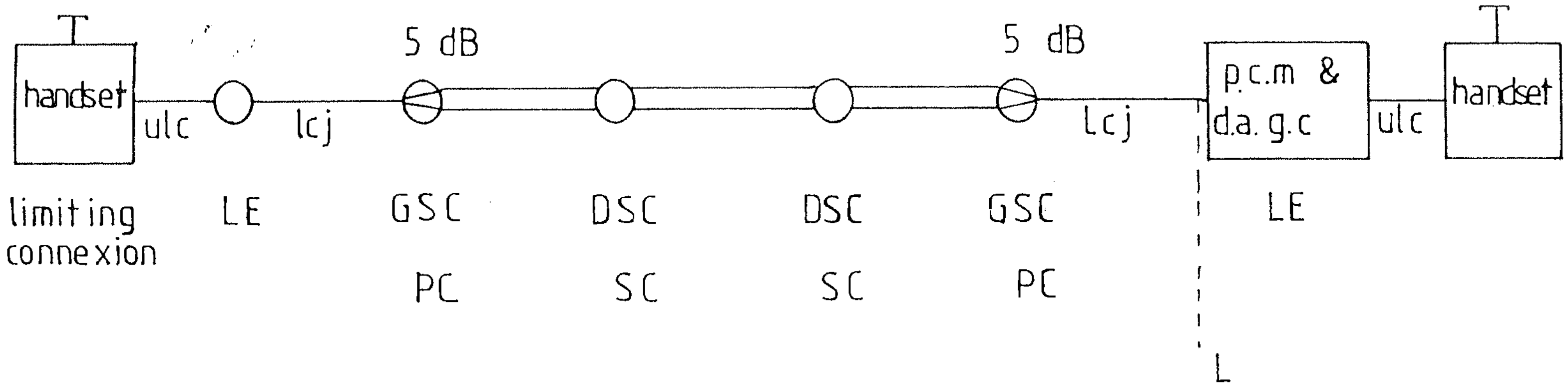


Fig. 6.3 Connection No. 1

Note:

ulc	unloaded cable
lcj	loaded cable junction
GSC	Group Switching Centre
DSC	Digital Switching Centre
PC	Primary Centre
SC	Secondary Centre
LE	Local Exchange
L	Point where voltage speech level is given, and noise.

zone, the d.a.g.c. will give on average a 9 dB gain.

The digital part is formed by a 3 dB loss system. In the T.C.A.M., it is represented by a 3 dB loss attenuator and a channel filter. In this connection, the digital side-line is 0 km long.

Fig. 6.4 gives D% and E+G% as a function of the noise. The parameters for this connection are:

SLR=10.49 dB; RLR=2.19 dB; OLR=27.87 dB; V_L=-40 dBV.

It can be seen that, at high levels of noise, the increase in (E+G%) is of the order of 10-15% for gain of 6 dB and between 11-20% for gain of 12 dB. As the level reaching the point L, Fig. 6.4 is very low, the results from the 0-12 dB d.a.g.c. and from the 0-6-12 dB d.a.g.c. should be considered identical.

The degree of difficulty, D% for any of the connections with d.a.g.c. is less than half that of the connection without it. If the noise is negligible, the degree of difficulty with a 0-6 dB gain d.a.g.c. is half of the original value and with a 0-6-12 gain device, about a quarter.

6.3.2.2. <u>Connection Number 2</u>

This connection is similar to the previous one, except that the subscriber line on the digital side will be of an

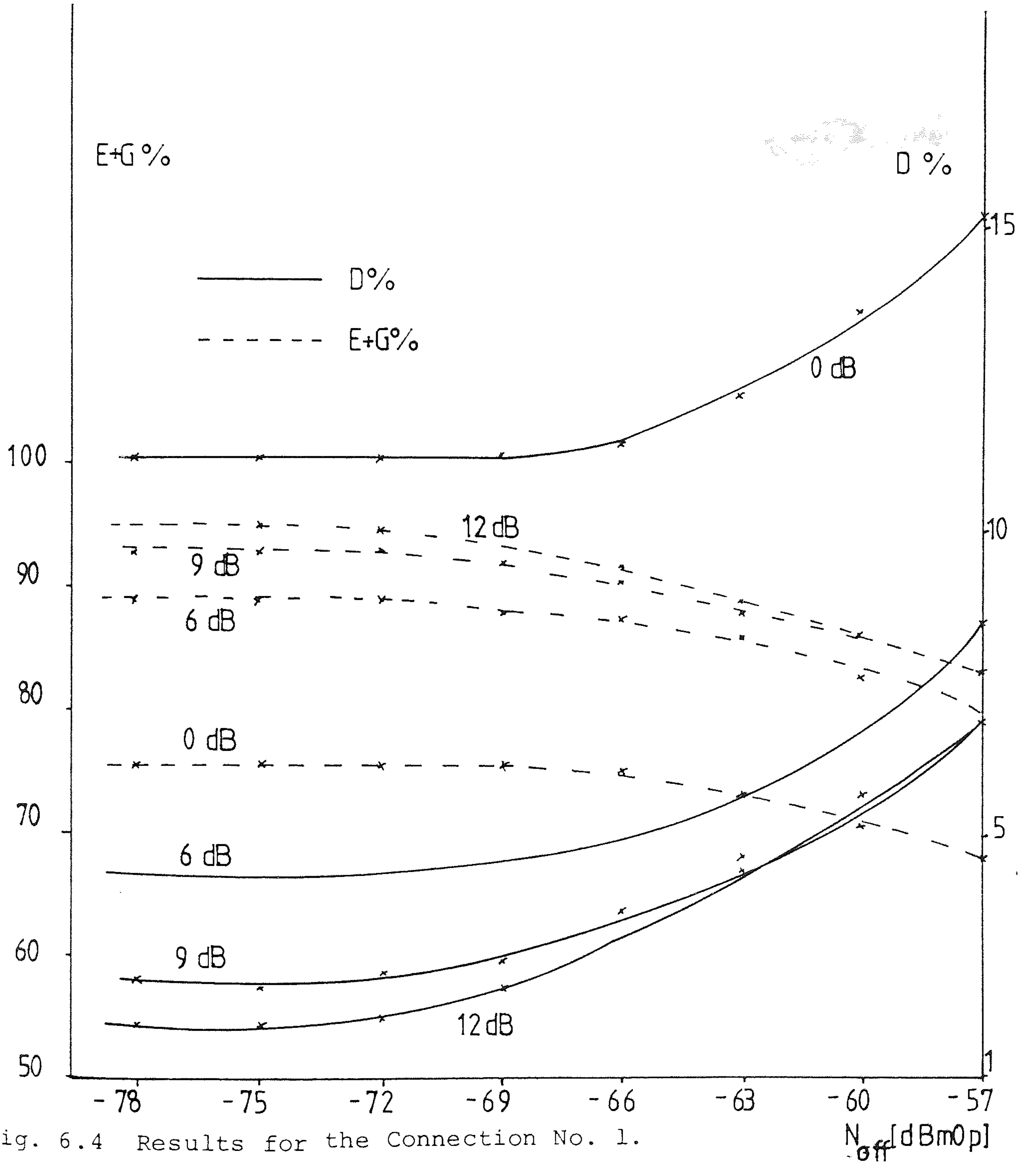


Fig. 6.4 Results for the Connection No. 1.

average length of 1.6 km. The curves are Fig. 6.5. The parameters for this connection are:

SLR=10.49; RLR=0.79 dB; OLR=28.3 dB; V_L = - 41.4 dBV

It can be seen that, at high levels of noise, the difference in performance between an 0-6 dB gain device and the 0-12 and 0-6-12 devices (these latter two will give the same results because it is expected that both will introduce 12 dB gain) is negligible.

When the noise is low, the 0-6 dB D% is twice the D% for 0-6-12 dB device. (E+G%) for 0-6-12 dB d.a.g.c. is 5% greater than the 0-6 dB.

6.3.2.3 Connection Number 3

This connection is similar to the previous one, except that the subscriber line on the digital side will be 5.9 km long. Fig. 6.6 shows the results. The parameters for this connection are:

SLR=10.49 dB; RLR=3 dB; OLR=31.76 dB; V_L=-45.6 dBV.

The conclusions are practically the same as for the previous connection. (D%) is less than half if the device is in. (E+G%) is 22% if the noise is high and 25% if the noise is low.

6.3.2.4 Connection Number 4

Here the subscriber handset is digital; therefore the

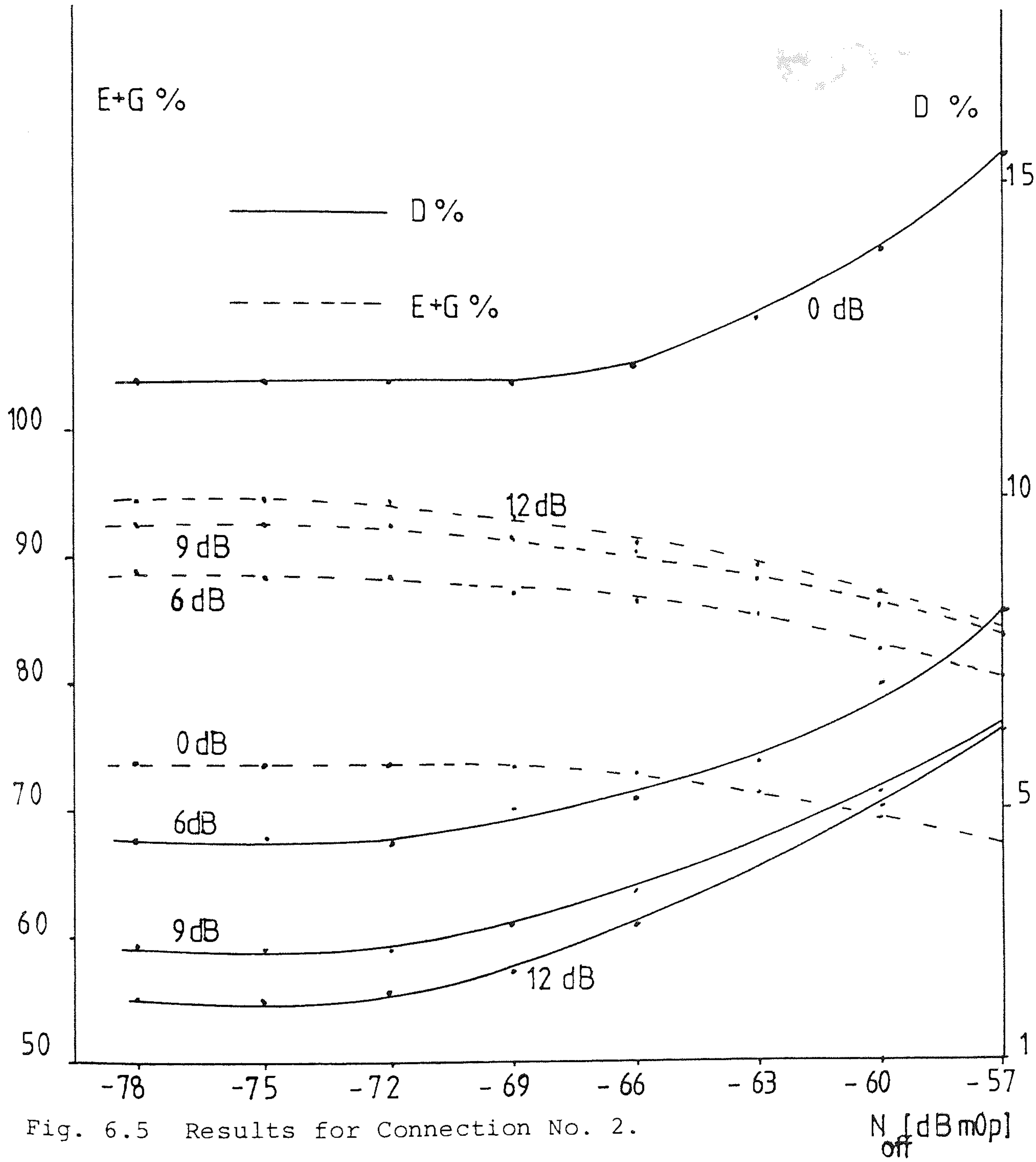


Fig. 6.5 Results for Connection No. 2.

