SMPTE RECOMMENDED PRACTICE

RP 155-1990

Audio Levels and Indicators for Digital Audio Records on Digital Television Tape Recorders

1 Scope

This practice specifies a reference amplitude to be used for the calibration of audio level indicators and to be recorded on the digital audio records of reference tapes intended for digital television tape recorders to facilitate the interchange of digital television tape recordings.

2 Normative references

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

ANSI S4.40-1985, Digital Audio Engineering — Serial Transmission Format for Linearly Represented Digital Audio Data

CCITT J.14, Relative Levels and Impedances on an International Sound-Programme Connection

3 Specifications

3.1 The reference signal shall be the digital representation of a 1000-Hz sine wave whose positive peaks attain the value 0CCD (hex) and whose negative peaks attain F333 (hex), considering only the 16 most significant bits. This is 20 dB below the system maximum.

3.2 Where present in a system in analog form, the reference signal shall be a 1000-Hz sine wave with an amplitude such that the system analog-to-digital converter generates a digital representation as in 3.1 above, ± 0.1 dB.

3.3 Digital television tape recorders which have level-indicating instruments with ballistics akin to the vu meter, as described in IEEE Std 152-1953, shall have a standard calibrated reading of zero when the reference signal is being recorded or reproduced.

3.4 Digital television tape recorders which have level-indicating instruments with ballistics akin to the peak program meter shall deflect to a point 9 dB below the reference mark corresponding to the maximum peak program signal according to CCITT Recommendation J.14.

3.5 When transmitting according to ANSI S4.40-1985, the most significant bit of the reference signal shall coincide with the most significant bit of the audio sample data.

3.6 Preemphasis shall not be used when recording the reference signal.

Annex A (informative)
Bibliography

IEEE Std 152-1953 (R1971), Volume Measurements of Electrical Speech and Program Waves

Approved December 7, 1990

PROPOSED SMPTE STANDARD

SMPTE 202M

Revision and Redesignation of ANSI PH22.202M-1984

for Motion-Pictures — B-chain Electro-acoustic Response — Dubbing Theaters, Review Rooms, and Indoor Theaters

1 Scope

This standard specifies the measurement methods and characteristic electro-acoustic frequency responses of the B-chain of motion-picture dubbing theaters, review rooms, and indoor theaters whose room volume exceeds 150 m^3 (5297 ft^3). It is intended to assist in standardization of reproduction of motion-picture sound in such rooms. It does not apply where the recorded sound is intended for reproduction under domestic listening conditions, i.e., for radio broadcasting, television broadcasting, tape, or disc. This standard does not cover that part of the motion-picture sound system from the transducer to the input terminals of the main fader, nor does it cover the electro-acoustic response characteristic of motion-picture theater sub-bass loudspeakers (sub-woofers).

2 Normative reference

The following standard contains provisions which, through reference in this text, constitute provisions of this standard. At the time of publication, the edition indicated was valid. All standards are subject to revision, and parties to agreements based on this standard are encouraged to investigate the possibility of applying the most recent edition of the standard indicated below.

ANSI S1.13-1971 (R1986), Methods for Measurement of Sound Pressure Levels

3 Definitions

3.1 complete sound reproduction system: Represented diagrammatically in figure 1 and used in indoor theaters and review rooms and in motion picture sound post-production facilities such as dubbing theaters, mix rooms, and ADR studios. The complete system is generally considered to consist of an A-chain and a B-chain.

3.2 preemphasized audio track: An audio record, either magnetic or photographic, containing high-frequency boost equalization which is intended for playback over deemphasized theater playback systems to curve N of this standard. A deemphasized theater playback system makes use of a combination of projector slit height losses, electrical filters, loudspeaker frequency response, screen losses, and auditorium acoustics to roll off high-frequency noise due to the magnetic or optical sound track.

3.3 wide-range audio track: An audio record, either magnetic or photographic, which is intended for playback over theater playback systems aligned to curve X of this standard. Such a track is typically recorded without fixed pre- and deemphasis. Frequently, companding noise reduction is employed, and is used with complementary decoding in playback, to reduce the effects of noise due to the magnetic or optical sound track.

