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(54) Analogue transmission of video signals

(57) A signal for transmission is compared 12 with a prediction signal to produce a prediction error signal which is amplitude limited 8 and compressed 14 before transmission in analogue form. The limiter output is added 26 to the production signal and the resultant applied to a prediction function circuit 24 to generate the prediction signal. At the receiver, the analogue signal is expanded 20 and applied to a similar adder 26 and prediction function circuit 24. The system is useful with MAC type television signals and enables a reduction of noise effects on fm transmission.

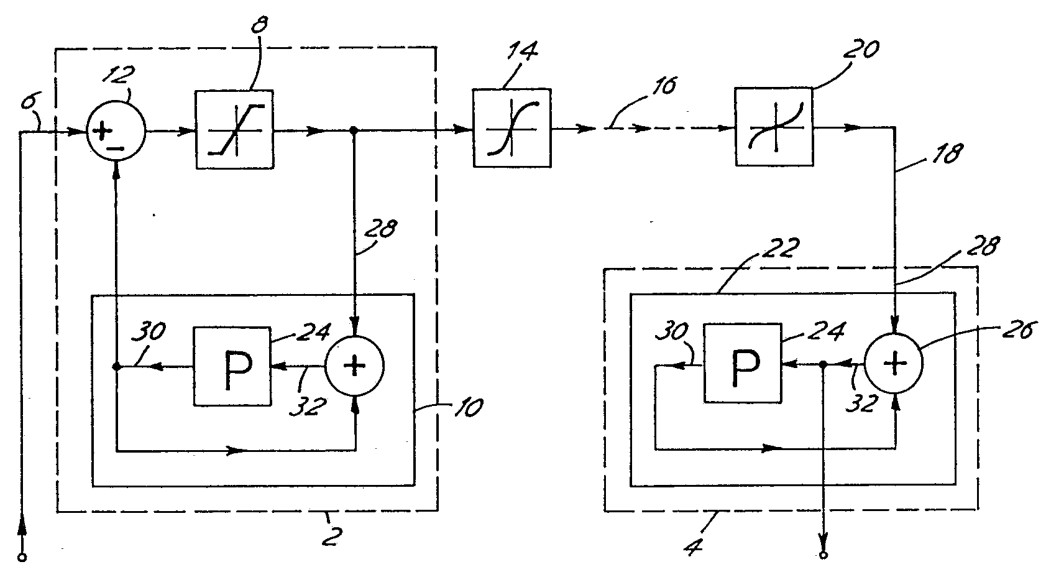

FIG. 1

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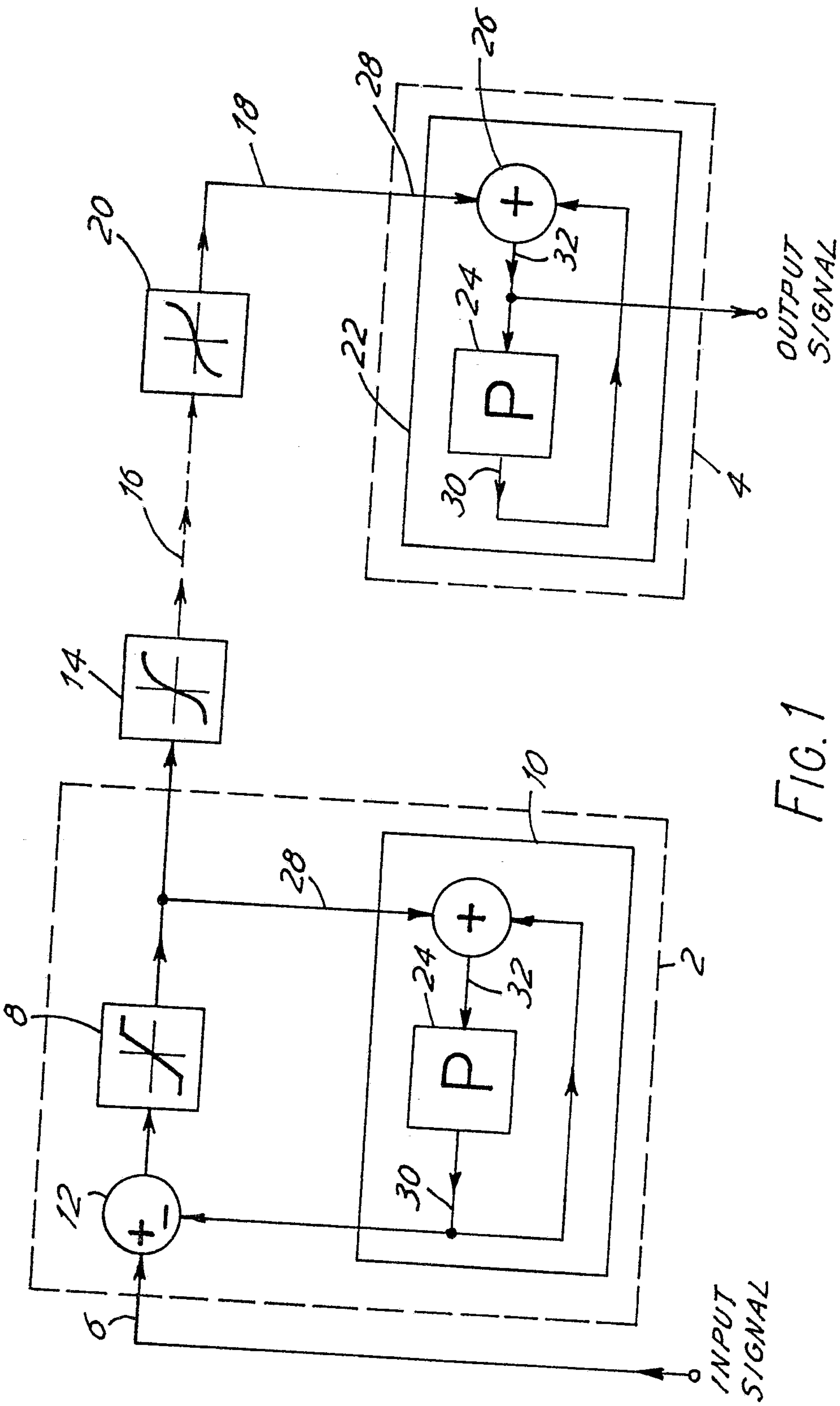