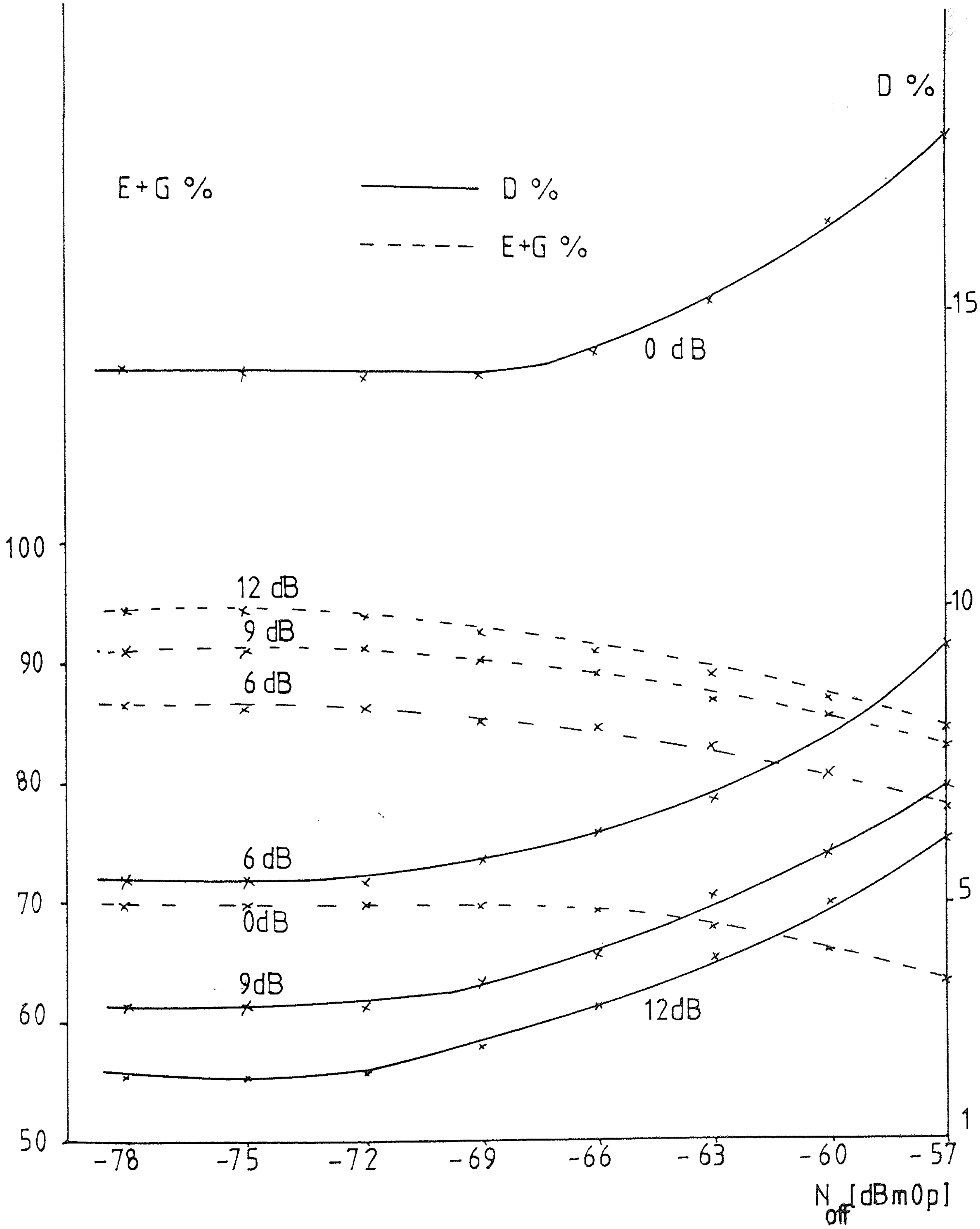


Fig. 6.6 Results for Connection Number 3

RLR is constant for all the subscribers connected to the digital end[4]. The results are in Fig. 6.7. Qualitatively, the curves as similar as the previous ones. Quantitively, (D%) is less than half if the d.a.g.c. is in. (E+G%) is about 10-20% greater. The parameters of this connection are:

SLR=10.49 dB; RLR=2.62 dB; OLR=31.95 dB; V_L=-44.84 dBV.

6.3.2.5 Connection Number 5

The analogue part is shortened to the average length, 1.6 km; keeping the digital side up to the subscriber. As a consequence the level at L is higher. The results are in Table 6.1.

It should be considered that there is a difference in performance between an 0-12 dB gain d.a.g.c. and an 0-6-12 gain d.a.g.c., due to the fact that the latter introduces an average gain of 9 dB.

6.3.2.6 Connection Number 6

The analogue part is 0 km long. Basically the results are the same, even though the RLR is larger. The results are in Table 6.2.

6.3.2.7 Connection Number 7

This connection is similar to number 4; but the P.C.M. link is considered as a 6 dB loss. The improvement due to

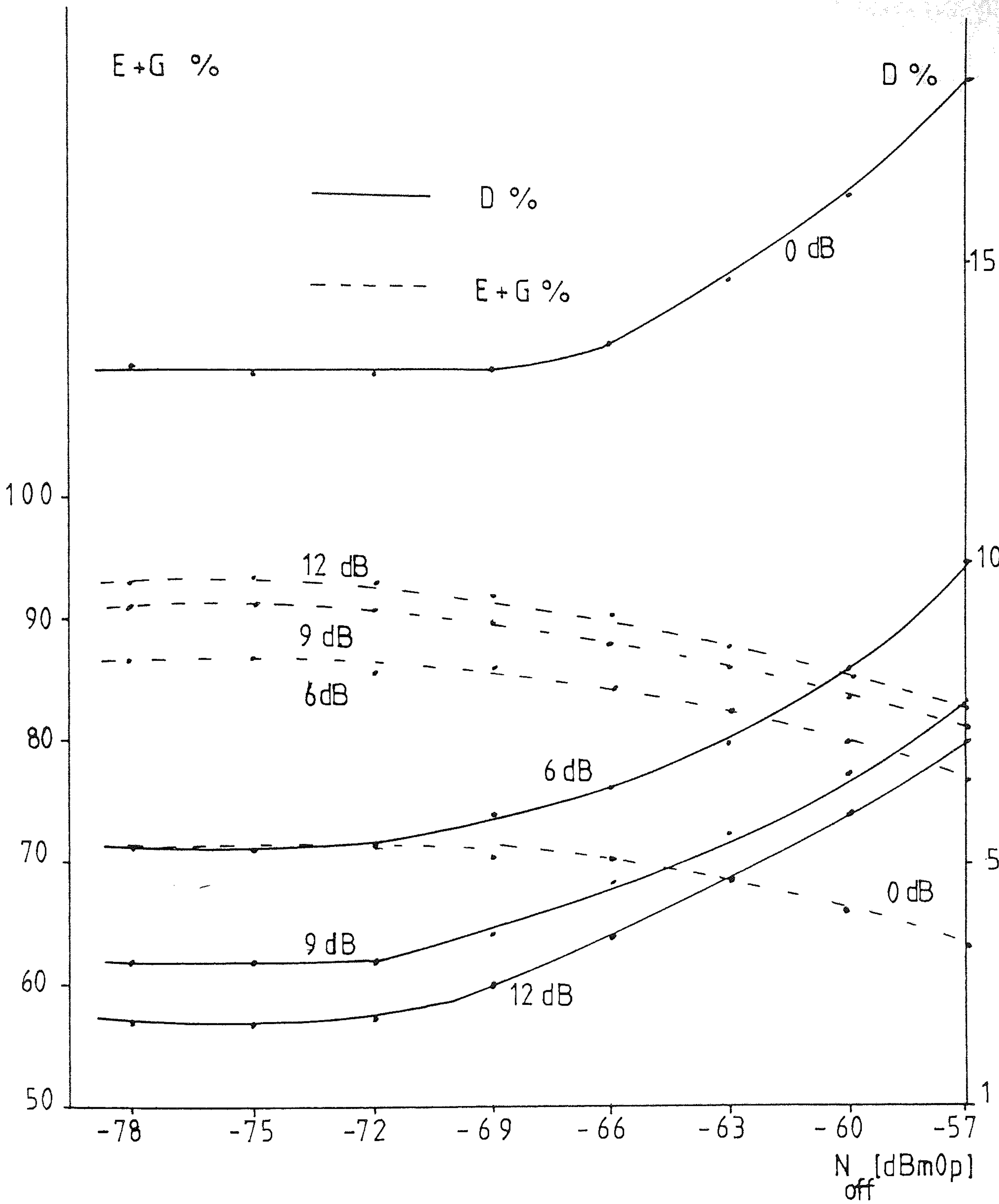


Fig. 6.7 Results for Connection Number 4

TABLE 6.1

Results of Connection Number 5

SLR(dB)	RLR(dB)	OLR(dB)	V_L (dBV)
4.77	2.62	26.56	-37.7
N_{off} (dBmOp)	(D%)	(E+G%)	D.A.G.C.
-57	15.8	66.7	OUT
	7.3	82.3	IN, 6 dB
	5.4	86.4	IN, 9 dB
	5.2	87	IN, 12 dB
-60	13.8	70	OUT
	6.2	84.7	IN, 6 dB
	4.5	88.5	IN, 9 dB
	4.2	89.2	IN, 12 dB
-63	12.4	72.6	OUT
	5.1	87.2	IN, 6 dB
	3.9	90	IN, 9 dB
	3.5	90.9	IN, 12 dB
-66	11.3	74.5	OUT
	4.5	88.5	IN, 6 dB
	3.2	91.7	IN, 9 dB
	2.7	92.7	IN, 12 dB
-69	10.9	75.3	OUT
	4.1	89.4	IN, 6 dB
	2.7	92.8	IN, 9 dB
	2.1	94.2	IN, 12 dB
-72	10.9	75.3	OUT
	3.8	90.1	IN, 6 dB
	2.4	93.5	IN, 9 dB
	1.8	95.1	IN, 12 dB
-75	10.9	75.3	OUT
	3.8	90.1	IN, 6 dB
	2.4	93.5	IN, 9 dB
	1.7	95.3	IN, 12 dB

TABLE 6.2

Results of Connection Number 6

SLR(dB)	RLR(dB)	OLR(dB)	V_L (dBV)
4.27	2.62	28.1	-36.2
N_{off} (dBmOp)	(D%)	(E+G%)	D.A.G.C.
-57	18.3	62.8	OUT
	8.6	79.8	IN, 6 dB
	6.3	84.5	IN, 9 dB
	5.7	85.7	IN, 12 dB
-60	15.9	66.5	OUT
	7.2	82.6	IN, 6 dB
	5.2	86.9	IN, 9 dB
	4.6	88.3	IN, 12 dB
-63	14.2	69.4	OUT
	5.8	85.6	IN, 6 dB
	4.4	88.7	IN, 9 dB
	3.8	90.2	IN, 12 dB
-66	12.9	71.6	OUT
	5.2	87	IN, 6 dB
	3.6	90.7	IN, 9 dB
	2.9	92.2	IN, 12 dB
-69	12.4	72.5	OUT
	4.7	88.1	IN, 6 dB
	3.1	92	IN, 9 dB
	2.3	94	IN, 12 dB
-72	12.4	72.5	OUT
	4.4	88.9	IN, 6 dB
	2.7	92.8	IN, 9 dB
	1.9	94.9	IN, 12 dB
-75	12.4	72.5	OUT
	4.4	88.9	IN, 6 dB
	2.7	92.8	IN, 9 dB
	1.8	95.1	IN, 12 dB

the d.a.g.c. is noted. Fig. 6.8 presents the results. The parameters for this connection are:

SLR=10.49 dB; RLR=5.62; OLR=34.95 dB; V_L=-47.8 dBV.

