SMPTE STANDARD

Calibration Reference Wideband Digital Pink Noise Signal

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Foreword

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SMPTE ST 2095-1 was prepared by Technology Committee 25CSS.

Intellectual Property

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Introduction

This section is entirely informative and does not form an integral part of this Engineering Document.

Pink noise is the most commonly used and ubiquitous test signal for the acoustic level and spectral calibration of sound systems. Pink noise was originally chosen for this task because it has characteristics that are similar to the way human spectral hearing perception works, which is that we roughly hear in an equal energy/octave manner.

Though pink noise can be generated by many different types of devices, there is no standard pink noise signal that all devices emulate. Nor is there a published standard that specifies exact parameters and values for calibration pink noise. Therefore, while many of today's field devices generate a usable pink noise signal, all of these signals are somewhat different. This presents a basic variable in the calibration process, as sound system calibrations designed to obtain the same results among many different systems are being performed with different test signals.

In the days of analog soundtracks, test tapes and test films were available from respected laboratories, so there was some agreement on the test signals that were utilized in the industry. Since the advent of digital sound reproduction, no standard has been in place for generating digital pink noise signals.

This document delineates the parameters and values for a standard digital pink noise signal that can be used for level and spectral calibration. In specifying these parameters and values, the subject of how to determine the level of a pink noise signal is discussed. See Annex C for an informative discussion of pink noise measurement methods.

Annex A describes an executable computer program file in the Python 2 programming language, and .wav files generated by that program, all provided separately. Annex B of this document includes a sample implementation of a compliant software-based pink noise generator presented as pseudocode.

Not addressed in this document are the various use cases where a pink noise signal may be employed, and the conditions that may affect the outcome. Such matters are best left to standards documents such as SMPTE ST 202:2012, SMPTE RP 200:2012, and AES17-1998 (r2009).

1 Scope

This standard defines a digital pink noise signal to be used in calibrating the sound pressure level and electroacoustic response of a cinema B-chain system. It also defines an example algorithm to generate compliant LPCM pink noise signals in DSP devices with sampling rates of 48.00 kHz and 96.00 kHz. It does not define pink noise signals that may be used for other purposes such as headroom, distortion, or other system measurements.

2 Conformance Notation

Normative text is text that describes elements of the design that are indispensable or contains the conformance language keywords: "shall", "should", or "may". Informative text is text that is potentially helpful to the user, but not indispensable, and can be removed, changed, or added editorially without affecting interoperability. Informative text does not contain any conformance keywords.

All text in this document is, by default, normative, except: the Introduction, any section explicitly labeled as "Informative" or individual paragraphs that start with "Note:”

The keywords "shall" and "shall not" indicate requirements strictly to be followed in order to conform to the document and from which no deviation is permitted.

The keywords, "should" and "should not" indicate that, among several possibilities, one is recommended as particularly suitable, without mentioning or excluding others; or that a certain course of action is preferred but not necessarily required; or that (in the negative form) a certain possibility or course of action is deprecated but not prohibited.

The keywords "may" and "need not" indicate courses of action permissible within the limits of the document.

The keyword “reserved” indicates a provision that is not defined at this time, shall not be used, and may be defined in the future. The keyword “forbidden” indicates “reserved” and in addition indicates that the provision will never be defined in the future.

A conformant implementation according to this document is one that includes all mandatory provisions ("shall") and, if implemented, all recommended provisions ("should") as described. A conformant implementation need not implement optional provisions ("may") and need not implement them as described.

Unless otherwise specified, the order of precedence of the types of normative information in this document shall be as follows: Normative prose shall be the authoritative definition; Tables shall be next; followed by formal languages; then figures; and then any other language forms.

3 Normative References

The following standards contain provisions that, through reference in this text, constitute provisions of this recommended practice. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this recommended practice are encouraged to investigate the possibility of applying the most recent edition of the standards indicated below.

SMPTE RP 200:2012, Relative and Absolute Sound Pressure Levels for Motion-Picture Multichannel Sound Systems — Applicable for Analog Photographic Film Audio, Digital Photographic Film Audio and D-Cinema.

AES17-1998 (R2009), AES Standard Method for Digital Audio Engineering — Measurement of Digital Audio Equipment.