FIG. 1

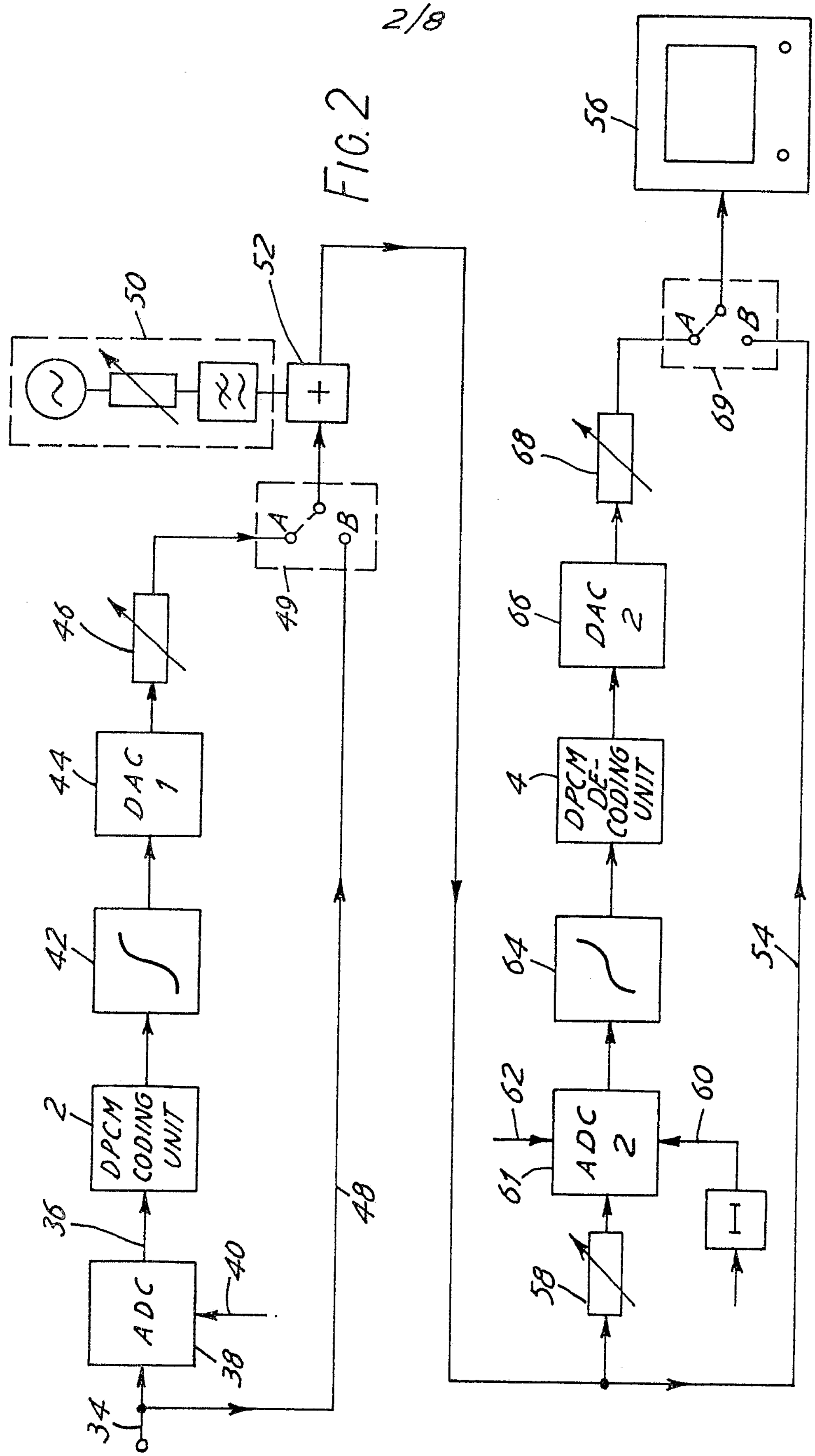
ADC
DPCM CODING UNIT
DAC 1
DAC 2
DPCM DE-CODING UNIT
ADC 2
A
B
34
36
38
40
2
42
44
46
48
49
50
52
54
56
58
60
61
62
64
4
66
68
69

FIG. 2

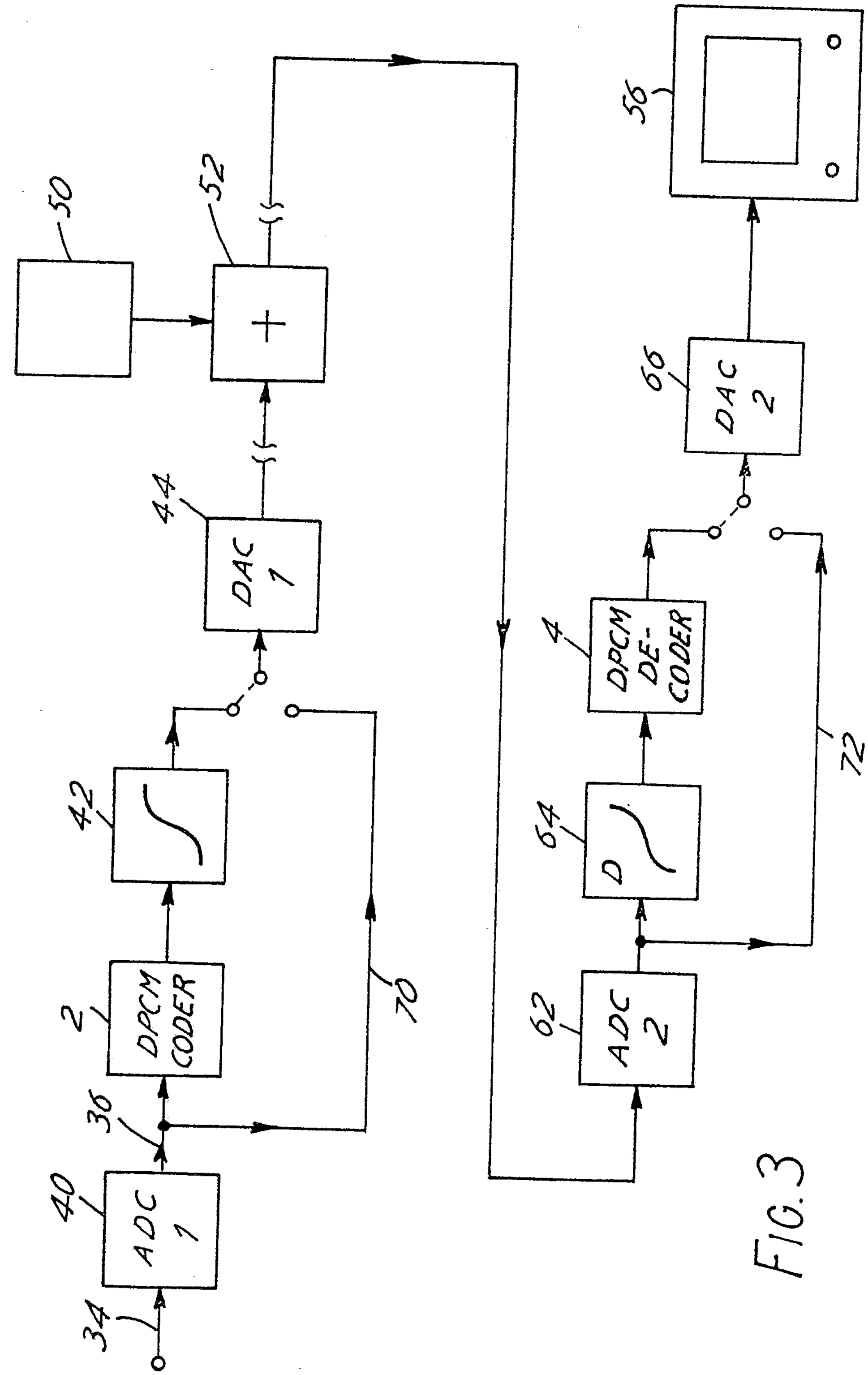
34
40
ADC 1
36
2
DPCM CODER
42
44
DAC 1
50
52
+
70
62
ADC 2
64
D
4
DPCM DE-CODER
66
DAC 2
56
72

