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ALABAMA AGRICULTURAL AND MECHANICAL UNIVERSITY
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PSEUDO NOISE TEST SET
OPERATION MANUAL

Technical Report
Contract NAS9-14770

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PSEUDONOISE TEST SET

OPERATION MANUAL

PSEUDONOISE TESTING THEORY

Until recently, the usual practice in end-to-end testing of communication systems has been to feed simple, deterministic signals into the system being tested and note the response. The response was then compared with a delayed and amplitude scaled version of the input signal to enable one to determine the distortion caused by the system and, thus, establish a system figure of merit. Unfortunately, most communication systems experience complex input signals and true simulations of system inputs are difficult to obtain. Even with the approximation of these complex signals (for instance, the sum of a number of independent sine waves), the deterministic approach has often proven unacceptable. Usually the number of component signals were inadequate to truly simulate the normal system input. As a result, performance was not evaluated. To overcome this deficiency, band-limited noise usually limited to the same operational bandwidth as the tested system, is now often used as a test signal and represents a very general approach to communication system performance evaluation. Yet, while band-limited noise is an excellent tool with which to simulate realistic data conditions, it has proven difficult to uniformly delay -- thereby making valid system input/output comparisons difficult to accomplish.

There are two basic ways to test a system utilizing band-limited noise as a test signal -- digital and analog. With the digital approach, band-limited noise is fed into a test system and the output monitored. The system input is sampled and digitized. These digitized samples are compared with digitized samples of the test system output yielding mean square values of the distortion introduced. Constant gain and average delay factors are compensated so that they do not contribute to the distrotion introduced by the system being tested since these factors are not usually considered as errors. There are two primary advantages of such a scheme: a complex test signal is used that is similar to the expected operational input spectrum, and all operations after sampling are digital. The primary disadvantages are that the effects of system spikes, such as FM "clicks," are often still missed.

In the analog system approach, band-limited noise is fed into a test system. The system output is compared on an rms basis with a delayed and scaled version of the system input. The obvious advantages are that the equipment is compact and relatively simple to build. The largest disadvantage is the difficulty in delaying the complex input without distortion

The pseudonoise (PN) test set was designed to provide a reliable, operationally simple unit which allows the previously mentioned analog rms end-to-end error measurement of most communication systems to be easily performed. It also provides a band-limited pseudorandom noise as input, thereby escaping the disadvantages of most deterministic signals while retaining the advantages of using true-band-limited noise. In addition, the PN test set has the capability to be used as a means with which the autocorrelation function and impulse response may be determined for certain systems.

Correctly defined noise signals give a good approximation of the data signals handled by communication systems. The noise signals must be stationary to assure that the measurements are repeatable. Noise can be defined by two unrelated parameters, power spectral density and the amplitude probability density function.

Power spectral density (PSD) describes how the energy in a noise signal is distributed in frequency. Generally, the noise power spectral density, as a function of frequency, can take almost any shape; however, the most common specification is that the power spectral density be constant with frequency. This describes "white" noise.

Probability density function (PDF) of a noise signal is defined as that function which, when integrated over an interval (X_1, X_2), gives the probability that the noise amplitude, X_n, lies in the interval (X_1, X_2). Specifically,

$$P(X_1 \leq X_n \leq X_2) = \int_{X_1}^{X_2} p(x)dx \ , \qquad (1)$$

and $p(x)$ is the amplitude probability density function, PDF.

True noise data signals are difficult to delay without introducing appreciable distortion. This problem is eliminated by using pseudorandom noise signals. PN signals that are produced by the test set are, in fact, not random but are completely definitive and repetitive. These PN signals closely resemble true random noise signals from both the power spectral and probability density function aspects.

A pseudorandom binary sequence can be generated by a shift register with an input dependent on feedback from certain register stages. The maximal length (ML) PN binary sequence is of length L and contains $2^n - 1$ bits before repetition of the sequence (where n is the number of register stages).

An important characteristic of pseudorandom sequences is the correlation property. If a binary sequence is pseudorandom, the correlation function of the sequence, and the sequence shifted with respect to itself, is constant for all shifts other than shifts of an integer number of sequence periods. Specifically, if vector A is a sequence $(a_1, a_2, \ldots, a_n)$, and B is a shifted version of this same sequence (with j shifts), then the correlation function (R) is:

$$R(j) = -\frac{1}{L} \sum_{i=1}^{L} (a_i \oplus b_i) \qquad (2)$$

where the $\oplus$ indicates modulo-2 addition. If the sequence is pseudorandom,

$$R(j = 0) = 1,$$
$$R(0 < j < L) = 1/L ,$$
$$R(j = L) = 1 \text{ where } L = 2^n - 1 . \tag{3}$$

This correlation property of a pseudorandom sequence is particularly important because it verifies that the signal generation approximates a random process.

The power spectrum envelope of a PN sequence is a $(\sin X/X)^2$ curve with the first zero occurring at the sequence clock frequency (w_k). If the PN sequence is lowpass filtered, with the cutoff frequency (w_0) restrained to $w_0 < w_k/n$, then the PDF of the resultant analog signal can be approximately Gaussian. However, a situtation will be described where the probability density function of the filtered sequence may not be Gaussian.

Noise signals generated by lowpass filtering of ML PN sequences are partially described by the cutoff frequency and order of the filter. The spectrum amplitude is approximately flat for $w < w_0$, the envelope is down 3 decibels (dB). At frequencies greater than w_0, the envelope rolls off at a rate of 6m dB/octave (where m is the data filter order). The spectrum is not continuous but consist of closely spaced spectral lines $\Delta w = \left[w_k \Big/ \left(2^n - 1\right)\right]$.

Figure 1 is a functional block diagram of the PN test arrangement. The filtered PN sequence is represented as a sum of sinusoids,

$$\sum_{i=1}^{\infty} c_i \cos (w_i t).$$

The filtered sequence is passed through the system with transfer function H(s), and the system output is $\sum_{i=1}^{\infty} A_i \cos (w_i t + \theta_i)$.

A scaled and delayed version of the original analog signal is

$$\sum_{i=1}^{\infty} B_i \cos (w_i t + \tau w_i). \tag{6}$$

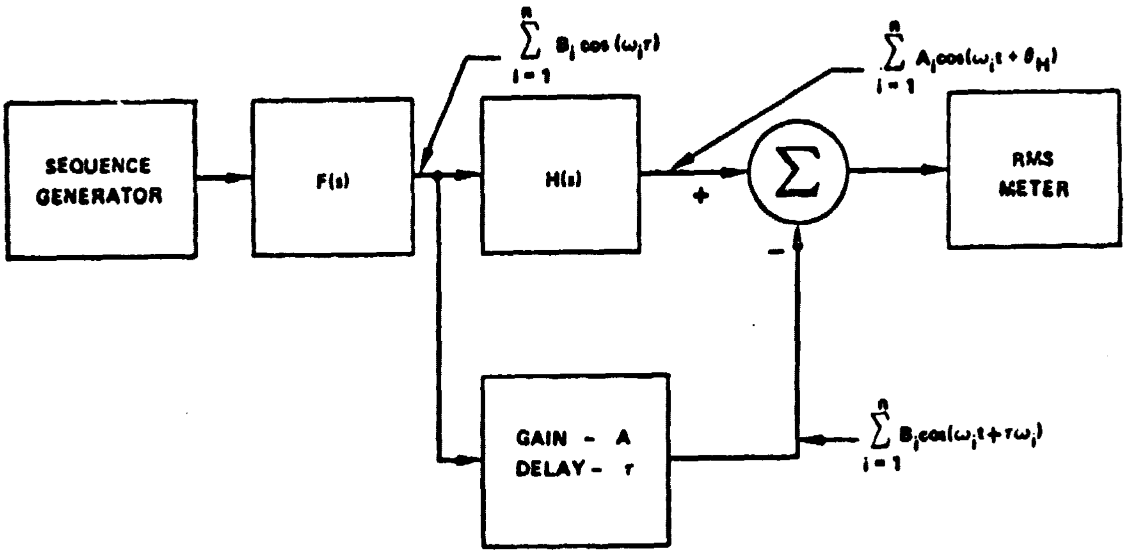


Figure 1. Block diagram of PN test arrangement.

