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DIGITAL SEISMIC RECORDER SPECIFICATION STANDARDS[1]

SEG Subcommittee of the Technical Standards Committee on digital seismic recorder specifications

FOREWORD

To help those SEG members who use digital seismic field recorders to compare the performances of various instruments, the Technical Standards Committee proposed that a set of standards be established to define critical characteristics of digital seismic recorders. Tests and analysis procedures for determining these characteristics were also to be recommended.

The subcommittee initially identified a number of items for action and divided the effort among the members. Inputs from those assignments were received and revised, and, after several reviews by the Technical Standards Committee and representatives of the industry in all areas of interest, the standards were approved by the SEG Executive Committee.

Considering the total spread of performance requirements and the variety of apparatus used to meet these needs, it would be unusual if all users and manufacturers were to agree on all specification definitions and test procedures that are recommended in these standards. Our position is that these specifications are meaningful, realistic, and capable of measurement as described and can therefore be a common basis for comparison and evaluation.

I. COVERAGE

A. These standards shall be applied to apparatus used for acquiring and recording seismic data for geophysical exploration.

B. Excluded from consideration in these standards are sensors, cables, energy sources, and magnetic recording tape formats.

C. No conflict with existing codes and lawfully enforceable regulations is to be construed in the standards herewith established.

II. ANALOG PARAMETERS

A. Input Characteristics

1. Input Impedance

a. Differential-mode input impedance is defined as the complex ratio of ac voltage to ac current for a sinusoidal input to the apparatus at its input terminals. It can be conveniently measured by applying a voltage whose peak value is within the input signal maximum range and determining the magnitude and phase of current drawn by the input circuit.

(1) If impedance varies over the seismic frequency range band specified for the apparatus, curves of magnitude Vs frequency and phase Vs frequency shall be drawn for this parameter.

b. Common-mode input impedance is defined as the complex ratio of ac voltage to ac current for a sinusoidal input applied between inputs and ground of apparatus. Inputs shall be connected together and a source applied to the inputs and to the ground of the apparatus through a series resistor not to exceed 1% of the measured impedance. Peak value of voltage applied shall be within the common-mode maximum range value and the magnitude and phase of current drawn by the circuit shall be measured.

(1) If impedance varies over the seismic frequency range band specified for the apparatus, curves of magnitude Vs frequency and phase Vs frequency shall be drawn for this parameter.

2. Maximum Input Signal Definition

a. Differential input

(1) Maximum Linear Input The maximum linear differential signal input is defined as the smallest RMS sine wave of voltage at the input of the instrument whose frequency is within the pass band of the instrument and causes the distortion limit specified for the instrument to be exceeded. Value shall be stated for each selection of available preamplifier gain.

(2)Maximum Nonlinear, Short-time Recovery Input
Maximum nonlinear differential short-time recovery input is defined as the amplitude of the maximum differential input pulse of voltage 16 ms duration applied at the input of the instrument such that the instrument can recover to linear operation within 200 msec after the end of the voltage pulse. The value for both polarities shall be specified. Values shall be specified for low-cut filters out and for the worst-case combination of filters which shall be identified.

b. Common-mode Input

(1) Maximum Linear Common-mode Input
Maximum linear common-mode input is defined as the smallest magnitude positive or negative common-mode voltage that causes a sine wave differential signal of 7.0 MV RMS or less whose frequency is within the passband of the instrument and which exceeds the distortion limit specified for the instrument.

(2) Maximum Common-mode Input
Maximum common-mode input signal is defined as the maximum pulse

of voltage 16 ms duration, both positive and negative, applied to shorted differential input of the instrument and referenced to common instrument ground such that the instrument can recover to its normal, specified differential operation at end of the voltage pulse within a 200 ms time.

3. Common-mode Rejection

Common-mode rejection quality of an instrument is defined as the ratio, expressed in decibels of output voltage referred to the input, to the common-mode input voltage, when differential input is shorted and driven with a sine wave of voltage, referenced to common instrument ground, for frequencies within the seismic bandpass of the instrument. The input test signal ' s amplitude should be one-half the maximum linear common-mode input voltage specified for the instrument. The common-mode rejection shall be calculated as a decibel measurement of the ratio of output voltage referred to the input to the test voltage; i.e.,

Vout* = Output voltage referred to input (divided by differential gain);
Vin = Input test common-mode voltage;
CMR = Common-mode rejection ratio in dB;

and

$$CMR = 20 \, Log_{10} \frac{Vout^*}{Vin}$$

Curves of common-mode rejection shall be drawn showing CMR in dB versus frequency for each preamplifier gain.