4 Terms and Acronyms

4.1 0 dB FS

The RMS level of a 997-Hz sine wave whose positive peak value reaches Full Scale Digital. (Note: This may be depicted as "0 dBFS" in some literature. The meaning is the same.)

4.2 Audio sampling rate

The number of audio samples per second, measured in Hz or kHz (one kHz being 1,000 Hz). For example, 48,000 samples per second can be expressed as either 48,000 Hz, or 48.00 kHz.

4.3 Average Responding Meter

An average responding meter measures the mean of the absolute value of sample values (or instantaneous values) but is calibrated to indicate the RMS value of the waveform when driven with a sine wave. Because the ratio of average to RMS varies with the type of waveform, such a meter can read the RMS value of sine waves accurately, but such a meter will read the RMS value of other waveforms incorrectly.

4.4 Crest factor

The ratio of peak value to RMS value of a waveform, in dB.

4.5 dB FSD

RMS value relative to FSD. (Note: This may be depicted as "dBFSD" in some literature. The meaning is the same.)

4.6 FSD

Full Scale Digital. The maximum numeric value capable of being stored in the bit depth of a digital signal path. Example: A full scale sine wave has positive peaks with a value of 7FFFFF (hex) and negative peaks with a value of 800000 (hex) in a 24-bit digital audio system.

4.7 Kurtosis

A measure of whether the data are peaked or flat relative to a normal distribution. The Kurtosis for a normal distribution is 3. The "excess kurtosis" for a normal distribution is 0.

4.8 Peak value

The absolute magnitude of the highest peak amplitude excursion of a signal.

4.9 Pink noise

Noise whose power spectral density (W/Hz) is inversely proportional to frequency and whose power per octave (W/octave) is constant.

4.10 Pinking filter

A filter that converts white noise to pink noise. A pinking filter has a frequency response of -3 dB/octave.

4.11 Root mean square (RMS or rms)

The square root of the mean of the squares of sample amplitudes. Different waveforms with the same RMS value will deliver the same amount of power. The RMS value of a sine wave is $1/\sqrt{2}$ times its peak value.

4.12 Skewness

A measure of the lack of symmetry. The skewness for a normal distribution is 0.

4.13 Unique signal period

Period over which no portion of a noise signal repeats itself.

4.14 White noise

Noise whose power spectral density (W/Hz) is constant and whose power per octave (W/octave) is proportional to frequency.

4.15 Wideband pink noise

A testing reference signal that includes all frequencies within the bandwidth of a measuring system.

5 Summary of Characteristics

Table 1 summarizes key characteristics of the Calibration Reference Wideband Digital Pink Noise Signal.

Table 1 – Digital pink noise characteristics

Parameter	Value	Reference
RMS Level, full spectrum	-18.5 dB FS	Section 6.2
RMS Level, 22.4 Hz – 22.4 kHz	-19 dB FS	Section 6.1
Target RMS Level, any 1/3-octave from 20 Hz to 16 kHz	-33.74 dB FS	Section 6.2
Crest factor	11.5 – 12 dB	Section 6.3
Pink Noise signal bandwidth	10 Hz – 22.4 kHz	Sections 7.2, 7.3
Energy uniformity	±0.25 dB for any 1/3-octave band from 20 Hz – 16 kHz	Section 7.4
Minimum unique signal period	10 seconds	Section 9.1

6 Signal Amplitude

6.1 Level Criteria (Informative)

In order to ensure consistency with the calibration level of dubbing theatres and the level of soundtracks produced over the past several decades, the digital pink noise level defined in this Standard is intended to be consistent with widely accepted reference noise signals that are currently used in movie soundtrack production.

It should be noted that historical discussions of pink noise level are based on SMPTE RP 200, which defines the reference level of wideband pink noise as being -20 dB FS when measured by an average-responding voltmeter with a bandpass filter of 22 Hz to 22 kHz.

Regarding meter type, SMPTE RP 200:2012 defines an average responding meter as “a meter which provides a voltage indication proportional to the average value of the rectified signal, with ballistics as described in IEC 60268-17.” In contrast, this Standard defines pink noise levels in the digital domain with RMS measurements. The difference in meter type affects the numerical level readings. For example, the -20 dB FS level per SMPTE RP 200 reads -19 dB FS with an RMS meter even though the actual signal level through the prescribed bandpass filter is equivalent.

