A Signal Combiner for Antenna Arraying

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The signal combiner performs the phasing and summing of a high data rate spacecraft telemetry signal received simultaneously at three DSN stations. The signals are combined at the subcarrier level after they have been microwaved to a common site. Before summing the signals the combiner delays them by the amount necessary to compensate for differences in phase due to unequal station-to-spacecraft distances and microwave delays. A tracking loop continuously adjusts for changes in signal delay so that correct phasing is automatically maintained throughout the spacecraft pass.

The signal combiner was successfully used in an experiment designed to improve the quality of real-time video data of the Mariner 10 spacecraft at its second Mercury encounter. Two 26-m antenna stations were arrayed with one 64-m antenna station at the Goldstone Deep Space Communications Complex. This combination produced an improvement in signal-to-noise ratio of 0.8 dB when compared to the 64-m antenna station by itself.

I. Introduction

A spacecraft signal received simultaneously at several tracking stations is affected by an independent noise contribution at each station. An improvement in signal-to-noise ratio (SNR) can therefore be obtained by combining the outputs from various receiving locations in an appropriate manner. This technique is sometimes referred to as "station arraying." Due to differences in spacecraft-to-station distances, the signal arrives with a different delay at each station. These delays vary with spacecraft position. Further delays, which are fixed but generally unequal, are incurred in relaying the signals to a common site. Compensation must be made for all of these delays if a meaningful combination of the signals is to be achieved.

Signal combination at the radio frequency (RF) or intermediate frequency (IF) level would be most desirable, but the technology for synchronizing the signals to the degree required is not yet available. Combining the signals at the subcarrier level, where synchronization requirements are less severe, is an attractive alternative. In fact, if the distance between stations is small and the data rate and subcarrier frequency are low, such that the delay difference amounts to only a small fraction of the bit time, the signals can be combined without delay compensation. This has been demonstrated by J. M. Urech (Ref. 1). At higher data rates and subcarrier frequencies a variable delay device is needed to phase the signals correctly. The design of the signal combiner was motivated by the problem of phasing the Mariner 10 117.6 kbits/s telemetry signal from three stations to an accuracy of 85 ns (1% of the bit time), with signal delays varying over a range of more than 50 μs during an 8-hr period. The signal combiner provides delay compensation that automatically tracks the changing signal delays and keeps the signals correctly phased throughout the spacecraft pass. The combiner also performs the summation of the properly phased signals.

II. Mariner 10 117.6-kbits/s TV Enhancement Experiment

Two 26-m antenna stations, DSSs 12 and 13, and one 64-m antenna station, DSS 14, were arrayed for the reception of the Mariner 10 high-rate TV data during the second Mercury encounter. The purpose of this experiment was to improve the quality of real-time video data and to demonstrate the arraying technology. Figure 1 describes the experiment configuration in general terms. The receiver baseband outputs from DSSs 12 and 13 are microwaved to DSS 14 via GCF 10, the Communications Switching Center located at DSS 12. The baseband outputs consist of a 177.1-kHz squarewave subcarrier modulated by 117.6-kbits/s telemetry data. The DSS 14 receiver baseband output is microwaved to GCF 10 and back to realize a fixed signal delay of 110 μs. This is necessary to assure that the signal from DSS 14 arrives later at the combiner input than the other two signals, at all times. At DSS 14 the three microwaved baseband signals are fed to the input of the signal combiner, which delays the signals from DSSs 12 and 13 by the amount required to synchronize them with the signal from DSS 14. Once properly phased, the signals are affected by a weighting factor according to their individual SNRs and combined in a summing amplifier. The weighting factors are chosen to optimize the SNR of the sum. The combined signal proceeds from the output of the signal combiner to the baseband input of the Subcarrier Demodulator Assembly (SDA). From there on, the signal is processed in the normal manner by the rest of the telemetry chain.