4. Instrument Noise

a. Introduction

Specification of "instrument" noise poses a dilemma for both instrument manufacturer and user. The user is concerned, and rightfully so, with noise introduced by the overall data collection and recording system, i.e., from the sensor to the signal recorded on magnetic tape. The instrument manufacturer, however, assumes no responsibility for spread cables or connectors; in many instances he is not responsible for assembly and/or installation of the total recording system. The instrument manufacturer generally delivers a set of instruments to the user who then adds auxiliary equipment (often from a variety of different manufacturers) and installs the system, with connecting cables, in a recording cab. Furthermore, it is not unusual for a set of instruments to be moved from one installation to another several times during its useful life. Consequently, the instrument manufacturer is reluctant to guarantee a noise specification "from the input panel to the tape". Thus we describe the test parameters that apply to the recording instrumentation alone. The results of the tests can be used as a reference for overall system tests conducted similarly.

b. Noise Measurement/Specification

(1) System Conditions

(a) Unless otherwise specified, it is assumed that the noise measurement is performed under normal production conditions, except that the system inputs are terminated at a specified point with a specified source impedance.

(2) Environmental Conditions

(a) Temperature Range and Humidity
The temperature range (°C) and the relative humidity range over which the specified noise level is met shall be specified.

(3) Source Resistance (s)
A noise specification shall be given for a number of values of source resistance. Suggested values are:

(a) 500 ohms
Historically, this is a popular value. Thermal noise due to this source termination is small (approximately 3.0 nanovolt / √Hz at 50°C) compared to instrument noise.

(b) 2000 ohms
This is a value corresponding to relatively long cables that are common in current operations. Its thermal noise is approximately 6.0 nanovolts / √Hz at 50°C. Note that over a bandwidth of 280 Hz, this produces 0.1 microvolt which is comparable to many instrument specifications.

(c) 8,000 ohms
For establishing a third value of source resistance with uniform ratios of noise voltage, this value results in 12 nanovolts / √Hz at 50°C.

(4) Filter Settings
A noise specification shall be given for the system with each anti-alias filter supplied with or offered for the system.

(5) Preamplifier Gain
A noise specification shall be given for the system with each value of preamplifier gain.

(6) Auxiliary Channels
Auxiliary channels that are not identical to the seismic data channels require a separate noise specification.

c. Measurement Technique/Calculations

(1) Terminate with specified value of low noise type construction resistor, e.g., wire-wound or metal film at specified point.

(2) Produce a magnetic tape recording with a minimum of 1024 samples per channel.

(3) A suggested method of calculation is as follows for any one channel:

Xi = Amplitude of ith sample,
dc = dc offset of analog signal,
RMS = total RMS value,
AC = RMS value of signal with dc removed,
N = Number of samples,

$$dc = \frac{1}{N} \sum_{i=1}^{N} Xi$$

$$RMS = \left(\frac{1}{N} \sum_{i=1}^{N} Xi^2 \right)^{1/2}$$

and

$AC = (RMS^2 - dc^2)^{1/2}$

(4) All calculated values shall be referred to terminated input terminals.

(5) Typical curves of noise power spectral density shall be provided

d. Noise Specification

It appears that most instrument manufacturers are driven, presumably by competitive pressure, to publish noise specifications which are difficult to meet and which do not necessarily correspond to field operating situations. The noise specification problem is aggravated by the continuing trend to more recording channels.

This standard specifies that the manufacturer shall define an "Absolute Maximum" noise level; i.e., no channel would ever exceed this value under the specified conditions. A statistical specification: "XX% of the channels will be less than .YY μV" shall accompany the above value with XX being greater than 90.

e. DC Offset Specification

(1) DC Offset shall be specified in the same fashion as noise: "Absolute Maximum DC Offset" for the specified conditions and termination, accompanied by a statement "XX% of the channels will be less than ZZ μV " where XX is greater than 90.

(2) DC offset shall be specified for each value of premplifier gain.