The difference between these two complex waveforms is evaluated on an rms basis by the PN test set to give a figure of merit relating to the system's ability to transfer the complex pseudonoise data signals.

Basically, the test set consists of a pseudorandom binary sequence generator, sequence delay circuitry, spectrum shaping filters, signal conditioning amplifiers, a difference amplifier, and a system clock. Figure 2 is a simplified block diagram of the PN test set.

Clock. The system clock is a free-running multivibrator. The clock output consists of a sequence of consecutive binary one's and zero's to the PN sequence generator. The one state is +5.0 Vdc and the zero state is ground level.

PN Sequence Generator. The sequence generator is a ML, 23 stage shift-register.

Delay Circuitry. The delay circuitry (channel) consists of a NAND gate matrix for coarse delay selection and clock vernier for fine delay control.

Spectrum Shaping Filters. Are provided to give data of various bandwidths and characteristics for system testing.

Conditioning Amplifier. Channels 1 and 2 conditioning amplifiers have gain and dc offset potentiometers provided for control of the amplifier outputs.

Difference Amplifier. The difference amplifier presents a high impedance to the differential inputs, and has a low output impedance.

Installation. The PN test set chassis is designed for mounting in a standard equipment enclosure. A power cable is provided for connection to a 115/125 Vac power source. All system interconnects are on the front panel.

REAL-TIME RMS ERROR MEASUREMENT

Measurement of real-time rms error due to waveform distortion in a communication system can be accomplished by interconnecting the system under test and the test set as shown in figure 3. The operator then adjusts the test set gain control, and delay control to compensate for gain and delay differences in the reference and system signals.

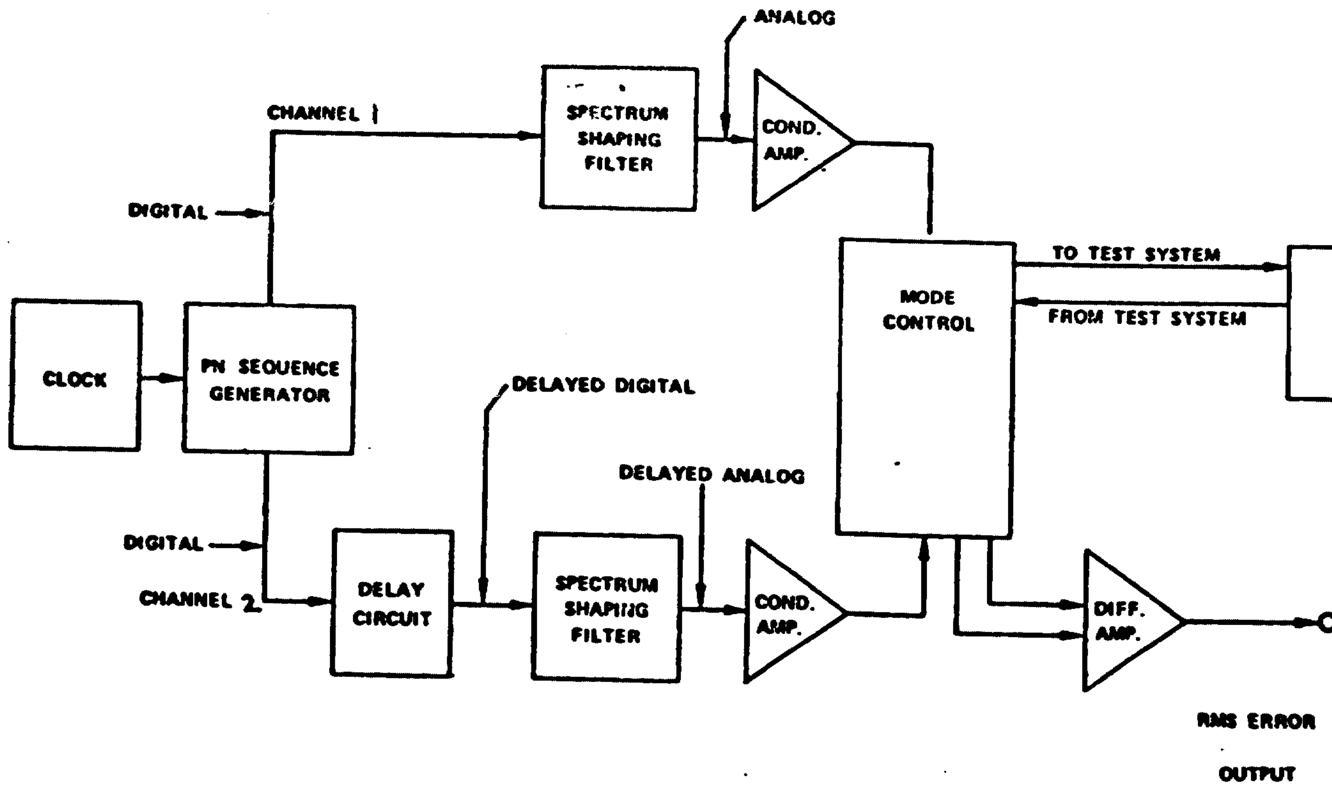


Figure 2. Performance Evaluator Block Diagram.

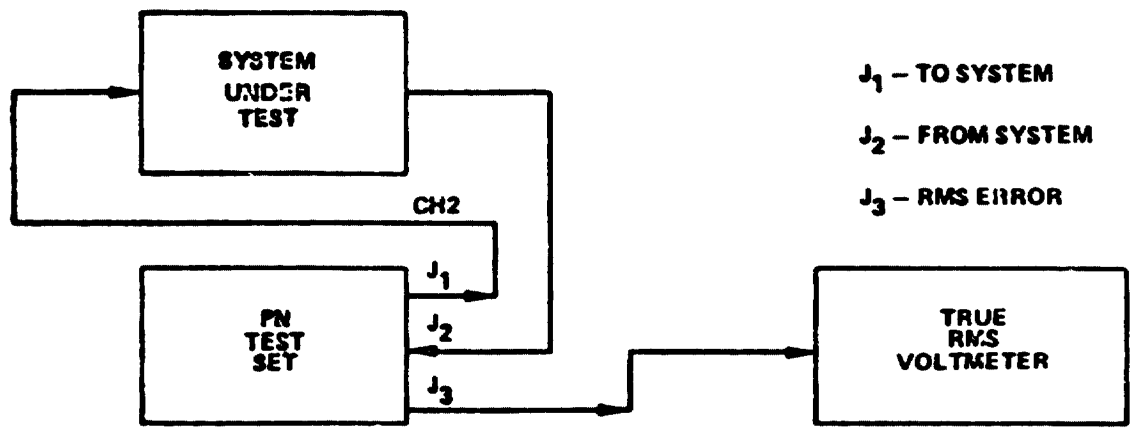


Figure 3. Test setup for real-time rms error measurement.