FIG. 3

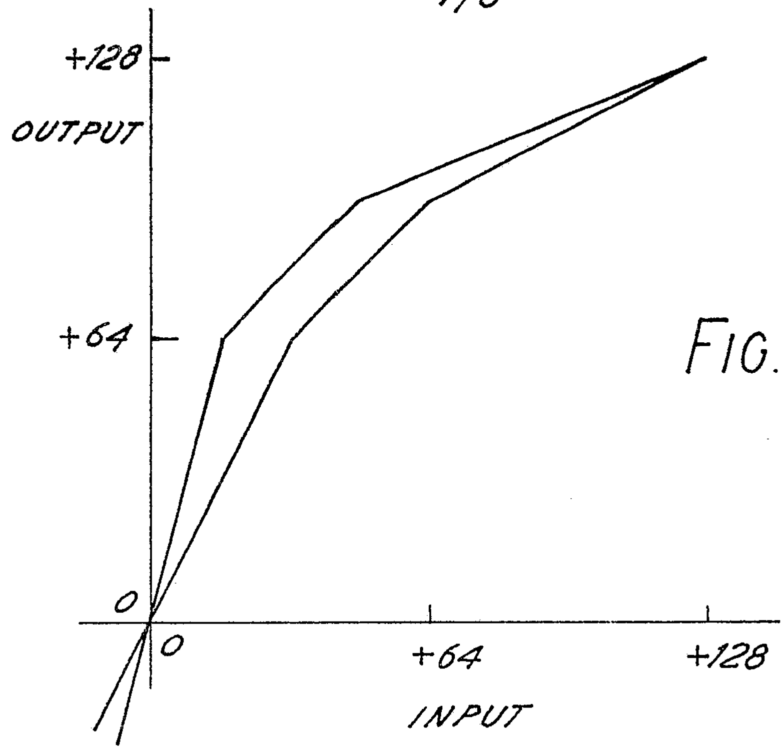

FIG.4

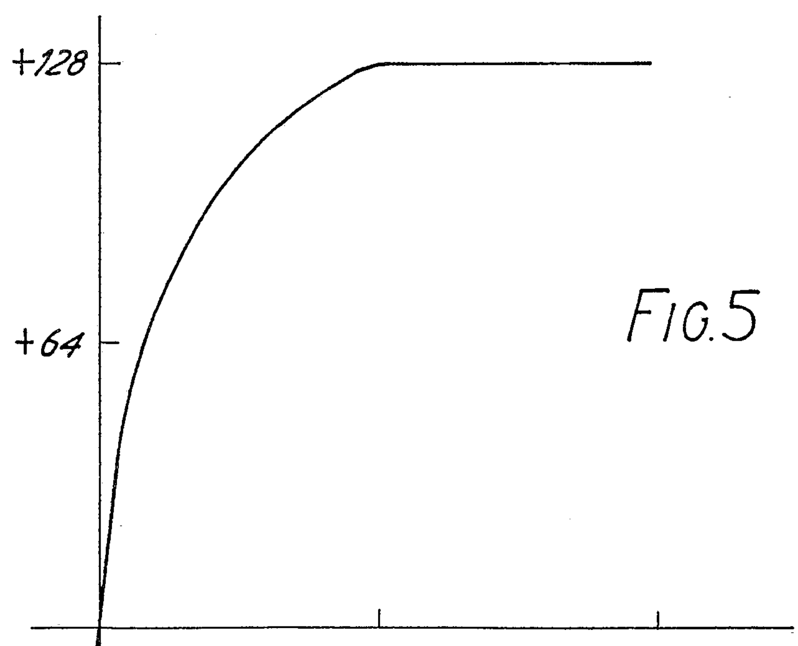

FIG.5

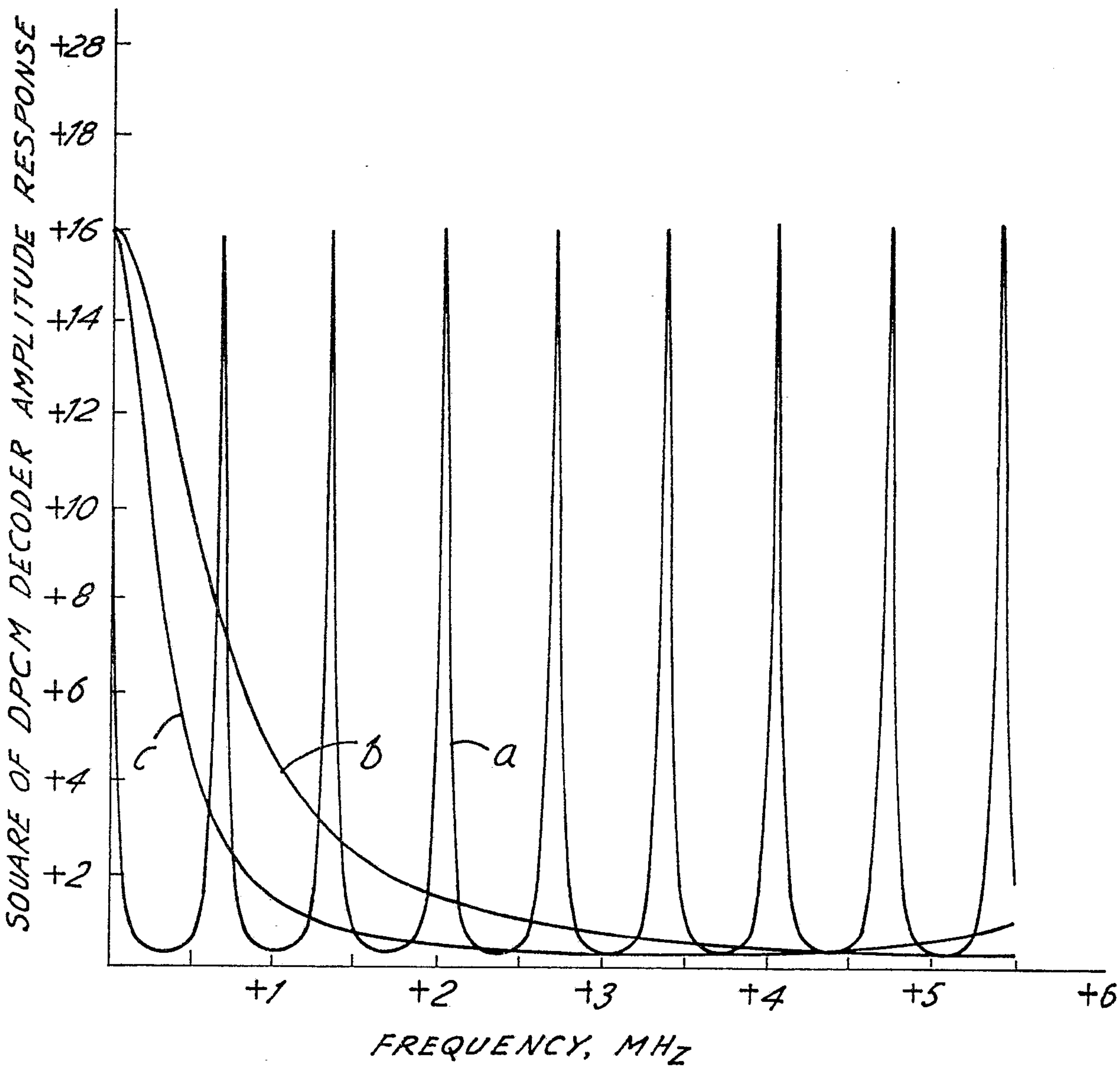
SQUARE OF DPCM DECODER AMPLITUDE RESPONSE
+28
+18
+16
+14
+12
+10
+8
+6
+4
+2
+1
+2
+3
+4
+5
+6
FREQUENCY, MHz
a
b
c