B. Seismic Recorder Filters

1. Digital seismic recorders are normally constructed with a wide variety of filters. These filters are designed to fill the needs of a variety of users. However, only anti-alias filters are usually included in the signal path to prevent aliasing of high frequency signals as a result of the sampling process.

This standard, therefore, requires that the amplitude and phase response of the basic system with anti-alias filters only, if included, be specified completely. Other filters available with the system shall be specified separately. The user may then select the particular combination of filters which he desires to use, and arrive at the amplitude and phase characteristics which are applicable.

Although the amplitude response of a recording system and its filters is of interest, phase and time delay characteristics are of primary importance, because the first objective of the seismic process is measurement of travel times. Therefore, time delay characteristics of a system and its filters must be fully and accurately specified. Knowledge of distortions in phase and time produced by recording systems allows the user to derive more accurate travel time measurements.

The phase delays produced by recording filters, as well as other time delays through the signal path, produce time errors in the data which are recorded on tape. Reproduce (playback) filters and other time delays through the reproduce circuitry, do not introduce time errors in the recorded data, but the reproduce system does introduce time errors in signals displayed on camera monitor records.

Since camera monitors are frequently used to check the timing accuracy of data recorded on tape, it is essential that reproduce time delays be known. Likewise, phase and time delay characteristics of auxiliary channels must be specified, as these channels are frequently used to record timing reference signals such as time break, gun breaks, and pilot sweep signals.

This standard requires that certain specified quantities include a tolerance. This provides to the user a measure of the similarity between channels for a given recording system. Methods of measuring amplitude and phase response characteristics, both in the laboratory and in the field, are discussed in the last section of this standard.

2. Basic Definitions

a. Nominal, Maximum, and Minimum Values

A nominal value or quantity is defined here as the mean average of all channels, or as a design center. Filter cut-off frequencies are specified as nominal values.

A maximum quantity is defined as an absolute maximum value, which will not be exceeded by any channel of the system.

A minimum quantity is defined as an absolute minimum value, which will be exceeded by all channels of the system.

Maximum and minimum quantities must take into account the effects of component variations, and must be applicable to the entire temperature and humidity range specified for the system.

b. Cut-Off Frequency

The cut-off frequency of a filter is defined as that frequency at which the filter's amplitude response is equal to 0.707 times the filter ' s maximum pass-band response. The cut-off frequency may also be referred to as the "3 dB point". All filter cut-off frequencies are to be specified at the 3 dB point. Identical cascaded filters will have multiples of 3 dB attenuation at their cut-off frequency such as 6 dB for two sections, etc. As

used herein dB refers to $20 \text{ Log}_{10} \dfrac{V_{IN}}{V_{OUT}}$.

Filter - cut-off frequencies must always be specified as nominal values, to the nearest 0.1 Hz. A cut-off frequency plus-or-minus tolerance must also be specified, to indicate the maximum range of values which the cut-off frequency may have. This tolerance may be expressed either in Hertz or as a percent of the cut-off frequency. The cut-off frequency of all filter channels must lie between the tolerance limits over the full temperature and humidity range of the system.

c. Filter Order or Slope

The filter slope, in dB/octave, must be specified for all filters having a constant slope, such as Butterworth and similar designs. The order, or number of poles and number of finite zeros, must be specified for all other types of filters.

d. Minimum Attenuation

Anti-alias filters and notch filters are designed to attenuate particular frequency bands. A measure of the effectiveness of such filters is the minimum attenuation to be expected at certain frequencies. The minimum attenuation must be specified, in decibels referred to the maximum passband response, at a specified frequency requiring that all channels of the system have attenuation over the full temperature and humidity range specified for the system.

e. Gain Tolerance

Because of tolerances on components used in filter construction, a filter's passband gain will usually vary somewhat from the design value. The amount of variation in passband gain must therefore be specified, either in plus-or-minus decibels, or as a percent of the typical gain value.

f. Phase Response

Phase response of a system or circuit is defined as the difference in phase between the system or circuit output and the input terminals, as measured using a sine wave signal source. An output which leads the input is to be specified as a positive phase shift, while a lagging output is to be specified as a negative phase shift.