The three participating stations were in the "listen-only" mode for the experiment. The baseband signal SNRs of the 26-m antenna stations were about 10 dB below that of the 64-m station. Assuming signal addition with optimum weighting (discussed in a later paragraph on "mixing ratio") and perfect phasing, the SNR of the sum is

$$SNR_s = SNR_1 + SNR_2 + SNR_3$$

where

SNR_s = SNR of the combined signal

SNR_1 = SNR of the signal from DSS 14

SNR_2 = SNR of the signal from DSS 13

SNR_3 = SNR of the signal from DSS 12

The improvement in SNR of the combined signal over that of the 64-m antenna station by itself is

$$\frac{SNR_s}{SNR_1} = \frac{SNR_1}{SNR_1} + \frac{SNR_2}{SNR_1} + \frac{SNR_3}{SNR_1}$$
$$= 1 + 0.1 + 0.1 = 1.2 = 0.8 \text{ dB}$$

The following definitions are useful for discussing the delay relationships:

DV_2 = the variable spacecraft signal delay seen at DSS 13 with respect to DSS 14 (DV_2 = 20 μs, means that the signal arrives at DSS 13 20 μs later than at DSS 14)

DV_3 = the variable signal delay seen at DSS 12 with respect to DSS 14

DF_2 = the fixed delay incurred in microwaving the signal from DSS 13 to DSS 14

DF_3 = the fixed delay incurred in microwaving the signal from DSS 12 to DSS 14

DF_1 = the fixed delay in microwaving the signal from DSS 14 to GCF 10 and back to DSS 14

DC_2 = the variable delay the signal combiner must impart to the DSS 13 signal in order to phase it with the DSS 14 signal

DC_3 = the variable delay the signal combiner must impart to the DSS 12 signal to phase it with the DSS 14 signal.

The signal combiner does not delay the signal from DSS 14.

If the three signals are to arrive at the summing point in phase, the total delay incurred by each must be the same. Therefore $DF_1 = DV_2 + DF_2 + DC_2 = DV_3 + DF_3 + DC_3$ so that

$$DC_2 = DF_1 - DF_2 - DV_2$$

$$DC_3 = DF_1 - DF_3 - DV_3$$

where

$$DF_1 = 110\ \mu s$$

$$DF_2 = 82\ \mu s$$

$$DF_3 = 55\ \mu s$$

$$\left.\begin{aligned} DV_2 &= -57 \text{ to } +6\ \mu s \\ DV_3 &= -43 \text{ to } +9\ \mu s \end{aligned}\right\} \text{for Mercury II encounter}$$

From the above it can be computed that the signal combiner had to provide delays ranging from 22 to 85 μs for the DSS 13 signal and 46 to 98 μs for the DSS 12 signal.

The rate of change of the variable delays was less than 10 $\mu s/h$ at all times, i.e., $dDV_2/dt < 10\ \mu s/h$ and $dDV_3/dt < 10\ \mu s/h$.

III. Brief Functional Description of the Signal Combiner

Figure 2 shows a block diagram of the signal combiner. Signals 1, 2, and 3 are the baseband signals from the three receiving stations to be combined. Signal 1, which arrives more delayed than the other two, is fed directly to the summing amplifier. Two identical but independently variable delay channels (Channel A and Channel B on the diagram) bring Signal 2 and Signal 3 into phase agreement with Signal 1 before they reach the summing amplifier, where the three signals are added to produce the combined output.

Briefly, each delay channel functions as follows: The signal to be delayed is sampled at a rate of 2.5 MHz. The samples are then converted from analog to digital form and stored in a first in/first out (FIFO) buffer for a controllable length of time (delay). The control for the buffer delay is derived from the correlation of the buffer output with Signal 1. After leaving the FIFO buffer the delayed samples are reconverted to analog and fed to the summing amplifier.

IV. Detailed Discussion of the Signal Combiner

The three combiner input signals are conditioned by three essentially identical input amplifiers provided with front panel adjustable attenuators and rms voltmeters to monitor the amplifier outputs. The signal + noise voltages out of all three amplifiers are set to the same level. This makes it easier to obtain known mixing ratios. The input amplifiers limit the signal bandwidths to 1 MHz to prevent aliasing in the sampling process that follows.

From the input amplifier, Signal 1 proceeds directly to the summing amplifier; however, at the same time a quantized version of the signal is produced by a sample-and-hold (S/H) followed by a 1-bit analog-to-digital (A-D) converter. The sampling rate is 2.5 MHz. This digitized version of Signal 1 is used only for the purpose of correlation with Signals 2 and 3.

After passing through its input amplifier, Signal 2 is sampled at a rate of 2.5 MHz and then quantized by an 8-bit A-D converter. The 2.5-MHz sampling rate was chosen so as not to degrade the 177.1-kHz squarewave subcarrier appreciably. The use of an 8-bit A-D converter makes the quantization error negligibly small.