FIG. 6

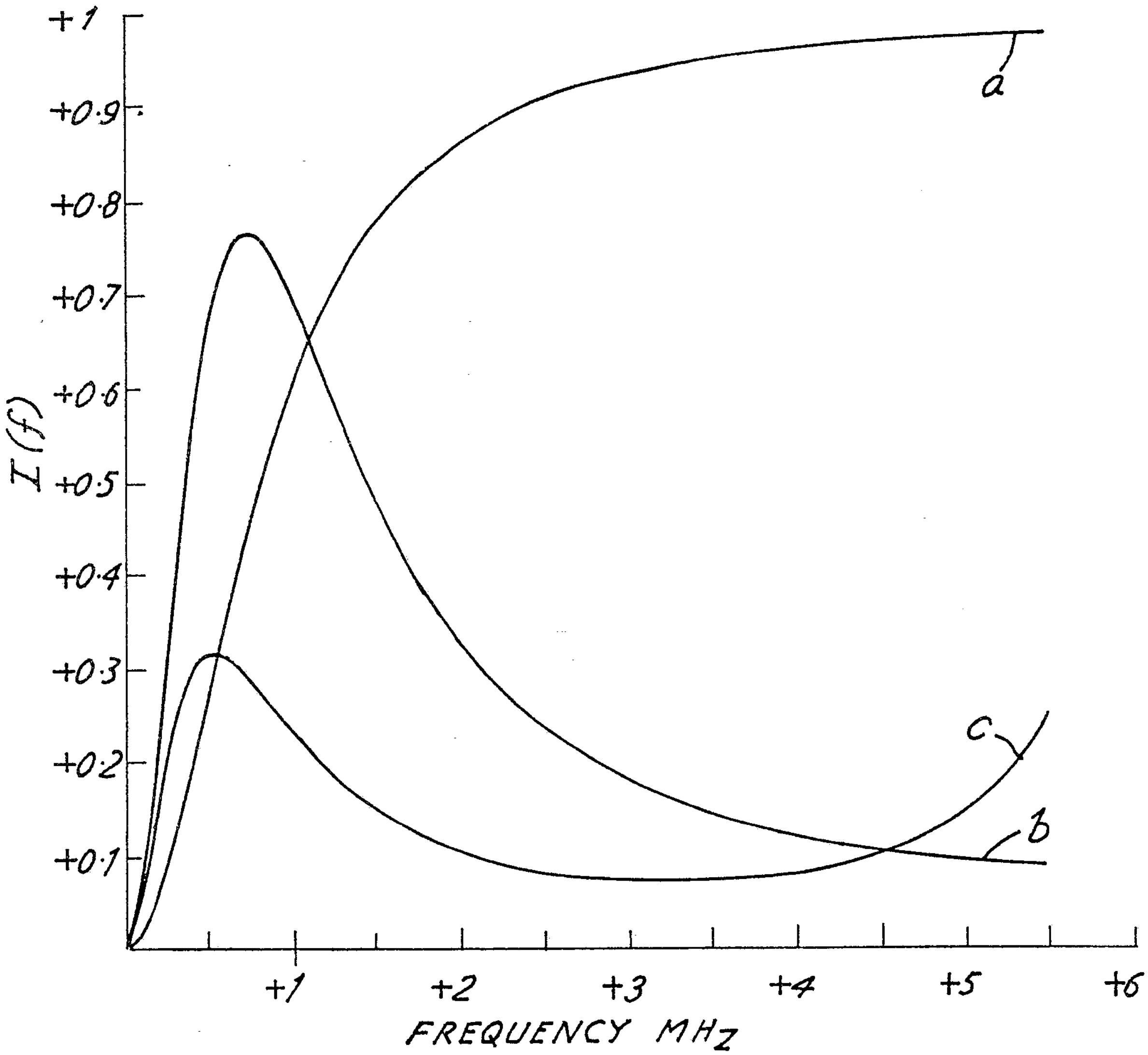

FIG. 7

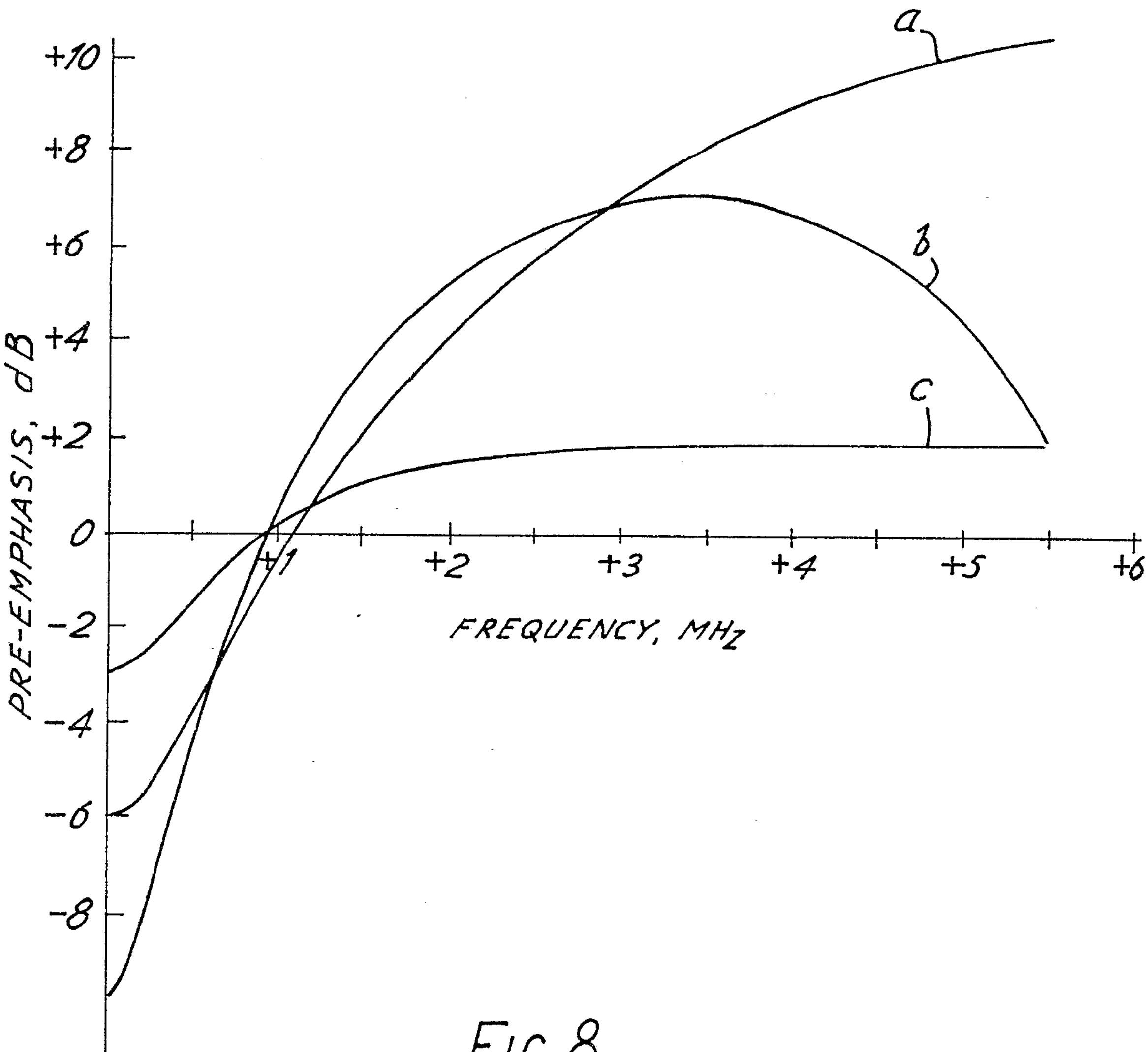

FIG. 8

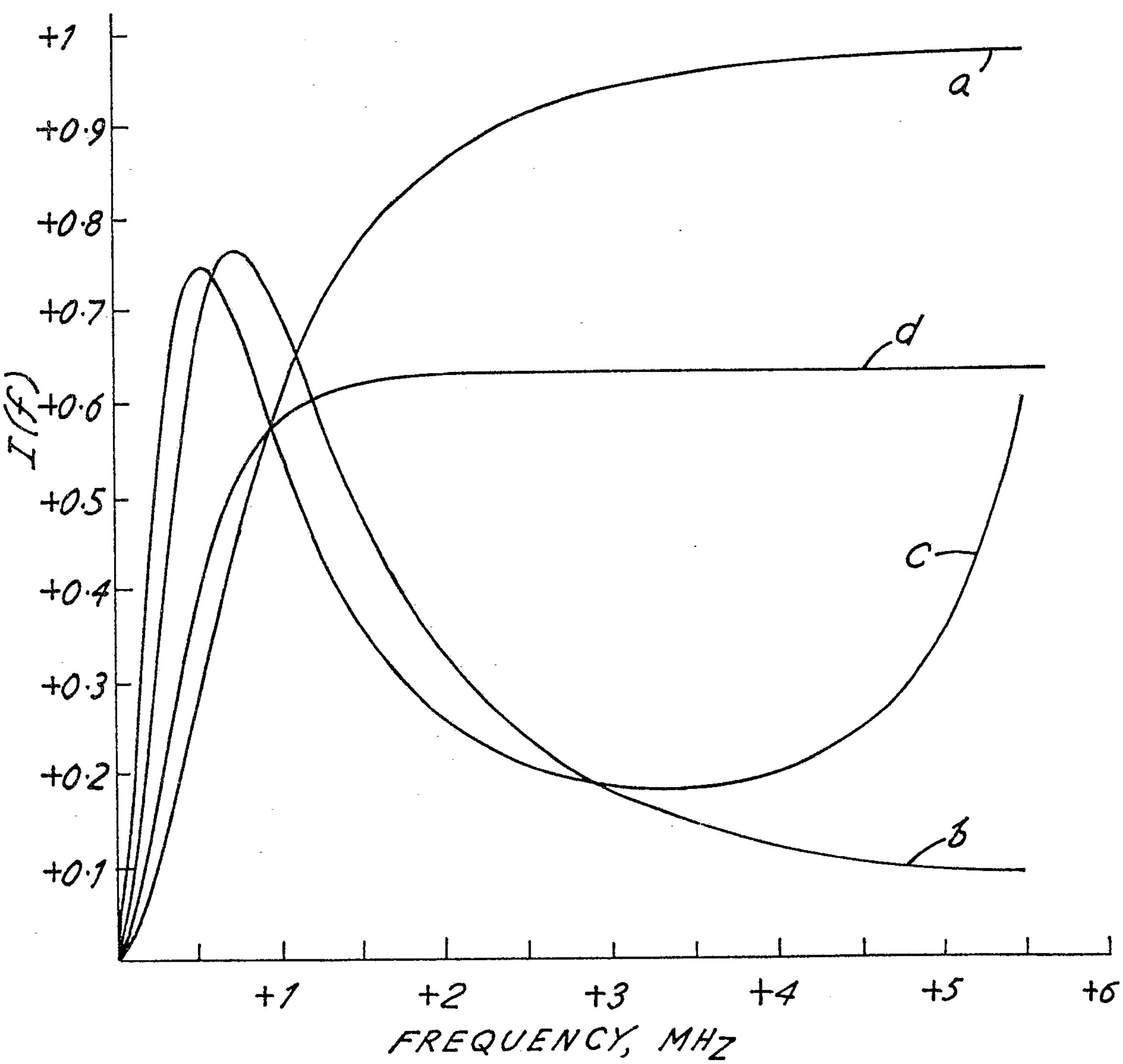

FIG. 9

SPECIFICATION

Analogue transmission of video signals

This invention relates to the analogue transmission of video signals using predictive techniques.

Video signals have in the past been transmitted over noisy channels using frequency modulation (f.m.) of a more or less wideband nature, thereby gaining a noise advantage at the expense of bandwidth, often with the use of pre-emphasis. This method has served well on point-to-point links and satellite circuits. However, if the available bandwidth is limited it becomes inadequate and unable to transmit the full spectrum of the signal. Therefore, for wide bandwidth signals such as high definition television (HDTV) broadcasting the conventional method is not very suitable.