The phase response of a seismic digital recording system includes the phase shifts introduced by all components of the system, including anti-alias filters, but not including low-cut, high-cut, notch, or reproduce filters. Low-cut, high-cut, notch, and reproduce filter phase shifts are to be specified separately.

g. Amplitude Response Curves

Curves of typical relative amplitude response (decibels) Vs log frequency

(Hz) are to be provided for the system, and separately for low-cut, high-cut, notch, and reproduce filters.
It is recommended that these curves be plotted on 3 cycle semilogarithmic graph paper, covering the frequency range from 1 to 1000 Hz for normal systems or 10 to 10,000 Hz for high resolution systems, and with at least 80 dB amplitude range for system and notch filter response curves, and at least 60 dB range for other filter curves. Low-cut and high-cut filter response curves may be combined if desired.

h. Phase Response Curves
Curves of typical phase shift in degrees linear scale Vs frequency (Hz) log scale to are to be provided for the system separately and for the low-cut, high-cut, notch, and reproduce filters.
System phase response curves must at least cover the frequency range from system low-frequency -20 dB response to high frequency -20 dB response. Low-cut and high-cut filter phase response curves may be combined if desired in addition to separate curves.

i. Group Delay Curves
Curves of typical group delay (milliseconds) Vs log frequency (Hz) are to be provided for the system. Group delay, the negative of the derivative of phase with respect to frequency, can be determined using the phase response curve slope from linear frequency scale phase data and converted to milliseconds by multiplying by 1000/360.

3. Anti-Alias Filters; System Response

a. Anti-Alias Filters
The following anti-alias filter characteristics are to be specified:

(1) Typical cut-off frequency and tolerance.

(2) Order or slope.

(3) Minimum attenuation above the alias (Nyquist) frequency. Alias frequency is:

$$\frac{1}{2 \text{ x sample interval in seconds}} \quad \text{Hz}$$

or

$$\frac{1}{2\,\Delta t} \qquad \text{where } \Delta t = \text{sample interval in seconds}$$

(4) Maximum passband ripple. If the anti-alias filter is of Chebyshev or other similar design having ripple in the pass-band response, the ripple amplitude must be specified. Ripple amplitude is defined as the

difference in decibels, between points of maximum and minimum amplitude in the passband.
If the filter's amplitude response contains a peak near the cut-off frequency, the maximum amplitude of this peak must be specified in decibels referred to the dc or low-frequency response of the filter.

b. System Response
The following characteristics, relating to system response including anti-alias filters, but not including low-cut, high-cut, notch or reproduce filters must be specified:

(1)Typical low-frequency cut-off frequency, measured with an input source resistance of 500 ohms; and including effects of input static filters normally supplied with the system, input transformer, preamplifier, filters, multiplexer, gain-ranging amplifier, offset hulling circuits, and digital filters which are present in the signal path between input terminals and digital tape.

(2) Typical low-frequency cut-off frequency, as in paragraph (1), measured with an input source resistance of 2000 ohms.

(3) Passband Gain Tolerance

(4) Nominal group delay through the system at a frequency equal to one-half the anti-alias cut-off frequency. This must be specified for each of the sample intervals normally available with the system, and include the effects of all phase delays present between input terminals and digital tape.

(5) Group delay tolerance. This must be specified as the maximum plus-or-minus variation in group delay from published group delay curves, in the frequency range from the low-frequency cut-off to the anti-alias filter cut-off frequency.

(6) System amplitude response curves, plotted as relative response (decibels) Vs log frequency (Hz). These curves must show the typical system amplitude response for all anti-alias filters normally supplied with the system, and with input source resistance of both 500 ohms and 2000 ohms; for the complete signal path between input terminals and digital tape.

(7) System phase response curves, plotted as phase shift in degrees Vs log frequency (Hz). These curves must show the typical system phase response for all sample intervals and anti-alias filters normally supplied with the system, and with input source resistance of both 500 ohms and 2000 ohms; for the complete signal path between input terminals and digital tape.

(8) System group delay curves plotted as time delay (milliseconds) Vs log frequency (Hz). These curves must show the nominal group delay for all sample intervals and anti-alias filters normally supplied with the system and with input source resistance of 500 and 2000 ohms. These curves must cover the frequency range at least from the system low frequency cut-off to the anti-alias filter cut-off frequency.