Bandwidth also affects the numerical level readings. The spectrum of the pink noise defined in this Standard extends to 10 Hz to better cover the low frequency range of modern loudspeaker systems. When measured without a bandpass filter, the RMS level reads 0.5 dB higher, -18.5 dB FS.

6.2 **RMS Value** (Normative)

The RMS value of the digital pink noise signal having the bandwidth defined in Section 7.1 shall be -18.5 dB FS.

The broadband level tolerances are as follows:

a. The value for any interval greater than or equal to the unique signal period shall be within ±0.10 dB.

b. The value of any one second interval shall be within ±0.75 dB.

c. The value of any 125 millisecond (ms) interval shall be within ±2.00 dB.

When analyzed on a 1/3-octave scale, the target level for any 1/3-octave band between 20 Hz and 16 kHz (inclusive) is -33.74 dB FS.

See Section 6.1 for an informative comparison to current analog pink noise signals.

6.3 **Crest Factor** (Normative)

The Crest Factor, measured over the unique signal period, shall be between 11.5 dB and 12 dB, inclusive.

6.4 **Statistical Criteria** (Informative)

The heart of a pink noise signal is the white noise source from which it is derived. Wideband noise must be uniformly distributed across the spectrum of interest. Another way to say this is it must be totally random. However, not all random noise is perfectly random. In a digital noise signal, true randomness implies the value of each sample within a defined range is equally probable; i.e., a uniform distribution, with each successive sample being statistically independent of the others. The predictability of computers makes it a challenge to generate perfect randomness in a compact algorithm. Section 9 describes the pseudorandom number generator used in the example pink noise algorithm.

Ideal randomness in a noise signal can be shown by how well it conforms to a normal distribution, also known as a Gaussian distribution — the bell-shaped curve. Kurtosis is one measure of distortion in the shape of the Gaussian curve. A high kurtosis distribution has a sharper peak and fatter tails, while a low kurtosis distribution has a more rounded peak and thinner tails.

Another form of distortion in the distribution curve is skewness, or asymmetry, wherein the tails on either side of the peak have different slopes, rather than being symmetrical mirror images.

Figure 1 shows the ideal Gaussian distribution for the example noise generator with kurtosis better than 3 ±0.2, and skewness better than 0 ±0.01. The *x* axis of the graph shows normalized full scale amplitude (min/max -1 to +1). The *y* axis is frequency of occurrence, normalized to max bin = 1.0. The graph shows the relative number of samples falling into 71 amplitude bins, each with a width of 0.01, between -0.35 to +0.35. The example noise generator algorithm limits the amplitude of peaks to +/- 0.335 and so the curve goes to 0 at +/- 0.35.

Skewness and Kurtosis figures are calculated directly from the sample values for pink noise signals generated by the example algorithm at 48 kHz and 96 kHz sample rates, with periodicity ranging from 512 K to 4096 K samples, using the SKEW() and KURT() functions in Microsoft Excel.

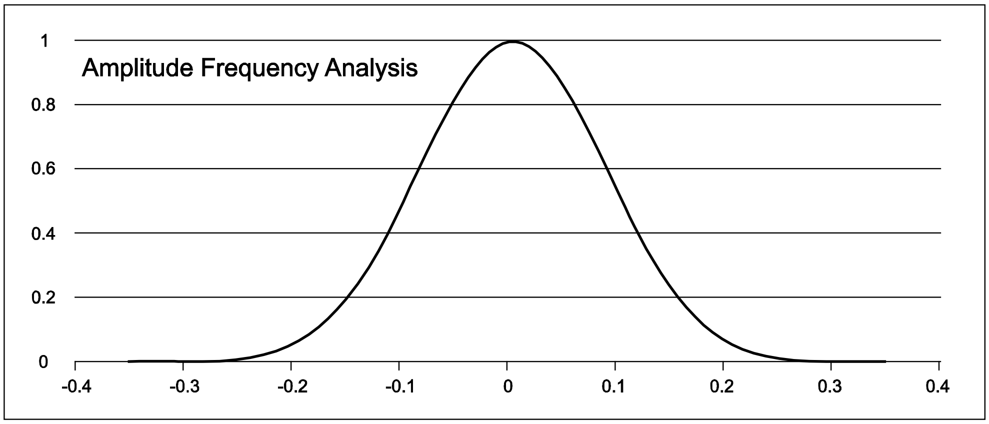

Figure 1 – Normalized amplitude frequency distribution

7 Signal Spectral Content

7.1 Bandwidth Characteristics

The bandwidth of the calibration reference wideband digital pink noise signal shall be 10 Hz to 22.4 kHz.