In the transmission of digital signals methods exist for achieving reduction in bit-rate. In particular, in differential pulse code modulation (DPCM), the properties of a source S and an observer O are exploited. If S generates information with some pattern to it, then similar apparatus at S and O can be arranged to form a prediction of the next piece of information. Therefore, only departures from the expected pattern need to be transmitted. This leads to a reduction in the transmitted bit rate, the reduction being inversely related to the randomness of the signal. The other consideration in such a system is whether or not an observer O is responsive to all the information generated at S. If not, then a further reduction in bit-rate can be obtained by transmitting only information which can be detected by O. We have appreciated that similar techniques can be used for analogue signal transmission systems with advantage.

This invention relates to the analogue transmission of video signals using a predictive technique as set out in the appended claims to which reference should now be made.

In an embodiment of the invention to be described, the predictors used are digital DPCM predictors and the signal is converted to an analogue signal prior to transmission.

The invention will now be described in more detail by way of example with reference to the drawings in which:

Figure 1 is a block diagram of an analogue signal transmission system based on DPCM-type predictors.

Figure 2 is a block diagram of an experimental arrangement illustrating the invention.

Figure 3 is a functional equivalent of Figure 2 to illustrate more clearly the role of the compandor and the expander.

Figure 4 shows a "piece-wise linear" companding law curve with an overall gain of 1.

Figure 5 shows a typical non-linear companding law with an overall gain of 2.

Figure 6 shows the square of DPCM amplitude response for three different predictors.

Figure 7 shows the curves for the noise visibility function I(f).

Figure 8 compares MAC pre-emphasis with two different types of predictor.

Figure 9 shows the effect of pre- and de-emphasis in noise and visibility.

Figure 1 shows a DPCM based analogue system comprising a coding unit 2 and a decoding unit 4.

The coding unit 2 has an input 6, a limiter 8, and a coding module 10 in a negative feedback loop around the limiter. To this end the input and the feedback output of the coding module 10 are combined in subtractor 12. The output of the feedback loop is the output of the coding unit 2 and enters a compander 14 prior to transmission. A transmission link 16 which includes fm modulation and demodulation is provided between the coding and decoding units.

An expander 20 at the end of the transmission link provides the input 18 for the decoding unit 4. The decoding unit comprises a coding module 22 identical to the coding module 10 in the coding unit.

The two coding modules 10 and 22 each comprise a predictor 24 and an adder 26. One input 28 is available. The input 28 enters the adder 26 and is combined with the output 30 of the predictor 24. The output 32 of the adder 26 is the input to the predictor 24. Two outputs are available from the module, either the output 32 of the adder 26 or the output 30 of the predictor 24. In the coding unit 2, output 30 is used and applied to the inverting input of subtractor 12 and in the decoding unit 4 the output 32 is used and constitutes the output of the decoding unit.

The limiter 8 in the coding unit is used to contain the prediction error swing. The limiter will, if the predictor is well chosen, allows the most common small values of the prediction error to pass through perfectly accurately. However, there will be occasional large values of the prediction error and these will be distorted by the limiter. The limiter can be linear within its predetermined range, or it can introduce increasingly greater degrees of distortion as the prediction error increases. Thus, the coding unit can be arranged to transmit most picture detail accurately and to distort only rarely occurring types of detail.

The non-linear compander 14 and the expander 20 at opposed ends of the transmission link 16 are used to concentrate the effects of noise where they matter least. Small signals with most picture detail are amplified prior to transmission with the associated addition of noise. They are attenuated at the receiver, as is the noise. Large signals are sent "real-size" and suffer the full impact of transmission noise.

In operation, the subtractor 12 compares the input signal with an estimate or prediction of its value provided by coding module 10. This difference signal or prediction error signal is applied to limiter 8 to give a limited prediction error signal which is applied to the compressor 14 and thence to the transmission link. Between its threshold limits the limiter has a linear response. The limited prediction error signal is also applied to adder 26 in the coding module where it is added to the previous prediction signal. The output 32 of adder 26 is then applied to the prediction function circuit 24 which generates a new prediction from it.

The modules 10 and 22 at the transmitter and receiver are essentially identical so that the coder can mirror the signals generated at the decoder and generate a new prediction on the same basis as used at the decoder.

At the receiver end of the transmission link, the signal is first expanded by the expander 20. The regenerated prediction error signal is the input to the coding module 22, as is the case at the transmitter. However, the output of the coding module is the output 32 of the adder 26. This output is therefore the sum of the predicted signal and the limited prediction error signal.

The actual signal transmitted can be arranged to occupy the same peak-to-peak swing as the original input signal 6. Its distribution within that range depends on: the predictor chosen,—the d.c. leakage of the predictor,—the maximum deviation from OV at the limiter output, and—the small signal gain of the compressor.

The factor α is the amount by which the predictor d.c. gain is less than unity, i.e. its gain is $1-\alpha$. Typical values for α might be in the range 1/32 to 1/4. The d.c. content of the prediction error signal will thus range from $-\alpha V$ to $+\alpha V$ where the original video signal swing extends from 0 to V volts and the prediction error can therefore swing between $-V$ and $+V$.

The limiter will limit the excursion of the prediction error signal to $-\beta V$ to $+\beta V$. Typical values for β may be in the range 1/5 to 1/2. Conveniently the signal may then be normalised to $-V/2$ to $+ V/2$ by dividing by 2β.

At the input to the compressor the d.c. content of the signal lies in the range $-\alpha V/2\beta$ to $+\alpha V/2\beta$ if limiting is neglected, which is likely as normally α is smaller than β. This will be increased somewhat in the transmitted signal since the amplification factor applied by the compressor 14 lies in the range 1 to γ where γ is greater than 1.

Thus as α tends to zero the d.c. content of the signal is reduced, but if β is also reduced and γ is increased to improve the worse advantage this built in 'pre-emphasis' is progressively cancelled out. Thus a compromise must be made.

At the receiver there is a complementary process of expansion, scaling and decoding, giving a combined small-signal gain of $2\beta/\gamma$ which is the factor by which noise will be multiplied before it is received by the decoder. The decoder gain at d.c. and very low frequencies is l/α, i.e. greater than one, tending towards unity for less-predictable high spatial frequencies.

Thus very roughly the noise magnification M can be defined as $2\beta/\alpha\gamma$ at d.c. and very low frequencies reducing to $2\beta/\gamma$ at high spatial frequencies. If one assumes the values:

$$\alpha = 1/32 \quad \beta = 2/5 \quad \gamma = 8$$

then the low frequency noise magnification is 3.2, equivalent to +10 dB, and the high frequency noise magnification is 0.1, equivalent to −20dB.

