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Cinematography — B-chain electro-acoustic reponse of motion-picture control rooms and indoor theatres — Specifications and measurements

Réponse électro-acoustique de la chaîne B des salles de contrôle et d'exploitation cinématographique — Spécifications et mesurages

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Foreword

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The procedures used to develop this document and those intended for its further maintenance are described in the ISO/IEC Directives, Part 1. In particular the different approval criteria needed for the different types of ISO documents should be noted. This document was drafted in accordance with the editorial rules of the ISO/IEC Directives, Part 2 (see www.iso.org/directives).

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For an explanation on the meaning of ISO specific terms and expressions related to conformity assessment, as well as information about ISO's adherence to the WTO principles in the Technical Barriers to Trade (TBT), see the following URL: Foreword — Supplementary information.

The committee responsible for this document is ISO/TC 36, *Cinematography*.

This third edition cancels and replaces the second edition (ISO 2969:1987), which has been technically revised.

Introduction

This International Standard is to be used in conjunction with the relevant standards which cover that part of the motion picture sound system from the transducer to the input terminals of the main fader.

In this International Standard, 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 labelled 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 are as follows: Normative prose is the authoritative definition; Tables are next; followed by formal languages; then figures; and then any other language forms.

Cinematography — B-chain electro-acoustic reponse of motion-picture control rooms and indoor theatres — Specifications and measurements

1   Scope

This International Standard specifies the measurement methods and characteristic electroacoustic frequency response of the B-chain of motion-picture dubbing theatres (mixing rooms), screening rooms, and indoor theatres whose room volume exceeds 125 m^3 (4,414 ft^3). It is intended to assist in standardization of monitoring and reproduction of motion-picture sound in such rooms. The goal is to have constant perceived loudness and frequency response from installation to installation, and from position-to-position within an installation. This International Standard does not cover that part of the motion-picture sound system extending from the transducer to the input source audio selector.

2   Terms and definitions

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

2.1
complete sound reproduction system
system used in indoor theatres and screening rooms and in motion-picture sound post-production facilities such as dubbing theatres, mix rooms and ADR control rooms

Note 1 to entry: The complete system in an indoor theatre or review room is generally considered to consist of an A-chain and a B-chain.

Note 2 to entry: Represented diagrammatically in Figures 1 and 2.

2.2
pre-emphasized audio track
audio record, either magnetic or photographic, containing high-frequency boost equalization, which is intended for playback over de-emphasized theatre playback systems

Note 1 to entry: Now very rarely used, but found on all films prior to the mid-1970s. Part of the playback de-emphasis was generated by use of Curve-N in previous versions of this standard (see 2.10 and A.10).

2.3
wide-range audio track
audio record, either magnetic, analogue photographic or digital, which is intended for playback over theatre playback systems aligned to this International Standard

Note 1 to entry: This characteristic was previously referred as Curve-X (see 2.9). Such tracks are recorded without fixed pre- and de-emphasis. Analogue wide-range soundtracks invariably use noise reduction companding technology.

2.4
A-chain (transducer system)
part of a motion-picture audio system extending as far as the input source selector, as shown in Figures 1 and 2