7.2 Low Frequency Roll-off Characteristics

The low frequency roll-off characteristics shall be such that the signal is attenuated by -3 dB at 10 Hz, and decreases at a minimum rate of 21 dB per octave below 10 Hz, which is the result of applying a conventional 4th-order Butterworth highpass filter with a cutoff frequency of 10 Hz to a pink noise signal with a 3 dB per octave slope. Note that a digital filter with these characteristics may be realized from the continuous time (analog) prototype using the matched pole-zero mapping method (matched Z transform) or equivalent.

7.3 High Frequency Roll-off Characteristics

The high frequency roll-off characteristics shall be such that the signal is attenuated by -3 dB at 22.4 kHz, and decreases at a minimum roll-off rate of 36 dB per octave above 22.4 kHz. It is noted that for sample rates of 48.00 kHz and 96.00 kHz, a digital filter realized from a 4th-order, continuous-time Butterworth lowpass filter prototype by means of the bilinear transform with frequency pre-warping will easily satisfy this specification.

7.4 Spectral Uniformity

The RMS signal level of any 1/3-octave band between 20 Hz and 16 kHz shall fall within ±0.25 dB of the nominal target level when measured over a period of time equal to at least one unique signal period.

8 Digital Signal Parameters

8.1 Sample Rate

The calibration reference wideband digital pink noise signal shall support sample rates of 48.00 kHz and 96.00 kHz.

8.2 Sample Word Size

The calibration reference wideband digital pink noise signal shall have a have an integer word size of at least 24 bits/sample.

9 Signal Duration

9.1 Minimum Unique Signal Period (Normative)

The minimum unique signal period for a calibration reference wideband pink noise signal is 10 seconds.

9.2 Signal Duration

To extend the duration of the signal for purposes of acoustic measurements, the unique signal period may be looped or repeated as required. Alternatively, the unique signal period can be extended beyond the minimum prescribed 10 seconds.

9.3 Measurement Implications of Signal Duration (Informative)

In acoustic measurements, factors such as reverberation and background noise often will necessitate measurement durations on the order of one minute or more for full settling and stability, especially at low frequencies. If the minimum unique signal period is repeated, it is expected to provide a stable measurement within the settling time of the room and measurement system. For cases where an extended signal with a longer unique signal period is used, measurement time may need to be longer for the measurement signal to converge to the specified tolerances.

10 Calibration Reference Wideband Digital Pink Noise Signal Algorithm (Informative)

10.1 Noise Generator

Annex A describes an available executable noise generator program, and Annex B provides a pseudocode version. The algorithm demonstrates a method of generating band-limited "pink" noise by filtering the output of a pseudorandom number generator (PRNG). The PRNG used is a linear congruential generator (LCG). The cyclicality of lower order bits in full-period LCGs with modulus 2^n (often regarded as a flaw in many other applications) makes it possible to limit its periodicity while maintaining a uniform probability distribution by simply discarding some of the higher order bits. This is useful for our purposes, as it makes it easy to support multiple sampling rates and obtain faster, more repeatable results in spectrum analysis and sound level measurements.

10.2 Spectral Shaping and Filtering

The filter network used to shape and band-limit the output spectrum consists of a "pinking" filter bank and a bandpass filter. The pinking filter is a parallel network of six, first-order lowpass filters. The bandpass filter is comprised of a fourth-order highpass and lowpass filter, implemented as two pairs of second-order (biquad) filters.

10.3 Scaling the Output Level

To obtain a desired output level, given specific highpass and lowpass filter cutoff frequencies, a gain stage can be applied anywhere in the signal chain. Alternately, scaling gain can be built into the pinking filter (saving one mult per iteration) by multiplying the beta/b1 coefficient of each component filter by a desired gain factor.

Annex A Example Band-Limited Pink Noise Generator Executable Script (Informative)

A.1 Python Script

Included with this standard is the file ST-2095-Generator.py. This file is an executable computer program in the Python 2 programming language. To execute this program you will need a Python interpreter, which might have been included by default with your operating system or else can be obtained from https://www.python.org/.

