Calibration of hydroacoustical instruments.

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A MICROPROCESSOR CONTROLLED TIME-VARIED GAIN AMPLIFIER - D. W. HOARE

INTRODUCTION

A 29.3 kHz time varied gain receiver amplifier has been developed for the Fisheries Laboratory at Lowestoft to be compatible with existing equipment to give the following two laws:

(1) $40 \log_{10} R + 2\alpha R$ dB

(2) $20 \log_{10} R + 2\alpha R + 20$ dB

Where R is the target range in metres and α is an attenuation factor having the units dB/m.

The first law is suitable for individual targets while the second is for scattering layers.

OVERALL SYSTEM

Two independent channels are fed from a common pre-amplifier stage which converts a balanced input into a single ended signal. Each channel has a digitally controlled gain stage using multiplying analog-to-digital converters. The variable gain stage is preceded by a front panel switchable gain stage (-20 to +30 dB in 10dB steps) and followed by a band pass filter and power output stage. The filters can be changed from 670 Hz to 2 kHz by changing a single board. The variable gain stages are controlled by a microprocessor which takes into account the attenuation constant setting of 0 to 9.9 dB/km in 0.1 dB steps and whether the range switch is set for short (3 to 150m) or long (12 to 600m). The microprocessor forms essentially a small computer having its program in a read only memory. A random access memory is used to store the required TVG law once it has been calculated for the particular settings required. A pulse from the transmitter scanning system to the computer initiates the time varying gain scan for the receiver. This is shown in Figure 1.

OPERATION OF SYSTEM

On switching the unit on, the gain is set to a minimum and an amber light is turned on to indicate that 510, 24 bit words are being calculated to give the required TVG law. Each of the 24 bit words is split into 13 bits (4 exponent & 9 mantissa) for the R^2 value and 11 bits (3 exponent & 8 mantissa) for the R value. The 510 values consist of 6 groups of 85, the step size increasing by a factor of 2 for each of the 6 groups. For short range 3 to 150 m is covered while for long range 12 to 600 m, the gain being set to a minimum until 3 to 12 m is reached as appropriate.

After about 6 seconds a green light replaces the amber to indicate that the unit is ready.
Should the short/long range switch or the attenuator setting be adjusted recalculation starts and the lights indicate this (when scanning it awaits the completion of the current scan before recalculation starts).

When the positive going edge of a scan pulse is received a scan starts and this can be restarted before a scan is completed for higher repetition rates. By operating a switch on the microprocessor board automatic scanning occurs at a rate of about 1 per second for test purposes.

Both the proportional to R & R^2 channels can be independently varied in gain or inhibited together. The gain can be varied in 10 dB steps from -20 to + 30 dB for the R channel and the R^2 on short range but is limited to 0 & -20dB on the R^2 long range (this is due to too much gain being present otherwise).

Should the system fail for any reason it can be reset by turning the mains switch off and on again but a built in test routine should automatically reset it should it fail.

SETTING UP PROCEDURE

Once the filters have been aligned the only adjustment required is that of the absolute gain by means of a trimpot on each channel.

GENERAL DESIGN CONSIDERATIONS

The specification aimed for was a relative gain accuracy of $\pm\frac{1}{2}$dB (absolute gain adjustable by ±3dB) and a noise level 50dB down on the maximum output of 5V subject to 1µV noise level referred to the input.

To meet the relative gain accuracy insufficient time was available at the start of a scan to compute the required next value as the scan proceeded. The gain changes by about 7dB in 1msec at around 3m on the R^2 range and as a 16 x 16 bit multiplication took about 2msec with the microprocessor used another approach was required. It was decided therefore to compute the complete gain profile and store this. As the gain changes rapidly initially, more values were calculated in this period, the gap being doubled after every 85 values up to a total of 510. The disadvantage of this method was that about 6 seconds delay is introduced after changing parameters and additional storage is required but neither seemed to be much of a penalty.

To obtain the low noise performance it is essential to get as much gain as possible before any dynamic switching occurs. The logical position for the preset switched gain was therefore before the TVG stages and to avoid problems with pick-up from leads to the front panel Reed relay switching was used. While this is reliable enough the Reed relays do add to the power dissipation of the unit.

Multiplying digital-to-analog (D-A) converters were selected to modify the gain and this is covered in a separate section below as it is the basis of the unit.

Filtering (Ref 1) of the signal to give the required bandwidth was essentially a separate design exercise. From noise considerations it had to take place as near the output as possible but due to the large gains involved some filtering was required to reduce the noise before the TVG stages. This was achieved using a simple tuned circuit which by making it act as part of the overall filtering could be made relatively narrow band thus aiding the performance.

