SMPTE ST 2095-1:2023
Revision of SMPTE ST 2095-1:2015

SMPTE STANDARD

Calibration Reference Wideband Digital Pink Noise Signal

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Approved September 27, 2023

Foreword

SMPTE (the Society of Motion Picture and Television Engineers) is an internationally-recognized standards developing organization. Headquartered and incorporated in the United States of America, SMPTE has members in over 80 countries on six continents. SMPTE's Engineering Documents, including Standards, Recommended Practices, and Engineering Guidelines, are prepared by SMPTE's Technology Committees. Participation in these Committees is open to all with a bona fide interest in their work. SMPTE cooperates closely with other standards-developing organizations, including ISO, IEC and ITU.

SMPTE Engineering Documents are drafted in accordance with the rules given in its Standards Operations Manual.

SMPTE ST 2095-1 was prepared by Technology Committee 27C. This second version of SMPTE ST 2095-1 differs from the first version, SMPTE ST 2095-1:2015, only insofar as:

- to improve the clarity of the normative provisions, in particular in recognition of a known 3 dB difference between two different industry use cases of the dBFS units;
- to consolidate informative text into informative annexes;
- to update its references to other documents; and
- to update the Example Band-Limited Pink Noise Generator Executable Script to be compatible with both Python 2.7 and Python 3 and to provide a warning in the event that the script is requested to create Pink Noise that is longer than the file size limit of the 32 bit RIFF format.

Intellectual Property

At the time of publication no notice had been received by SMPTE claiming patent rights essential to the implementation of this Engineering Document. However, attention is drawn to the possibility that some of the elements of this document may be the subject of patent rights. SMPTE shall not be held responsible for identifying any or all such patent rights.

Introduction

This clause 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 has been no standard Pink Noise signal that all devices emulate. Nor has there been 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, many 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, and until this document, 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 A, Annex B, Annex E, and Annex F for an informative discussion of Pink Noise measurement methods.

Annex C describes an executable computer program file in the Python 2.7 or Python 3 programming language, and .wav files generated by that program. The executable Python program is provided separately (see Annex G). Annex D 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 might be employed, and the conditions that might affect the outcome. Such matters are best left to Engineering Documents such as SMPTE ST 202, SMPTE RP 200, ST 2096-1, and ST 2096-2.

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 as defined in in SMPTE ST 202:2010. It also defines an example algorithm to generate compliant LPCM (Linear Pulse Code Modulation) Pink Noise signals in DSP (Digital Signal Processing) devices with sampling rates of 48 kHz and 96 kHz. It does not define Pink Noise signals that might 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 clause 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; then formal languages; then figures; and then any other language forms.

3 Normative references

There are no normative references in this document.

4 Terms and definitions

For the purposes of this document, the following terms and definitions apply:

4.1
Unique Signal Period

period over which no portion of a signal repeats itself

4.2
Root Mean Square
RMS
rms

square root of the arithmetic mean of the set of squared digital audio sample amplitudes over a period at least as long as the Unique Signal Period

Note 1 to entry: Example: The RMS of a sine wave is $\frac{1}{\sqrt{2}}$ times (−3.01 dB) its Peak Value.

4.3
Peak Value

highest sample amplitude of a sampled signal

Note 1 to entry: For sampled signals (discrete time-series signals), oversampling or reconstructing to a continuous time (analog) signal via a suitable lowpass filter might cause the true peak value to exceed the Peak Value of the original discrete time-series signal.

4.4
Crest Factor

ratio of Peak Value to RMS value of a waveform, expressed in dB

4.5
Full Scale
FS

maximum sample amplitude that can be represented by the digital modulation system

Note 1 to entry: Respectively, the maximum sample value, minimum sample value, and maximum sample amplitude for signed integer representations are: $2^{(n-1)} - 1$, $-2^{(n-1)}$, and $2^{(n-1)} - 1$ where n is the sample word-length in bits (bit depth).

4.6
dBFS

deciBels (dB) relative to Full Scale (4.5)

Note 1 to entry: The definition of dBFS for the purposes of this standard is not the same as that defined by AES17-2020 for the purposes of its scope, which is for measuring digital audio equipment.

4.7
dBFS RMS

deciBels (dB) RMS relative to Full Scale (4.5)

4.8
dBFS(AES17)

deciBels (dB) RMS relative to $\frac{1}{\sqrt{2}}$ times Full Scale (4.5)

Note 1 to entry: This means $x\ dBFS\ RMS = x + 3.01\ dBFS(AES17)$.

Note 2 to entry: The definition of dBFS(AES17) for the purposes of this standard is the same as dBFS as defined by AES17-2020 for the purposes of its scope, which is for measuring digital audio equipment. Rather than describing a ratio to Full Scale, dBFS(AES17) describes a ratio to the RMS of a sine wave with a peak amplitude equal to Full Scale; the Crest Factor of a sine wave is 3.01 dB. dBFS(AES17) is included in this standard to add clarity and remove ambiguity to the provisions of this standard by acknowledging the two common, but differing, industry use-cases of dBFS.