The theory of this measurement concept is based on the definition of rms error:

$$e_{rms} = \left(\overline{e^2\ (t)}\right)^{1/2} . \tag{7}$$

True-readings rms voltage measuring devices give an indication proportional to an average power absorption over a period T_i, the instrument time constant. Specifically,

$$e_{rms}\ (t) = \sqrt{\frac{1}{T_i} \int_0^t e^2\ (\tau)\ e^{-k\ (t\ -\ \tau)}\ d\tau} . \tag{8}$$

If the approximation is made that

$$\varepsilon^{-k(t\ -\ \tau)} = 1,\ t - \tau < T_i \tag{9}$$

$$\varepsilon^{-k(t\ -\ \tau)} = 0,\ t - \tau > T_i\ , \tag{10}$$

then

$$e_{rms}(t) = \left\{\overline{\left(e^2\ (t)\right)_{T_i}}\right\}^{1/2} . \tag{11}$$

Figure 4 is a block diagram of the system internal to the PN test set. For the case shown in the figure, the true-reading rms voltmeter indicates

$$e_{rms}\ (t) = \left\{\overline{\left(e_1\ (t)\ -\ e_2\ (t\ -\ \tau)\right)_{T_i}^{\ 2}}\right\}^{1/2} \tag{12}$$

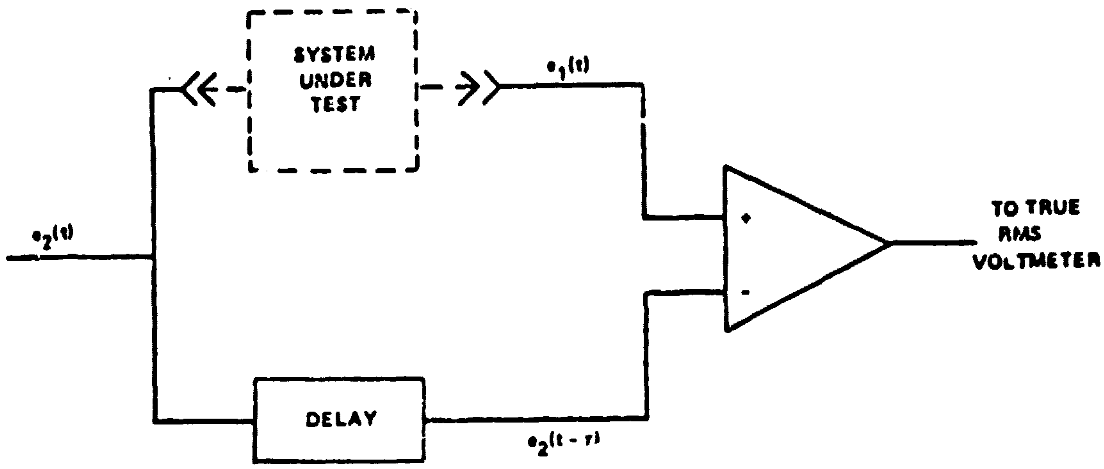


Figure 4. Error detection section of the test set.

or

$$e_{rms}^2(t) = \overline{\left(e_1^2\ (t)\right)}_{T_i} + \overline{\left(e_2^2\ (t - \tau)\right)}_{T_i} - 2\overline{\left(e_1\ (t)\ e_2\ (t - \tau)\right)}_{T_i} \quad . \tag{13}$$

Time Delay Measurement. Average time delay in a communication system will measured using the arrangement of figure 5.

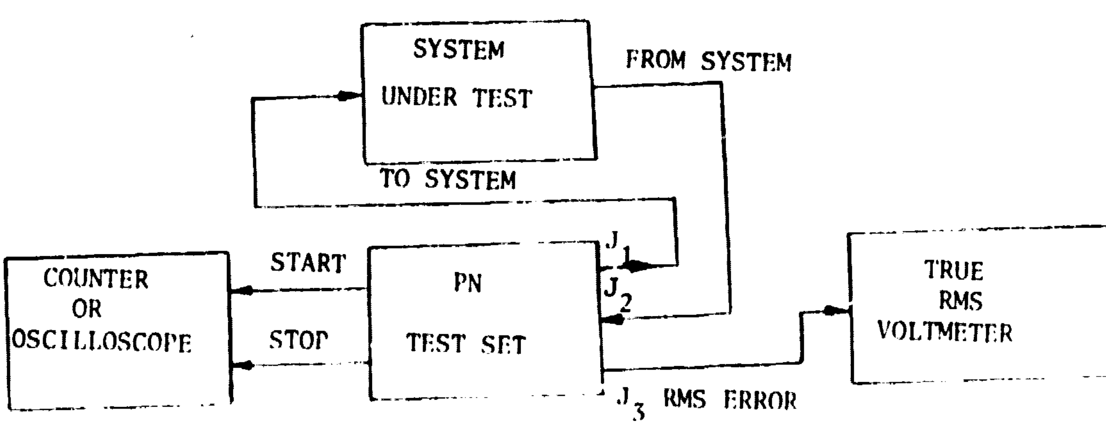


Figure 5. Test setup for time delay measurement.

The measurement procedure is to minimize the RMS error by adjusting reference channel gain and time delay. Start-stop pulses, separated by a period of time corresponding to the time delay introduced in the reference channel, will provide a trigger to a counter or oscilloscope for the average delay determination.

Correlation Function Determination. Determination of the correlation function of a communication system is accomplished with the instrument by the following procedure:

The correlation function for periodic waveforms is by definition

$$R_{e_2 e_1}(\tau) = \frac{1}{T_o} \int_{-T_o/2}^{T_o/2} e_1(t)\, e_2(t-\tau)\, d\tau \quad , \tag{14}$$

where T_o is the basic period of the waveforms.

The rms error as measured by the instrument will be

$$e_{rms}^2 = \overline{\left(e_1^2(t)\right)}_{T_o} + \overline{\left(e_2^2(t-\tau)\right)}_{T_o} - 2R_{e_2 e_1}(\tau) \quad , \tag{15}$$

and

$$R_{e_2 e_1}(\tau) = 1/2 \left\{ e_{1rms}^2 + e_{2rms}^2 - e_{rms}^2 \right\} . \tag{16}$$

From the preceding equation, the correlation maximum occurs at the minimum rms error and by measuring system input, system output, and minimum rms error, the correlation of the system input and output is obtained.

<u>Impulse Response Determination</u>. Determination of a communication system impulse response is accomplished by the following procedure:

1. Determine the system correlation function in accordance with the previously described method.

2. Measure the system time delay (τ_{min}) in accordance with the previously described method.

3. For values of delay different from τ_{min}, measure $Re_2\, e_1\, (\tau)$ and obtain impulse response from equation (20).

Theory: Determine impulse response for the system with input, $e_i\, (t)$, and output $e_o\, (t)$.

1. The system correlation function is

$$Re_i\, e_o\, (\tau) = \int_{-\infty}^{\infty} e_i\, (t)\, e_o\, (t - \tau)\, dt \tag{17}$$

also

$$Re_i\, e_o\, (\tau) = \int_{-\infty}^{\infty} Re_i\, e_i\, (t)\, h\, (t + \tau)\, dt \tag{18}$$

where $h(t + \tau)$ is Green's function.

2. If the input is white noise, or looks like white noise to the system under test, then

$$Re_i\, e_i\, (t) = k\delta\, (0)\ , \tag{19}$$

and

$$Re_i\, c_o\, (\tau) = kh\, (\tau)\ . \tag{20}$$

Therefore, assuming the system under test is linear, the cross-correlation function is proportional to the system impulse response.

ASPECTS OF THE SYSTEM DESIGN

Two technical problems were of particular importance to the design of the system evaluator. They were the extreme measurement accuracies required to tune the data filters to a mismatch of less than 1% and a problem with skewing of the amplitude distribution of the pseudoanalog noise signal for large ratios of sequence clock frequency to data filter cutoff frequency.

For the purpose of developing an improved design procedure for matched data filters for pseudonoise testing, a filter component sensitivity analysis was performed. The analysis covered passive Butterworth and Chebyshev approximation type filters. The result was a mathematical procedure for isolating the error sensitivity of a component on comparison basis and for standardizing the tuning procedure.

