Open-Loop Receiver/Predetection Recording System for the DSN

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A configuration for recording the telemetry IF output of an open-loop receiver and reproducing the data at minimal loss, incorporating a digital time-jitter compensator on playback is described. Results of tests performed at CTA 21 in a DSN-compatible environment show a total system degradation of less than 0.5 dB (ST_b/N_0) at bit rates as low as 16 bps and over a range from -2 to 9 dB (ST_b/N_0).

I. Introduction

A practical configuration for an open-loop receiver/predetection recording system for use in the DSN with existing standard hardware is described. The breadboard model was implemented in tests at CTA 21 in a DSN-compatible environment. Results demonstrated the feasibility of recording/reproducing telemetry data from an open-loop receiver with a total system loss of less than 0.5 dB (ST_b/N_0) at bit rates as low as 16 bps and over a range from -2 to 9 dB (ST_b/N_0). Description of the analog hardware required to frequency down-convert and band-limit the telemetry IF output on record and to up-convert on playback is followed by a brief analysis of the time-jitter compensator introduced earlier (Ref. 1). Results of tests using the experimental configuration at CTA 21 are presented.

II. Test Configuration

Figure 1 is a block diagram of the experimental setup. The programmable Simulation Conversion Assembly (SCA) modulates the test transmitter with simulated data and feeds receiver 2 through a low-noise amplifier. The IF output of receiver 2, operating in an open-loop mode, is bandpass-filtered to a total bandwidth of 80 kHz and frequency-translated to a center frequency of 50 kHz. The filtering is necessary for image rejection and to ensure that the recorded signal spectrum remains within the bandwidth capability of the breadboard time-jitter compensator used on playback. This bandpass output is then amplified, linearly added to a synchronizing sinusoid at a frequency of 201 kHz and recorded at 76.2 cm/s (30 ips) on an Ampex FR 1400 tape unit. During recording (switches positioned at "B"), the signal is simultaneously frequency up-converted to S-band and sent through Receiver 1 and the subcarrier demodulator assembly (SDA), where the data are detected. The detected bit stream is acquired and tracked by a software loop in the Telemetry and Command Processor Assembly (TCP) computer, utilizing a Mariner test program, DOI 5081TP. Estimates of ST_b/N_0 and calculated bit error rates corresponding to the recorded data are generated. These statistics, when compared with the program output generated in the direct mode (switch at "A"), show the degradation introduced in the frequency down-conversion, filtering, and up-conversion phases.

The tape is played back on an Ampex FR 2000 unit, which exhibits about an order-of-magnitude better instantaneous phase stability than the FR 1400 recorder. Present memory size limitations of the breadboard time-jitter compensator prevent its use exclusively with the FR 1400 units (Ref. 1).

The compensated playback signal (with the synchronizing sinusoid removed) is then up-converted to S-band (switch at "C") and sent through Receiver 1 and the data detection hardware, and performance statistics are again generated. Losses introduced in the record/playback system and total system losses are determined.

III. Open-Loop Receiver

Figure 2 is a block diagram of a portion of the Block 3-C receiver used in an open-loop configuration. The standard Block 3-C receiver can be operated as an open-loop receiver (OLR) simply by "shorting" the tracking filter and operating the voltage-controlled oscillator (VCO) manually. Receiver gain control is also accomplished manually. In this configuration (see Fig. 2), the signal is frequency-translated to 10.040 MHz using an offset in the VCO and then filtered by a special 80-kHz bandpass filter which is inserted in the telemetry B channel. The linear output of the telemetry IF amplifier is then translated to 50 kHz for output to the time-jitter compensator (TJC).

The main areas of concern in terms of preventing signal degradation are (1) maintain the gain distribution so as to prevent saturation on noise, (2) provide filtering adequate to pass the data with minimum sideband losses yet sufficient to eliminate image noise, and (3) provide sufficient gain so as to minimize thermal noise contributions from all following stages. Saturation on noise, which results in signal suppression, is prevented by maintaining the rms noise levels at least 20 dB below the 1-dBm compression limit of all OLR components. This is achieved by first setting the gain equal to that which results from the input signal level in the normal mode of receiving. The variable attenuator in the telemetry B channel is then adjusted so that the rms signal-plus-noise power equals -13 dBm as measured at the linear output of the 10-MHz IF amplifier. The square-wave subcarrier frequency was set at 1.7 kHz for data rates of 16 bps and at 3.197 kHz for 2048 bps, to remain within the filter passband of 70 kHz. With these subcarrier frequencies, either 19 or 11 sidebands were passed. Figure 3 shows the characteristics of the filter.