2.5
B-chain (final chain)
part of a motion-picture sound reproduction system, as shown in Figures 1 and 2, commencing at the input source audio selector and terminating in the listening area
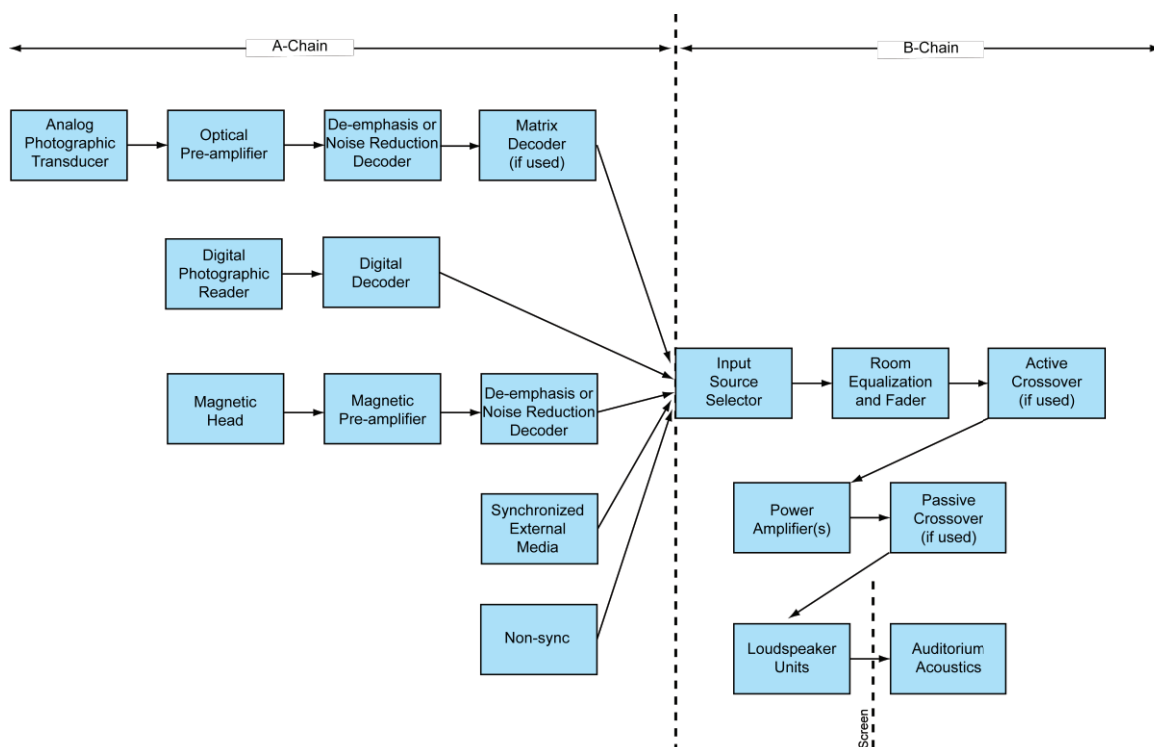
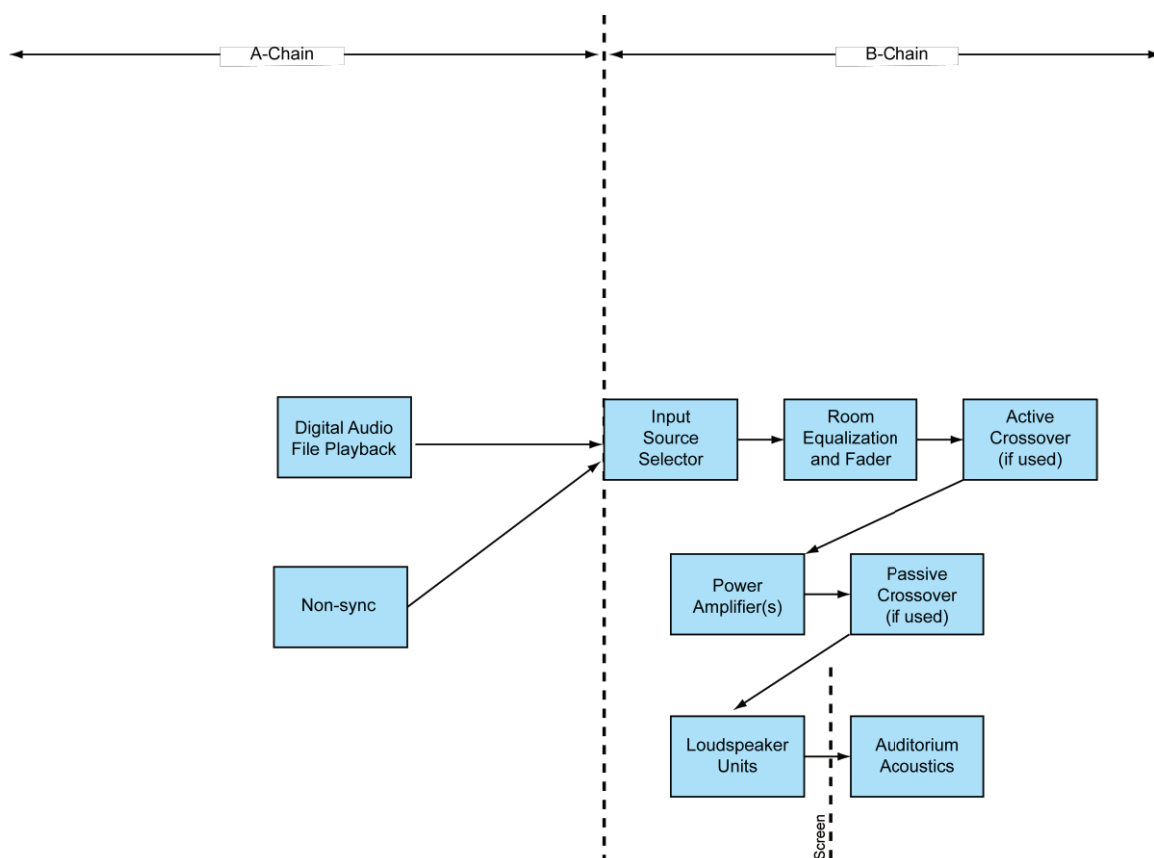**Figure 1 — Complete theatrical audio reproducing chain — Traditional Film Formats**