The program accepts a single required argument, the name of the file to be created. Execution is thus very simple:

```
$ python ST-2095-Generator.py example-file.wav
```

The generated WAV file will contain a single channel of noise at a sample rate of 48.00 kHz, 24 bits per sample, or a single channel of noise at a sample rate of 96.00 kHz, 24 bits per sample. These and other parameters can be adjusted by altering the values of the constant variables found at the head of the program.

A.2 Computationally Efficient Pseudorandom Number Generation

The Python script employs a linear congruential generator (LCG) to create a pseudorandom number generator (PRNG)

The generalized form of an LCG PRNG is:

$r_{n+1} = a \cdot r_n + c \pmod{m}$

where

m is the modulus

r_n , r_{n+1} , … are the pseudorandom output

a is a constant (the multiplier) between 0 and m - 1

c is a constant (the adder or stepper) between 0 and m - 1

r_0 is a seed -- can be any number between 0 and m - 1

Selection of the three constants (a, c, and m) is a critical factor in the statistical performance of the generator and must be approached carefully. The particular stepper and adder values used in this example have been found to work well with modulo 2^{19} though 2^{24} at sample rates ranging from 44,100 to 96,000 samples/second.

LCGs are typically implemented using storage bit truncation (overflow or "wrapping") in integer math operations as a natural (essentially free) modulo function, making them very fast and light and well suited to implementation on a variety of hardware platforms. However a common source of bottlenecks in general, when using integer PRNGs in floating point DSP applications, can be the scaling and type conversion steps. In some cases, simply remembering to multiply by the reciprocal of a divisor, rather than dividing, can alleviate a performance problem. Otherwise, it could be desirable to construct a scaled, signed floating-point number bitwise, from the integer output of the PRNG.

Since we know that the maximum output value of the LCG in this example algorithm is always a positive power of 2 minus 1, we could scale it to fit the mantissa (significand) of a 32-bit IEEE 754 floating point number by simply shifting all the bits to the left or right. Masking the resulting number with 0x40000000, by means of a bitwise OR, scales the exponent bits to yield a positive number between 2 and 4. Once this bit sequence is recast to a floating point number, subtracting 3.0 from the result yields a fractional number between 1 and -1. See the example in Figure A.1.

32-bit Signed integer: MSB is sign bit, Max value = 2^31 - 1 = 2147483647 = 0x7FFFFFFF

randMax	Shift by	31	30	29	28	27	26	25	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
2^19 - 1	<< 4	0	0	0	0	0	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
2^20 - 1	<< 3	0	0	0	0	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
2^21 - 1	<< 2	0	0	0	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
2^22 - 1	<< 1	0	0	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
2^23 - 1		0	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
		|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|	|

Hex 0x40000000

		31	30	29	28	27	26	25	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
	Binary	0	1	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

Layout of IEEE 754 32-bit Float (Result of Bitwise OR)

		31	30	29	28	27	26	25	24	23	22	21	20	19	18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
		0	1	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
		Sign	Exponent (8 bits)								Mantissa (23 bits)																						