4. Notch Filters

The following notch filter characteristics are to be specified:

a. Nominal frequency of maximum attenuation and tolerance, for all notch filter frequencies normally available with the system, e.g., 50 Hz, 60 Hz. In the case of cascaded notch designs, both frequencies of maximum attenuation must be specified. The notch frequencies and tolerance must be specified to the nearest 0.1 Hz.

b. Nominal-3 dB bandwidth, specified to the nearest 0.1 Hz.

c. Nominal-40 dB bandwidth specified to the nearest 0.01 Hz.

d. Minimum attenuation at the nominal center frequency of filter, e.g., 50.00 Hz, 60.00 Hz.

e. Curves of typical amplitude response (decibels) Vs log frequency (Hz), and curves of typical phase response to linear scale Vs log frequency (Hz). These curves must be plotted separately from the system response curves.

5. Lowcut Recording Filters

The following low-cut filter characteristics are to be specified:

a. Nominal cut-off frequencies and tolerance

b. Slope

c. Passband gain tolerance

d. Phase tolerance - This must be specified as the maximum plus-or-minus percent of variation in phase from published phase curves, over the frequency range from the filter cut-off frequency to the highest anti-alias filter frequency.

e. Curves of typical amplitude response (decibels) Vs log frequency (Hertz) and curves of typical phase response to linear scale Vs log frequency (Hertz). These curves must at least cover the frequency range from low-cut filter -20 dB response frequency to the highest anti-alias filter cut-off frequency. These curves must be plotted separately from the system response curves.

6. High-Cut Recording Filters

If high-cut filters, in addition to anti-alias filters, are provided with the system, the following characteristics must be specified:

a. Nominal cut-off frequencies and tolerance

b. Slope

c. Passband gain tolerance

d. Group delay and tolerance

e. Phase tolerance - This must be specified as the maximum plus-or-minus percent of variation in phase from published phase curves, over the frequency range from the system low-frequency cut-off frequency to the filter cut-off frequency.

f. Curves of typical amplitude response (decibels) Vs log frequency (Hz), and curves of typical phase response to linear scale Vs log frequency (Hz) scale. These curves must at least cover the frequency range from system low-frequency cut-off to the high-cut filter cut-off frequency. These curves must be plotted separately from the system response curves, but may be combined with the low-cut filter curves.

7. Reproduce Filters

a. Analog High-Cut (Smoothing) Filters
The following high-cut filter characteristics are to be specified:

(1) Nominal cut-off frequencies and tolerance

(2) Slope

(3) Curve of group delay and tolerance

(4) Curves of typical amplitude response (decibels) Vs log frequency and phase response to linear scale Vs log frequency.

b. Analog or Digital Lowcut Filters
The following low-cut filter characteristics are to be specified:

(1) Type - analog or digital

(2) Nominal cut-off frequencies, and tolerance if analog

(3) Slope

(4) Phase response and tolerance

(5) Curves of typical amplitude response (decibels) Vs log frequency; and phase response to linear scale Vs log frequency.

c. Reproduce Delays

(1) The time delay, in milliseconds, introduced by demultiplex and playback must be specified, for each sample interval normally available with the stem.

(2) The read-after-write delay, in milliseconds, must be specified for each combination of sample interval, number of channels, and tape format normally available with the system.

Note: The relationship between read-after-write delay and tape speed is not sufficient, unless the relationship between tape speed and sample interval, number of channels, and tape format is also provided.

8. Auxiliary Channels

The following auxiliary channel characteristics shall be specified:

a. Nominal low-frequency cut-off frequency and tolerance

b. Low-frequency slope

c. Nominal high-frequency cut-off frequency and tolerance

d. High-frequency slope

e. Curves of typical amplitude response and phase response

f. Special purpose channels such as for Timebreak and Uphole signals shall be fully described if different from other auxiliary channels.

9. Testing and Measurement

The most convenient, and possibly most accurate, method of measuring the amplitude response and phase response of a seismic digital recorder is by means of Fourier analysis of the impulse response of the system.

Most recent digital systems have built-in impulse or step function generators, so that it is quite convenient to record the impulse response of a system with any combination of filters.

Computer programs which perform the necessary Fourier transforms and produce outputs of relative amplitude, phase response, and time delay as a function of frequency, either in tabular form or as graphical plots, are available or can be obtained by most users and manufacturers. Other methods of obtaining amplitude response and phase response are quite time-consuming and may not include the effects of digital filters or other circuits within the signal path. In addition, the impulse response test may be performed equally well in the field and laboratory, provided only that the system is equipped with a suitable built-in pulse generator. It should be noted that the impulse response method depends on the system linearity being sufficient to provide accurate results.