4.9
White Noise

stochastic signal having a continuous spectrum with equal energy per equal linear interval of frequency

4.10
Pink Noise

stochastic signal having a continuous spectrum with equal energy per equal logarithmic interval of frequency, and with a Gaussian probability distribution of instantaneous amplitude

4.11
Pinking Filter

filter with a −3 dB/octave response that converts White Noise to Pink Noise

5 Calibration Reference Wideband Digital Pink Noise Signal characteristics

A Calibration Reference Wideband Digital Pink Noise Signal shall have the characteristics defined in Table 1 of this standard.

Table 1 — Calibration Reference Wideband Digital Pink Noise Signal characteristics

Characteristic	Value
Unique Signal Period	≥10 seconds
Duration	≥Unique Signal Period
Bandwidth	10 Hz to 22400 Hz (−3 dB points)
Band roll-off:	
- low frequency roll-off	≥21 dB per octave below 10 Hz
- high frequency roll-off	≥36 dB per octave above 22.4 kHz
Crest Factor	11.5 dB to 12 dB (inclusive)
Level	−21.5 dBFS RMS −18.5 dBFS(AES17)
Spectral and temporal uniformity:	
- level of each 1/3-octave band from 20 Hz to 16 kHz	−36.74 dBFS RMS ±0.25 dB −33.74 dBFS(AES17) ±0.25 dB
- level tolerance of any period greater than or equal to the Unique Signal Period	±0.10 dB
- level tolerance of any one-second period	±0.75 dB
- level tolerance of any 125 ms interval	±2.00 dB
Sample rate	48 kHz or 96 kHz
Sample word size	≥24 bits (integer representation)

Annex A
(informative)

Level and characteristics

A.1 Level

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 have been used in movie soundtrack production.

Note 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 dBFS 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.” Note that such an average responding meter is calibrated to indicate the RMS value of a sine wave and such a meter therefore only measures the RMS value of sine waves correctly and will measure the RMS value of other waveforms incorrectly. In contrast, this Standard defines Pink Noise levels with RMS measurement. The difference in meter type affects the numerical level readings. For example, the −20 dBFS level per SMPTE RP 200 reads −22 dBFS RMS (−19 dBFS(AES17)) 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, especially subwoofers. The RMS level of this wider spectrum reads 0.5 dB higher, −21.5 dBFS RMS (−18.5 dBFS(AES17)).

A.2 Statistical criteria

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 state 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. Annex B 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 A.1 below 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 tends to 0 at ±0.35.

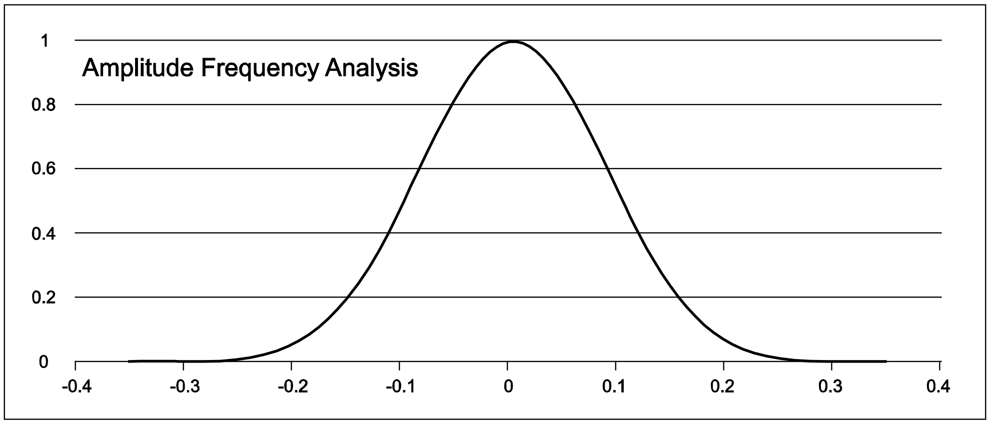

Figure A.1 — Normalized amplitude frequency distribution

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.

A.3 Low frequency roll-off characteristic

The low frequency roll-off characteristics defined in Table 1 can be achieved by 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 can be realized from the continuous time (analog) prototype using the matched pole-zero mapping method (matched Z transform) or equivalent.

A.4 High frequency roll-off characteristic

Note that for sample rates of 48 kHz and 96 kHz, a digital filter realized from a 4^{th}-order, continuous-time Butterworth lowpass filter prototype by means of the bilinear transform with frequency pre-warping will satisfy the high frequency roll-off characteristics as defined in Table 1.

A.5 Signal duration

To extend the duration of the signal for purposes of acoustic measurements, the Unique Signal Period can be looped or repeated as required. Alternatively, the Unique Signal period can be extended beyond the minimum prescribed Duration as defined in Table 1.

A.6 Measurement implications of signal duration

In acoustic measurements, factors such as reverberation and background noise will often 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 might need to be longer for the measurement signal to converge to the specified tolerances.