Figure A.1 – Example of bitwise to floating point conversion

Annex B Pseudocode Listing for Band-Limited Pink Noise Generator (Informative)

```
// Band-Limited Pink Noise Generator
// Produces band limited, pink noise from pseudorandom numbers
// Inputs:
//    int SampleRate
//    int Period
//

//    float HpFc
//    float LpFc
//    float MaxPeak
// Revised 2015-01-04 by Calvert Dayton

float maxAmp = 10.0^(MaxPeak / 20.0);
float pi = 3.141592654;

// Initialize variables for generating a random number
int randMax = Period - 1;
int seed = 0;
float white = 0.0;
float scaleFactor = 2.0 / float(randMax);
int randStep = 52737;
// 1024k Samples at 9600 or 88200 sample rate is a special case
if (Period == 1048576 and SampleRate > 48000) { randStep = 163841 };

// Calculate omegaT for matched Z transform highpass filters
float w0t = 2.0 * pi * HpFc / float(SampleRate);

// Disaster check: Limit LpFc <= Nyquist
if (LpFc > float(SampleRate / 2.0) { LpFc = float(SampleRate / 2.0) };

// Calculate k for bilinear transform lowpass filters
float k = tan(( 2.0 * pi * LpFc / float(SampleRate)) / 2.0);
float k2 = k * k;

// Calculate biquad coefficients for bandpass filter components
hp1_a1 = -2.0 * exp(-0.3826835 * w0t) * cos(0.9238795 * w0t);
hp1_a2 = exp(2.0 * -0.3826835 * w0t);
hp1_b0 = (1.0 - hp1_a1 + hp1_a2) / 4.0;
hp1_b1 = -2.0 * hp1_b0;
hp1_b2 = hp1_b0;
```

```

hp2_a1 = -2.0 * exp(-0.9238795 * w0t) * cos(0.3826835 * w0t);
hp2_a2 = exp(2.0 * -0.9238795 * w0t);
hp2_b0 = (1.0 - hp2_a1 + hp2_a2) / 4.0;
hp2_b1 = -2.0 * hp2_b0;
hp2_b2 = hp2_b0;

lp1_a1 = (2.0 * (k2 - 1.0)) / (k2 + (k / 1.306563) + 1.0);
lp1_a2 = (k2 - (k / 1.306563) + 1.0) / (k2 + (k / 1.306563) + 1.0);
lp1_b0 = k2 / (k2 + (k / 1.306563) + 1.0);
lp1_b1 = 2.0 * lp1_b0;
lp1_b2 = lp1_b0;

lp2_a1 = (2.0 * (k2 - 1.0)) / (k2 + (k / 0.541196) + 1.0);
lp2_a2 = (k2 - (k / 0.541196) + 1.0) / (k2 + (k / 0.541196) + 1.0);
lp2_b0 = k2 / (k2 + (k / 0.541196) + 1.0);
lp2_b1 = 2.0 * lp2_b0;
lp2_b2 = lp2_b0;

// Initialize delay line variables for bandpass filter
float w = 0.0;
float hp1w1 = 0.0;
float hp1w2 = 0.0;
float hp2w1 = 0.0;
float hp2w2 = 0.0;
float lp1w1 = 0.0;
float lp1w2 = 0.0;
float lp2w1 = 0.0;
float lp2w2 = 0.0;

// Initialize delay lines for pink filter network
float pink = 0.0;
float lp1 = 0.0;
float lp2 = 0.0;
float lp3 = 0.0;
float lp4 = 0.0;
float lp5 = 0.0;
float lp6 = 0.0;

// For each iteration of the noise generator

// Generate a pseudorandom number using linear congruential PRNG (LCG).
// Bitwise AND with randMax forces sign bit positive and zeroes any unwanted bits.
seed = (1664525 * seed + randStep) & randMax;
```

```
// Scale to a real number in the range -1.0 <= white <= 1.0
white = float(seed) * scaleFactor - 1.0;

// Run pink filter; a parallel network of first-order LP filters, scaled to
// produce an output signal with target RMS = -21.5 dB FS (-18.5 dB AES FS)
// when bandpass filter cutoff frequencies are 10 Hz and 22.4 kHz.
lp1 = 0.9994551 * lp1 + 0.00198166688621989 * white;
lp2 = 0.9969859 * lp2 + 0.00263702334184061 * white;
lp3 = 0.9844470 * lp3 + 0.00643213710202331 * white;
lp4 = 0.9161757 * lp4 + 0.01438952538362820 * white;
lp5 = 0.6563399 * lp5 + 0.02698408541064610 * white;
pink = lp1 + lp2 + lp3 + lp4 + lp5 + lp6 + white * 0.0342675832159306;
lp6 = white * 0.0088766118009356;

// Run bandpass filter; a series network of 4 biquad filters
// Biquad filters implemented in Direct Form II
w = pink - hp1_a1 * hp1w1 - hp1_a2 * hp1w2;
pink = hp1_b0 * w + hp1_b1 * hp1w1 + hp1_b2 * hp1w2;
hp1w2 = hp1w1;
hp1w1 = w;

w = pink - hp2_a1 * hp2w1 - hp2_a2 * hp2w2;
pink = hp2_b0 * w + hp2_b1 * hp2w1 + hp2_b2 * hp2w2;
hp2w2 = hp2w1;
hp2w1 = w;

w = pink - lp1_a1 * lp1w1 - lp1_a2 * lp1w2;
pink = lp1_b0 * w + lp1_b1 * lp1w1 + lp1_b2 * lp1w2;
lp1w2 = lp1w1;
lp1w1 = w;

w = pink - lp2_a1 * lp2w1 - lp2_a2 * lp2w2;
pink = lp2_b0 * w + lp2_b1 * lp2w1 + lp2_b2 * lp2w2;
lp2w2 = lp2w1;
lp2w1 = w;

// Limit peaks to ± MaxAmp
if (pink >  MaxAmp) {pink =  MaxAmp};
if (pink < -MaxAmp) {pink = -MaxAmp};

// Do something with the output sample stored in "pink" before repeating
```