Another problem of importance concerned the skewing of the graphical representation of the probability density of a filtered pseudorandom digital sequence. Ideally, the distribution should approximate a normal curve as illustrated in figure 6. It has been observed that for ratios of digital clock to filter cut-off frequency, that are large compared to the number of stages in the sequence generator, a skewing of the distribution occurs as illustrated in figure 7.

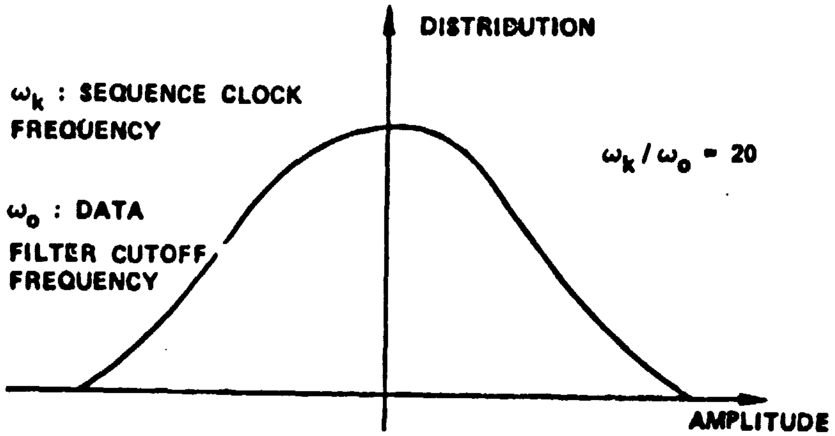


Figure 6. Pseudorandom approximation to the normal distribution.

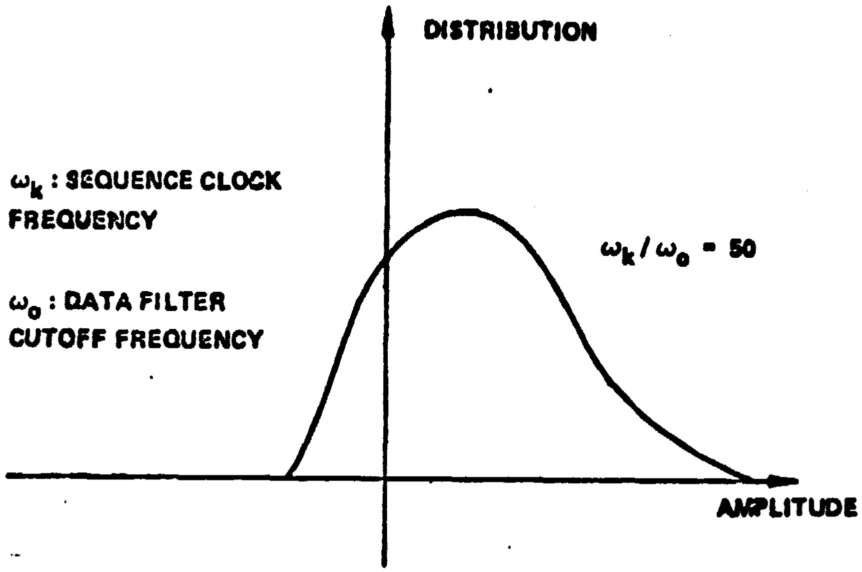


Figure 7. Skewed density function for large ratios of sequence clock to data cutoff frequency.

An investigation of the causes of this skewing was undertaken in an effort to correct the problem. A literature search revealed that R.P. Gilson observed the phenomena in question and reported the results of an experimental program concerning filtered maximum length sequences. Gilson's report confirmed that the skewing problem was basic to filtered ML sequences, and not just a property of the particular ML generator in question. James H. Lindholm reported the results of an analysis of the statistics of M-tuples from ML sequences. If the impulse response of the data filter is approximated as a weighted sum of the last M-bits from the sequence generator, Lindholm's results can be applied to the problem in question. Lindholm showed that the third moment of the distribution of weights of M-tuples from a maximum length sequence is related to the number of trinomials of order M-1 or less that contain the sequence characteristic equation as a factor. In the case of the PN test set, the M-tuple length corresponds to the ratio of sequence clock to data filter cutoff frequency.

An investigation was made to determine if there are any sequence generators with approximately 20 stages that would have a characteristic polynomial that is a factor of few trinomials of order 50 or less. It was determined that the 23-stage sequence generator with feedback from stages 23, 17, 11, and 5 has such a characteristic polynomial. A computer simulation of the filtered sequence, with a filter impulse response period of 60 clock periods, was performed. Figure 8 is the result of this simulation. The solid curve is the approximated normal distribution and the ● marks give the weights from the simulation. The approximation is satisfactory for the purposes of pseudonoise testing, and this sequence generator was used in the PN test set.

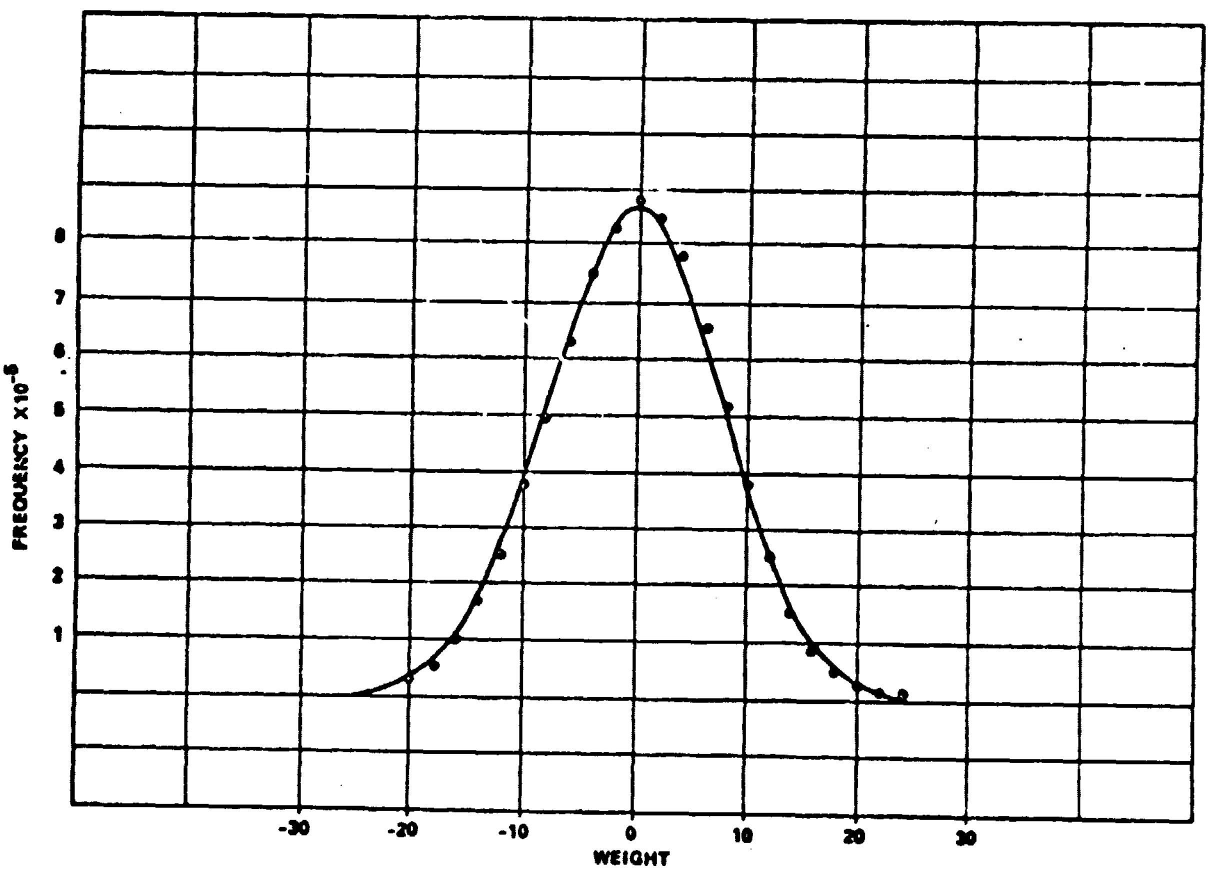


Figure 8. Normal probability and observed weights.