The OLR output level of -13 dBm insured more than adequate gain to overcome thermal noise contributions introduced in following stages of analog hardware.

Figure 4 shows the block diagram of the breadboard upconverter. It is a dual-conversion unit employing doubly balanced commercial mixers. Dual conversion is employed to space S-band signal harmonics so that they are several channels away from the desired signal. The first local oscillator (LO) of 2.457 MHz mixed with the 50-kHz input signal provides harmonic spacing at S-band of 2.507 MHz, which is more than adequate since receiver channel spacing is 370 kHz. The choice of 2.507 MHz is also convenient in that summing it with the second LO of 2292.49 MHz results in an S-band signal in receiver channel 14, which is already available in both receivers and one of the test transmitters at CTA 21. Not coincidentally, the second LO frequency is in receiver channel 7, which also is currently available in a test transmitter at CTA 21.

To prevent saturation in the upconverter, the rms signal-plus-noise levels into both mixers are set at -16 dBm. The 50-dB output attenuator is necessary to prevent saturating the Block 3-C phase-locked receiver used for data recovery. The bandpass filter in the unit has a 1-dB bandwidth of 100 kHz centered at 2.507 MHz.

IV. Record/Playback System

Figure 5 shows the experimental setup for performing recording of 10-MHz telemetry IF. Additionally, an amplifier and a linear mixer are required to add in a sine-wave sync signal at a frequency at least twice the baseband width. Since the record/playback speed ratio is unity, the tape speed at record is set to the value that provides sufficient bandwidth for accommodating the signal spectrum. Table 1 lists the available tape speeds and associated bandwidths for the Ampex FR 1400 tape unit. With

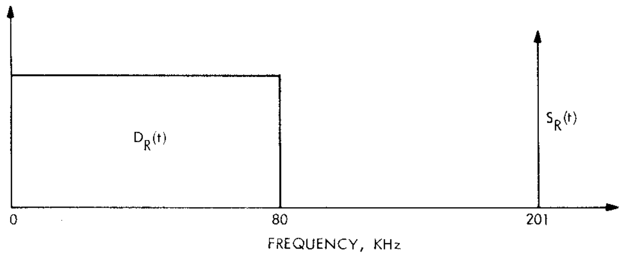
$$D_R(t) = \text{data signal at record}$$

$$S_R(t) = \text{timing signal at record}$$

where

$$S_R(t) = A_R \sin(\omega_0 t + \theta_R), \qquad \left(\frac{\omega_0}{2\pi} = 210\,\text{kHz}\right)$$

the input recorded spectrum ideally has the form shown below:



On playback, the output becomes $D_p(t) + S_p(t)$, where now

$$S_p(t) = A_p \sin\left(\omega_0 t + \theta_p + \int \omega_f(t)\,dt\right) + \text{noise}$$

The signal has thus been frequency-modulated by the random flutter frequency $\omega_f(t)$, which may be negative, and experimentally was found to be limited to about a 100-Hz bandwidth.

The instantaneous playback frequency is

$$[\omega_0 + \omega_f(t)]$$

Since both $S_p(t)$ and $D_p(t)$ are recorded/reproduced on the same tape track, they are both equally modulated by $\omega_f(t)$. Therefore, $S_p(t)$ can be used to digitally sample $D_p(t)$ and to clock the samples into a first-in/first-out buffer memory. After the memory is half full, the data are clocked out at a constant rate equal to the frequency of $S_R(t)$ (or ω_0), thereby removing the flutter modulation. This is the function of the TJC.