Annex C Measuring Pink Noise Amplitude (Informative)

C.1 The Role of Sine Waves in Noise Measurement

Sine waves have been found to be essentially unusable for acoustic level calibration measurements inside reflective spaces due constructive and destructive interference affecting the SPL at the microphone. The broader spectrum and random nature of noise signals makes them the preferred choice for acoustic level measurements.

It is, however, this very randomness and the scalable bandwidth of noise, coupled with the desire for a pink spectrum rather than white, that presents certain challenges in the definition and use of noise signals for sound system calibration.

Sine waves posses several useful properties, among them a narrow frequency spectrum, a stable amplitude over time, and a well defined crest factor. These properties minimize uncertainty in level measurement compared with other periodic waveforms such as square waves which require much wider bandwidth and can exhibit overshoots or tilt in the analog domain.

Since the crest factor of a sine wave is 3.01 dB, it is possible to create a sine wave whose positive peaks reach FSD in the digital domain, and to translate this level to a known quantity in the analog domain. That known quantity is defined as 0 dB FS. The RMS value of any analog waveform emanating from a digital system can thus be quantified relative to 0 dB FS, and by extension to dB FSD.

C.2 Sine Wave and Pink Noise Amplitude Considerations

The normative noise signal level criteria described in Section 6 are based on digital measurements made with modern computer-based tools. However, these same pink noise signals are intended for use in acoustic and analog electronic environments where they will be measured with a variety of analog input devices.

The measurement of analog audio signals, which includes sine waves and noise, involves the conversion of the waveform to a DC voltage suitable for deflecting an analog meter movement or translating to a numerical value. All AC voltmeters are calibrated to display the RMS value of a sine wave. Regardless of the display type, it is the method of converting the analog signal to DC that fundamentally affects the accuracy of noise measurements. Most AC voltmeters and SPL meters are one of two types: average responding or true RMS responding. Both types of meters can accurately measure the RMS value of a sine wave, but only the true RMS meter can accurately measure the RMS value of non-sinusoidal waveforms, e.g. noise or square waves.

C.3 Metering Considerations

Modern professional SPL meters use true RMS detectors, while common consumer SPL meters use average-responding detectors. When measuring noise, average-responding meters can read approximately one decibel lower than true RMS meters, which can vary due to meter characteristics and the crest factor of the noise itself. To avoid such uncertainties, all measurements of RMS noise level relative to this document use true RMS conversion.

Annex D Explanation of Parameter and Tolerance Choices (Informative)

Following are explanations of the reasoning behind the choices for parameters and tolerances in the respective paragraphs of the document.

6.1 Level Criteria: As explained in this informative paragraph, prior to this project there had been different interpretations of just what was meant by the "-20 dB" signal referred to in SMPTE RP 200 and elsewhere. AES 17 and IEC standards present a definition of dB FS, which is an RMS-only entity referred to a 997-Hz sine wave with a peak value corresponding to the absolute maximum sample value obtainable for a given sample word size. Note that in this case, a square wave can be constructed which has an RMS value of +3 dB FS, however many mathematics programs define dB relative to full scale as a peak value, such that a full scale square wave would have an RMS value which equals its peak value. Further complicating these matters, legacy metering had been done with "average" reading meters, such that wide-band signals cannot be expected to show true RMS values.

6.2 RMS Value: For well over a decade, one particular calibration pink noise has been widely accepted for calibration purposes, which calculate out to being roughly -18.5 dB FS. As such, the committee felt that it would not be appropriate to change the actual RMS value of the standard digital calibration signal. The rest of the numbers seen in Section 6.2 reflect bounds which empirically are in correspondence to the tolerances encountered with metering schemes typically employed for measuring SPL.