The rms distortion measured by the PN test set can be related to several other parameters that reflect communication system quality for particular types of applications and data. This includes voice quality, digital link bit-error-rate (uncoded), and television picture quality. In the case of the digital link, the relationship of bit-error-rate to rms distortion is straightforward if the signal is corrupted only by additive Gaussian noise.

The arrangement for real-time rms distortion analysis is shown below along with the channel under test (digital link with additive noise).

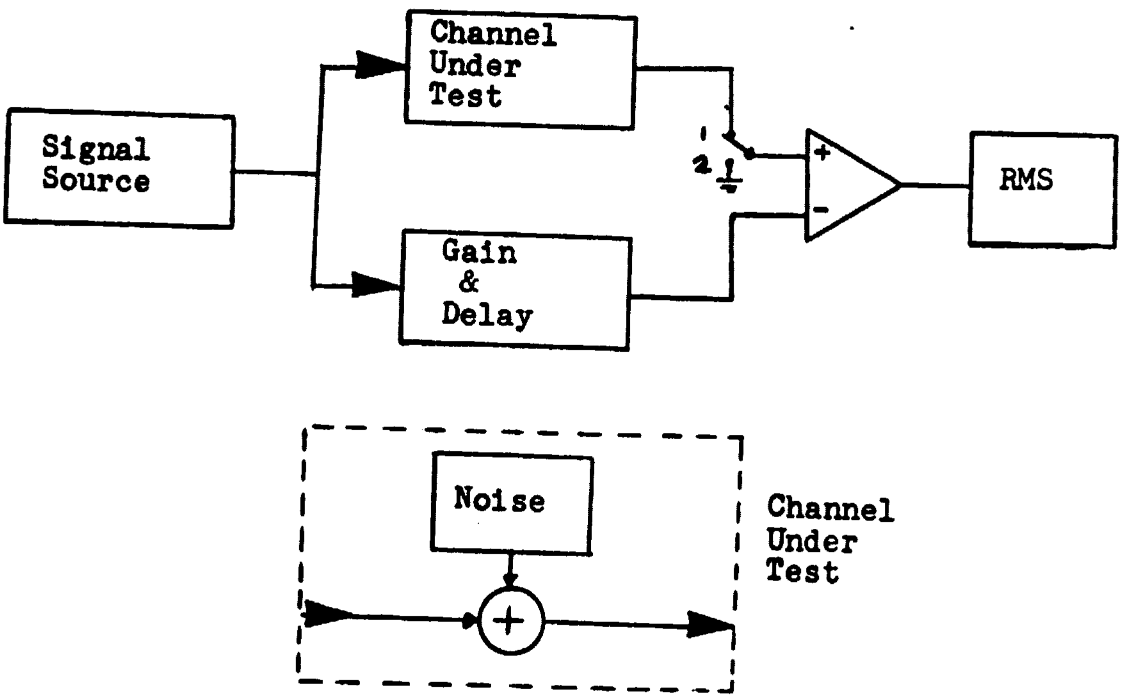


Figure 9. Digital Link Test Arrangement.

The signal e_1 is measured with the switch in position 1 with gain and delay adjusted for minimum rms error, and e_2 is measured with the switch in position 2. The normalized rms distortion is

$$\varepsilon = e_1/e_2 = \sqrt{\frac{1}{S/N}}$$

where S/N is the signal-to-noise ratio of the link. As an example, if the binary system using the channel was PSK modulated with coherent detection, the probability of a bit error as a function of rms distortion is

$$P_e = \frac{1}{2} \ \mathrm{erfc}(1/\varepsilon)$$

This is shown plotted below.

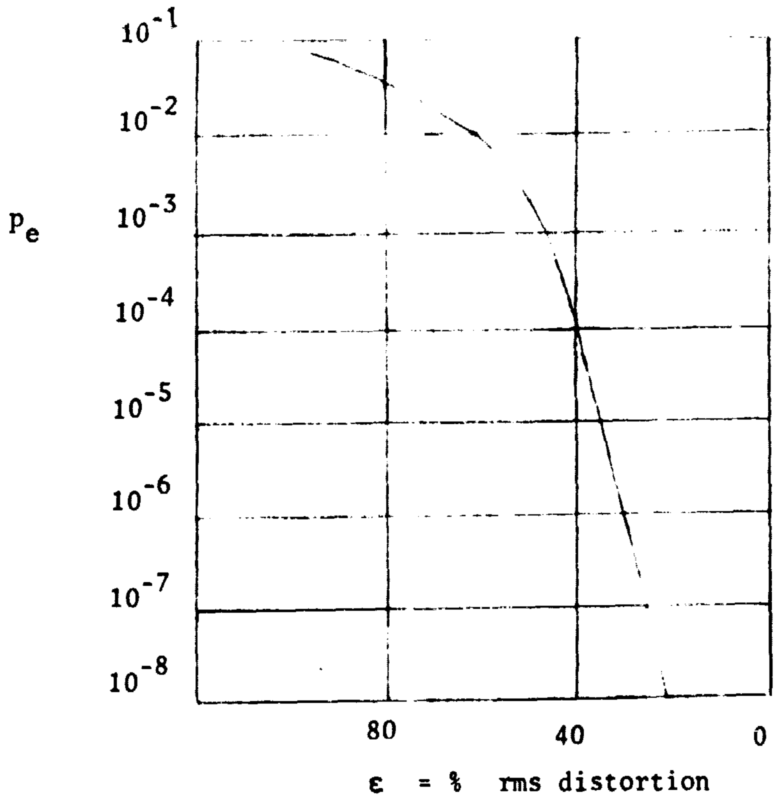


Figure 10. Error Probability as a Function of RMS Distortion. Noise Corrupted PSK Channel with Coherent Detection.

The technique for rms distortion measurement used in the PN test set requires near identical filters in the data and reference channels. Any difference in the transfer function of these two filters causes error in rms distortion measurement.

The filter-set tuning procedure yields filters with rms mismatch between .1% and 1% depending on filter type and order. The resulting instrument measurement error as a function of the rms error of the system under test is shown in the table below:

SYSTEM ERROR	MEASUREMENT ERROR .1% FILTERS	MEASUREMENT ERROR 1% FILTERS
1%	.5%	41.4%
2%	.125%	11.8%
3%	.05%	5.4%
4%	.03%	3.1%
5%	.02%	2.0%
6%	.01%	1.4%

The table shows that .1% filter match is an overdesign since the measurement uncertainty is much below acceptable levels. In fact, for testing most communications systems, (system error $\geq$ 2%) a 1% filter set mismatch gives acceptable results.

PSEUDONOISE TEST SET DESCRIPTION

The PN test set generates pseudorandom signals by filtering a pseudorandom digital sequence as shown in figure 11. Two channels are formed, a data channel which provides a signal to the system under test, and a reference channel as shown in figure 12. The complete rms distortion measurement system is shown in figure 13.

A block diagram showing all components of the PN test set is shown in figure 14.

The test-set has two modes of operation:

(1) Low-rate, 50 kHz clock

(2) High-rate, 20 MHz clock

The low-rate mode is used for the generation of data bandwidths up to 4 kHz. This includes the low-rate telemetry data and simulated voice data.

The high-rate mode is used for the generation of data bandwidths up to 4 MHz. This simulates a television video signal (high-rate science data).