3.4 A-chain (transducer system): That part of a motion-picture audio system, as shown in figure 1, extending from the transducer to the input terminals of the main fader.

3.5 B-chain (final chain): That part of a motion-picture sound reproduction system, as shown in figure 1, commencing at the input terminals of the main fader and terminating in the listening area defined in figures 3 and 4 at which sound pressure measurements are taken.

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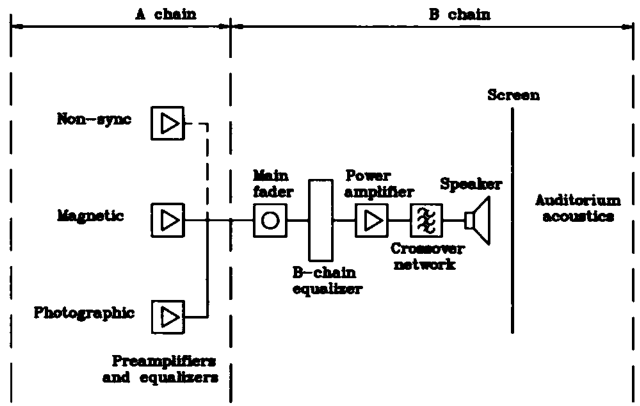


Figure 1 – Complete theatrical sound reproduction system

3.6 pink noise: A stochastic signal having a continuous spectrum with equal energy per equal logarithmic interval of frequency, and with a Gaussian probability distribution of instantaneous amplitude. (See 4.4.)

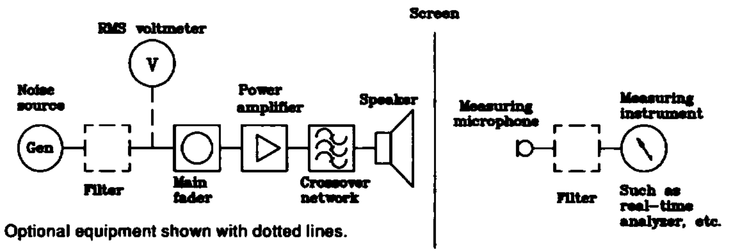
3.7 wide-band pink noise: Pink noise having a bandwidth exceeding the normal acoustic frequency range. A suitable test signal should have a frequency response flat to within ±0.5 dB when measured in ⅓-octave bands with center frequencies from 25 Hz to 20 kHz with an integrating averaging technique.

3.8 electro-acoustic response: The electro-acoustic response of the B-chain is the spatially averaged sound pressure level measured in ⅓-octave bands expressed in decibels with respect to reference level when wide-band pink noise is applied to the input terminals of the main fader, preceding the power amplifier. (See figure 1.) The reference level is the average level of the electro-acoustic response of the ⅓-octave bands over the frequency range from 400 Hz to 2 kHz. The electro-acoustic response is computed as a spatial average over the listening area using one of the methods given in annex A.4.

4 Method of measurement

4.1 The electro-acoustic response shall be measured with the equipment arranged in accordance with figures 2, 3, and 4. (See annex A.)

4.2 Sound pressure level (SPL) vs. frequency measurements (see annex A) shall be made as follows:



NOTE – Optional equipment shown with dotted lines.

Figure 2 – Method of measurement of B-chain

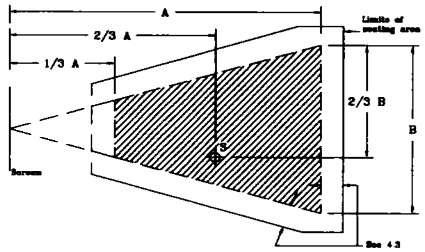


Figure 3 - Theater auditorium

(a) In dubbing theaters or mix rooms, at each of the principal listening positions, such as at the position of each of the mixing personnel, and at the producer's location.

(b) In review rooms and review theaters, at a sufficient number of positions to cover the listening area and to reduce the standard deviation of measured position-to-position response to less than 3 dB, which will typically be achieved with four positions.