Figure 6 details the playback arrangement. The sync signal is removed from the output signal track in a phase-locked loop tracking filter with a tracking bandwidth of 100 Hz. The sampled signal is written into the digital buffer at the modulated rate and read out at a constant rate, equal to the frequency of the originally recorded sinewave sync signal. The output is then passed through a digital-to-analog converter and low-pass filter with a cutoff frequency of half the sync-signal frequency, thus providing a reconstructed and compensated version of the baseband signal. The recorded clock (write clock) directly triggers the analog-to-digital converter (ADC). Therefore, the buffer input rate is

$$\left(\frac{\omega_0 + \omega_f(t)}{2\pi}\right)\left(\frac{\text{samples}}{\text{sec}}\right)$$

The output data rate is

$$\left(\frac{\omega_0}{2\pi}\right)\left(\frac{\text{samples}}{\text{sec}}\right)$$

Figure 7 details the buffer memory organization. A $1K \times 6$ bit array of shift registers is partitioned into 8 independent read/write access lines. Data are shifted through the ith column of registers by either a read or write clock pulse appropriately gated by R1 or W1, each high for 128 read and write clock pulses, respectively. The only forbidden state is when both W1 and R1 are high simultaneously, resulting in an overflow (or underflow) condition. This occurs if the total number of input samples in time T in excess of output samples for the same time T exceeds the buffer capacity. That is, in general, for no overflow, require

$$\frac{1}{2\pi}\int_0^T |\omega_f(t)|\,dt < \left(\frac{N-1}{N}\right) M$$

for an M sample memory partitioned by N unique read/write lines. Here, $N = 8$, $M = 1K$.

V. Experimental Results

Table 2 summarizes the tests performed and their results. The subcarrier frequency (F_{sc}) was originally set to 8.1 kHz. However, with a total available bandwidth of 80 kHz, centered at 50 kHz, only the first and third harmonics were detected in tests 1 and 2. This sideband suppression introduced about 0.3 dB (ST_b/N_0) data degradation. With the lower subcarrier frequency of 1.7 kHz, sufficient harmonics were passed, thus removing this possible source of error.

Owing to the problem of accurately measuring the signal-to-noise ratio into the receiver, the calculated direct mode bit error rate (BER) was averaged and plotted on the theoretical BER versus ST_b/N_0 curve to determine the ST_b/N_0 operating value. The playback averaged BER was then plotted on the same curve at the direct ST_b/N_0 value to give an estimate of the data degradation. Figure 8 shows the theoretical BER versus ST_b/N_0 curve with the direct and playback results plotted.

It can be seen that the measured total system loss is within 0.5 dB (ST_b/N_0). Plots of the probability density of occurrence of values of bit error rates are shown in Figs. 9 through 14. Each value of P_D represents the cumulative average number of bit errors that occurred within the range of values represented in a one scale interval of the abscissa. With sufficient statistics it is expected that the curves follow a binomial distribution. The curves of Figs. 9, 11, and 14 best illustrate this, while an insufficient number of BER statistics caused the misleading "bimodal" forms shown in Figs. 10, 12, and 13. The curves point up the fact that the OLR/record/playback system does not introduce any additional data degradation in the form of altering the error distributions, such as causing burst errors, which become more important when coded data are handled.

Overflows in the TJC occurred on an average of twice per one hour playback. This resulted in about a 10% total loss of data typically in an hour's run, due mainly to the time required for the software data loop to reacquire after an overflow. With an automatic reset (not available with the breadboard TJC), the amount of data lost in an overflow, in seconds, is given by

$$\frac{\left(\frac{N-1}{N}\right)M}{\frac{\omega_0}{2\pi}}$$

In the tests this quantity equals $(7/8 \times 1024/201 \times 10^3)$, or about 4.5 msec.

VI. Conclusions

A breadboard model of an open-loop receiver/predetection recording system has been effectively demonstrated at CTA 21 in a DSN-compatible environment. Results indicate a total system degradation within 0.5 dB (ST_b/N_0), measured over a range of signal-to-noise and data rate values, including tests performed close to the real-time system data detection threshold.

Reference

1. Slekys, A. G., "Implementation of a Flutter Compensator for DSN Predetection Recording," in *The Deep Space Network Progress Report*, Technical Report 32-1526, Vol. XVI, pp. 132–139, Jet Propulsion Laboratory, Pasadena, Calif., Aug. 15, 1973.