The low-rate pseudorandom sequence generator includes 33-stages of digital delay with feedback from stages 23, 17, 11, and 5. The resulting sequence is 8388607 bits in length, and at a 50 kHz clock-rate repeats every 167.8 seconds. With a 100 Hz data filter, the ratio of clock-rate to data-bandwidth is 500. In order for the statistics of the 100 Hz bandwidth analog data signal to approximate those of true random signal, the characteristic polynomial of the sequence generator must be a factor (mod-2) of very few trinomials of order 500 or less. It has been shown that the (23, 17, 11, 5, 0) polynomial is a factor of no trinomials of order 500 or less, so this generating structure is ideal for the test-set application.

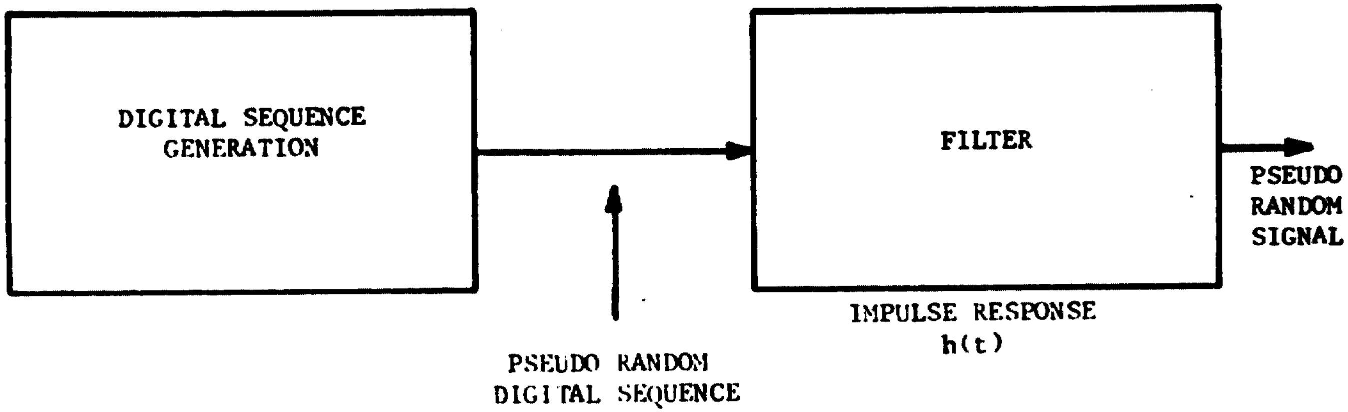


Figure 11. Pseudorandom Signal Generation

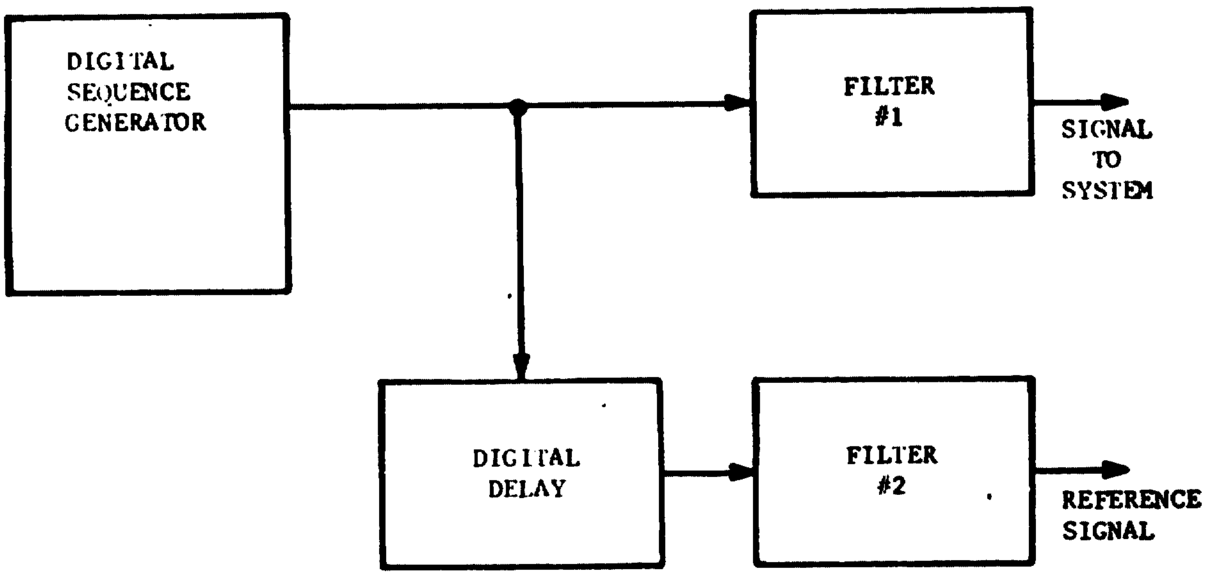


Figure 12. Data and Reference Signal Generation

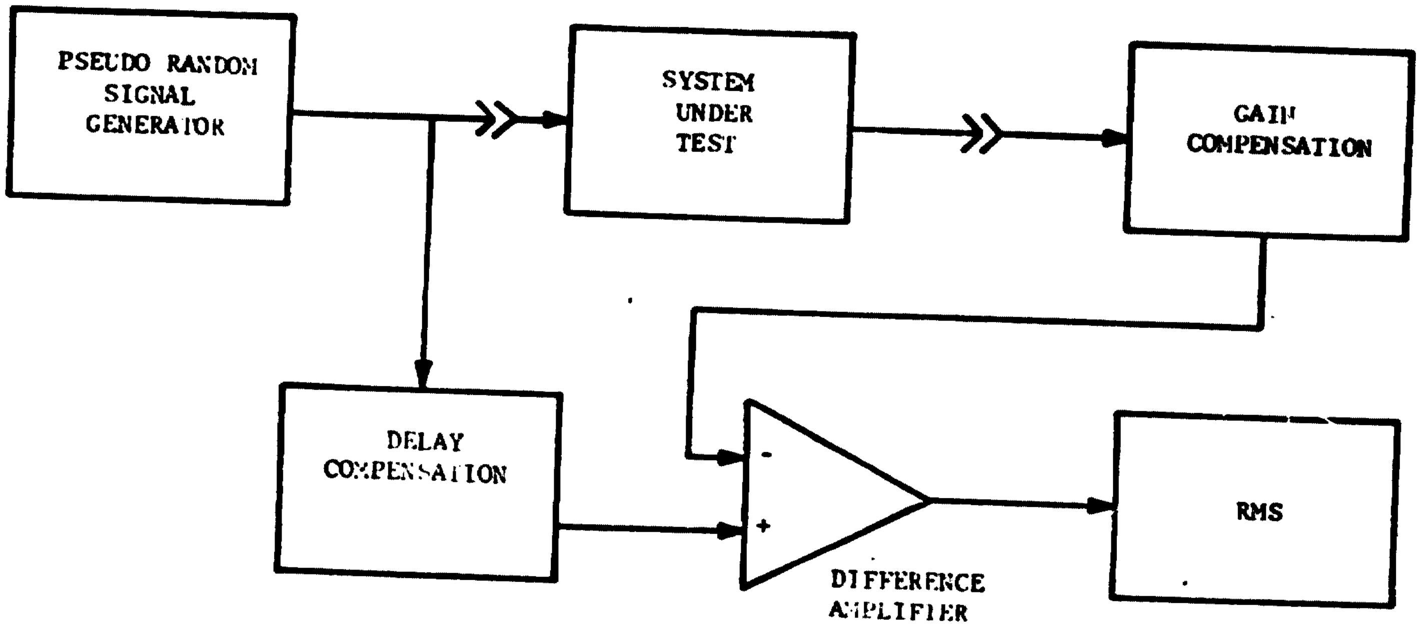


Figure 13. RMS Distortion Measurement

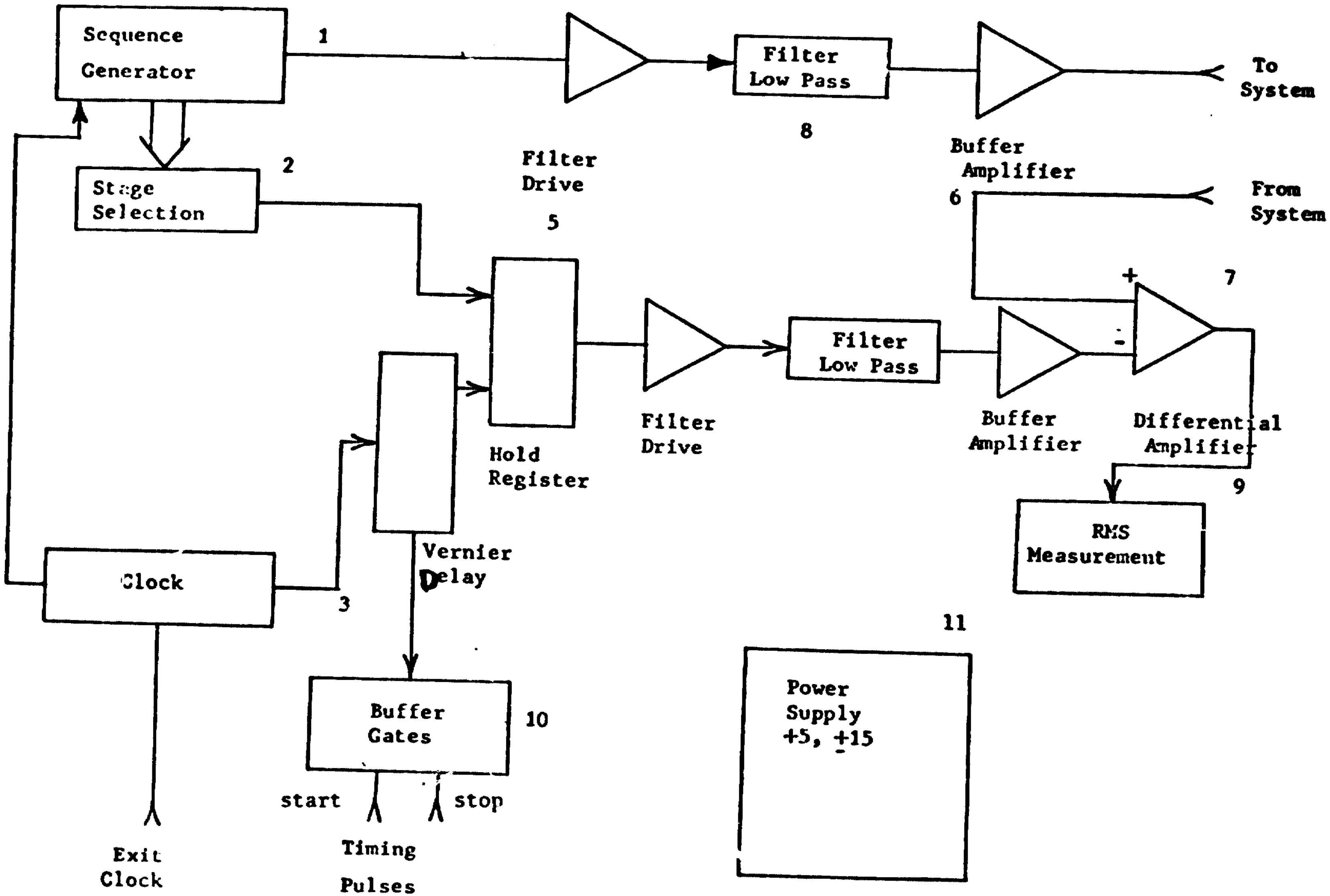
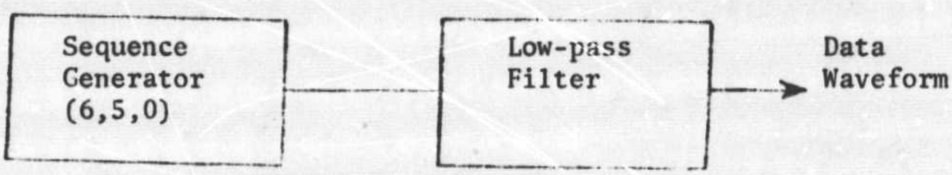


Figure 14 PM Test Set Block Diagram

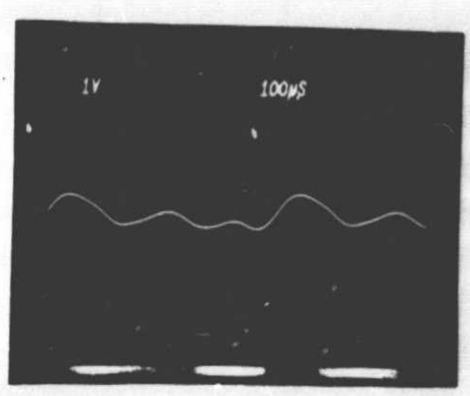