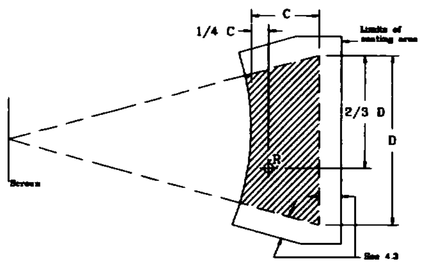


Figure 4 - Theater balcony

(c) In indoor theaters, at position S as shown in figure 3 and position R as shown in figure 4 should it exist, and at a sufficient number of other positions to reduce the standard deviation of measured position-to-position response to less than 3 dB, which will typically be achieved with four positions, but avoiding those aberrant locations described in A.4.

4.3 It is recommended that measurements be made at a normal seated ear height between 1.0 m and 1.2 m (3.3 ft and 4.0 ft), but not closer than 150 mm (6 in) from the top of a seat, and not closer than 1.5 m (4.9 ft) to any wall and 5.0 m (16.4 ft) from the loudspeaker(s).

Table 1 - B-chain characteristics

Center frequencies of ⅓-octave bands	Level Curve N	Curve X	Tolerances +	-
Hz	dB		dB	
40	−8	−2	3	7
50	−5	−1	3	6
63	−3	0	3	5
80	−1	0	3	4
100	0	0	3	3
125	0	0	3	3
160	0	0	3	3
200	0	0	3	3
250	0	0	3	3
315	0	0	3	3
400	0	0	3	3
500	0	0	3	3
630	0	0	3	3
800	0	0	3	3
1000	0	0	3	3
1250	0	0	3	3
1600	0	0	3	3
2000	0	0	3	3
2500	−1	−1	3	3
3150	−2	−2	3	3
4000	−3	−3	3	3
5000	−5	−4	3	3
6300	−8	−5	3	3
8000	−11	−6	3	3
10000	−14	−7	3	4
12500	—	−9	3	5
16000	—	−11	3	6

4.4 A suitable single loudspeaker auditorium sound pressure level with wide-band pink noise is 85 dB SPL C-weighted and slow reading. (See annex A.10.) The measured level in any ⅓-octave band can be used directly if it exceeds the background noise in the band by at least 10 dB. If the background noise is between 4 dB and 10 dB below the test signal, the measurement may be corrected using the techniques described in ANSI S1.13-1971, table 4. (See annex A.4.)

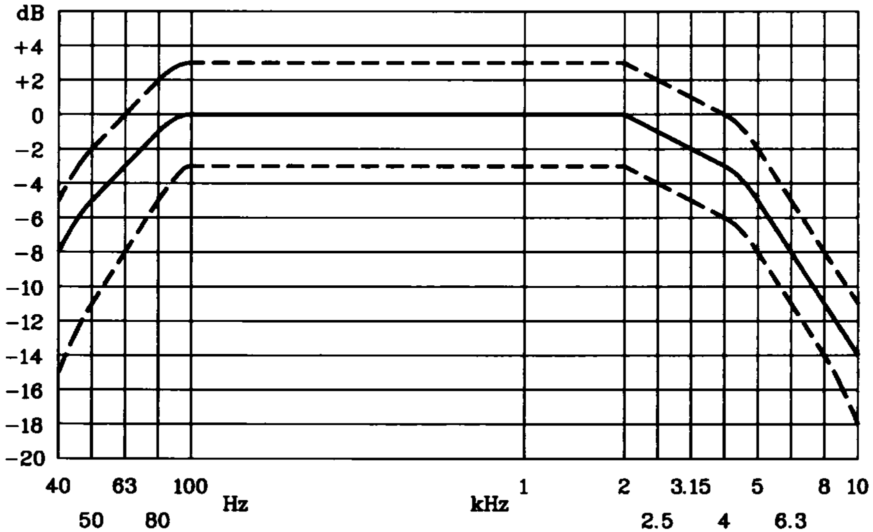
4.5 A system for playing contemporary stereo films will generally employ four wide-range channels: screen left, center, and right loudspeaker systems, and a surround channel loudspeaker system generally employing a number of individual loudspeakers spaced around the room for uniform coverage. Each of these channels is to be measured and adjusted in turn.

5 Characteristic amplitude responses with respect to frequency

5.1 With the sound system set for playback of preemphasized audio tracks, the electro-acoustic response of the B-chain shall be to curve N in table 1 and figure 5 within the tolerances given.