It would be desirable for equipment manufacturers to standardize on pulse width and amplitude. This standard, therefore, requires that all new digital recording equipment should contain an impulsive source which produces a positive pulse having the following characteristics: pulse width equal to one-half the sample interval for all sample intervals normally supplied with the equipment; and pulse amplitude equal to one-half the maximum linear differential signal input for all values of preamplifier gain normally supplied with the equipment. The pulse shall be initiated at start of digitizing. If a negative pulse follows the initial pulse it shall be at least 1024 samples delayed.

C. Gain

1. Preamplifier gain is that portion of analog amplification that is not variable during a recording. It may have several values that are selectable prior to a record, but is not changed during a record. Channel to channel deviation is controlled by accuracy of calibration. Channel to channel deviation errors can be held to less than 2% without undue burden to the system observer.

a. Nominal values shall be given in numerical value as the ratio of output/input at a specified frequency at least an octave from roll-off.

b. Tolerance of gain from the nominal value shall be given in plus-or-minus percent of nominal numerical value for any selectable gain without manual adjustment.

2. Variable gain is that portion of analog gain which changes during recording in response to the level of signal applied.

a. Where a gain factor is coded in binary form, including multiples of binary, identification of quantified gain steps, shall be integral powers of two.

b. The magnitude of gain steps and number of gains provided shall be specified for the system.

c. Use of multiples of 6 dB to describe gain is recognized as approximate only and gains are precise multiples of two.

3. Gain Step Accuracy

a. Generally, the gain step accuracy of gain ranging seismic amplifiers is governed by the state of the art (usually 0.05% to 0.1% at low gain). Measurement errors are a problem; and, with high gains in many cases, the measurement error is greater than the gain error. Primary causes of measurement error are noise, dc offset, and sample time interval. Some steps to minimize measurement errors are:
Use as long a recording as possible.
Make calculations over an integral number of cycles. Use all sample rates.
Use a low frequency test signal.
Use caution and knowledge of the instrument and test procedure when interpreting results.
Average results for data from same amplifier if possible.
One method of recording test data utilizes record pairs. Each successive pair contains a common amplitude signal at adjacent gain steps; or, if slipped one record, common gain at adjacent signal amplitude steps. Processing can be programmed to check the gain step accuracy in one case or the source signal voltage divider in the other case.

b. The computer program shall perform the following functions:

(1) Pick first zero crossing after 256 samples and select last zero crossing so that an integral number of cycles are used.

(2) Calculate dc offset from arithmetic mean and subtract this from each sample.

(3) Calculate RMS (root-mean square) value.

(4) Calculate and print amplitude ratio for each record pair. Convert ratio to decibels if desired.

(5) Calculate and print amplitude mean and each channel deviation from mean for each record.

(6) Calculate and print % error in gain step

D. Interchannel Crossfeed

1. Interchannel crossfeed is defined as the coupling of signal or noise from one channel to another.

2. Crossfeed is measured by recording an input signal that will produce a near full scale output on all channels except the one being tested and then comparing average amplitudes of driven channels with that of the one under test. The procedure is repeated until all channels have been tested. Input of channel being tested must be terminated with the proper impedance at the recording system input terminals. Amplifiers should be in the gain ranging mode in one test procedure and in minimum fixed gain in a second procedure.

3. Installation crossfeed measurements shall be similarly made, Inputs for this test should be made from input panel of recording truck, so that all input wiring, filter and switching ahead of recording system will also be included.

4. The computer program should perform the following functions:

a. Calculate RMS value of each channel.

b. Calculate average of RMS values of all driven channels.

c. Calculate crossfeed isolation as: Avg. output of driven channels Output of channel being tested. Crossfeed isolation can also be expressed in dB as:

$$20 \text{ Log}_{10} \frac{\text{Avg. output 0f driven channels}}{\text{Output of channel being tested}}$$

d. Repeat above calculations until crossfeed isolation has been determined for all channels.

e. Tabulate results.

5. For routine verification of crossfeed performance of systems with large numbers of channels, it is an accepted practice to record first the near full-scale signal on all the odd channels while all the even channel inputs are terminated; and then to record near full-scale signal on all the even channels while all the odd channels are terminated. The results are computed as outlined in 4 above.