The 8-bit samples then pass through a serial FIFO buffer. The buffer output rate is determined by a fixed 2.5-MHz clock, while the buffer input rate is governed by a separate variable clock of approximately 2.5 MHz but asynchronous with respect to the output clock. Making the input clock slightly faster than the output clock will cause the number of samples stored in the buffer (buffer fill) to increase, since more samples are put into the buffer than are withdrawn. Conversely, an input clock lower in frequency than the output clock will cause the buffer fill to decrease. The length of time a sample remains in the buffer (delay) is a function of buffer fill and the phase difference between input and output clock. Therefore, the buffer delay can be continuously adjusted by controlling the input clock frequency. The approximate buffer delay can be calculated using the formula

$$\text{BUFFER DELAY} = \text{BUFFER FILL} \times \text{CLOCK PERIOD}$$

The maximum delay that can be achieved obviously depends on the length or capacity of the buffer. The buffers used in the signal combiner are 8 bits wide × 512 words long. With a clock period of 0.4 μs the delay would be $512 \times 0.4\ \mu s = 204.8\ \mu s$ for a completely full buffer and zero for an empty one. In practice, however, due to internal propagation time limitations the buffer cannot be run completely full or completely empty at 2.5 MHz, so that the actual maximum delay is about 180 μs and the minimum is 20 μs.

Parallel and series connected Fairchild 3341 FIFO memories (Ref. 2) were used to implement the delay buffers. A digital front panel display of the buffer fill is provided as an acquisition aid. At the output of the FIFO delay Signal 2 is correlated with Signal 1. This correlation, which is maximum when the signals agree in phase, is digitally displayed on the front panel. Furthermore the difference of the correlation of Signal 1 with Signal 2 advanced 90 deg of subcarrier phase and that of Signal 1 with Signal 2 retarded 90 deg is also developed. This difference, designated "quadrature" correlation, crosses zero when Signal 2 agrees in phase with Signal 1 and is, therefore, suitable as a control function for the delay tracking loop. Values for both the correlation and the quadrature correlation are generated once a second.

The quadrature correlation passes through a front panel controlled digital attenuator, which provides a means to manually adjust the loop gain.

A 16-bit digital-to-analog (D-A) converter translates the digital quadrature correlation values into an analog voltage, which is applied to the input of a voltage-controlled oscillator (VCO). The D-A converter output is also displayed by a front panel voltmeter.

The VCO consists of a John Fluke 644A frequency synthesizer adjusted so that the input voltage controls its frequency over a range of 20 Hz around a center frequency of 2.5 MHz. The VCO can also be switched to manual control, which is required during acquisition. The output frequency of the VCO determines the FIFO buffer input clock rate, thus closing the loop.

The control loop functions as follows: It is assumed that Signal 2 is initially in phase with Signal 1. If Signal 2 now advances (in phase) with respect to Signal 1 the quadrature correlation increases. This causes the VCO input voltage and, therefore, its output frequency to rise, making the FIFO buffer input clock rate higher. As a consequence there is an increase in buffer fill and delay, retarding Signal 2. The action of the loop tends to bring Signal 2 back into phase with Signal 1. The phasing error is less than 50 ns.

At the output of the FIFO buffer, Signal 2, which is now correctly phased with Signal 1, is reconverted to analog by an 8-bit D-A converter and then applied to the input of the summing amplifier.

As can be seen in Fig. 2, Signal 3 is processed by Channel B in the same manner as Signal 2 is by Channel A.

Signal 2 and Signal 3 are weighted by gains α_2 and α_3, respectively, before they are summed with Signal 1. These gains are chosen to optimize the SNR of the combined signal. The selection of α_2 and α_3 is discussed in a later paragraph. The summing amplifier itself consists of an operational amplifier followed by a 50-Ω cable driver and a front panel adjustable attenuator. An rms voltmeter is provided to monitor the combined output. Figure 3 is a photograph of the signal combiner.