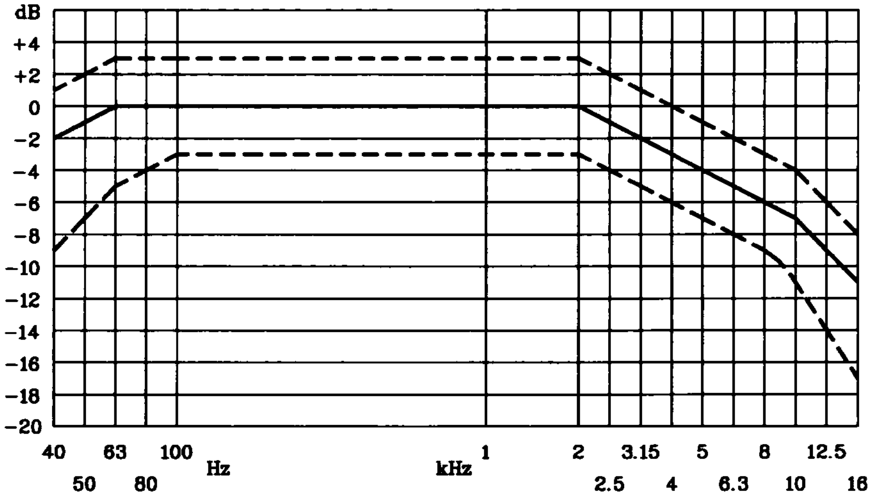
5.2 With the sound system set for playback of wide-range audio tracks, the electro-acoustic response of the B-chain shall be to curve X in table 1 and figure 6 within the tolerances given.

5.3 It is recognized that many older sound systems currently present in theaters cannot meet the centerline of the standard over the fully extended frequency range. The response standard has been updated over the years to account for the changes in technology which permit a wider frequency range, but note the precaution on excessive equalization of these older systems in A.8.



NOTE – Tolerances are based upon ⅓-octave measurements. If ⅟₁-octave measurements are used, reduce the tolerance by 1 dB.

Figure 5 - Curve N of B-chain characteristic



NOTE – Tolerances are based upon ⅓-octave measurements. If ⅟₁-octave measurements are used, reduce the tolerance by 1 dB.

Figure 6 - Curve X of B-chain characteristic

Annex A (informative)
Additional data

A.1 Factors outside the scope of this standard

Compliance with this standard is a necessary but not sufficient condition for the achievement of high-quality sound reproduction in review rooms and theaters. Subjective judgments of sound quality are influenced not only by the frequency response of the B-chain which is the subject of this standard, but also by such factors as:

(a) A-chain performance, including frequency response, signal-to-noise, wow and flutter, and the like;

(b) electrical performance of the sound system, including headroom to clipping, hum and noise, and the like;

(c) room acoustics, including reverberation time vs. frequency, echoes and behind screen reflections, background noise, and intrusive noise;

(d) placement of loudspeaker sources vs. picture;

(e) loudspeaker distortion, and many others.

Therefore it is essential that the A-chain be correctly aligned within the tolerances of existing standards by the use of the appropriate photographic or magnetic test film and that relevant electrical deemphasis be applied (see annex B). For monitoring during recording, where magnetic masters are prepared with preemphasis for making photographic negatives, see SMPTE 214M-1984. The other factors listed above should also be given due attention.

This standard was prepared in the belief that an extended and uniform frequency response is a fundamental component of good sound quality.

A.2 Preliminary checks

Preliminary checks for gross acoustic errors should be made prior to measuring the electro-acoustic response as described in this standard. Typical checks include verification that the loudspeaker being measured is close enough to the screen to avoid any behind-screen echoes, and verification of speaker polarity. A wide-band pink noise test signal can be sent to combinations of loudspeakers (L and C, L and R, C and R) as a simple verification of consistent loudspeaker polarity. The correct polarity (in-phase) condition is the one producing the greater bass response from the sum.

Evaluation of uniformity of loudspeaker distribution patterns can be crudely evaluated by ear using a wide-band pink noise test signal. A more exhaustive numerical analysis of uniformity can be derived by analyzing the point-to-point response as measured in A.5.