Annex B
(informative)

Calibration Reference Wideband Digital Pink Noise Signal algorithm

B.1 Noise generator

Annex C describes an available executable noise generator program, and Annex D 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.

B.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, 1st-order lowpass filters. The bandpass filter is comprised of a 4th-order highpass and lowpass filter, implemented as two pairs of 2nd-order (biquad) filters.

B.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 C
(informative)

Example Band-Limited Pink Noise generator executable script

C.1 Python script

Included with this standard is the file ST-2095-1-noise-generator_v1-4.py. This file is an executable computer program in the Python 2.7 or Python 3 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-2095-1-noise-generator_v1-4.py example-file.wav
```

The generated WAV file will contain a single channel of noise at a sample rate of 48 kHz, 24 bits per sample, or a single channel of noise at a sample rate of 96 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.

C.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, and 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} through 2^{24} at sample rates ranging from 44100 to 96000 Hz.

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 from 2 to 4. Once this bit sequence is recast to a floating-point number, subtracting 3.0 from the result yields a fractional number from −1 to 1. See the example in Figure C.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
		ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ	ǀ
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 C.1 — Example of bitwise to floating-point conversion

Annex D
(informative)

Pseudocode listing for band-limited Pink Noise generator

```
// Band-limited Pink Noise Generator
// Produces band-limited, Pink Noise from pseudorandom numbers
// Inputs:
//     int SampleRate:       the output sample rate in samples/sec
//     int Period:           the periodicity of the output signal in samples
//                           must be a power of 2 in length, <= 2^31. Typical
//                           values are 524288 or 1048576.
//     float HpFc:           the highpass filter cutoff frequency in Hz
//     float LpFc:           the lowpass filter cutoff frequency in Hz
//     float MaxPeak:        limit threshold for peaks in dBFS (±1.0 = 0 dBFS)

// convert MaxPeak to amplitude
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; // Starting value for PRNG; any positive integer < 2^31
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; // (precalculating k^2 makes for cleaner code)

// 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 Pinking 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; // "&" Denotes a bitwise AND
// Scale to a real number in the range -1.0 <= white <= 1.0
white = float(seed) * scaleFactor - 1.0;

// Run Pinking Filter; a parallel network of first-order lowpass filters, scaled to
// produce an output signal with target RMS = -21.5 dBFS (-18.5 dBFS(AES17))
// 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 E
(informative)

Measuring Pink Noise amplitude

E.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 to constructive and destructive interference affecting the sound pressure level (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 possess 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.

It is possible to create a sine wave whose positive peaks reach 0 dBFS in the digital domain and to translate this level to a known quantity in the analog domain. That quantity is equivalent to 0 dBFS(AES17). Since the Crest Factor of a sine wave is 3.01 dB, the RMS of this sine wave is equivalent to -3.01 dBFS. The RMS value of any analog waveform emanating from a digital system can thus be quantified relative to 0 dBFS(AES17), and by extension to 0 dBFS.

E.2 Sine wave and Pink Noise amplitude considerations

The normative noise signal level criteria described in Clause 5 are based on digital measurements made with 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 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 rectifying the analog signal 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.

E.3 Metering considerations

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 F
(informative)

Explanation of parameter and tolerance choices

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

Crest Factor: Crest Factor describes the relative value of a signal's absolute peak referenced to the signal's RMS value. It was felt important to constrain this parameter because leaving it unbounded meant a random sample could potentially have a value at or beyond FS, which could cause overload or clipping of the signal chain. Constraining the Crest Factor to 12 dB sets 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 the Pinking Filter. Constraining it to 12 dB involves only minimal removal of over-valued samples from the bitstream.

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

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, ensuring 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 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.

Low frequency roll-off characteristics: In keeping with the bandwidth constraints, 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.

High frequency roll-off characteristics: Also in keeping with the bandwidth requirements, 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.

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.

Sample rate: Both 96 kHz and 48 kHz sample rates are accommodated in this standard in order to be compliant with the provisions of SMPTE ST 428-2 DCDM Audio Characteristics.

Sample word size: In keeping with DCI audio specifications, a PCM word size of 24 bits is specified as the nominal data format for the calibration Pink Noise standard.

Minimum Unique Signal Period: Two factors were considered 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 indicates 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.

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.

Annex G
(informative)

Additional elements

Table G.1 lists non-prose elements of this document.

Table G.1 — Non-prose element

Non-prose element	Description
a	st2095-1a-2023.zip (informative) Containing: 1. ST-2095-1-noise-generator_v1-4.py Band-Limited Pink Noise Generator Executable Script, details as stated in Annex C Containing: Two versions of Calibration Reference, Band-Limited, Wideband Digital Pink Noise Signal as .wav files; filters: 10 to 22400, 10 seconds. 1. 96 kHz 96000_10-22400_2048k_-18_5.wav 2. 48 kHz 48000_10-22400_512k_-18_5.wav

Bibliography (informative)

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

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.

SMPTE ST 202:2010 - Motion-Pictures — Dubbing Theaters, Review Rooms and Indoor Theaters — B-Chain Electroacoustic Response.