The low-rate mode includes continuously variable reference delay capability up to 640 μsec. (with the 50 kHz internal clock). This delay range will be achieved with 32 steps of 20 μsec. delay and ± 10 μsec. vernier delay adjustment. The vernier delay will be achieved digitally, with no clock frequency adjustment required.

The high-rate pseudorandom sequence generator includes 33 stages of digital delay with feedback from stages 6 and 5. The resulting sequence is 63 bits in length, and at the 20 MHz clock-rate repeats every 3.15 μsec. The ratio of clock-rate to data-filter bandwidth varies from 5 (4MHz data) to 20 (1 MHz data).

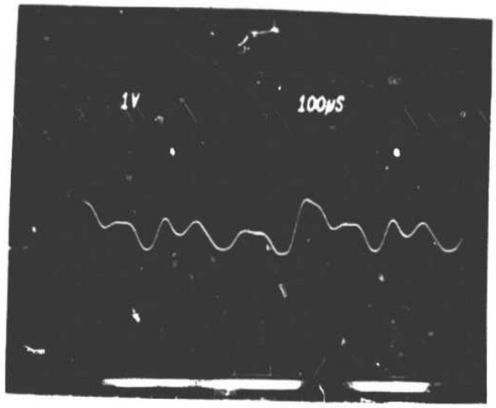
An experiment was performed to indicate the waveforms that could be expected in the high-rate mode. The block diagram below shows the method of waveform generation.



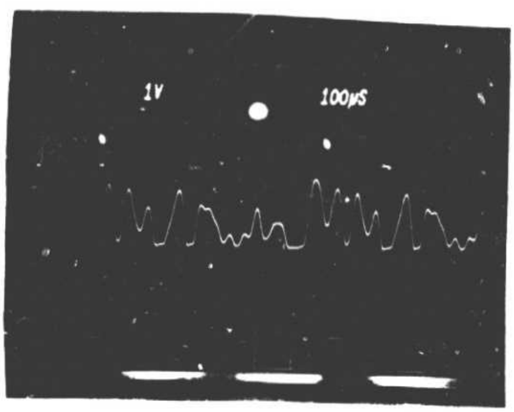
The experiment was run at a sequence clock-rate of 100 kHz, but the ratios of clock-rate to data-bandwidth were set equal to the values designed into the high-rate mode of the test-set. The results are shown below:



Data-ratio = 20
Simulates 1 MHz data



Data-ratio = 10
Simulates 2 MHz data



Data-ratio = 5
Simulates 4 MHz data

These waveforms, scaled to the high-rate frequency, are the test waveforms and are intended to simulate the high-rate science data.