A.3 Theater changes from curve N to curve X

If a theater wishes to change from curve N to curve X, it is necessary to make suitable adjustments to the photographic A-chain in order to reproduce conventionally recorded audio records. A companding noise reduction system is normally used for recording and reproducing photographic tracks when used with curve X.

A.4 Qualifying the accuracy of measurements

A.4.1 Type of measurement

Measurements of sound fields from loudspeakers in rooms can take many forms. Tone burst, fast Fourier transform, time-delay spectroscopy, and maximum length sequence analysis may all prove useful, especially during the design phase of a loudspeaker system. Much of the analysis conducted with these methods has the object of reducing the effect of room acoustics on the measurements. Analysis of pink noise with a constant-percentage bandwidth real-time spectrum analyzer, such as a ⅓-octave real time analyzer, on the other hand, includes the influence of room acoustics and has been found to be most useful in day-to-day alignment of sound systems. Traditional real-time analysis has been improved in reliability by the method outlined in this standard through the use of spatial and time averaging, which can yield typical differences as small as ± 1 dB from one setup of the equipment to another.

A.4.2 Background noise

See 4.4 above.

A.4.3 Maximum sound pressure level caution

The sound pressure level shall not be so great as to risk damage to loudspeakers.

A.4.4 Microphone response, directivity, and mounting

The microphones used for theater measurements are subjected to three sound fields, all of which must be measured appropriately. They are the direct sound field from the loudspeaker, early reflections, and the reverberant sound field. Substantial errors can be introduced by using microphones which have large diaphragms, or which have cavities in front of the diaphragm, primarily because their response to direct sound fields and diffuse sound fields are different. Therefore, small diameter calibrated microphones are preferred for accuracy over large diameter types, but large diameter ones can be used for approximate conditions so long as their calibration is known, and the angle of incidence of the direct sound is equal to that of the calibration conditions. (But there may well be a difference in calibration for screen vs. surround loudspeaker systems due to the different nature of the sound fields from these sources.)

Pressure calibrated measurement microphones are preferred to free-field types, since free-field microphones are generally used for measurements where sound from one direction predominates, such as in anechoic measurements. Pressure measurement microphones are typically adjusted for flat response for diffuse-field sound, and their response rises on axis. Since many measurements are made of typical systems at around the critical distance, where the sound pressure contribution of the direct and reverberant sound fields are equal, it is important to find that angle between the direct sound and the diaphragm for which the response is the flattest. This angle is 90° in typical 12.5 mm (0.49 in) pressure measurement microphones, so they would normally be used pointed at 90° relative to the direct sound. Using even smaller diameter microphones has the advantage of reducing the difference in response of on-axis sound and diffuse-field sound. Using typical recording microphones causes problems, since their calibration for mixed sound field conditions is usually unknown.

Some microphone mounting hardware and configurations in common use may cause errors up to ± 2 dB in measured frequency response of the direct sound, due to sound reflections from the mounting equipment entering the capsule. The best mounting hardware has small dimensions and is arranged so that first order reflections from it are reflected away from the microphone capsule.

The frequency response of the measurement microphone shall be known through calibration under conditions similar to its use. In particular, the measurement microphone shall be adequately omnidirectional and calibrated to be flat when measuring a mixture of direct and diffuse sound fields using the same mounting arrangement used in practice, and the angle of flattest direct field response shall be known from the calibration procedure and employed in making measurements.

A.4.5 Spatial averaging

A spatial average of different positions within the room, yet falling within the placements given in 4.2 and 4.3, greatly improves the reliability of equalizing the sound system, due to lessening the influence of specific room modes in the bass, and reducing the effect of lack of uniformity of high-frequency output of loudspeakers in the treble.

Care should be taken that none of the microphone placements used in calculating the spatial average are extraordinary. Positions should be avoided which are exactly on lateral or transverse theater centerlines, or are under the lip of a balcony. Microphone positions employed in a spatial average shall be distributed among a range of positions in lateral and transverse directions to minimize the influence of any particular room mode, but the points should lie within the requirements of 4.2 and 4.3. The minimum spacing of the microphones in an average shall be 1.0 m (3.3 ft).