V. Control Loop Acquisition Procedure

The delay tracking loop acquisition must be performed manually. The loop is opened by switching the VCO control to manual. Monitoring the FIFO buffer fill and varying the VCO frequency by hand, the operator must adjust the delay to within $\pm$ 2.8 μs of the value for correct phasing. The loop can then be closed by switching VCO control back to automatic. If the delay is more than 2.8 μs away from the right value when the loop is closed, false lock will result. This is due to the periodicity of the subcarrier correlation which allows the loop to lock at intervals of 360 deg of the 177.1-kHz subcarrier. Only one of these points, namely where both the subcarrier and the data correlate, is correct. It is possible to find this point by slowly varying the buffer delay and searching for the highest correlation peak with the aid of the correlation display. However, since this procedure can be somewhat time consuming, a Fortran program was written to predict the FIFO buffer fill as a function of time to an accuracy of $\pm$ 2 counts, which corresponds to a delay of $\pm$ 0.8 μs and is better than required. The program computes the variable delays from the station coordinates and antenna pointing angle predicts. It then combines variable and fixed delays to determine the FIFO buffer fill as a function of time and prints it at user selectable intervals. Armed with the program output, an operator can accomplish the task of acquisition in a few seconds.

VI. Best Mixing Ratio

The three inputs to the summing amplifier of Fig. 4 are three versions of the same signal affected by independent noise contributions.

$$S = \text{signal power}$$

$$N = \text{noise power}$$

$$SNR = \text{signal-to-noise ratio}$$

Inputs 2 and 3 are weighted by gains α_2 and α_3, respectively. The weighting should be such that SNR_s is maximum. Since the signals correlate and the noise does not

$$SNR_s = \frac{(\sqrt{S_1} + \alpha_2\sqrt{S_2} + \alpha_3\sqrt{S_3})^2}{N_1 + \alpha_2^2 N_2 + \alpha_3^2 N_3} \qquad (1)$$

SNR is maximum if

$$\frac{\partial SNR_s}{\partial \alpha_2} = \frac{\partial SNR_s}{\partial \alpha_3} = 0$$

and

$$\left(\frac{\partial^2 SNR_s}{\partial \alpha_2^2} \cdot \frac{\partial^2 SNR_s}{\partial \alpha_3^2}\right) - \left(\frac{\partial^2 SNR_s}{\partial \alpha_2 \partial \alpha_3}\right)^2 > 0 \text{ and } \frac{\partial^2 SNR_s}{\partial \alpha_2^2} < 0$$

The above conditions result in the system of equations:

$$\alpha_2 = \frac{\sqrt{S_2}(N_1 + \alpha_3^2 N_3)}{N_2(\sqrt{S_1} + \alpha_3\sqrt{S_3})}$$

$$\alpha_3 = \frac{\sqrt{S_3}(N_1 + \alpha_2^2 N_2)}{N_3(\sqrt{S_1} + \alpha_2\sqrt{S_2})}$$

which is solved by

$$\alpha_2 = \frac{N_1\sqrt{S_2}}{\sqrt{S_1}N_2} \qquad \alpha_3 = \frac{N_1\sqrt{S_3}}{\sqrt{S_1}N_3}$$

These are the values of α_2 and α_3 that maximize SNR_s and substituting them into Eq. (1) yields

$$SNR_{s\,(\max)} = SNR_1 + SNR_2 + SNR_3$$

It can be shown that similar results hold for n inputs with

$$\alpha_n = \frac{N_1\sqrt{S_n}}{\sqrt{S_1}N_n}$$

and

$$SNR_{s\,(\max)} = SNR_1 + SNR_2 + \cdots + SNR_n$$

For the Mariner 10 television (TV) enhancement experiment a value of 0.3 was used for α_2 and α_3 with the signal combiner input attenuator set such that

$$S_1 + N_1 = S_2 + N_2 = S_3 + N_3$$

The sensitivity of SNR_s to α_2 and α_3 is low, so that their accuracy is not very critical.

VII. Conclusion

The signal combiner performed as predicted during the preencounter tests and the Mercury II encounter. The combined signal showed an improvement in ST_b/N_0 of 0.8 ± 0.1 dB over the DSS 14 signal by itself. No significant operational difficulties were encountered. The Mariner 10 TV enhancement experiment has clearly demonstrated that antenna arraying is practical and can be used to improve telemetry performance. A combination of two 64-m antennas, if they were available within the same complex, could produce a 3-dB improvement in signal-to-noise ratio.

References

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2. "The 3341 First In/First Out Serial Memory," *Optimos Manual*, pp. 74–79, Fairchild Semiconductor, Mountain View, Calif., September 1972.