6.3.2.8 Connection Number 8

This connection is equivalent to that of Fig. 6.9.

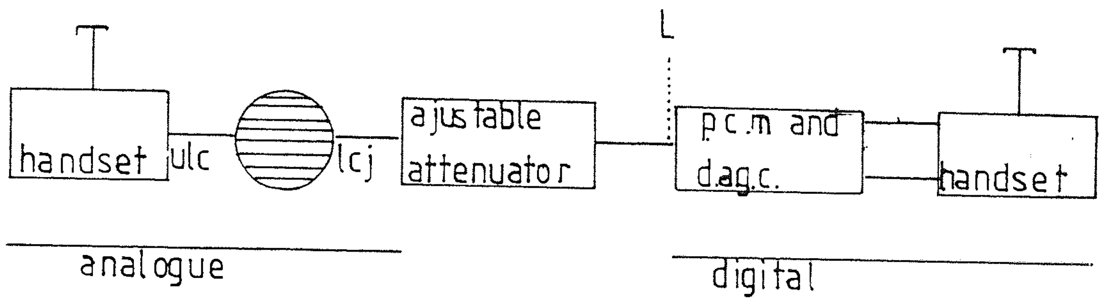


Fig. 6.9 Connection Number 8

The purpose of this is to gather information about the relationship between (D%) and (E+G%) with the speech level reaching the listener when the d.a.g.c. is installed. The programme was run several times with different values for the attenuation. The results are given in Fig. 6.10. It may be observed that the improvement at high levels of signal is small and that, if the noise level is high, the performance is slightly worse with it than without. If we go to low levels with the d.a.g.c. installed, the score is always better. Actually, the performance is better from

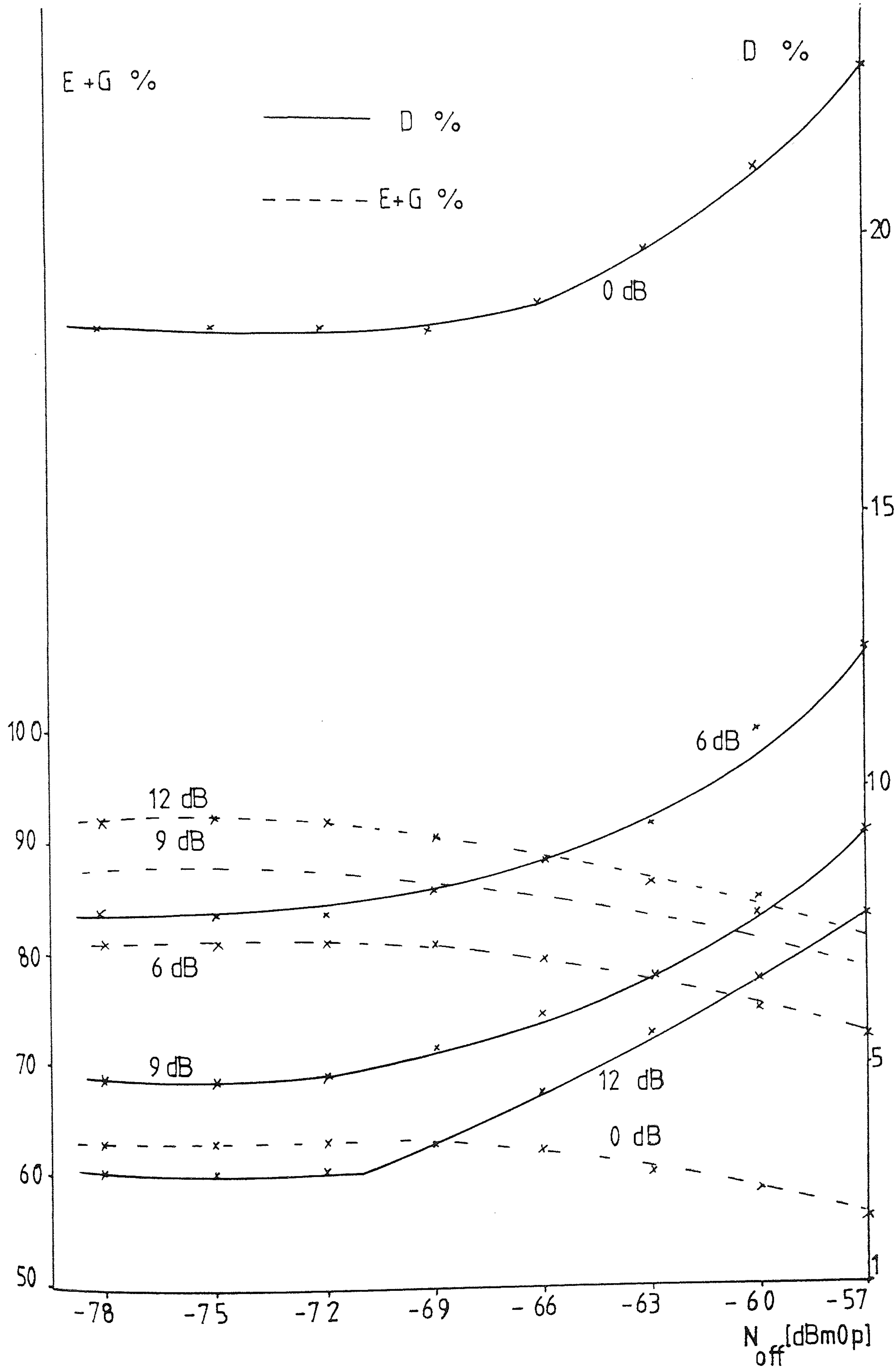


Fig. 6.8 Results for Connection Number 7.

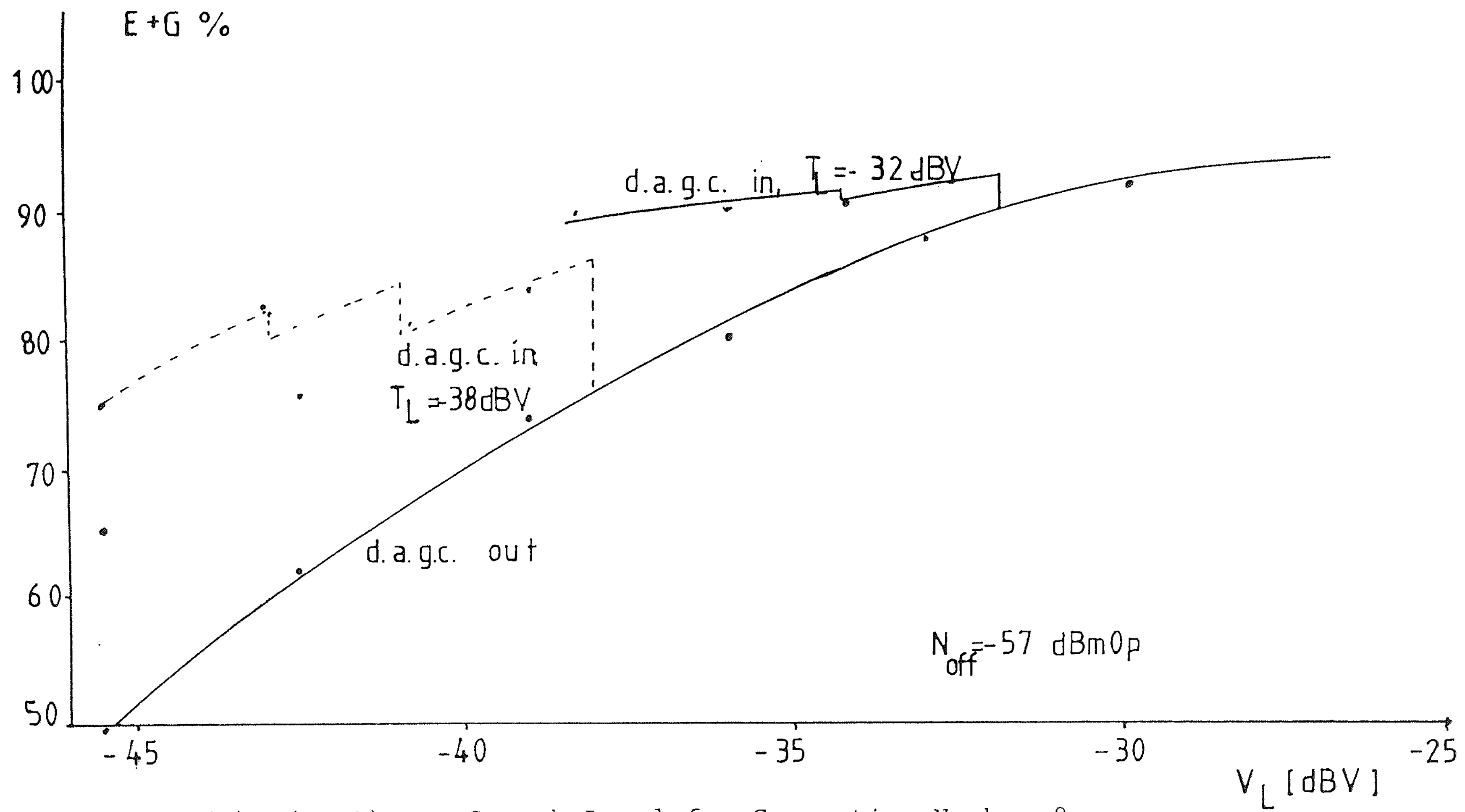


Fig. 6.10(a) (E+G%) vs. Speech Level for Connection Number 8.