The calculation of a spatial average shall be done by the sum of the squares of the sound pressure levels as follows:

$$L = 10\log_{10}\left[\frac{1}{N}\sum_{k=1}^{N}\text{antilog}_{10}\left(\frac{L_k}{10}\right)\right]$$

where N is the number of positions and L_k is the sound pressure level at each position. For four positions, the ⅓-octave by ⅓-octave average would be computed as follows:

$$SPL = 10\log_{10}\left[\left(\frac{1}{4}\right)\left(10^{\frac{L_1}{10}} + 10^{\frac{L_2}{10}} + 10^{\frac{L_3}{10}} + 10^{\frac{L_4}{10}}\right)\right]$$

where L_1 equals the sound pressure level in a ⅓-octave band at position 1, L_2 equals the sound pressure level in the same ⅓-octave band at position 2, etc. If the range of sound pressure levels is within 4 dB, simple arithmetic averaging may be used. Large standard deviations may indicate significant acoustic or loudspeaker coverage problems.

A.4.6 Temporal averaging

Stochastic signals such as pink noise cause a fluctuating sound pressure level. The level fluctuations become more severe as the bandwidth of measurement is decreased and as the center frequency of the measurement is lowered. In order to obtain high accuracy with such a nonsteady-state test signal, it is useful to perform temporal averaging on the data obtained from a ⅓-octave band spectrum analyzer. At least two methods are widely used for temporal averaging, RC-type averaging in the detector circuit of the analyzer, and calculated averaging in an integrating real-time analyzer. With a calculated averaging method, accuracy can be very high if the measurement is adequately long. The minimum averaging time of a conventional real-time analyzer should be such that measurements even at low frequencies are readable with an accuracy better than the tolerances of the standard. It is recommended that measurements be time-averaged over a period of not less than 20 s in the lowest frequency bands for accuracy within ± 1 dB.

A.5 Methods of measurement

At least two methods of measurement are recognized as providing appropriate data for the evaluation of the electro-acoustic response of the B-chain. For each of the methods, generate wide-band pink noise, and apply to each loudspeaker channel, left, center, right, and surround, in turn. The methods of measurement are:

(a) Measure the electro-acoustic response with a set of four typically calibrated microphones connected to a microphone multiplexer switch (not a mixer), the output of which is connected to an audio-frequency ⅓-octave band spectrum analyzer. Position the set of calibrated microphones according to A.4.5. Temporally average the data for a sufficient amount of time to produce a standard deviation under 1 dB.

(b) Measure the electro-acoustic response with a calibrated microphone and an audio-frequency ⅓-octave-band spectrum analyzer at each of a number of locations and compute the spatial average, as specified in A.4.5.

Other methods which conform to the accuracy of the given methods may be employed, such as use of a ⅓-octave band filter set and a voltmeter, measuring each ⅓-octave band level for each response position in turn and mathematically computing the averages. Measurement in whole-octave bands is now rarely employed, because of the ready availability of ⅓-octave analysis equipment.

Note that the pink noise source should be an electrical noise generator, not an optical or a magnetic pink noise test film, since the use of test films in aligning the B-chain will cause accumulating errors, and in many theaters, the active or passive A-chain deemphasis cannot easily be disabled.

A.6 Acoustical and psychoacoustical effects

The electro-acoustic response resulting from a loudspeaker situated behind a motion-picture screen, or from an array of loudspeakers used for the surround sound channel in the auditorium, is affected by various factors before the sound is perceived by a listener. These include the following acoustical and psychoacoustical effects:

(a) Attenuation of high frequencies caused by the screen for the wide-band screen channels. With conventional theater loudspeakers and normally perforated screens, the attenuation on axis is 3 dB at between 5 kHz and 8 kHz, with a 6 dB/octave characteristic rolloff at higher frequencies. This high-frequency attenuation is typically less off axis by a small amount. Old screens, where acoustic transmission is degraded by air-borne particulate matter clogging the perforations, can severely degrade high-frequency performance.

(b) A room gain reverberation component added to the direct signal. It should be noted that since reverberation in large rooms takes a finite time to build up, this component is only measurable with quasi-steady-state signals, such as pink noise. For accurate measurement, the sound field must have reached stasis for the reverberant component to add fully to the direct sound. This component has a frequency response proportional to the reverberation time vs. frequency characteristic, and will be most significant on sustained program material like held musical chords.