Table 1. Tape speed vs bandwidth and recording time for the Ampex FR1400 tape unit

Tape speed, cm/s (in./s)	Recording bandwidth, kHz	Recording time, h
304.8 (120)	1500	0.25
152.4 (60)	750	0.5
76.2 (30)	375	1
38.1 (15)	187	2
19.05 (7½)	93	4
9.52 (3¾)	46	8
4.76 (1⅞)	23	16

Table 2. Experimental data summary

Test	F_{sc}, kHz	Uncoded rate, bps	Mode	BER	ST_b/N_0, dB	Loss, dB	BER values	Bits per BER value
T7	3.197	2048	Direct	0.000016	9.40	–	37	100,000
			Playback	0.000020	9.25	0.15	63	100,000
T3	1.70	16	Direct	0.004061	5.50	–	17	1000
			Playback	0.003966	5.50	0	54	1000
2	8.1	16	Direct	0.041107	1.80	–	298	96
			Playback	0.046761	1.50	0.30	115	96
1	8.1	16	Direct	0.049992	1.35	–	249	96
			Playback	0.058408	0.95	0.40	224	96
T6	1.7	16	Direct	0.063121	0.70	–	34	1000
			Playback	0.067182	0.50	0.20	43	1000
T5	1.7	16	Direct	0.113225	−1.35	–	71	1000
			Playback	0.127155	−1.85	0.50	35	1000

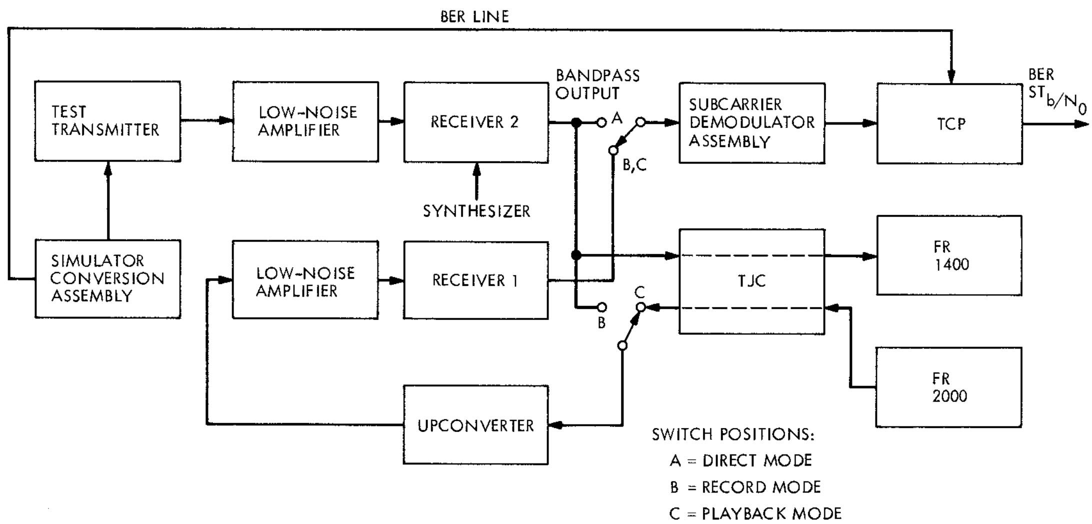


Fig. 1. Block diagram of experimental setup

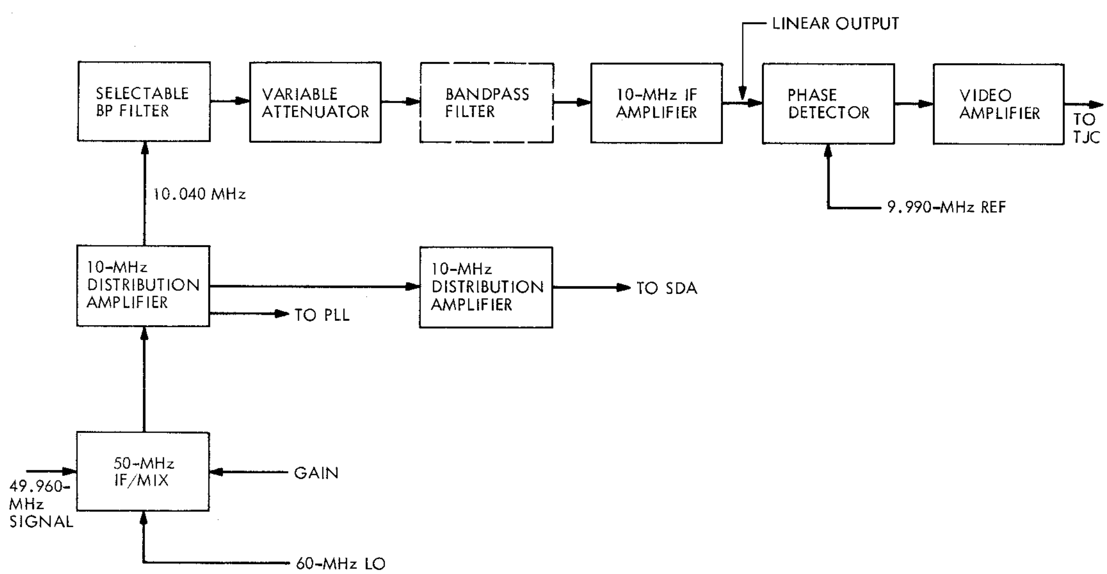


Fig. 2. Partial diagram of the Block 3-C receiver showing open-loop configuration