6.3 Crest Factor: Crest factor describes the relative value of a signal's absolute peak referred to the signal RMS value. It was felt important to constrain this parameter because leaving it un-bounded meant a random sample could potentially have a value at or beyond FSD which could cause overload or clipping of the signal chain. Constraining the crest factor to 4.0 (or 12 dB) gives us a properly bounded amplitude which ensures at least 6 dB of headroom free of clipping, and if there ever is a need to evaluate a system at a higher drive level, this facilitates operating as much as 6 dB above calibration level still with no chance of exceeding the linear range of the signal chain. This value also corresponds closely with the natural peak-to-average ratio of a random noise signal subjected to pink filtering. Constraining it to 12 dB involves only minimal removal of over-valued samples from the bit stream.

6.4 Higher order statistical properties: The tolerances presented were drawn from analysis of signals generated which met all of the other criteria, and are presented here for informative purposes only.

7.1 Signal Bandwidth characteristics: Because this pink noise calibration signal can be used for frequency response analysis, it is essential that the signal bandwidth extend beyond the bounds of system operation. This is so that the spectral content within the system operating bandwidth can be held within the specified tolerances, and system behavior at the extremes of the operating band can be quantified. It is well documented that many modern cinema soundtracks can contain significant energy at frequencies outside of the traditional B-chain frequency range. If one were to consider a system which might have a peaking resonance at 15 Hz or 20 kHz, and the test signal were to be restricted to a narrower bandwidth, such response anomalies could not be ascertained. By extending the signal bandwidth to include these frequencies, such system behavior can be revealed and addressed.

7.2 Low frequency roll off characteristics: In keeping with the bandwidth constraints in Section 7.1 above, analysis showed that a 4^{th}-order filter provided acceptable in-band flatness while sufficiently attenuating below cutoff, the minimum slope of 21 dB/octave reflects the measured slope just below the corner frequency.

7.3 High frequency roll-off characteristics: Also in keeping with the bandwidth requirements of Section 7.1, much as noted for the LF roll-off, a 4^{th}-order filter was specified and it is noted that the bilinear transform implementation provides a steeper roll off than the 24 dB/octave typically expected of a 4^{th} order filter as Nyquist frequency is approached.

7.4 Spectral Uniformity: Because the signal can be used for RTA frequency response analysis, it needs to converge to a flat trace when viewed with a 1/3 octave analyzer. The 0.25 dB tolerance was selected such that net results with accuracy on the order of ±0.5 dB can reasonably be expected.

8.1 Sample rate: DCI specifications prescribe PCM audio at 96-kHz sample rate and 24 bits, Because many legacy systems still use 48,000 samples per second, it was determined that both 96 kHz and 48 kHz sample rates would be accommodated in the standard.

8.2 Sample word size: In keeping with DCI audio specifications, a PCM word size of 24 bits was decided to be the nominal data format for the calibration pink noise standard.

9.1 Minimum unique signal period: Two factors came into consideration when this aspect of the signal was being discussed. First, the periodic nature of pseudo-random signals possessing a short period (less than 5 seconds) is more audibly noticeable and could be disconcerting to anyone more accustomed to using a random noise signal. Second, harmonic analysis tells us that the minimum period of a signal dictates the spacing of the harmonic components of any periodic signal. In order to achieve a good density of spectral content down to the lowest frequencies of interest, it was determined that harmonic spacing needed to be less than 0.1 Hz, which necessitates a 10 second minimum period. It is also important to note that this is a minimum specification, and longer period pseudo-random sequences are fully acceptable, provided that the other parameter constraints laid out within this standard are met.

9.2 Signal duration: Because room acoustic measurements are subject to settling time and averaging, the signal in nearly all cases must be available for a duration beyond that specified as the minimum period. Since the spectral content and RMS values are met by a signal with the minimum period, repeating such a signal is an acceptable means of extending the duration of the signal to satisfy the measurement requirements. Alternatively, running a longer pseudo-random sequence, effectively extending the periodicity beyond the minimum specification, is also an acceptable means of generating a signal of longer duration. Either technique will lead to consistent results for extended duration room measurements.