(c) High-frequency attenuation in the air, proportional to signal path length. This process applies to direct sound through only one path, and to the reverberant component through the average composite path length of sound. The result is that with all other factors held constant, for steady-state signals in more reverberant spaces, the sound will be duller, with rolled off high frequencies, since the average path length of the reverberation component is longer in the more reverberant room.

(d) Rows of seats have been shown to cause dips in the frequency response of sound fields at near grazing incidence. The dip is usually in the range of 80 Hz to 125 Hz, depending on the dimensions.

(e) All published experimenters have found that in a large room, a flat response near-field loudspeaker is subjectively best matched by a distant loudspeaker having a rolled-off high-frequency response in steady-state measurements. The source of the need for this measured rolloff appears to be the differing interaction of the sound field with the head, pinnae, and ear canal between a distantly originating sound field, and one originating close by, or by differences between transient and steady responses caused by the mechanism described in (b) above. Since the need for such an apparent rolloff with steady state signals is shown, this standard documents the response for best interchangeability of product across many auditoriums.

To account for (b) and (c) above, the measured characteristic to maintain subjectively identical response will differ slightly according mainly to auditorium size and reverberation time. The measured response by the method of this standard should have a slightly attenuated high-frequency characteristic in a large theater when compared with table 1 and figures 5 and 6, when comparing auditoria using the same absorption/frequency characteristics, due to the naturally longer reverberation time of the larger theater. In the same way, there should be a slightly elevated response in a small theater. Table A.1 gives approximately suitable correction factors which should be added numerically to the characteristic given in table 1 and figures 5 and 6.

Table A.1
Approximate correction factors for auditorium size, dB

Frequency	Number of seats					
kHz	30	150	500	1000	1500	2000
2.0	0	0	0	0	0	0
4.0	1.0	0.5	0	−0.5	−1.0	−1.5
8.0	2.0	1.0	0	−1.0	−2.0	−3.0

Corrections for auditorium size are not normally required below 2 kHz, as a result of a flatter reverberation vs. frequency characteristic typical at mid-range frequencies, and the longer integration time of the ear at low frequencies. More accurate determination of the above correction factors for a particular auditorium can be deduced from measurement of the reverberation time vs. frequency characteristic.

Whenever possible, the electro-acoustic response should be measured with the auditorium's typical operational humidity, since humidity variations are a significant component of day-to-day variation in properly operating modern sound systems.

A.7 Troubleshooting

With good acoustic design and modern loudspeakers with uniform coverage, not only should the overall spatially averaged electro-acoustic response fall within the tolerances specified in 5, but the response for each position should also fall within those tolerances.

Provided that the B-chain meets the tolerances specified, the electro-acoustic frequency response for sound reproduction should be satisfactory for all types of photographic and magnetic recordings, provided the additional items outlined in A.1 are given attention.

Care should be taken that deviations from the curve, although within the tolerance area, do not cause a tonal imbalance. Broad, low-Q effects have been shown to be of more perceptual significance than narrower, high-Q effects, which have even greater amplitude. For example, a situation where the overall bass response is at one extreme of the tolerance, and the treble response at the other, should be avoided.

If extreme amounts of electrical equalization are required to bring the response into conformity with the standard, or electrical equalization is required which is significantly atypical for the loudspeaker system in use, each element of the B-chain should be examined to determine the cause of the problem, which may be included in the following list:

(a) faulty power amplifier;

(b) incorrect or faulty loudspeaker performance;

(c) incorrect location, orientation, or directivity of the loudspeaker;

(d) room acoustical defects;

(e) incorrect adjustment of the loudspeaker crossover network (relative level of the bass and treble loudspeaker drivers), crossover wiring polarity reversal, relative time displacements between drivers due to different geometries;

(f) obscured perforations in the screen;

(g) dated loudspeaker design, unable to perform according to current specifications.

If the B-chain cannot be brought into conformance with the curve X characteristic described in this standard, but it is found possible to achieve conformance to the curve N characteristic, there still may be playback problems when playing material intended for a curve N environment. Sound records are preemphasized according to the intended playback environment — those intended for playback over curve N systems will have greater high-frequency preemphasis than those intended for curve X systems. This increased preemphasis may reveal power amplifier or loudspeaker deficiencies in a B-chain which are not evident when the system is being tested with a flat frequency response test signal such as pink noise.