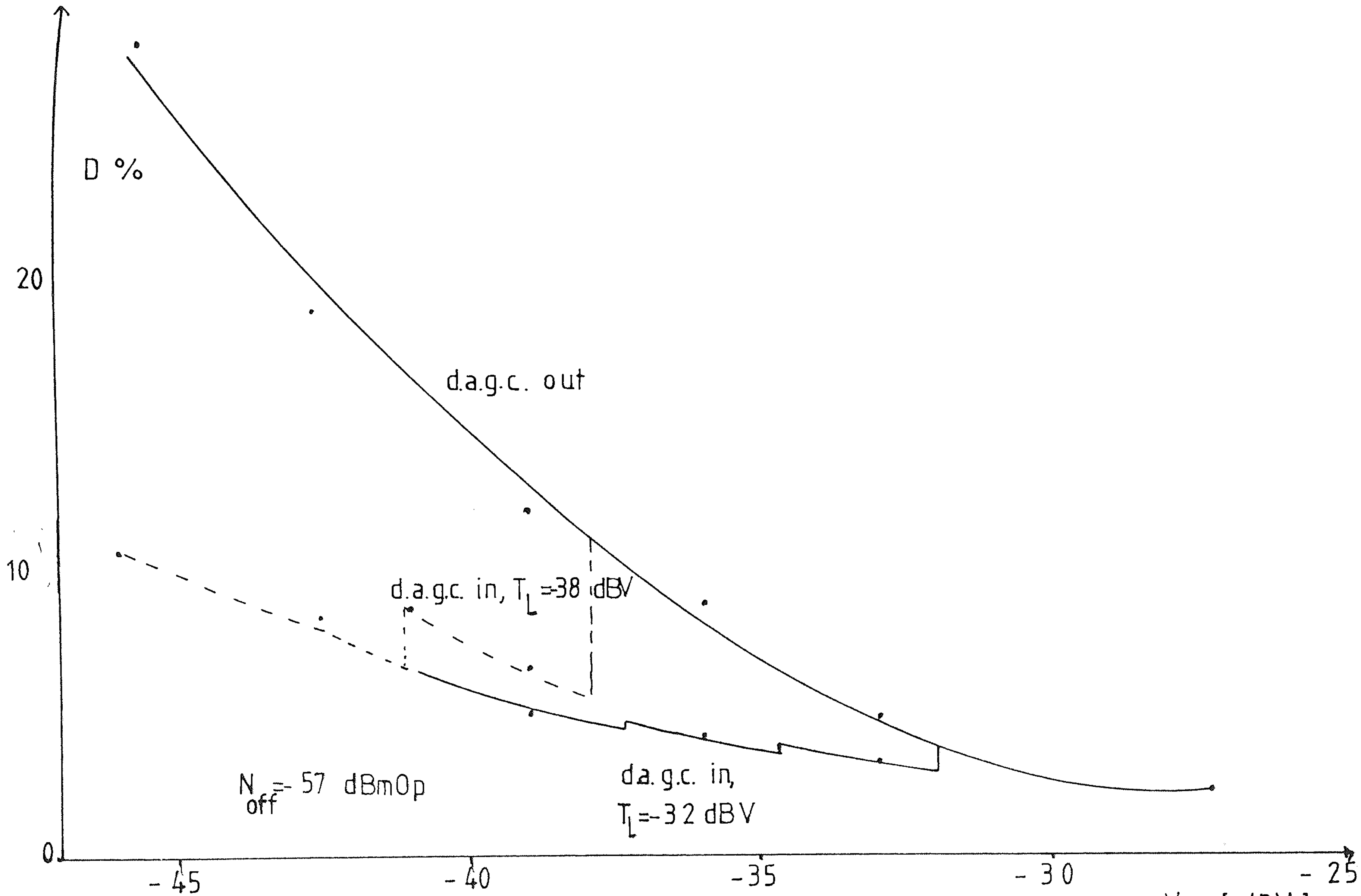


Fig. 6.10(b) (D%) vs. Speech Level for Connection Number 8.

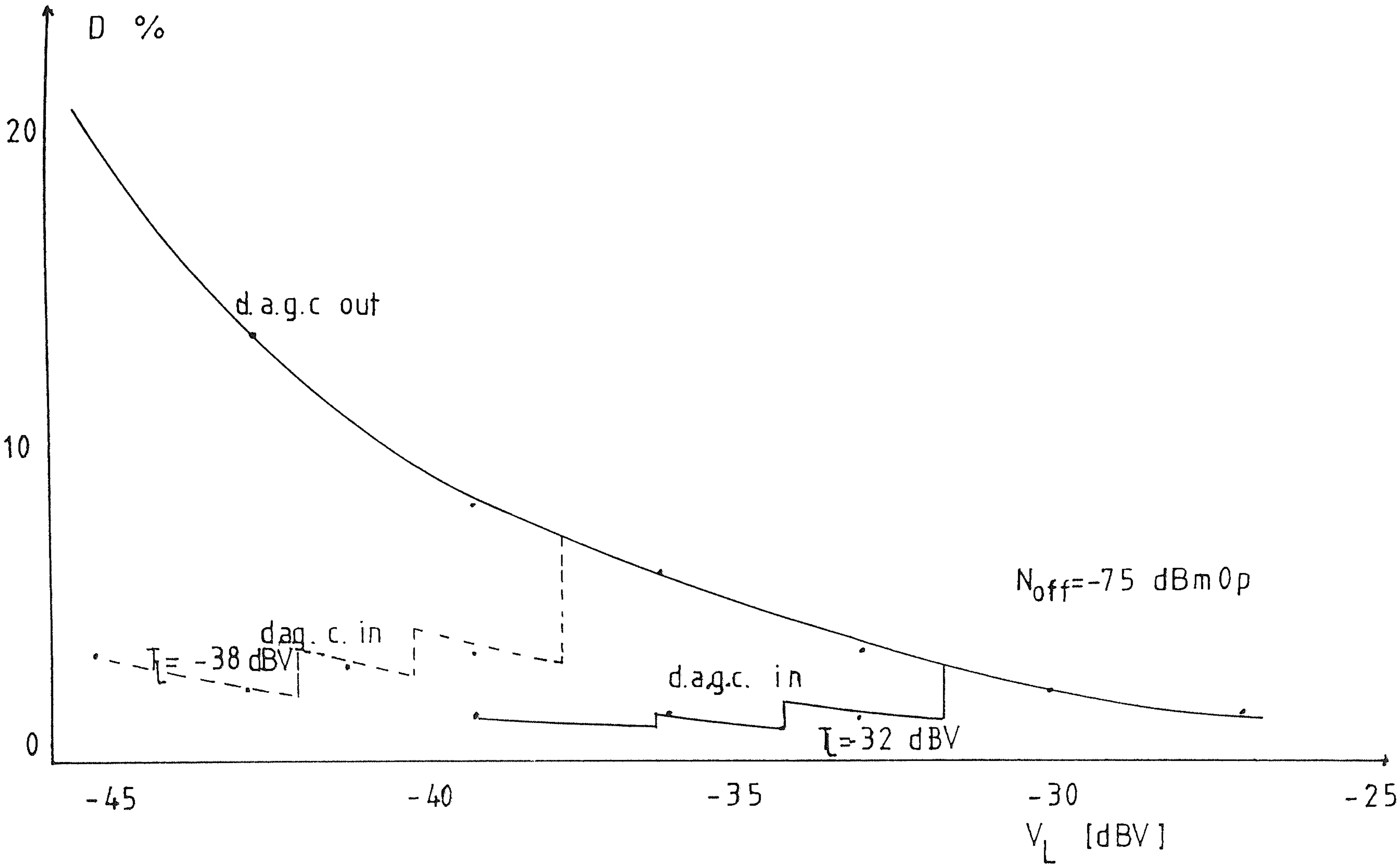


Fig. 6.10(c) (D%) vs. Speech Level for Connection Number 8

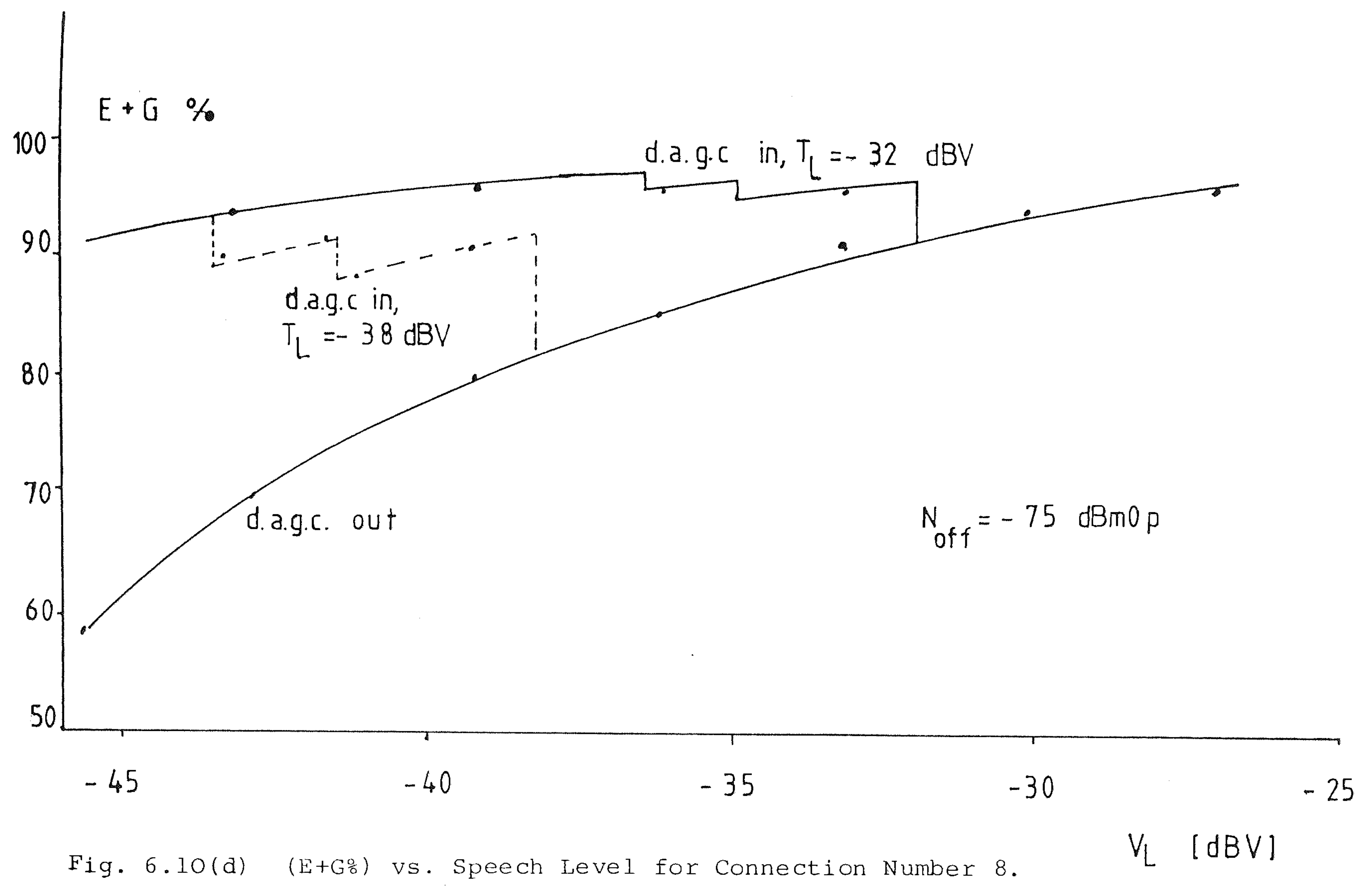


Fig. 6.10(d) (E+G%) vs. Speech Level for Connection Number 8.

about -30 dBV downwards; however, it must be taken into account that the effect of the adaptive-like nature of the d.a.g.c. is not a parameter, and in the subjective experiments people rejected it at these levels.

For this connection, the SLR and RLR are 10.49 dB and 2.62 dB. The analogue part is a limiting connection and the digital side is that of Connection Number 4. The OLR varies as the attenuator of Fig. 6.9 does. The OLR varies from 12.61 dB up to 35.51 dB.