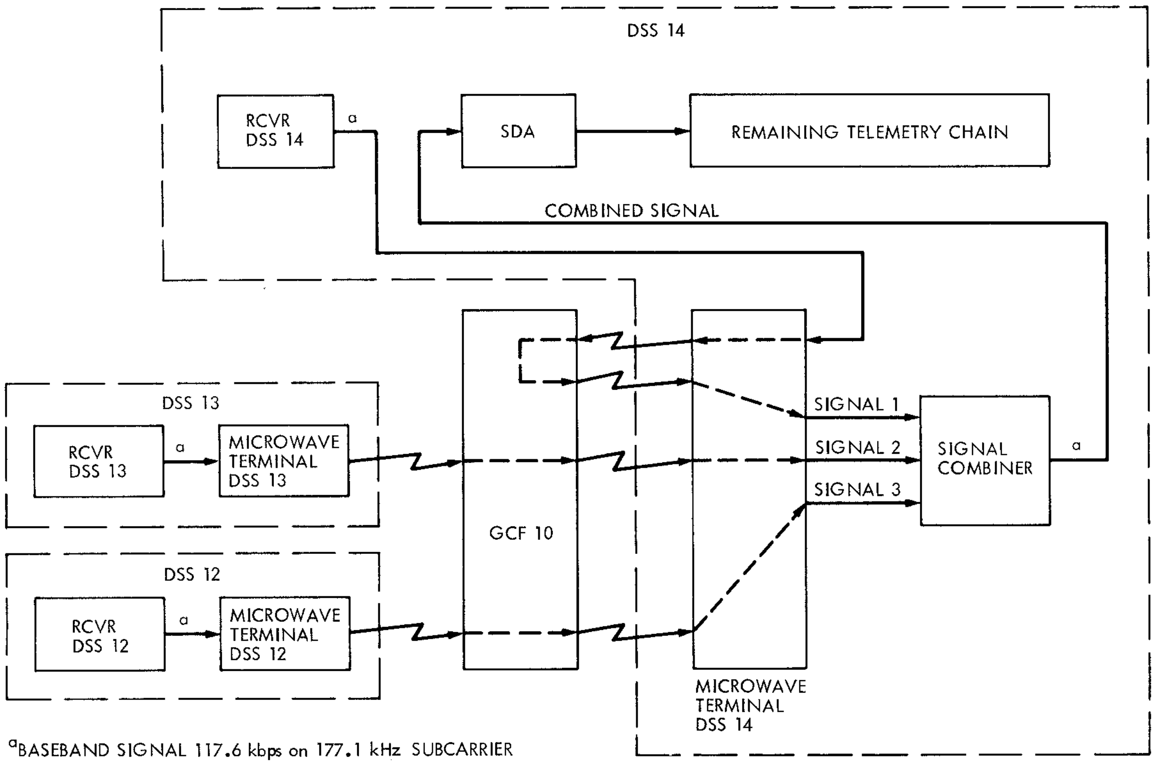

Fig. 1. MVM TV enhancement experiment configuration

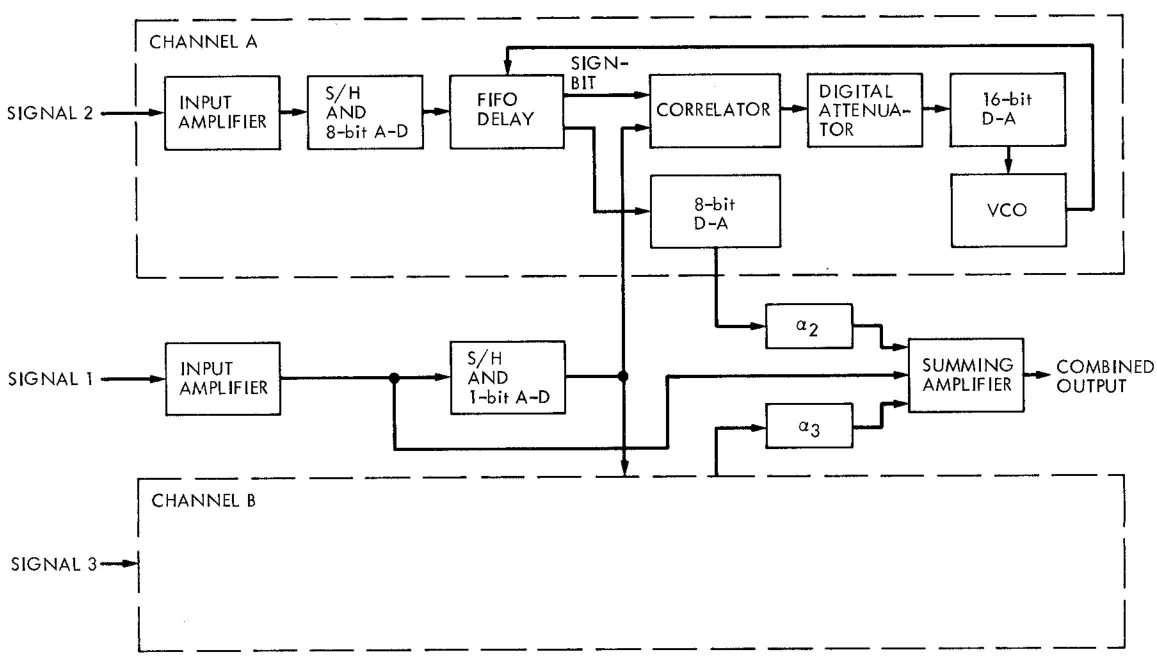