To alleviate the effect of microprocessor noise at the output the mechanical layout used the power supply unit to separate the amplifiers from the processor.

MULTIPLYING DIGITAL-TO-ANALOG CONVERTER STAGES

This essentially worked on the basis of a coarse (exponent) stage followed by a fine (Mantissa) stage. The gain required was calculated essentially as a binary floating point number. The exponent part fixes the gain of a coarse stage to the required power of two and the Mantissa part fixes the gain of a following stage as shown in Figure 2.

The fine gain stage is easy to design as essentially the multiplying D-A acts as an attenuator and the amplifier is really just a buffer stage. The coarse stage uses the multiplying D-A in the feedback path of the amplifier and as the gain of the stage increases, at 30 kHz, it does not require a very high gain before gain-bandwidth product becomes a real problem. A high performance amplifier was used in this location but even so several such stages were needed to get the total gain variation required (24dB/stage maximum).

It might appear that an easier solution would be to use the fine gain type of stage to perform the entire function. This could be achieved by attenuating the large signals and not attenuating the weak signals so much and then amplifying the output, with a fixed gain, up to a reasonable level. The problem with this is that switching breakthrough from the digital part of the D-A converter gave a noise spike of about $1V_{p-p}$ having a duration of 2μsec at the output for both of the above stages. In the case of the coarse gain stage this has to be compared with almost the full output signal while for the fine gain stage unless the signal is already large as with the system shown it can lead to a very poor singal-to-noise performance. (This would be the case if additional gain were required).

An estimate of the signal-to-noise performance due to this break through with the system used can be made as follows.

Consider the wider bandwidth of 2kHz.

The maximum signal will be in the range 2.5 to 5V.

The 2μsec noise spike will be attenuated by a factor of about 250 due to the filter (2kHz passes 500μsec pulse). $\therefore$ Poorest maximum signal-to-noise ratio $\simeq 20 \log_{10} \left[\frac{2.5 \times 2 \times \sqrt{2} \times 250}{1} \right]$ dB $\simeq$ 65dB.

i.e. 15 dB in hand on design requirement.

The overall variable gain stages are shown in Figure 3.

<u>MICROPROCESSOR</u> An 8080A microprocessor was used with a crystal to give accurate time. The basic system which is essentially a small computer is shown in Figure 4. While the values for the output store are being computed the interrupt system is disabled but once they are available the interrupt is used to start each real time scan.
The complete program for the unit took about 760 bytes. A wire wrapped construction was used due to the relatively large number of connections to be made in a small space and it is worth noting that with an 8085 microprocessor the size could be considerably reduced.

<u>PERFORMANCE</u> Measurements made by Lowestoft (Ref 2) suggest that the unit was within specification on the short ranges and met the noise performance on the long ranges. With regard the long range accuracy particularly on the R^2 channel accurate measurements have yet to be made but measurement made indicate that it should be within ±1dB.

<u>CONCLUSION</u> A microprocessor controlled time varied gain amplifier has been demonstrated. More use could be made of the microprocessor and it is probable that future models of the amplifier would be given information on the sea temperature and salinity to compute the attenuation factor. While this is only a small additional step it indicates that further processing could be performed by the device.

Ref 1 Filters designed by A. R. Pratt - Loughborough University of Technology

Ref 2 Measurements made by B.Robinson - Fisheries Laboratory, Lowestoft.

Unit constructed and tested by T.Unwin - Loughborough University of Technology.