A telephone connection with a d.a.g.c. in presents a degree of difficulty, D%, which is less than half of that without it. Being about half if the noise is high; and less if the noise is low. (E+G%) is 10-30% greater if the device is in. The least improvement is obtained if the level is near the quiet than preferred level and the greatest if the level is much quieter than preferred and low noise.

CHAPTER 7

FURTHER WORK AND CONCLUSIONS

7.1 FURTHER WORK

One of the areas where further research may be carried out is in the possibility of using the first utterence of a speaker to decide if gain is needed. This is based upon the fact that people noted the introduction of gain during the listening tests. This is so because the sample of speech, during an interval of T seconds, could present a level which is outside the average level and therefore requires less or more gain. For example, if the present gain is 6 dB, the next stage may ask for just 0 dB, or 12 dB, even though in the long term the requirement is 6 dB. This adaptive-like effect could have been the cause why people voted against the insertion of gain at quiet levels.

Some care should be taken due to the possiblity of a talker speaking very softly at the beginning of a conversation which will lead to excessive gain being inserted. However, this effect can be reduced by dealing only with 'much quieter' lines rather than just quieter.

Once the device has adjusted the gain at the beginning of the conversation, it could be used to process other channels. This could lead to the processing of more than 30 speech channels. Attention should be paid to the congestion of the device which may occur.

Another solution to the problem of people's awareness of the action of the d.a.g.c. would be the use of a 'hysteresis' d.a.g.c.; whose system characteristic (E+G%) v.s. input voltage will be like that of Fig. 7.1. Further investigation will be necessary to determine the width of the hysteresis loop, $\Delta\omega$. This solution is equivalent to the overlapping of the regions of gain settings, whereas in the solution which was adopted, there is a sharp division around that line where the adaptive-like effect is felt.

It could be argued that this solution may give the same results as having the gain adjusted at the beginning of the conversation and remaining at this value during it. This is a subject of study; but it would be expected that if $\Delta\omega$ is wide enough, they would be practically the same solution.

A further point to be looked into is the treatment given to the room noise. During subjective tests, it was considered fixed and out of the control of the system designer. At the moment, it is being studied how the ambient noise from one end affects the other and how this depends on the handset shape, for instance.

The d.a.g.c. which uses three threshold levels and one counter could be built, or simulated, and tested to compare its behaviour with that developed for this thesis.

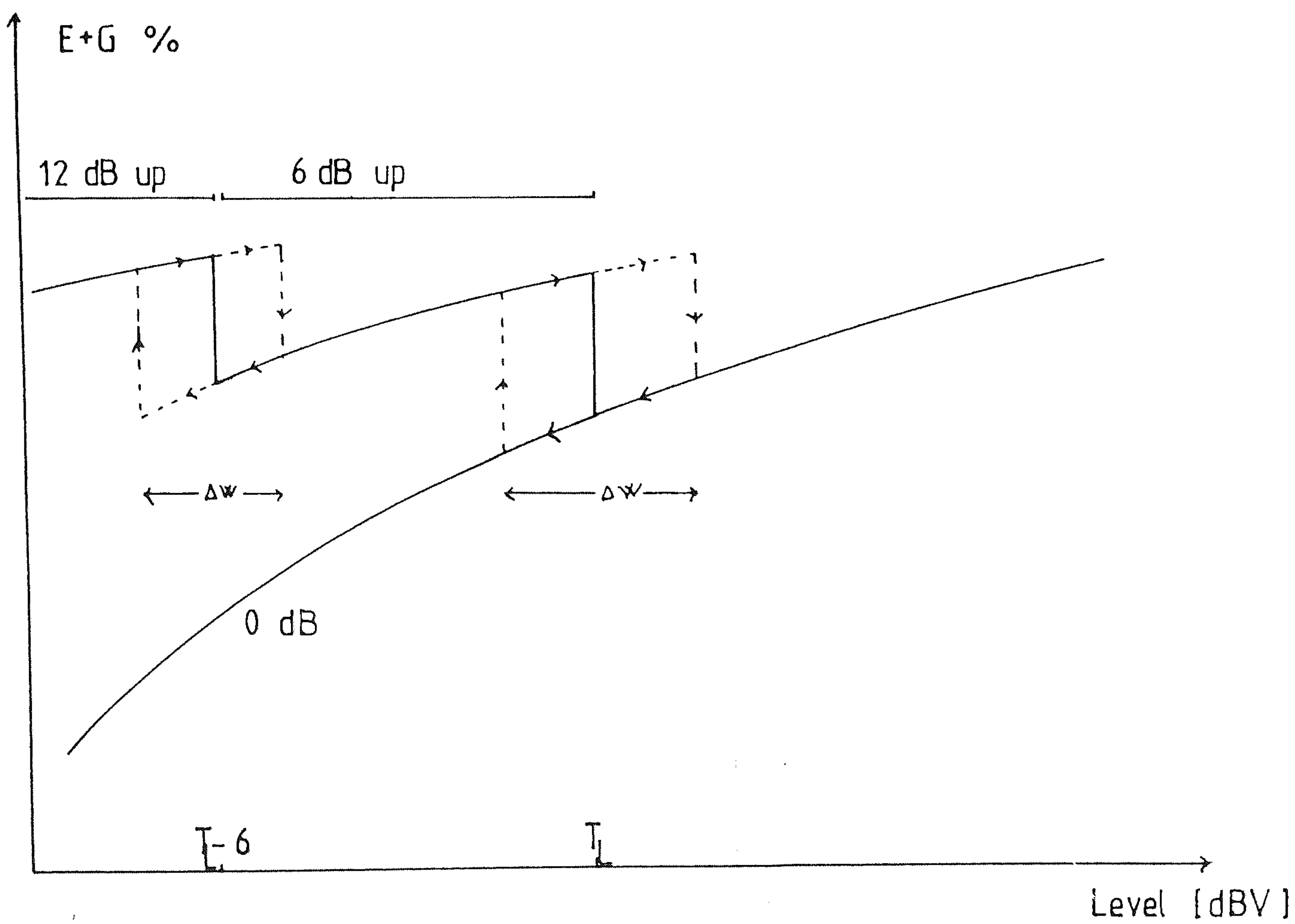


Fig. 7.1 Hysteresis d.a.g.c.

A d.a.g.c. which introduces just one gain (apart from 0 dB) can be tested subjectively. From the T.C.A.M. assessments there is little difference between the improvements using a 0,6 dB only or 0,12 dB only compared with the performance of the 0, 6, 12 dB device. Gains other than 6 dB or 12 dB could be selected; but attention should be paid to the additional quantizing distortion if the gain is not an integer power of 2 [35].

7.2 CONCLUSIONS

A d.a.g.c. which uses a time-shared processing system to control the speech level in all channels of a P.C.M. system is possible. A simple d.a.g.c. in closed-loop configuration is not advisable, because the device oscillates when a sine-wave is used as input. As pure tones are used as test signals for lining up circuits and as carriers for data transmission, any d.a.g.c. should operate satisfactorily for any channel which is transmitting sinusoidal signal. Of the solutions studied to avoid oscillation, the one adopted was the simpler to implement, that of open-loop configuration with 1 threshold level.

The additional quantizing distortion introduced by the d.a.g.c. is about 2 dB if its dealing with low levels and the gain introduced is power of 2. At about middle levels, there is no degradation because the device is processing

A-law P.C.M. signals. At high levels, the distortion is greater than 10 dB; but it comes mainly from the clipping of the signal.

The insertion of gain into a telephone analogue-digital network gives improvement in the transmission performance of a national telephone network which as the analogue part producing a speech level in the output of the decoder having both a low level and a poor signal-to-quantization noise ratio. However, when listeners noticed changes in gain during the reception of speech their opinion concerning preferred level were affected. Accordingly if the achievement of a comfortably loud listening level was accompanied by noticeable changes in gain, a quieter level was preferred if this avoided perceptable changes in loudness.

The preferred time constant was between 100 ms and 333 ms. Above this value, action of the device was rejected, possibly due to the fact that the perception of the adaptive-like effects was intolerable. A lower value of time constant will be impracticable because of the inaccuracy when measuring the speech level. Furthermore, because the telephone speech contains pauses, a parameter taken directly from the P.C.M. samples will change very slowly.

The degree of difficulty (D%) of a telephone connection with a d.a.g.c. is less than half of that without it. Being about half if the noise is high; and less if the noise is low. (E+G%) is 10-30% greater if the device is in. The lowest improvement if the level is near the quiet than preferred level; being the highest if the level is much quieter than preferred and low noise.

In the appendix A, it is shown that the instability is avoided in two ways, first, if the digital path is up to the telephone handset, and therefore the side-tone loudness rating is very high, or, second, provided that the balance-return loss is greater than 6 dB if the P.C.M. link is 3 dB loss, as at the present, and greater than 3 dB if the P.C.M. link is 6 dB loss, as it will be in the future [45], if the digital path is not up to the telephone handset.

APPENDIX A

INSTABILITY PROBLEM

APPENDIX A

INSTABILITY PROBLEM

When the listener is connected via a two-wire analogue local line to digital exchange the instability problem can be studied using Fig. A.1.

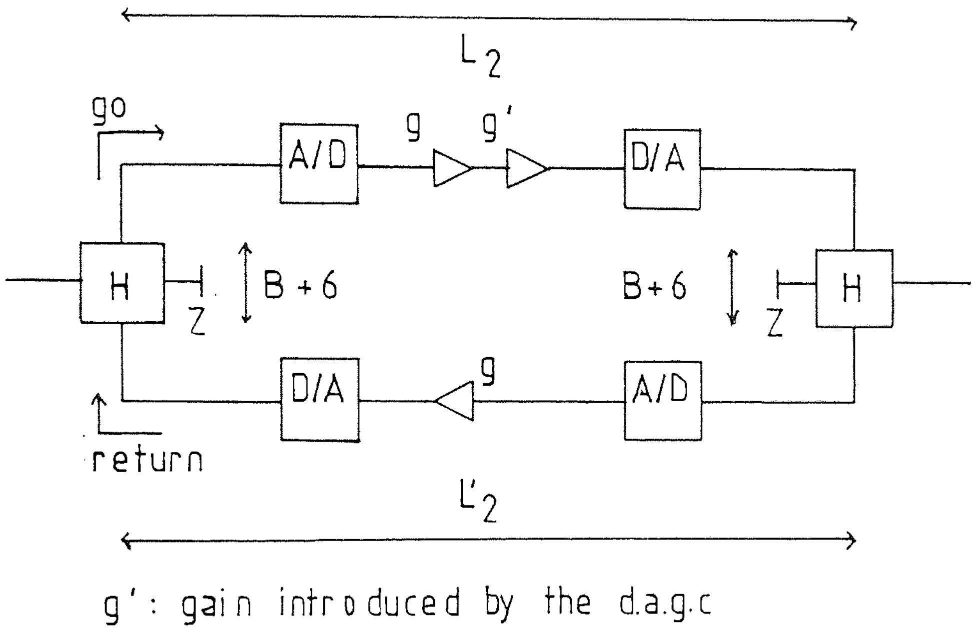


Fig. A.1 Four-wire circuit of d.a.g.c.

g, net gain of one side of the 4-wire circuit without d.a.g.c.
g' is gain introduced by the d.a.g.c. (6 dB or 12 dB).

L_2 = 6-g when the d.a.g.c. is out; if the d.a.g.c. is in then L_2 = 6-g-g', L'_2 = 6-g., and,

B is the balance-return loss.