Fig. 2. Signal combiner block diagram

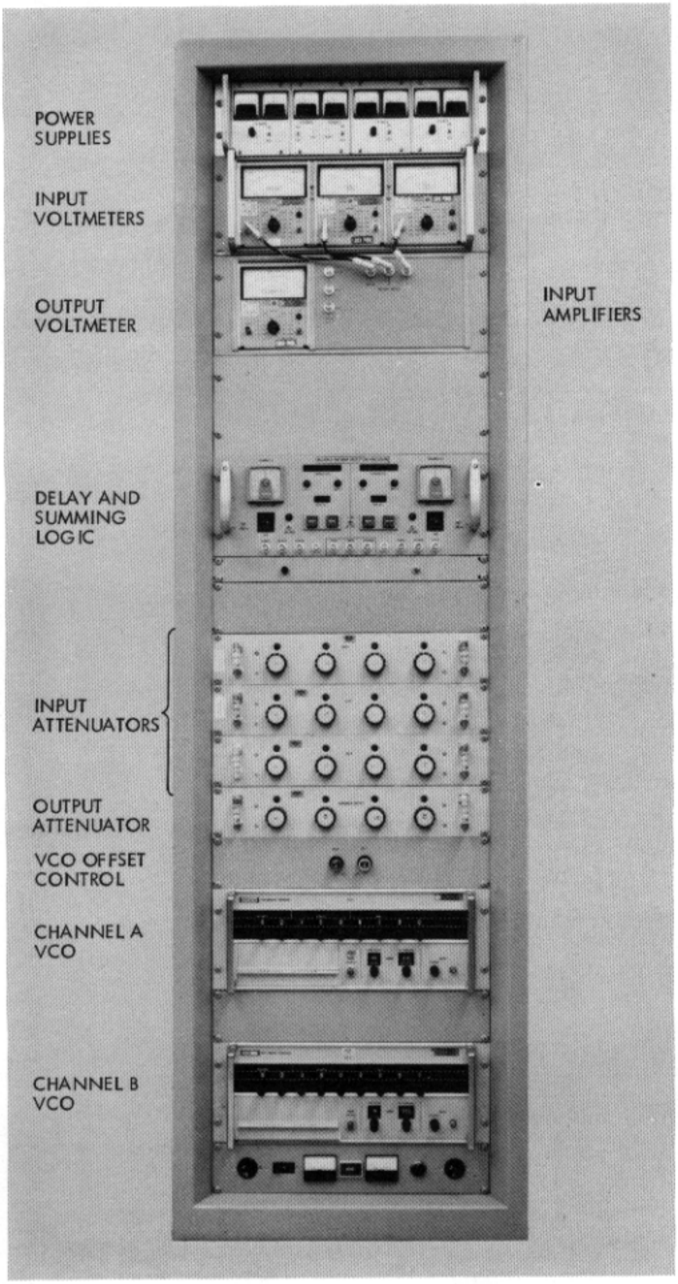

Fig. 3. Signal combiner

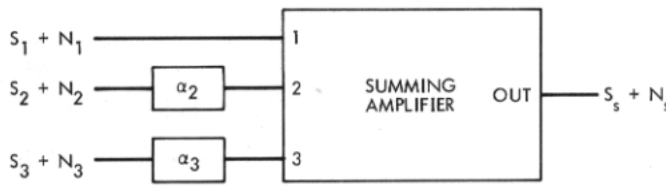

Fig. 4. Mixing ratio