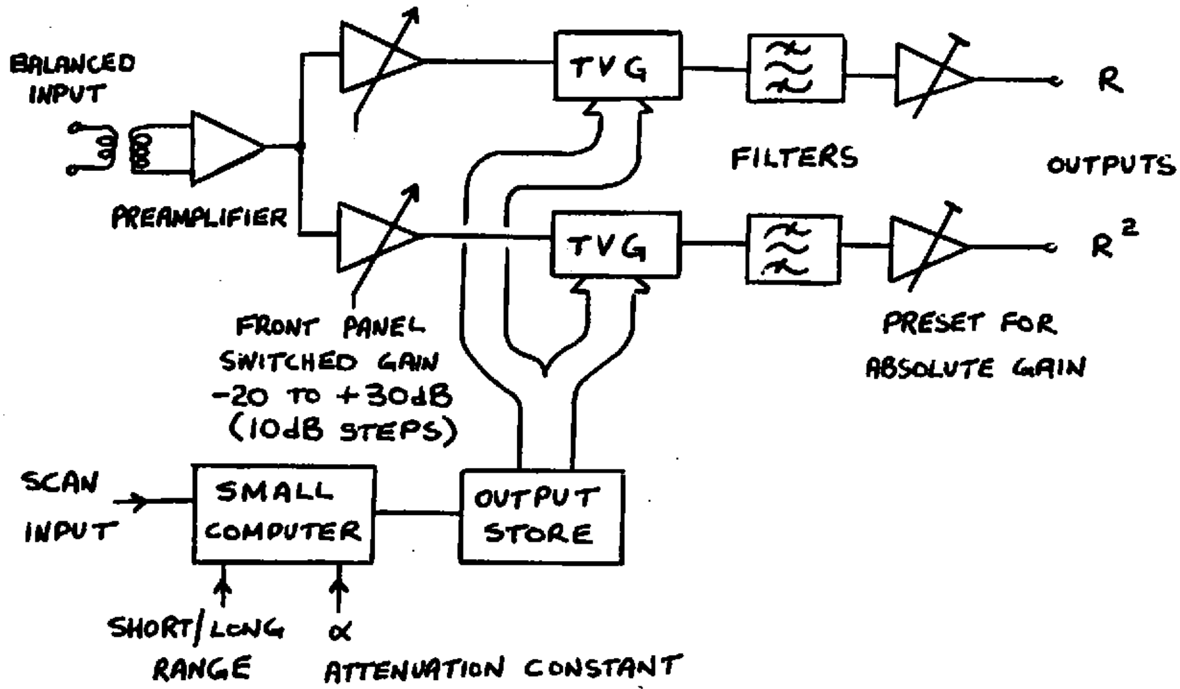

FIGURE 1 OVERALL SYSTEM

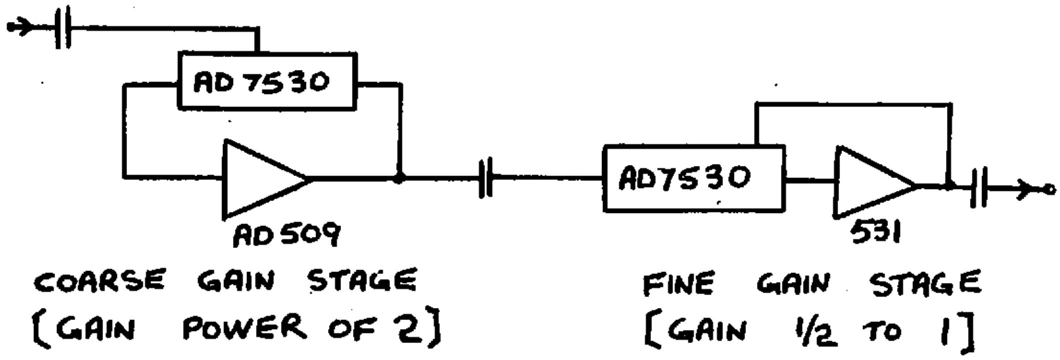

FIGURE 2 TVG STAGES