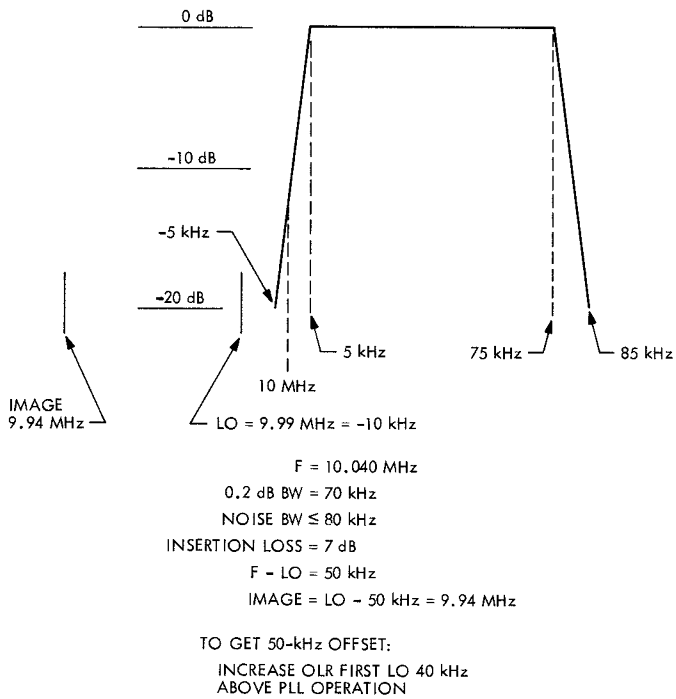


Fig. 3. Open-loop receiver bandpass filter

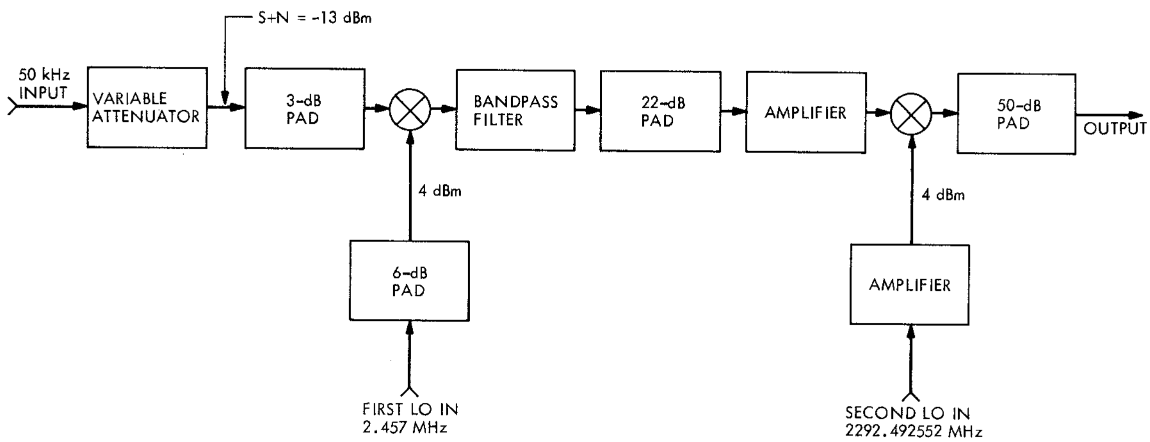


Fig. 4. Upconverter block diagram