Figure 2 shows an experimental set up to test the circuit of Figure 1. The DPCM coding and decoding units 2 and 4 are the same as these shown in Figure 1. The input 34 to the circuit is the luminance signal from a PAL coder. This analogue input is converted to an 8-bit digital signal by an analogue-to-digital converter 38. This ADC has a sampling frequency of 13.5 MHz generated by the clock input 40. The DPCM coding unit 2 is linearly range limited and the resultant prediction error is then compressed in the compressor 42. The signal is then converted back to analogue form by a digital-to-analogue converter (DAC) 44 and passed through a variable gain amplifier 46 with a gain G1. An analogue bypass loop 48 is provided to the whole coding process and either this or the coded prediction error signal as selected by switch 49 is combined with the output of a triangular noise generator 50 in an adder 52. The bypass loop 48 is used to simulate normal FM transmission, but without pre-emphasis and de-emphasis for comparison purposes.

The receiver end of the circuit is essentially the reverse of the above described transmitter end. An analogue bypass loop 54 is provided and either the signal through this or a decoded signal can be viewed on a monitor 56. The decoding process starts with a variable gain amplifier 58 of gain G2. An ADC 61 receives the signal and requires a clock pulse 60 and a sync pulse 62 to generate an internal clamp timing pulse. The output of this ADC is the input to an expander 64. This expander provides the input to a DPCM decoding unit 4 identical to the decoding unit in Figure 1. After this decoding, the signal passes through a digital-to-analogue converter 66 and then a variable gain amplifier 68 with gain G3. Either this signal or the signal on the analogue bypass line 52 can be received on the monitor 56 as selected by switch 69.

Figure 3 shows a functional equivalent circuit of Figure 2. In this the analogue bypass loops 48 and 54 are replaced by digital bypass loops 70 and 72 at the coder and decoder respectively, between the ADC and DAC convertors at the transmitting and receiving ends. The variable gain amplifiers are also omitted from this equivalent circuit.

The three variable gain amplifiers 46, 58 and 68 with gains G1, G2 and G3 are adjusted as follows. With the DPCM coder and decoder connected back to back, G3 is adjusted to give a signal equal in amplitude to the unprocessed analogue input. G1 is adjusted so that the peak-to-peak amplitude of the processed signal is equal in amplitude to the luminance signal, and G2 is adjusted to give unity gain between the compander and the expander.

The parameters varied in the experimental arrangement of Figure 2 are:

(a) the prediction algorithm,
(b) the range of the prediction error,
(c) the shape and overall gain of the companding law

In initial tests a piece-wise linear, non-linear companding law was used of the sort shown in Figure 4. Even without the addition of noise this form of law gives rise to very visible patterning in high frequency areas which is not present with a companding law with little or no non-linearity. Consequently, a more continuous form of law of the type shown in Figure 5 is needed. This does not give such visible pattering. The patterning with the laws of Figure 4 and 5 is probably caused by the fact that the sampling phase at the second ADC does not correspond exactly with the sampling phase at the first ADC. In addition the non-linearity tends to introduce some out-of-band components for high amplitude signals which can be removed by the low-pass filters in the signal path. With a linear "companding" law the quality of the decoded picture seems to be unaffected by variation in the phase of sampling at ADC2, 62 in Figure 3.

The DPCM equipment was programmed initially with a nine-element, 2-dimensional predictor which had been used for luminance coding of YUV signals in 140 Mbit/s. The coefficients are given in Table 1 below. When triangular FM noise is added to the analogue prediction error signal this gives rise to low frequency (noise) patterning on the output picture which is more visible than the impairment produced by the same level of FM noise added to the unprocessed signal. A reduction of this noise visibility can be achieved by reducing the d.c. gain of the predictor, e.g. from 0.98 to 0.75 this being achieved by reducing the amplitude of all coefficients accordingly. This shortens the error decay length of the predictor and reduces the visibility of the noise patterning. However, the noise is still more visible than for the same noise added to the unprocessed signal even for a range of companding laws.

Repetitions of the experiment with other 2-dimensional predictors with their d.c. gains adjusted to be equal to 0.75 show that the noise visibility is a minimum for a simple previous-sample predictor, in which the predictors 24 are a simple one-element delay (1T). Also, the use of a non-linear companding law rather than a linear law does not give any overall improvement in picture quality. The introduction of a larger companding gain for small differences compared to large differences reduces the noise visibility in plain picture areas but at the expense of noise on picture edges. This edge noise appears as noise streaks which extend from the edges into the surrounding plain areas.

With previous sample prediction and with the peak-to-peak range of prediction error limited to half the range of the input luminance signal and using a linear companding law such that the amplitudes of the processed and unprocessed luminance signals are the same, the noise visibility is less for the differential signal by a factor of about 3 dB.

Another result is that the noise visibility is reduced by changing to second-previous-sample (2T) rather than previous-sample prediction. However, this change causes some slope overload because of clipping of the now larger prediction error in the DPCM loop. To avoid clipping, the range of the prediction error signal can be increased to 0.77 of the input signal range. Consequently the gain of the compandor has to be reduced to keep the peak-to-peak amplitude of the transmitted signal constant. After these changes there is little to choose between the 1T and 2T differential coding system.

These results can be better understood by considering the effects of differential coding and decoding in the frequency domain.

The frequency response D(f) of the decoder is given by:

$$D(f) = 1/k(1 - P(w))$$

$$\text{Where } P(w) = \sum_{n=1}^{N} a_n \exp(-jwnT) = \text{Power Spectral Density}$$

a_n is the weight given to this sample in the prediction,

k is the overall compandor gain (assumed linear) at the coder,

and T is the sample interval.

At a given baseband frequency f, the FM noise amplitude is proportional to f and therefore the noise power density at the decoder output will be proportional to $f^2.|D(f)|^2$. The function $|D(f)|^2$ should therefore be small for large values of f. This implies that P(w) should not have a value close to unity, i.e. the prediction should be poor, at high frequencies.

If a predictor 0.75 of the adjacent sample from the previous line is used, then the square of the decoder amplitude response as a function of frequency would be as shown in Figure 6 curve (a). It can be seen that the FM noise power density is magnified by a factor of 16 at all multiples of line frequency and consequently differential coding using this predictor increases overall noise visibility.

A previous sample (1T) predictor is more suitable for this form of differential coding because the prediction accuracy decreases steadily with frequency. The function $|D(f)|^2$ for a 1T predictor is shown in curve (b) of Figure 6. Curve (c) of this figure also shows $|D(f)|^2$ for a 2T predictor. In order to understand why, with other parameters unchanged, 2T prediction gives lower noise visibility than 1T prediction it is necessary to consider the noise visibility weighting curve.

A generally accepted noise power weighting function W(f) for the luminance component is given by:

$$w(f) = 1 + (2\pi\tau f)^2$$

where $\tau = 200$ns
(see e.g. Bailey, W. F., Proceedings of the Institute of Radio Engineers 1954,42,1 and Hacking, K., BBC Research Dept. Report 1965/24).