6. Suggested test frequencies are:

10 Hz and 40 Hz	4 ms sample interval
10 Hz and 80 Hz	2 ms sample interval
10 Hz and 160 Hz	1 ms sample interval

E. Harmonic Distortion

1. Harmonic distortion is the nonlinear distortion of a system characterized by the appearance in the output of harmonics of a pure sinusoidal input. For seismic recording systems, this specification shall be a worst case condition. Testing must cover the range of permissible input signal levels to determine the worst case. Although harmonic distortion is generally more severe at low frequencies, tests shall cover the typical seismic bandwidth. Inaccuracies due to power line interference can be minimized by careful choice of the input frequencies. A possible list of such frequencies is shown below:

4.5 Hz	18.0 Hz	72 Hz
6.5 Hz	26.0 Hz	104 Hz
9.0 Hz	36.0 Hz	144 Hz
13.0 Hz	52.0 Hz	208 Hz

2. In making harmonic distortion tests, it is most important that the test oscillator have the lowest distortion possible (less than 0.05%). Four examples are the Tektronix Model SG-505, Hewlett-Packard Model 239A, Sound Technology Model 1400A, or GUS PTU. Internal oscillators in some systems are adequate, but may not offer a wide range of frequencies. Source impedance driving the amplifiers should be on the order of the geophone-cable impedance used normally. A thorough evaluation will include various filter combinations as well as wide band. Record length should be at least 1500 samples and sample interval appropriate to the fundamental and harmonics expected.

3. In computer analysis for harmonic distortion, as little data manipulation as possible should take place before the analysis; preferably only demultiplexing. Assuming sine wave signals recorded by the system, one method of computer analysis may involve a program which performs the following steps:

a. Pick zero crossings of the data.

b. Find average period of the sine wave.

c. Determine and print frequency of fundamental.

d. Calculate the dc offset from arithmetic mean over an integer number of periods.

e. Print the value of dc offset, and subtract it from the raw data.

f. Calculate sin and cos transforms (i.e., cross correlate with computer generated sin and cos functions) for the fundamental and each harmonic of interest.

g. Calculate the Fourier amplitude coefficient.
AH = square root of sum of the squares of sin and cos transforms.

h. For each harmonic of interest, calculate and print:
Percent Harmonic Distortion equals:
100 Fourier amplitude of harmonic Fourier amplitude of fundamental

i. Calculate and print total harmonic distortion by:

$$\text{(1)} \quad THD = 100 \times \text{Sqrt}\left(\frac{(A_2^2 + A_3^2 + A_4^2 + \ldots A_N^2)}{A_1^2} \right)$$

4. For systems which incorporate a low-distortion test oscillator synchronized to the system clock, computer analysis can utilize a Fast-Fourier-Transform. Sufficiently accurate results can only be obtained if the frequency of the test oscillator is an integer multiple of the frequency increment calculated by the Fast-Fourier-Transform.

F. F. Dynamic Range

1. Dynamic range of a system (usually expressed in decibels) is the ratio of the largest signal that can be recorded, with specified system distortion, to the smallest signal that

can be detected above the system noise (nominally when RMS values of signal and noise are equal).
Since the two parameters in the ratio are functions of combined fixed and variable gain, dynamic range should be specified as a chart or graph Vs system gain, or at least minimum and maximum values should be specified.

2. To measure dynamic range where variable gain can be manually controlled, two records should be made at each gain setting of interest-

    a. With an oscillator driving the amplifiers to produce the largest acceptable output level.

    b. With amplifier inputs terminated in a resistance of 500 ohms or 2,000 ohms to approximate geophone/cable impedance in use.

3. For each recording channel, the computer program should:

    a. Calculate the RMS value of recorded maximum output signal.

    b. Calculate the RMS value of recorded noise.

    c. Calculate the signal-to-noise ratio in decibels.

    d. Print the signal-to-noise ratio or dynamic range for each channel.

4. To measure dynamic range of floating point amplifier systems where variable gain cannot be manually controlled:

    a. Choose a test oscillator frequency in the upper part of the pass band to drive the amplifiers.

    b. Make a series of records using -10 dB or -20 dB increments from the maximum acceptable input signal level down to approximately one microvolt.

    c. For each recording channel, the computer program should:

        (1) Calculate the RMS value of the recorded output signal.

        (2) Apply a digital low-pass filter with unity output in the pass band, and a rejection of at least 100 decibels at the test oscillator frequency.