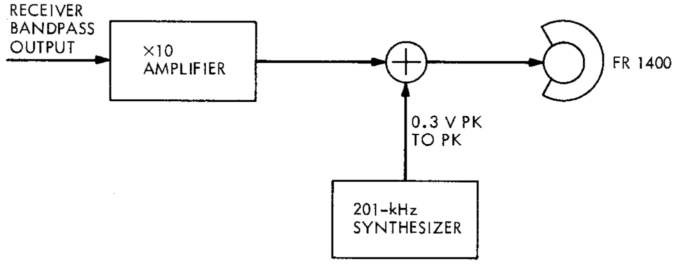


Fig. 5. Recording setup

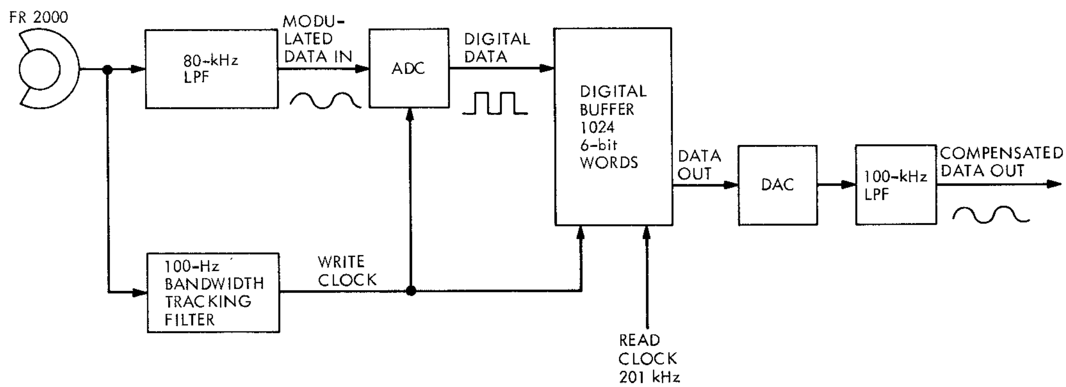


Fig. 6. Playback setup with TJC

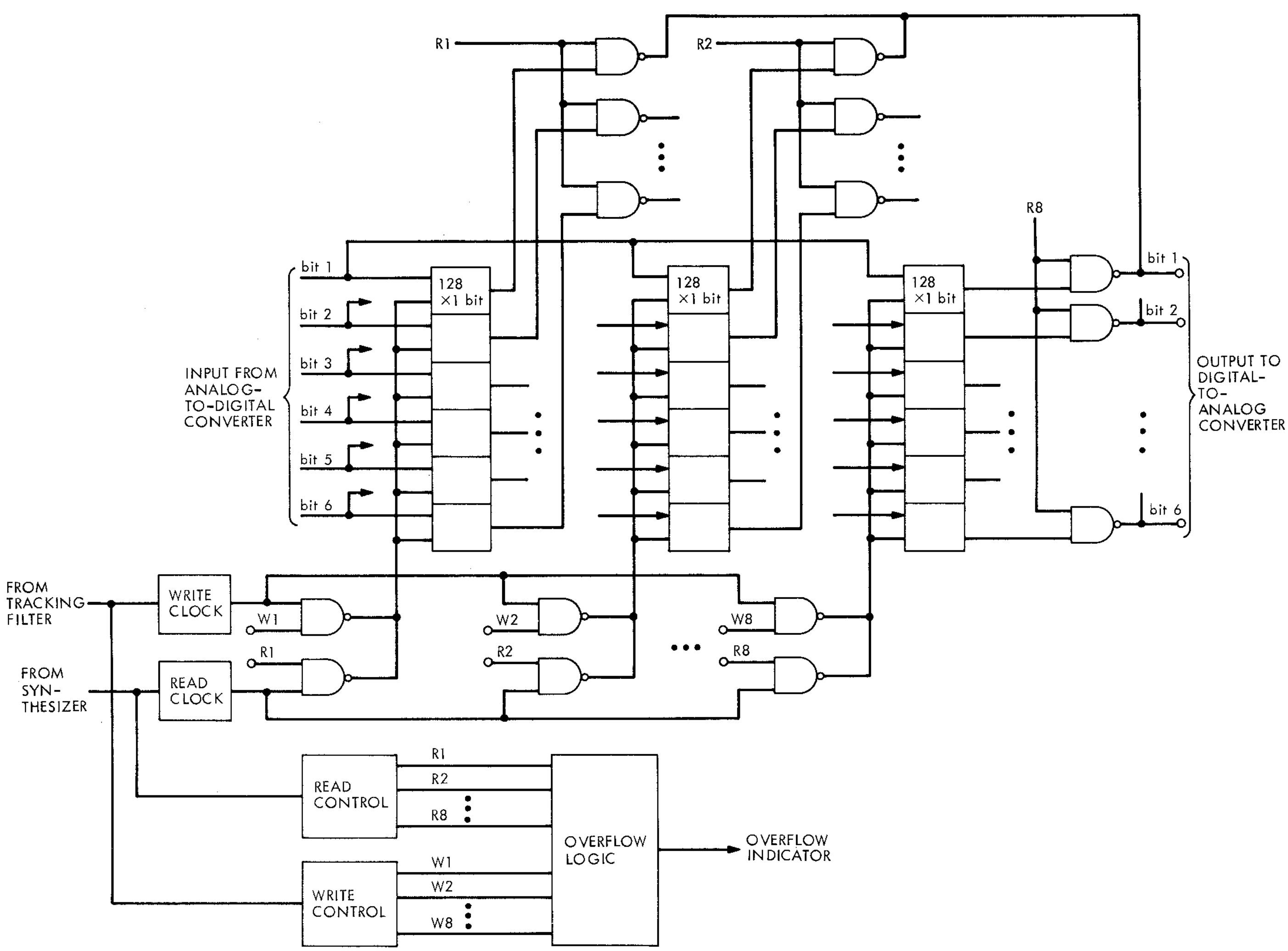