PSEUDONOISE TEST SET OPERATION

Figure 15 shows the pseudo-noise test set front panel. The front panel controls are divided into two sections.

(1) Data Channel Controls

(2) Reference Channel Controls

In addition to the controls, BNC connectors for interfacing the instrument to the system under test and to other test equipment are provided on the front panel.

The operator of the test set works from left-to-right in making a measurement. The sequence of operation is as follows:

DATA CHANNEL

(1) POWER -- Master switch for test set power supplies

(2) CLOCK -- Can be switched to INT for internal clock, or EXT for external clock. In the EXT position, clock signals must be provided via the EXT clock connector. The PN test set includes a circuit to monitor the clock bus and give a positive indication of a proper clock on the bus. This indicator is a green LED adjacent to CLOCK. This is intended for convenience in the external clock mode and gives an indication of proper clock drive.

(3) DATA -- FILTER The test set has space in the filter bay for four filter pairs. One of these filter pairs is selected to operate on the data signal by this switch. Below and to the right of this switch are four plates mounted on the front panel. On each of these plates is printed a description of the particular filters in the test set, and the LED adjacent to the place indicates which has been selected. The data filter switch also selects the test set rate mode. Positions 1 and 2 set the low-rate mode, and positions 3 and 4 set the high-rate mode.

(4) GAIN DC OFFSET-- The amplitude and DC offset of the data signal provided to the TO SYSTEM connector are controlled by these pots. These controls are provided for convenience in interfacing to a communications systems to be tested.

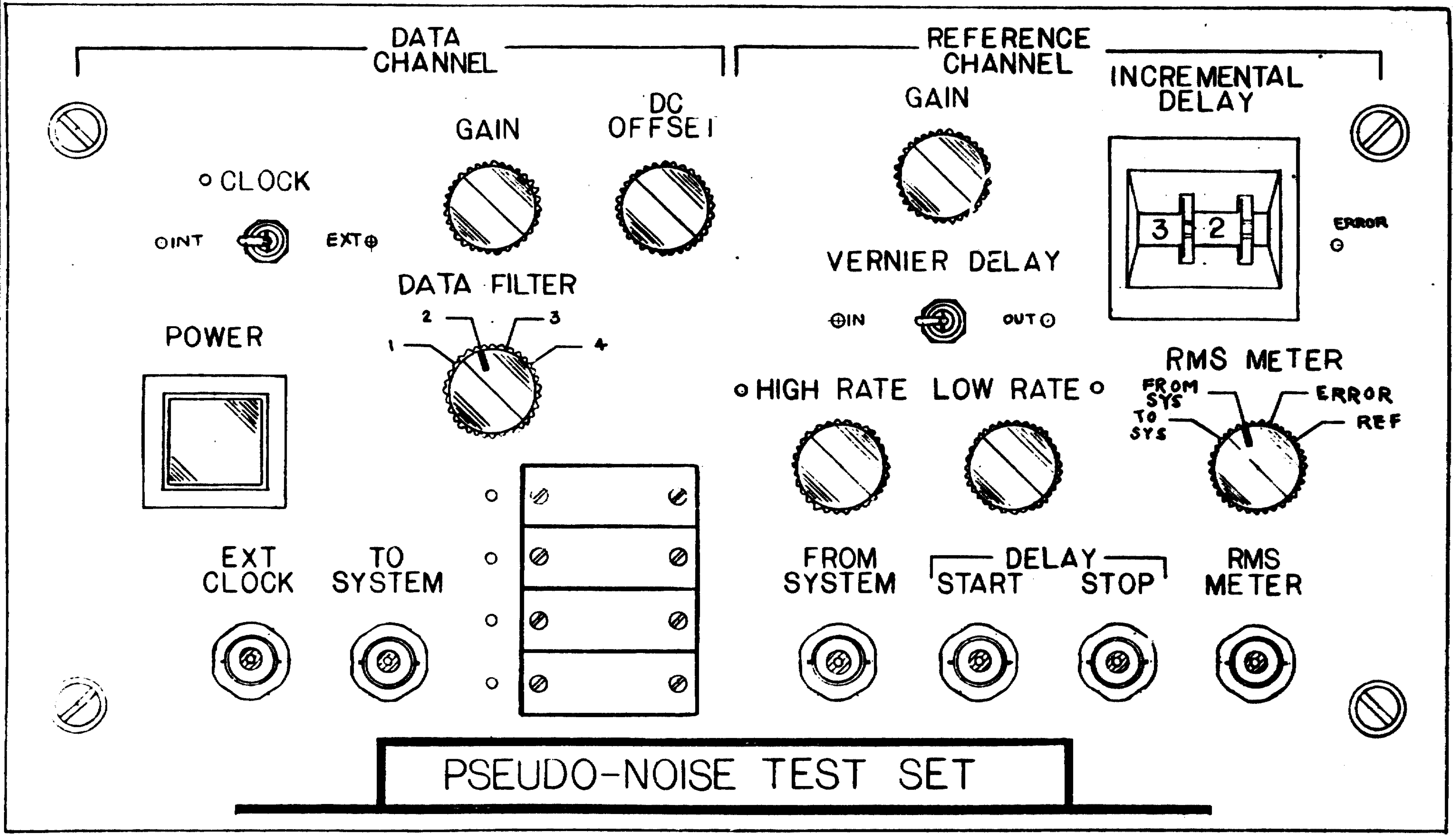


Figure 15. Pseudo-Noise Test Set Front Panel

REFERENCE CHANNEL

(5) GAIN	--	Controls reference channel amplitude, and is adjusted to minimize rms error.
(6) INCREMENTAL DELAY	--	Adjusts the reference channel delay in incremental steps to match the delay of the system under test and minimize rms error.
(7) ERROR	--	The test set incremental delay line has 33 taps (settings 00 to 32). The MSB of the incremental delay switch has a stop at 3, but the LSB has no stop. Improper settings of 33 through 39 are indicated by the red error LED.
(8) VERNIER DELAY	--	Switch fine delay control in or out with LED's indicating position of the switch. This switch is operational in the low-rate mode only.
(9) HIGH RATE LOW RATE	--	Pots to control the vernier delay in the high rate and low rate test set modes. An LED adjacent to each lable indicates the test set mode.
(10) RMS METER	--	Selects the signal patched to the true-rms voltmeter. With the switch in the ERROR position, reference channel delay and gain are adjusted to minimize rms error. Once minimized, the rms error is measured, and normalized by the signal measured in the REF position. Also provided is the capability to switch the meter TO SYS (to the system under test) and FROM SYS (from the system under test).

CONNECTORS

(11) EXT CLOCK	--	External clock input
(12) TO SYSTEM	--	Data signal to the input of the communications system under test.
(13) FROM SYSTEM	--	Output of the communications system under test.
(14) DELAY START DELAY STOP	-	Provides start-stop pulses so that with a counter the exact value of vernier delay can be measured.
(15) RMS METER	--	To true rms voltmeter.

The rms error measurement is performed by monitoring the error signal on the rms voltmeter and adjusting the reference channel gain and delay to minimize the rms error. When the error has been minimized, read the error voltage V_E and reference voltage V_{REF}. The percent rms distortion is V_E/V_{REF} x 100%.

In the low rate mode, system delay is measured by using an interval counter to measure the vernier delay using the START and STOP timing pulses. The total time delay is

$$T_D = I\ (20) + T_V - 10\ \mu SEC,$$

where T_V is the vernier delay interval and I is the setting on the incremental delay switch.

In the high rate mode the true delay is

$$T_D = I/_{fc}$$

Where fc is the clock frequency. This frequency is adjustable, and may be measured with a counter from the STOP or START connectors.