Some high-frequency loudspeaker drivers exhibit more distortion than others; this may cause a subjective change in high-frequency response which will not be evident from the test procedures described in this standard.

Because the measurements deal only with the steady-state properties of the auditorium, acoustical defects such as backstage or auditorium echoes are not accounted for in the measurement procedure. Attempts to use these measurement results as a basis for major equipment redesign in a theater found defective have to be preceded by ascertaining that no grave acoustical faults are present. Possible problems are listed in A.1. Methods for finding or eliminating such faults are not covered in this document.

A.8 Equalization

Adjustment of the electro-acoustic response to curve X for record monitoring and playback of wide-range sound tracks will normally require some electrical equalization, typically ⅓-octave, which approximately corresponds to the critical bands of human hearing. The following points should be noted:

(a) A crossover network, if used, should be adjusted, if adjustable, to the smoothest response before equalization is attempted. If an electronic crossover is employed before the power amplifiers, its gain, and the gain of the following power amplifiers, should be set with a due consideration for the best headroom and signal-to-noise ratio.

(b) Equalization above 8 kHz should not normally be attempted with older loudspeaker designs which exhibit rapid rolloff beyond 8 kHz.

(c) Strong equalization of low-frequency modes should be avoided, since the low-frequency modes, particularly below 100 Hz, are often very narrow response anomalies which cannot normally be corrected with the relatively broader ⅓-octave-band equalizer without audible consequences at frequencies away from the modal one.

(d) A correction may be necessary to the basic curve promulgated by this document when equalizing the surround channel due to the differences between screen and surround loudspeakers. Among these differences are the facts that surround channel loudspeakers are routinely placed closer to typical seating locations, increasing the ratio of direct-to-reverberant sound fields; the screen loudspeakers approximate a point source, whereas the surround channel uses an array of loudspeakers; and the radiation patterns of the loudspeakers are different between screen and surround loudspeakers. The correction necessary will tend towards an increase in the measured high-frequency response of surround loudspeakers compared to screen loudspeakers.

A.9 Uniformity of sound pressure level from different loudspeakers

Applying the same voltage of wide-band pink noise at the input of the B-chain for each loudspeaker system in turn should yield spatially averaged sound pressure levels which are equal within ± 1 dB (see A.4) as measured on a C-weighted and slow reading sound level meter. In the event that the auditorium exhibits strong low-frequency room modes which could significantly affect wide-band noise level measurements, A-weighting should be used when matching loudspeaker levels.

A.10 Spectrum level

While 85 dBC is commonly used as a wide-band signal level reference (see 4.4), on some analyzers it may be convenient to use as a preferred sound pressure a decade level such as 70 dB SPL in each ⅓-octave band in the range 50 Hz to 2 kHz. In this case, with the overall response on curve X, the sum of all the one-third octave bands will add to approximately 83 dBC.

A.11 Future work

As first noted in A.6, it is recognized that more uniform perceived loudness and spectral balance from installation to installation would be promoted by accounting for the following factors:

(a) reverberation time vs. frequency;

(b) loudspeaker radiation pattern vs. frequency;

(c) the consequent direct-to-reverberant ratio at listening positions within the space due to (a) and (b);

(d) an accurate method to quantify the measured difference in frequency response between surround and screen channels as described in A.8.

The goal is to have constant perceived loudness and frequency response from installation to installation, and from position to position within an installation, using this standard and contemporary dubbing stage practice as a reference. This goal may be promoted by keeping constant within the ear canal the spectrum level and response throughout one and across many installations. As equipment becomes available capable of making reverberation time measurements, the results of such measurements, combined with information on the loudspeakers, may be used to add correction factors to the curves to promote greater uniformity.

Annex B (informative)
Bibliography

ANSI S1.11-1986, Specifications for Octave-Band and Fractional Octave-Band Analog and Digital Filters

ANSI PH22.208M-1984, Motion-Picture Film (35-mm) — Magnetic Audio Reproduction Characteristic

ANSI/SMPTE 214M-1984, Motion-Picture Film (35-mm) — Photographic Audio Reproduction Characteristic

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