There are two possibilities: the P.C.M. junction has a

3 dB loss or a 6 dB[45] loss.

A.1 3 dB LOSS P.C.M.

When the d.a.g.c. is working, the singing margin[48,49] SM, is given by

$$B+6-g-g'+B+6-g=SM$$

To avoid bad effects such as hollow sounds, singing, etc.[8,48,49],

$$SM>6$$

therefore,

$$2B+2(6-g)-g'>6 \tag{A.1}$$

$$6-g=L_2 = L_2' = 3$$

$$2B+6-g'>6$$

The worst case is when g' = 12 dB, then,

$$2B>6+12-6$$

To avoid instability, therefore B>6.

As the d.a.g.c. will be in operation after the subscriber has been connected, it is not expected that B will be 0. In general, B should be much higher than 6[51].

A.2 6 dB LOSS P.C.M.

(A.1) becomes:

$$2B+12-g'>6$$

then, instability is avoided if B>3.

When the listener is connected to a digital exchange via a digital local line, which is the equivalent to a four-wire transmission system then there is no problem of instability.

It is expected that the side-tone will be almost absent[4]; however, some side-tone should be added to show that the telephone is not dead.

APPENDIX B

PUBLICATIONS BY THE AUTHOR RELATED TO THIS WORK

APPENDIX C

SOFTWARE FOR THE D.A.G.C. IN OPEN-LOOP CONFIGURATION

APPENDIX G

CENTRAL PROGRAM

```
*TT:=DX1:DAGCOL.FOR

      PROGRAM DAGCOL
      DIMENSION IFECHA(5)
      COMMON/LOCB/IGB,ITCB(4)
      COMMON /LOCC/IGC,ITCC(3)
      COMMON/LOCD/IGD,ITCD(4)
C     PROGRAMME TO SIMULATE A DIGI-
C     TAL AUTOMATIC GAIN CONTROL
C     (DAGC) IN OPEN LOOP CONFIGURA-
C     TION. THE FORTRAN PART WILL PUT
C     TOGETHER THE ACTUAL DAGC'S.
C     THESE ARE IN MACRO.
      DATA IA/'A'/
      DATA IB/'B'/
      DATA IC/'C'/
      DATA ID/'D'/
      DATA IBLANK /' '/
      IFECHA(5)=IBLANK
      CALL DATE(IFECHA)
      WRITE (7,200) (IFECHA(I),I=1,5)
200   FORMAT ('?*** DATE: ',5A2,'***')
      CALL FROM
 1    WRITE (7,5)
 5    FORMAT ('  WHICH SYSTEM?. A=0 DB, B=0/6/12/F DB'/
     1'$ C=0/12 DB OR  D=0/6/12 DB:')
 6    READ (5,10) ISYST
 10   FORMAT (A1)
      IF(ISYST.EQ.IA) GO TO 20
      IF(ISYST.EQ.IB) GO TO 30
      IF(ISYST.EQ.IC) GO TO 40
      IF(ISYST.EQ.ID) GO TO 50
      WRITE (7,15)
 15   FORMAT (' WRONG ENTRY. TRY AGAIN, PLEASE')
      GO TO 6
 20   WRITE (7,25)
 25   FORMAT ('  0 DB ONLY')
      CALL A
      GO TO 1
 30   WRITE (7,35)
 35   FORMAT ('  0,6 & 12 DB, FIXED, ONLY'/
     1'$ TYPE IN TIME CONSTANT IN MS ')
      READ (5,36) ITIMEC
 36   FORMAT (I6)
      ITCB(1)=1.94*ITIMEC
      ITCB(2)=0.48*ITIMEC
      ITCB(3)=8.00*ITIMEC
      ITCB(4)=0.03*ITIMEC
      CALL B
      WRITE (7,90) IGB
      GO TO 1
 90   FORMAT (O6)
 40   WRITE (7,45)
```

continued...

```
 45    FORMAT ('   0 & 12 DB ONLY'/
     1'$ TYPE IN TIME CONSTANT IN MS   ')
      READ (5,36) ITIMEC
      ITCC(1)=1.72*ITIMEC
      ITCC(2)=8.00*ITIMEC
      ITCC(3)=0.054*ITIMEC
C
C
C
      CALL C
      WRITE (7,90) IGC
      GO TO 1
 50   WRITE (7,55)
 55   FORMAT ('   0,6 & 12 DB ONLY'/
     1'$ TYPE IN TIME CONSTANT IN MS ')
      READ (5,36) ITIMEC
      ITCD(1)=1.94*ITIMEC
      ITCD(2)=0.48*ITIMEC
      ITCD(3)=8.00*ITIMEC
      ITCD(4)=0.03*ITIMEC
      CALL D
      WRITE (7,90) IGD
      GO TO 1
      END
*TT:=DX1:B.MAC
```

6 dB STEP LOOK-UP TABLE

```
.TITLE PROM
.GLOBL  PROM
.MCALL ..V2..,.REGDEF
..V2..
.REGDEF
PROM:MOV        #0,R5
        ;****************************************
        ;*I.PART. LOADING THE COMPUTER MEMORY TO *
        ;*HAVE IT WORKING AS A PROM. THE  PDP-11 *
        ;*    WILL BE A 1024-WORD LOOK-UP TABLE    *
        ;****************************************
START:  MOV     #000400,R0      ;SETTING COUNTER
        MOV     #000000,R1      ;FIRST VALUE TO BE STORED
        MOV     #120000,R2      ;FIRST ADDRESS FOR 0 DB
LNG:    MOV     R1,(R2)+        ;FORMING  A 256-FIGUERE TABLE
        INC     R1
        SOB     R0,LNG
        MOV     #000140,R0      ;-6 DB TABLE. FIRST 96 FIGUERES
        MOV     #000020,R1      ;FIRST ADDRESS TO BE STORED
        MOV     #121000,R2
LA:     MOV     R1,(R2)+        ;FROM 121000 TO 121276
        INC     R1
        SOB     R0,LA
        MOV     #000020,R0
        MOV     #000160,R1
        MOV     #121300,R2
LA1:    MOV     R1,(R2)+        ;FROM 121300 TO 121376
        MOV     R1,(R2)+
        INC     R1
        SOB     R0,LA1
        MOV     #000140,R0
        MOV     #000220,R1
        MOV     #121400,R2
LA2:    MOV     R1,(R2)+        ;FROM 121400 TO 121676
        INC     R1
        SOB     R0,LA2
        MOV     #000020,R0
        MOV     #000360,R1
        MOV     #121700,R2
LA3:    MOV     R1,(R2)+
        MOV     R1,(R2)+        ;FROM 121700 TO 121776
        INC     R1
        SOB     R0,LA3
        MOV     #000020,R0
        MOV     #122000,R2
LG:     MOV     #000000,(R2)+   ;FROM 122000 TO 122036
        SOB     R0,LG
        MOV     #000000,R1
        MOV     #000140,R0
        MOV     #122040,R2
LG1:    MOV     R1,(R2)+        ;FROM 122040 TO 122336
        INC     R1
        SOB     R0,LG1
        MOV     #000020,R0
        MOV     #000140,R1
        MOV     #122340,R2
```

continued...

```
LG2:    MOV     R1,(R2)+         ;FROM 122340 TO 122376
        ADD     #000002,R1
        SOB     R0,LG2
        MOV     #000020,R0
        MOV     #122400,R2
LG3:    MOV     #000200,(R2)+    ;FROM 122400 TO 122436
        SOB     R0,LG3
        MOV     #000140,R0
        MOV     #000200,R1
        MOV     #122440,R2
LG4:    MOV     R1,(R2)+         ;FROM 122400 TO 122736
        INC     R1
        SOB     R0,LG4
        MOV     #000020,R0
        MOV     #000340,R1
        MOV     #122740,R2
LG5:    MOV     R1,(R2)+         ;FROM 122740 TO 122776
        ADD     #000002,R1
        SOB     R0,LG5
        MOV     #000040,R0
        MOV     #123000,R2
LMG:    MOV     #000000,(R2)+    ;FROM 123000 TO 123076
        SOB     R0,LMG
        MOV     #000120,R0
        MOV     #000000,R1
        MOV     #123100,R2
LMG1:   MOV     R1,(R2)+         ;FROM 123100 TO 123336
        INC     R1
        SOB     R0,LMG1
        MOV     #000010,R0
        MOV     #000120,R1
        MOV     #123340,R2
LMG2:   MOV     R1,(R2)+         ;FROM 123340 TO 123356
        ADD     #000002,R1
        SOB     R0,LMG2
        MOV     #000010,R0
        MOV     #123360,R2
        MOV     #000140,R1
LMG3:   MOV     R1,(R2)+         ;FROM   123360 TO 1223376
        ADD     #000004,R1
        SOB     R0,LMG3
        MOV     #123400,R2
        MOV     #000040,R0
LMG4:   MOV     #000200,(R2)+    ;FROM 123400 TO 123476
        SOB     R0,LMG4
        MOV     #123500,R2
        MOV     #000200,R1
        MOV     #000120,R0
LMG5:   MOV     R1,(R2)+         ;FROM 123500 TP 123736
        INC     R1
        SOB     R0,LMG5
        MOV     #000010,R0
        MOV     #000320,R1
        MOV     #123740,R2
LMG6:   MOV     R1,(R2)+         ;FROM 123740 TO 123756
        ADD     #000002,R1
```

continued...

```
        SOB     R0,LMG6
        MOV     #000340,R1
        MOV     #000010,R0
        MOV     #123760,R2
LMG7:   MOV     R1,(R2)+            ;FROM 123760 TO 123776
        ADD     #000004,R1
        SOB     R0,LMG7
        RTS     PC
        .END    START
```

3 dB STEP LOOK-UP TABLE

```
START:  MOV     #000400,R0          ;SETTING COUNTER
        MOV     #000000,R1          ;FIRST VALUE TO BE STORED
        MOV     #120000,R2          ;FIRST ADDRESS FOR 0 DB
LNG:    MOV     R1,(R2)+            ;FORMING  A 256-FIGUERE TABLE
        INC     R1
        SOB     R0,LNG
        MOV     #000140,R0          ;-6 DB TABLE. FIRST 96 FIGUERES
        MOV     #000020,R1          ;FIRST ADDRESS TO BE STORED
        MOV     #121000,R2
LA:     MOV     R1,(R2)+            ;FROM 121000 TO 121276
        INC     R1
        SOB     R0,LA
        MOV     #000020,R0
        MOV     #000160,R1
        MOV     #121300,R2
LA1:    MOV     R1,(R2)+            ;FROM 121300 TO 121376
        MOV     R1,(R2)+
        INC     R1
        SOB     R0,LA1
        MOV     #000140,R0
        MOV     #000220,R1
        MOV     #121400,R2
LA2:    MOV     R1,(R2)+
        INC     R1
        SOB     R0,LA2
        MOV     #000020,R0
        MOV     #000360,R1
        MOV     #121700,R2
LA3:    MOV     R1,(R2)+
        MOV     R1,(R2)+            ;FROM 121700 TO 121776
        INC     R1
        SOB     R0,LA3
        MOV     #000020,R0
        MOV     #122000,R2
LG:     MOV     #000000,(R2)+       ;FROM 122000 TO 122036
        SOB     R0,LG
        MOV     #000000,R1
        MOV     #000140,R0
        MOV     #122040,R2
LG1:    MOV     R1,(R2)+            ;FROM 122040 TO 122336
        INC     R1
        SOB     R0,LG1
        MOV     #000020,R0
        MOV     #000140,R1
        MOV     #122340,R2
LG2:    MOV     R1,(R2)+            ;FROM 122340 TO 122376
        ADD     #000002,R1
        SOB     R0,LG2
        MOV     #000020,R0
        MOV     #122400,R2
LG3:    MOV     #000200,(R2)+       ;FROM 122400 TO 122436
        SOB     R0,LG3
        MOV     #000140,R0
        MOV     #000200,R1
        MOV     #122440,R2
```