Fig. 7. Memory organization 1K × 6 bit array

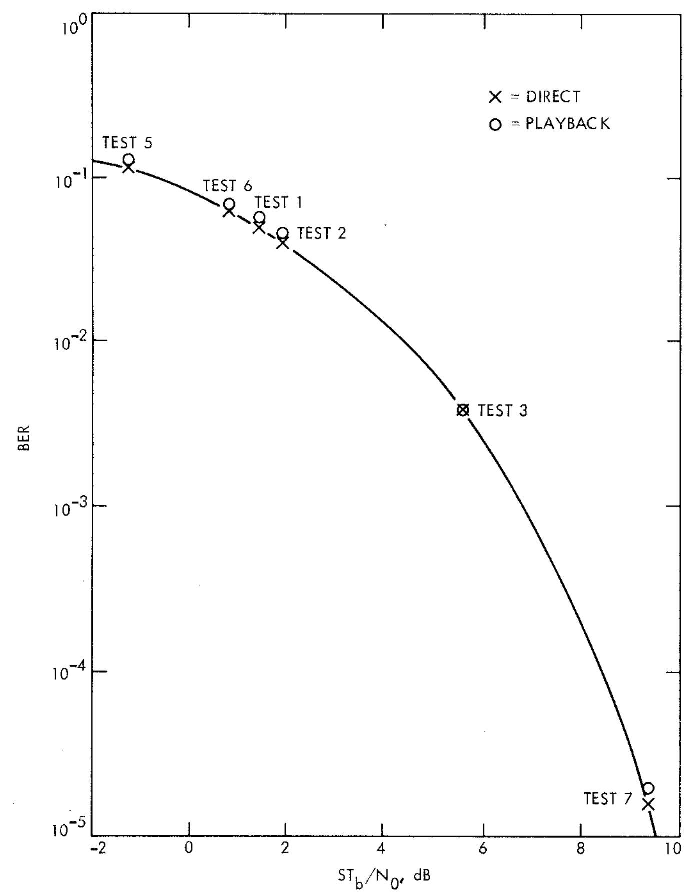


Fig. 8. Experimental results

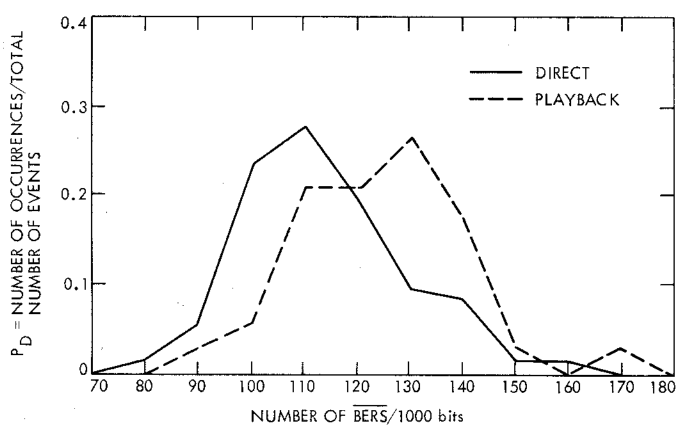


Fig. 9. Test 5 results (−1.35 dB)