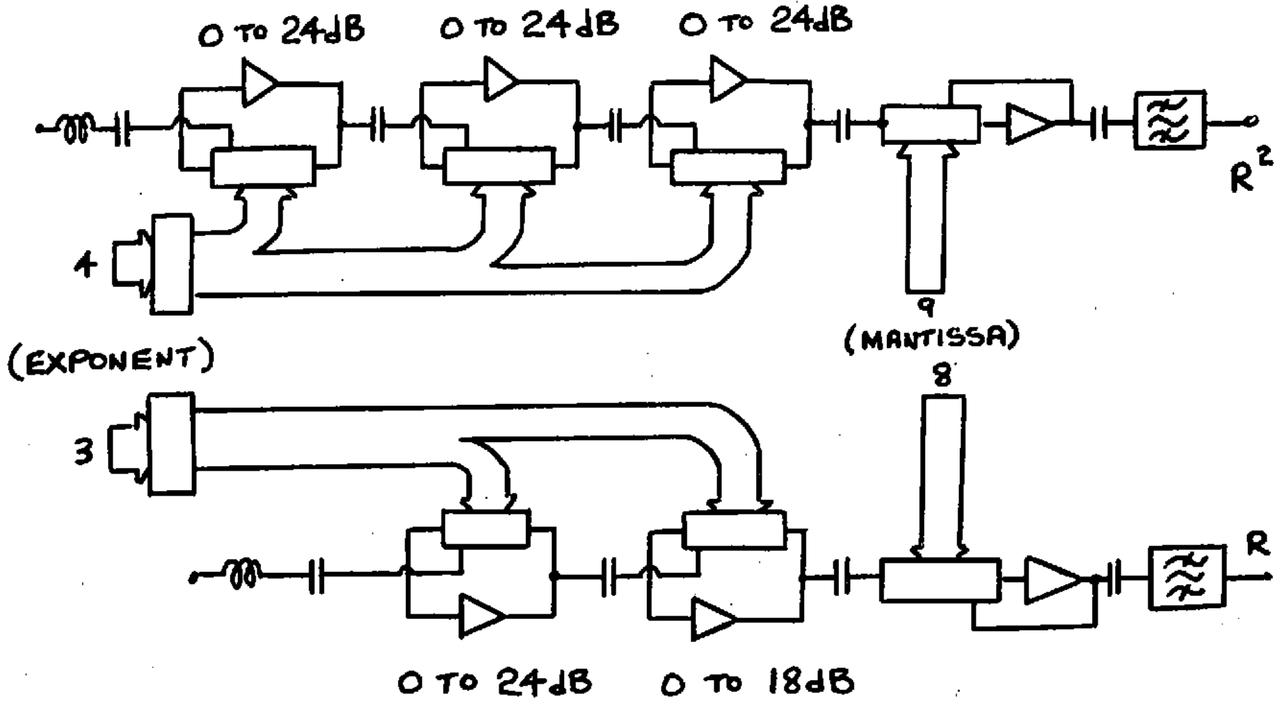

FIGURE 3 TVG AMPLIFIER SYSTEM

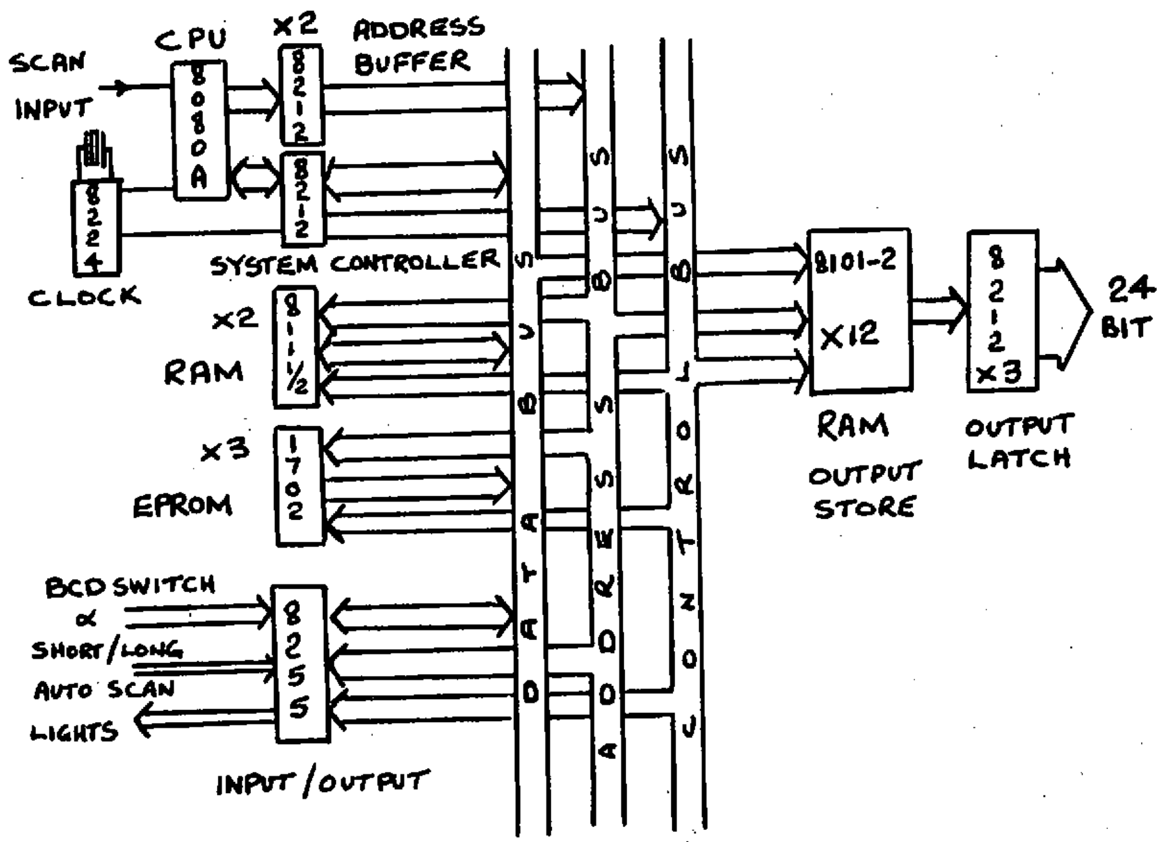

FIGURE 4 MICROPROCESSOR - SMALL COMPUTER