        (3)Calculate the RMS value of the noise which remains after the filter is applied by the equation in section II.A. 4.c(3).

        (4) Calculate signal-to-noise ratio in decibels.

        (5) Print signal-to-noise ratio or dynamic range for each channel.

III. DIGITAL PARAMETERS

A. Mantissa and Gain Code

1. In describing the number of binary bits associated with the mantissa of a digitized quantity, the convention of using one bit to designate the sign (polarity) shall be obeyed so that the number is stated as "sign plus (N-1) bits".

2. Where a gain factor or exponent is coded in binary form including multiples of binary to identify quantified gain steps, the steps shall be integral powers of two.

3. Total number of bits used for gain code or exponent shall be described along with the step's magnitude and number of steps provided.

B. A/D Converter

1. Accuracy

Accuracy is defined as the ability of the converter to digitize input signals within a stated maximum error, input to output, expressed as a fraction of full scale, with gain and offset errors adjusted to zero.

a. A precise DC voltage near full scale shall be applied to the input of converter to be tested. This voltage shall be attenuated at the input of converter to exercise all discrete bit values between the Most Significant Bit (MSB) values (1/2 full scale) and the Least Significant Bit (LSB) of converter. These discrete values would be single bits representing 1/2, 1/4, 1/8...1/n full scale where n is the converter resolution. Test shall be run for both plus and minus values. These values shall be recorded on tape, computer-analyzed for errors, and displayed as a per cent of full scale.

2. Linearity

Linearity is defined by the maximum deviation of an actual converter output from its theoretical value for any input over the full range of converter with gain and offset errors adjusted to zero. A differential linearity of ±½ LSB means that the size of each bit over range of converter is 1 LSB ±½ LSB.

a. Linearity of the converter may be tested in the field by furnishing precision DC voltages to converter with an instrument such as a Dial-A-Source and studying output patterns of digital output values.

b. Linearity of the converter may be tested more precisely by inputting a linear ramp voltage which exercises all bit values between zero and full scale. The non-linear deviation between actual and theoretical values is analyzed by computer and printed in a suitable format.

c. In practice, computer analysis of the system distortion test and converter accuracy test suffice to verify A/D converter linearity for seismic recordings.

3. Offset Error

Offset is defined as converter output for zero signal input. This is normally adjustable to zero or removed in computer processing.

4. Resolution

Resolution of an A/D converter is an expression of the largest change in input which is required to increment or decrement output from one code to the next adjacent code. It is expressed as a percent of full scale.

C. Time Parameters

1. Error of system time standard shall be expressed as a plus-or-minus percent value for its stated environmental range.

2. Sample intervals shall be specified in milliseconds. The number of channels available at each sample interval shall be specified for the system configurations.

3. Sampling skew time from channel 1 to last channel shall be specified for each available sample rate.

4. Tape transport speed shall be specified in inches per second with maximum error expressed as plus-or-minus percent.

IV. RECORDER

A. Tape recordings on commercial standard layouts shall conform to one of the SEG family of tape formats where feasible.

B. Exceptions to SEG formats on commercial digital tape layouts shall be completely documented and sample tapes made available by the manufacturer upon request.

C. ANSI Standards for tape layout shall apply to 800 BPI, 1600 BPI, and 6250 BPI recordings for compatibility with commercial computer transports.

D. Unique digital recording systems and formats are the sole responsibility of the manufacturer but specifications and standards for these apparatus should be maintained to permit transfer and exchange of data readily with commercial computing hardware and software.

V. ENVIRONMENTAL

A. Power input

1. All power source requirements shall be specified as voltage nominal, maximum and minimum and current nominal, maximum and minimum, and frequency nominal, maximum and minimum.

2. Configurations of equipment with different power requirements shall be specified separately.

3. Voltage, current, and frequency shall be measured at the external connection to the apparatus.

B. Temperature

1. Operating ambient temperatures shall be specified for continuous duty over the entire range of input power.

2. On/off operating cycle ambient temperature specifications shall be for the entire range of the input power and for an unlimited number of cycles. Durations of the ON and OFF periods of each cycle shall be specified.

C. Humidity
Ambient environment allowed relative humidity in percent shall be specified for operating Systems.

D. Storage

Non-operating temperature and humidity limits for unlimited storage shall be specified for the system.