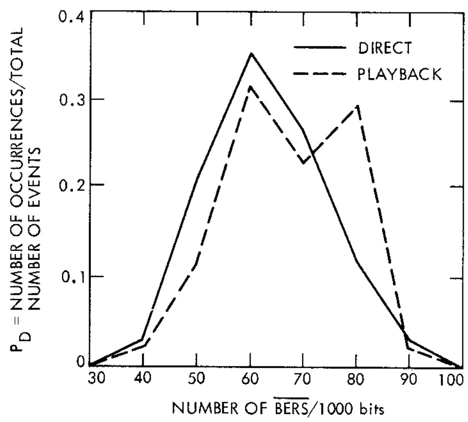


Fig. 10. Test 6 results (0.7 dB)

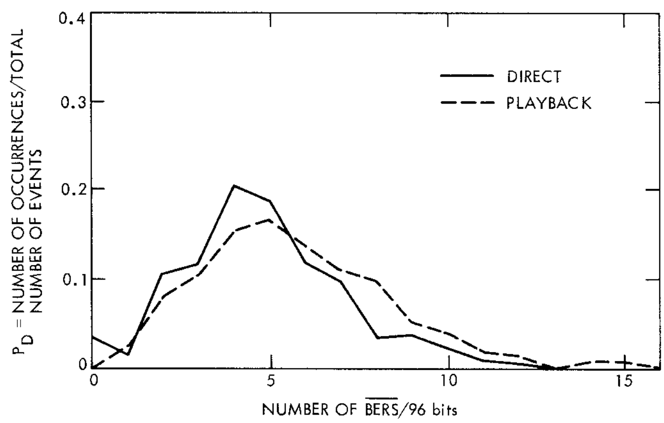


Fig. 11. Test 1 results (1.35 dB)

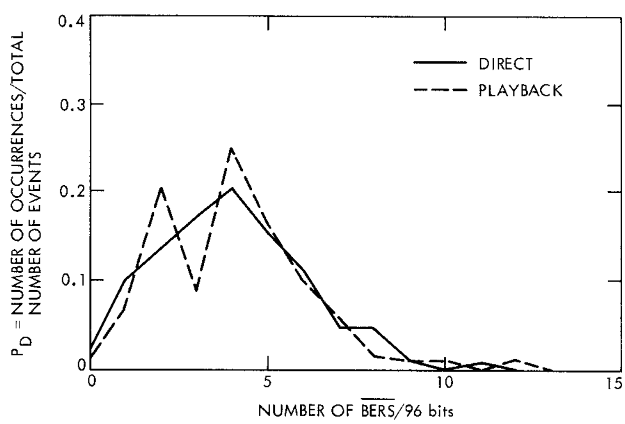


Fig. 12. Test 2 results (1.8 dB)

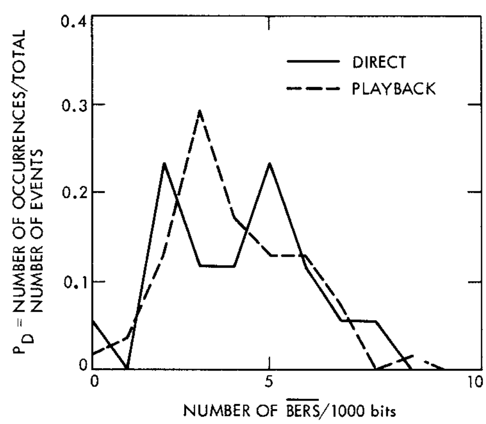


Fig. 13. Test 3 results (5.5 dB)

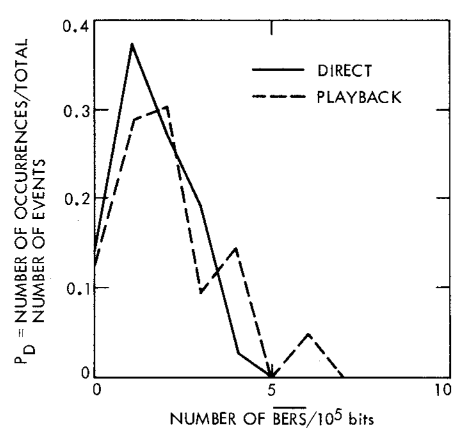


Fig. 14. Test 7 results (9.4 dB)