continued...

```
LG4:    MOV     R1,(R2)+            ;FROM 122400 TO 122736
        INC     R1
        SOB     R0,LG4
        MOV     #000020,R0
        MOV     #000340,R1
        MOV     #122740,R2
LG5:    MOV     R1,(R2)+            ;FROM 122740 TO 122776
        ADD     #000002,R1
        SOB     R0,LG5
        MOV     #000040,R0
        MOV     #123000,R2
LMG:    MOV     #000000,(R2)+       ;FROM 123000 TO 123076
        SOB     R0,LMG
        MOV     #000120,R0
        MOV     #000000,R1
        MOV     #123100,R2
LMG1:   MOV     R1,(R2)+            ;FROM 123100 TO 123336
        INC     R1
        SOB     R0,LMG1
        MOV     #000010,R0
        MOV     #000120,R1
        MOV     #123340,R2
LMG2:   MOV     R1,(R2)+            ;FROM 123340 TO 123356
        ADD     #000002,R1
        SOB     R0,LMG2
        MOV     #000010,R0
        MOV     #123360,R2
        MOV     #000140,R1
LMG3:   MOV     R1,(R2)+            ;FROM   123360 TO 1223376
        ADD     #000004,R1
        SOB     R0,LMG3
        MOV     #123400,R2
        MOV     #000040,R0
LMG4:   MOV     #000200,(R2)+       ;FROM 123400 TO 123476
        SOB     R0,LMG4
        MOV     #123500,R2
        MOV     #000200,R1
        MOV     #000120,R0
LMG5:   MOV     R1,(R2)+            ;FROM 123500 TP 123736
        INC     R1
        SOB     R0,LMG5
        MOV     #000010,R0
        MOV     #000320,R1
        MOV     #123740,R2
LMG6:   MOV     R1,(R2)+            ;FROM 123740 TO 123756
        ADD     #000002,R1
        SOB     R0,LMG6
        MOV     #000340,R1
        MOV     #000010,R0
        MOV     #123760,R2
LMG7:   MOV     R1,(R2)+            ;FROM 123760 TO 123776
        ADD     #000004,R1
        SOB     R0,LMG7
        .ASECT
        .=124000
```

continued...

```
.WORD 11,12,12,13,14,14,15,16
.WORD 17,17,20,22,23,25,26,27
.WORD 31,32,32,33,34,34,35,36
.WORD 37,37,40,42,44,45,46,47
.WORD 51,51,52,53,54,54,55,56
.WORD 57,57,60,62,63,65,67,67
.WORD 71,72,72,73,74,74,75,76
.WORD 77,77,100,101,103,105,106,107
.WORD 111,111,112,113,114,114,115,116
.WORD 117,117,120,122,123,125,126,127
.WORD 131,132,132,133,133,135,135,136
.WORD 137,137,140,142,143,145,146,150
.WORD 151,152,153,153,154,155,155,156
.WORD 157,157,160,161,161,162,163,164
.WORD 164,165,166,166,167,170,170,171
.WORD 172,173,173,174,175,175,176,177
.WORD 211,212,212,213,214,214,215,216
.WORD 217,217,220,222,223,225,226,227
.WORD 231,232,232,233,234,234,235,236
.WORD 237,237,240,242,244,245,246,247
.WORD 251,251,252,253,254,254,255,256
.WORD 257,257,260,262,263,265,267,267
.WORD 271,272,272,273,274,274,275,276
.WORD 277,277,300,301,303,305,306,307
.WORD 311,311,312,313,314,314,315,316
.WORD 317,317,320,322,323,325,326,327
.WORD 331,332,332,333,333,335,335,336
.WORD 337,337,340,342,343,345,346,350
.WORD 351,352,353,353,354,355,355,356
.WORD 357,357,360,361,361,362,363,364
.WORD 364,365,366,366,367,370,370,371
.WORD 372,373,373,374,375,375,376,377
.WORD , , , , , , ,
.WORD 0,0,1,2,3,5,6,7
.WORD 11,12,12,13,14,14,15,16
.WORD 17,17,20,21,23,25,26,27
.WORD 31,31,32,33,34,34,35,36
.WORD 37,37,41,42,43,46,46,47
.WORD 51,52,52,53,54,54,55,56
.WORD 57,57,60,62,63,65,66,67
.WORD 71,72,72,73,74,74,75,76
.WORD 77,77,100,102,103,104,106,107
.WORD 111,112,112,113,113,114,115,116
.WORD 117,117,120,122,124,125,126,127
.WORD 131,132,132,133,134,135,135,136
.WORD 137,137,140,142,143,145,146,150
.WORD 151,152,154,155,157,160,161,163
.WORD 164,166,167,170,172,173,176,177
.WORD 200,200,200,200,200,200,200,200
.WORD 200,200,201,202,203,205,206,207
.WORD 211,212,212,213,214,214,215,216
.WORD 217,217,220,221,223,225,226,227
.WORD 231,231,232,233,234,234,235,236
.WORD 237,237,241,242,243,246,246,247
.WORD 251,252,252,253,254,254,255,256
.WORD 257,257,260,262,263,265,266,267
.WORD 271,272,272,273,274,274,275,276
.WORD 277,277,300,302,303,304,306,307
```

continued...

```
.WORD 311,312,312,313,313,314,315,316
.WORD 317,317,320,322,324,325,326,327
.WORD 331,332,323,333,334,335,335,336
.WORD 337,337,340,342,343,345,346,350
.WORD 351,352,354,355,357,360,361,363
.WORD 364,366,367,370,372,373,376,377
.WORD , , , , , ,,
.WORD , , , , , , ,
.WORD , , , , , , ,
.WORD 0,0,1,2,3,5,6,7
.WORD 11,12,12,13,14,14,15,16
.WORD 17,17,20,22,23,25,26,27
.WORD 31,32,32,33,34,34,35,36
.WORD 37,37,40,42,43,45,46,47
.WORD 51,51,52,53,54,54,55,56
.WORD 57,57,60,62,63,65,66,67
.WORD 71,71,72,73,74,74,76,77
.WORD 77,77,100,102,103,105,105,110
.WORD 111,112,112,113,114,114,115,116
.WORD 117,120,121,122,123,125,126,130
.WORD 131,132,134,135,137,140,143,146
.WORD 151,154,157,162,164,167,172,175
.WORD 200,200,200,200,200,200,200,200
.WORD 200,200,200,200,200,200,200,200
.WORD 200,200,200,200,200,200,200,200
.WORD 200,200,201,201,203,205,206,207
.WORD 211,212,212,213,214,214,215,216
.WORD 217,217,220,222,223,225,226,227
.WORD 231,232,232,233,234,234,235,236
.WORD 237,237,240,242,243,245,246,247
.WORD 251,251,252,253,254,254,255,256
.WORD 257,257,260,262,263,265,266,267
.WORD 271,271,272,273,274,274,275,274
.WORD 277,277,300,302,303,305,305,310
.WORD 311,312,312,313,314,314,315,316
.WORD 317,320,321,322,323,325,326,330
.WORD 331,332,334,335,337,340,343,346
.WORD 351,354,357,362,364,367,372,375
.WORD 0,0,0,0,0,0,0,0
.WORD 0,0,0,0,0,0,0,0
.WORD 0,0,0,0,0,0,0,0
.WORD 0,0,0,0,0,0,0,0
```

O dB GAIN

```
.TITLE A
.GLOBL A
.MCALL ..V2..,.REGDEF
..V2..
.REGDEF
A:DRCSR=167770
DROUT=167772
DRIN=167774
START:  MOV     #000776,R0
        MOV     #120000,R3          ; NO GAIN
YS:     ADD     R3,R0               ; SPEECH, NO DECISION
        MOV     @R0,DROUT
LOOP:   BIT     #100000,DRCSR       ;SPEECH?
        BEQ     LOOP
        MOV     DRIN,R0             ; SPEECH ARRIVED
        BIT     #000200,DRCSR
        BEQ     YS
        RTS     PC
        .END    START

*^C

.R SWAP
```

O, 6 dB GAIN D.A.G.C.

```
.TITLE B
.GLOBL  B
.MCALL ..V2..,.REGDEF
..V2..
.REGDEF
        .CSECT  LOCB
GB:     .BLKW              ; HOLDS GAIN FOR SIMUL
PB1:    .BLKW              ; COEFF 1
PB2:    .BLKW              ; COEFF 2
PB3:    .BLKW              ; COEFF 3
PB4:    .BLKW              ; COEFF 4
B:      DRCSR=167770
DROUT=167772
DRIN=167774
        ;*************************************
        ;*II.PART. SIMULATION  OF  DIGITAL AUTOMA*
        ;*TIC GAIN CONTROL FOR PCM TELEPHONY   IN*
        ;*      OPEN-LOOP           CONFIGURATION *
        ;*************************************
START:  MOV     #000776,R0
        MOV     PB1,R4          ; 0-DB LOWER LIMIT
NG:     MOV     #120000,R3      ; NO GAIN
NS:     CLR     R1
        CLR     R2
YS:     ADD     R3,R0           ; SPEECH, NO DECISION
        MOV     @R0,DROUT
LOOP:   BIT     #100000,DRCSR   ; SPEECH?
        BEQ     LOOP
        MOV     DRIN,R0         ; SPEECH ARRIVED
        BIT     #000200,DRCSR   ; PRINT-OUT?
        BEQ     CON             ; NO, THEN CARRY ON
        MOV     R3,GB           ; GAIN OUT
        RTS     PC
CON:    BIT     #000400,R0      ; NEGATIVE SAMPLE?
        BEQ     WW              ; WRONG WORD
        CMP     #000756,R0      ; L.T.L.= AT -48 DB
        BLOS    YS
        CMP     PB3,R1          ; TIME CONSTANT
        BEQ     X
        INC     R1
        CMP     #000644,R0      ; THRESHOLD LEVEL
        BLOS    YS
        INC     R2              ; N.D.S.
        BR      YS
X:      CMP     R4,R2
        BLOS    NG              ; 0 DB
        MOV     #122000,R3      ; 6 DB
        BR      NS
WW:     CMP     #000356,R0
        BLOS    YS
        CMP     PB3,R1          ; TIME CONSTANT
        BEQ     X
        INC     R1
        CMP     #000244,R0      ; THRESHOLD LEVEL
        BLOS    YS
        INC     R2
        BR      YS
        .END    START
```