The FM noise power density is proportional to f^2 and this is modified by the decoder de-emphasis function $|D(f)|^2$. It is a reasonable assumption that the overall noise visibility V(f) is then given by

$$V(f) = \int_0^{5.5\text{ MHz}} I(f)df$$

where $I(f) = \text{const. } f^2. |D(f)|^2.W(f)$

The function I(f) inside this integral is plotted in Figure 7. Curve (a) gives this function for FM noise with no differential coding, i.e. $D(f) = 1$. Curve (b) gives I(f) for 1T prediction (with d.c. gain equal to 0.75) and a linear compandor with a gain of 2. The area under curve (b) is smaller than that under curve (a) by a factor of 2.7, i.e. the model predicts that the noise should be less visible overall by a factor of 4.4 d8 using 1T differential coding. This is in fairly good agreement with the value of 3 dB found by experiment.

Curve (c) of Figure 7 shows the integrand for differential coding using 2T prediction with the same value of d.c. gain and compandor gain. It can be seen that the area under curve (c) is smaller than that under curve (b) which explains why the noise visibility decreases in changing from 1T to 2T prediction. However for 2T prediction the compandor gain k has to be reduced from 2 to 1.3 to allow for the increased range of difference signal. This would multiply curve (c) by a constant factor of 2.4 giving a total area almost equal to curve (b) for 1T prediction.

An important advantage of the differential technique described is that the resulting pre-emphasis cannot cause overdeviation of the FM carrier because the prediction-error signal is clipped inside the DPCM loop.

The amount of pre-emphasis introduced by the differential coding is given by

$$10 \log (k^2 |(1 - P(w))|^2) \text{ dB}$$

This is plotted in Figure 8(a) for 1T prediction with k=2 and (b) for 2T prediction with k=1.3. Figure 8(c) also shows the preemphasis applied to MAC signals for DBS transmissions (adjusted for non time-compressed video); i.e. MAC signals in which the luminance is time compressed by a factor of 1.5, $f_1 = .84$ MHz and $f_2 = 1.5$ MHz). This pre-emphasis response is described by

$$10 \log 0.5(1 + f^2/f_1^2)/(1 + f^2/f_2^2)$$

where the basic bandwidth of the signal $f_1 = 0.56$ MHz, and the sampling frequency applied to the signal $f_2 = 1.0$ MHz.

The effects of the three pre-emphasis and de-emphasis characteristics of Figure 8 on the integrand I(f) are compared in Figure 9, (a) with no pre- and de-emphasis, (b) for a 1T prediction with k=2, (c) for a 2T prediction with k=1.3, and (d) for MAC preand de-emphasis. It can be seen that the DBS de-emphasis does not give any prominent peaks in the noise visibility/frequency function but that the reduction in noise visibility at high frequencies is smaller than with the differential techniques. The amount of preemphasis at high frequencies is limited by the requirement to avoid overdeviation of the FM carrier. Using this model the DBS deemphasis function gives a reduction in overall noise visibility of 1.4 dB, which is in agreement with measured values of between 1 and 2 dB.

The peaks in the function I(f) for the differential systems can be reduced by reducing the d.c. gain of the predictor. However, as the prediction accuracy is reduced then the range of the prediction error before clipping must be increased and the compandor gain reduced. In the limit when the prediction gain is zero, this corresponds to the case of no pre-emphasis. For values of the d.c. prediction gain close to unity, the low-frequency peaks in I(f) become large causing an increase in overall noise visibility. Frobably, a value of d.c. gain of around 0.75 is optimum.

It is therefore demonstrated that a predictive method of transmitting analogue signals with an embodiment such as shown in Figure 1 is a sensible proposition. The arrangement described here uses digital methods to code and decode the signals, however, this is not a pre-requisite of the invention. The best overall picture quality and noise visibility is achieved here with simple previoussample prediction and linear "companding" rather than complicated multi-element prediction and non linear companding.

TABLE 1

Coefficient values for 140 Mbit/s 9-element 2D-predictor

(d.c. gain = 0.98, Previous sample = $C_{1,0}$)

Coefficient	Value
$C_{1,0}$	1.247
$C_{2,0}$	-0.797
$C_{3,0}$	0.635
$C_{4,0}$	-0.506
$C_{5,0}$	0.284
$C_{6,0}$	-0.013
$C_{-1,1}$	0.094
$C_{0,1}$	0.422
$C_{1,-1}$	-0.385

CLAIMS

1. A method of transmitting an analogue signal comprising the steps of receiving an input signal, comparing the input signal with a prediction signal to produce a prediction error signal, adding the prediction error signal and applying a prediction function to the resultant to provide the prediction signal, compressing the prediction error signal, and transmitting the compressed

signal in analogue form.

2. A method according to claim 1, including amplitude limiting the prediction error signal.

3. A method according to claim 1 or 2, in which the compressed signal is transmitted in fm form.

4. A method according to any preceding claim, in which the input signal is digitised and the signal converted back to analogue form prior to transmission.

5. A method according to claim 4, in which the prediction function comprises the previous sample subject to a predetermined attenuation.

6. A method according to any preceding claim, in which the signal is a composite colour television signal.

7. A method of receiving an analogue transmitted signal comprising the steps of expanding the received analogue signal adding the expanded signal and a prediction signal to provide an ouput signal, and applying a prediction function to the output signal to generate the prediction signal.

8. Apparatus for transmitting an analogue signal comprising an input for receiving an input signal, means connected to the input for comparing the input signal with a prediction signal to produce a prediction error signal, means for adding the prediction error signal to the prediction signal, means connected to the output of the adding means for applying a prediction function to the output of the adding means to produce the prediction signal which is applied to the comparing means and the adding means, a compressor connected to compress the prediction error signal, and means connected to the compressor for transmitting an output signal in analogue form.

9. Apparatus according to claim 8, including an amplitude limiter connected to the output of the comparing means to limit the prediction error signal before application to the compressor and the adding means.

10. Apparatus according to claim 8 or 9, in which the transmitting means comprises an fm transmitter.

11. Apparatus according to claim 8, 9 or 10, including an analogue to digital converter connected to digitise the input signal and a digital to analogue converter connected to convert the signal to analogue form for transmission.

12. Apparatus according to claim 11, in which the prediction function means comprises means for taking the previous sample subject to a predetermined attenuation.

13. Apparatus for receiving an analogue transmitted signal comprising an input for receiving the transmitted signal, an expander for expanding the received signal, adding means connected to the expander to receive the expanded signal and also receiving a prediction signal, and means connected to the output of the adding means for applying a prediction function to the output of the adding means to produce the prediction signal which is applied to the adding means, the output of the adding means providing a decoded output signal.

14. Apparatus for transmitting an analogue signal substantially as herein described with reference to Figure 1 of the drawings.

15. Apparatus for receiving an analogue signal substantially as herein described with reference to Figure 1 of the drawings.

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