<u>O, 12 dB GAIN D.A.G.C.</u>

```
.TITLE C
.GLOBL  C
.MCALL ..V2..,.REGDEF
..V2..
.REGDEF
        .CSECT  LOCC
GC:     .BLKW              ; HOLDS GAIN FOR SIMUL
PC1:    .BLKW              ; COEFF 1
PC2:    .BLKW              ; COEFF 2
PC3:    .BLKW              ; COEFF 3
C:      DRCSR=167770
DROUT=167772
DRIN=167774
        ;*************************************
        ;*II.PART. SIMULATION  OF  DIGITAL AUTOMA*
        ;*TIC GAIN CONTROL FOR PCM TELEPHONY   IN*
        ;*      OPEN-LOOP          CONFIGURATION *
        ;*************************************
START:  MOV     #000776,R0
        MOV     PC1,R4         ; 0-DB LOWER LIMIT
        MOV     #120000,R3     ; NO GAIN
NS:     CLR     R1
        CLR     R2
YS:     ADD     R3,R0          ; SPEECH, NO DECISION
        MOV     @R0,DROUT
LOOP:   BIT     #100000,DRCSR  ; SPEECH?
        BEQ     LOOP
        MOV     DRIN,R0        ; SPEECH ARRIVED
        BIT     #000400,R0     ; NEGATIVE SAMPLE?
        BEQ     WW             ; WRONG WORD
        CMP     #000764,R0     ; L.T.L.= AT 20
        BLOS    YS
        CMP     PC2,R1         ; TIME CONSTANT
        BEQ     D
        INC     R1
        CMP     #000562,R0     ; THRESHOLD LEVEL
        BLOS    YS
        INC     R2             ; N.D.S.
        BR      YS
D:      CMP     R4,R2
        BLOS    ZGOSL          ; 0 DB
        CMP     PC3,R2
        BLOS    G12
        BR      NS
G12:    MOV     #123000,R3     ; 12 DB
ZGOSL:  ADD     R3,R0          ;
        MOV     @R0,DROUT
SL:     BIT     #100000,DRCSR  ; SAMPLE?
        BEQ     SL             ; NO,  THEN GO BACK
        MOV     DRIN,R0
        BIT     #000200,DRCSR  ; PRINT OUT?
        BEQ     ZGOSL
        MOV     R3,GC          ; GAIN OUT
        RTS     PC
WW:     CMP     #000364,R0
        BLOS    YS
        CMP     PC2,R1         ; TIME CONSTANT
        BEQ     D
        INC     R1
        CMP     #000262,R0     ; THRESHOLD LEVEL
        BLOS    YS
        INC     R2
```

<u>O, 6, 12 dB GAIN D.A.G.C.</u>

```
.TITLE D
.GLOBL D
.MCALL  ..V2...,REGDEF
..V2..
.REGDEF
        .CSECT  LOCD
GD:     .BLKW            ; HOLDS GAIN FOR SIMUL
PD1:    .BLKW            ; COEFF 1
PD2:    .BLKW            ; COEFF 2
PD3:    .BLKW            ; COEFF 3
PD4:    .BLKW            ; COEFF 4
D:      DRCSR=167770
DROUT=167772
DRIN=167774
        ;*************************************
        ;*II.PART. SIMULATION  OF  DIGITAL AUTOMA*
        ;*TIC GAIN CONTROL FOR PCM TELEPHONY   IN*
        ;*       OPEN-LOOP           CONFIGURATION *
        ;*************************************
START:  MOV     #000776,R0
        MOV     PD1,R4          ; 0-DB LOWER LIMIT
NG:     MOV     #120000,R3      ; NO GAIN
NS:     CLR     R1
        CLR     R2
YS:     ADD     R3,R0           ; SPEECH, NO DECISION
        MOV     @R0,DROUT
LOOP:   BIT     #100000,DRCSR   ; SPEECH?
        BEQ     LOOP
        MOV     DRIN,R0         ; SPEECH ARRIVED
        BIT     #000400,R0      ; NEGATIVE SAMPLE?
        BEQ     WW              ; WRONG WORD
        BIT     #000200,DRCSR   ; PRINT OUT?
        BEQ     CON             ; NO, THEN CARRY ON
        MOV     R3,GD           ; GAIN OUT
        RTS     PC
CON:    CMP     #000764,R0      ; L.T.L.= AT 20
        BLOS    YS
        CMP     PD3,R1          ; TIME CONSTANT
        BEQ     X
        INC     R1
        CMP     #000576,R0      ; THRESHOLD LEVEL
        BLOS    YS
        INC     R2              ; N.D.S.
        BR      YS
X:      CMP     R4,R2
        BLOS    NG              ; 0 DB
        CMP     PD2,R2          ; SECOND PARAMETER
        BLOS    G6              ; 6 DB
        CMP     PD4,R2
        BLOS    G12
        BR      NS
G12:    MOV     #123000,R3      ; 12 DB
        BR      NS
G6:     MOV     #122000,R3
        BR      NS
A6:     MOV     #121000,R3
        BR      NS
WW:     CMP     #000364,R0
        BLOS    YS
        CMP     PD3,R1.         ; TIME CONSTANT
        BEQ     X
        INC     R1
```

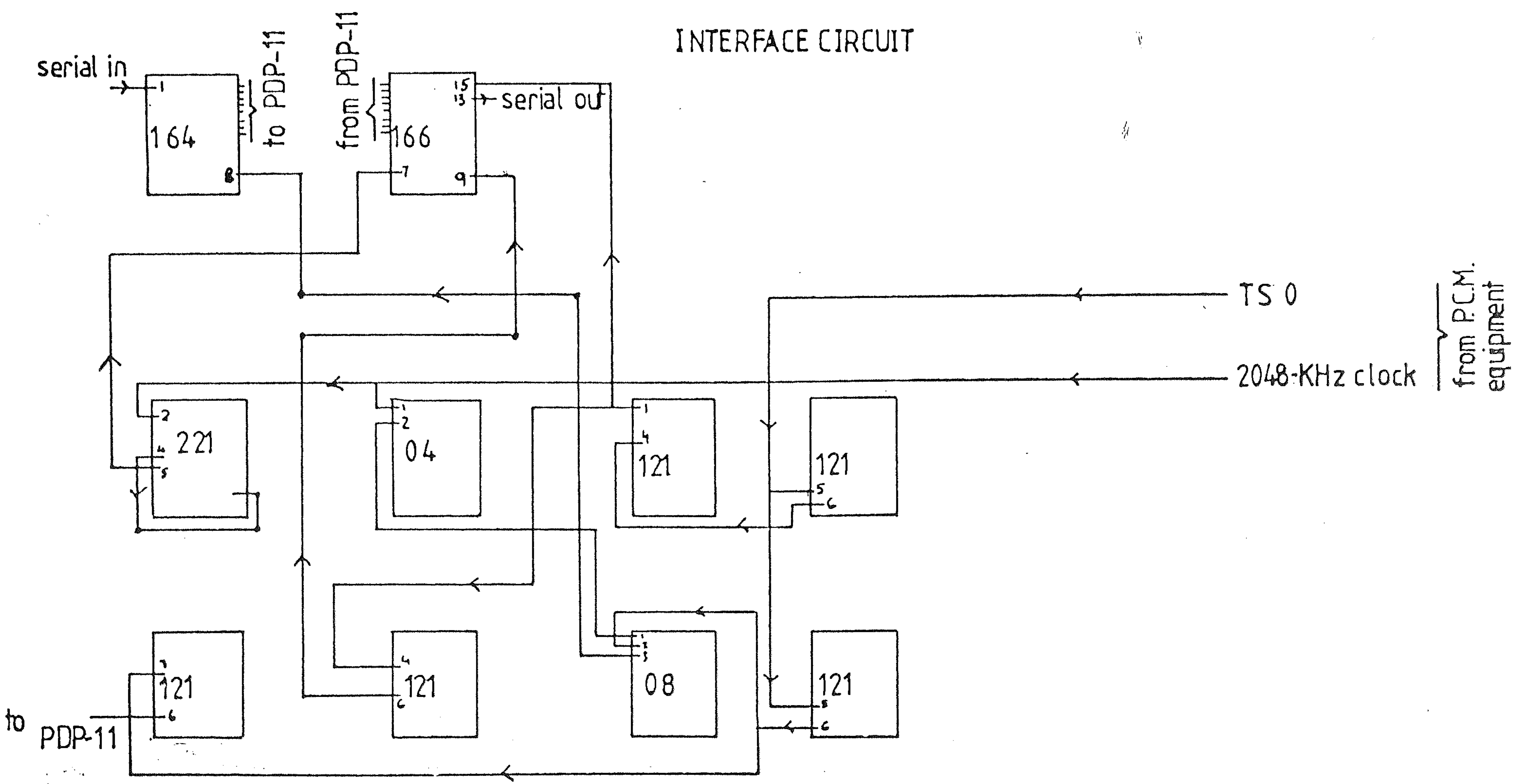
continued...

```
        CMP     #000176,R0      ; THRESHOLD LEVEL
        BLOS    YS
        INC     R2
        BR      YS
        .END    START
```

APPENDIX D

INTERFACE CIRCUIT

INTERFACE CIRCUIT



APPENDIX D

NOTES TO THE INTERFACE CIRCUIT:[52]

04	:	Inverter I.C.
08	:	AND Gate I.C.
121	:	Monostable multivibrator.
211	:	Dual monostable multivibrator.
164	:	8-bit parallel output serial shift register.
166	:	8-bit shift register.

APPENDIX E

LIST OF SYMBOLS

APPENDIX E

LIST OF SYMBOLS

A	parameter of the A-law, its inverse, 1/A separates the line or companding from the logarithmic.
x/X	input value-to-limiting value ratio.
y/Y	output value-to-limiting value ratio.
P(v)	probability cumulative exponential distribution.
x_r	instantaneous speech voltage/R.M.S. level.
T	time constant.
Tr	real time constant.
TL	threshold level.
TL1	threshold level 1.
TL2	threshold level 2.
TL3	threshold level 3.
Wi	weight i (i=1,2 or 3).
Mo	mid-value.
Nv	new value.
Nv(nT)	new value during nT
L1	limit 1.
L2	limit 2.
Sn	number of samples in time constant T.
U	digital representation of an analogue sample V.
V	amplitude of an analogue sample.
K	gain.
Xd	digital number representing KV.
z	number of bits of the device used in closed-loop configuration.
p(nT)	percentage of time the signal is above a threshold level during period nT.

Kp(nT)	setting value which depends on p(nT).
Ko(nT)	real setting value during period nT.
$\tilde{y}$	R.M.S. output level.
Δy	width between the limits of the zone of the d.a.g.c.
p2,p3	percentages which define a reference talker.
p	probability of reaching TL.
Kp(nT)g	Kp(nT) correspondent to gain.
Kp(nT)a	Kp(nT) correspondent to loss.
Vl	sinewave amplitude.
θ(nT)	angle at which a sinewave starts to be above a threshold level, during period nT.
t	period for determining the presence of a sinewave.
N, N_1, N_2, N_3	number of samples above T.
S_1, S_2, S_3	control bits.
nq	quantization error power.
E( )	mean or expected value.
v	input to the d.a.g.c.
u	output of the d.a.g.c.
X_c	compressed signal.
V_s	quantizing step.
M	total number of segments.
m	number of digits to represent segment number.
n	number of digits to represent quantizing step.
Q	signal-to-noise ratio.
L_s	segment number.
N_s	total number of quantizing steps.
p(v)	probability density exponential distribution.

S	input signal power.
N_{off}	speech-off noise power.
Ne	subjective effect of Noff and na noises.
D%	percentage of people likely to experience some difficulty in conversing.
E+G%	percentage of people likely to give opinion Good or Excellent.
V_L	level at listener ears.
SLR	sending loudness rating.
RLR	receiving loudness rating.
OLR	overall loudness rating.
Δw	hysteresis loop width.
B	balance-return loss.
12,1'2	attenuation between 2 wire lines.
SM	singing margins.
g	net gain of a 4-wire connection, without d.a.g.c
g'	gain introduced by the d.a.g.c.

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