Figure 2 — Complete theatrical audio reproducing chain — Digital Cinema

2.6
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 (see 3.4)

2.7
wide-band pink noise
pink noise having a bandwidth exceeding the normal acoustic frequency range

Note 1 to entry: A suitable test signal should have a frequency response flat to within ±0,5 dB when measured in 1/3-octave bands with centre frequencies from 25 Hz to 20 kHz with an integrating averaging technique.

2.8
electroacoustic response
<B-chain> spatially and temporally averaged sound pressure level measured in 1/3-octave bands expressed in decibels with respect to reference level (see A.9) when wide-band pink noise is applied to the input source selector (see Figures 1 and 2)

Note 1 to entry: The electroacoustic response is computed as a spatial and temporal average over the listening area using one of the methods given in A.4.

2.9
Curve-X
X-Curve
B-chain characteristic referred to as Curve-X for wide-range sound tracks, also known as X-Curve

Note 1 to entry: This characteristic typically required some high-frequency equalization boost when older loudspeakers were in use, but is now easily achievable with contemporary loudspeakers. All contemporary practice is targeted to the X-Curve.

2.10
Curve-N
B-chain characteristic referred to as Curve-N for use with loudspeakers with much poorer high-frequency response than those typically now in use (see A.10)

3   Method of measurement

3.1   The electroacoustic response shall be measured with the equipment arranged in accordance with Figures 3 and 4 (see Annex A).

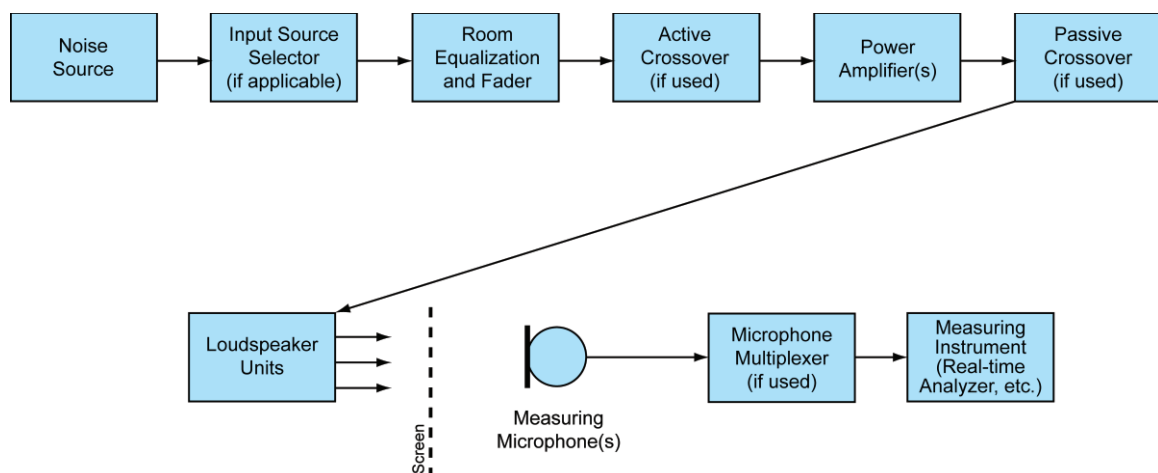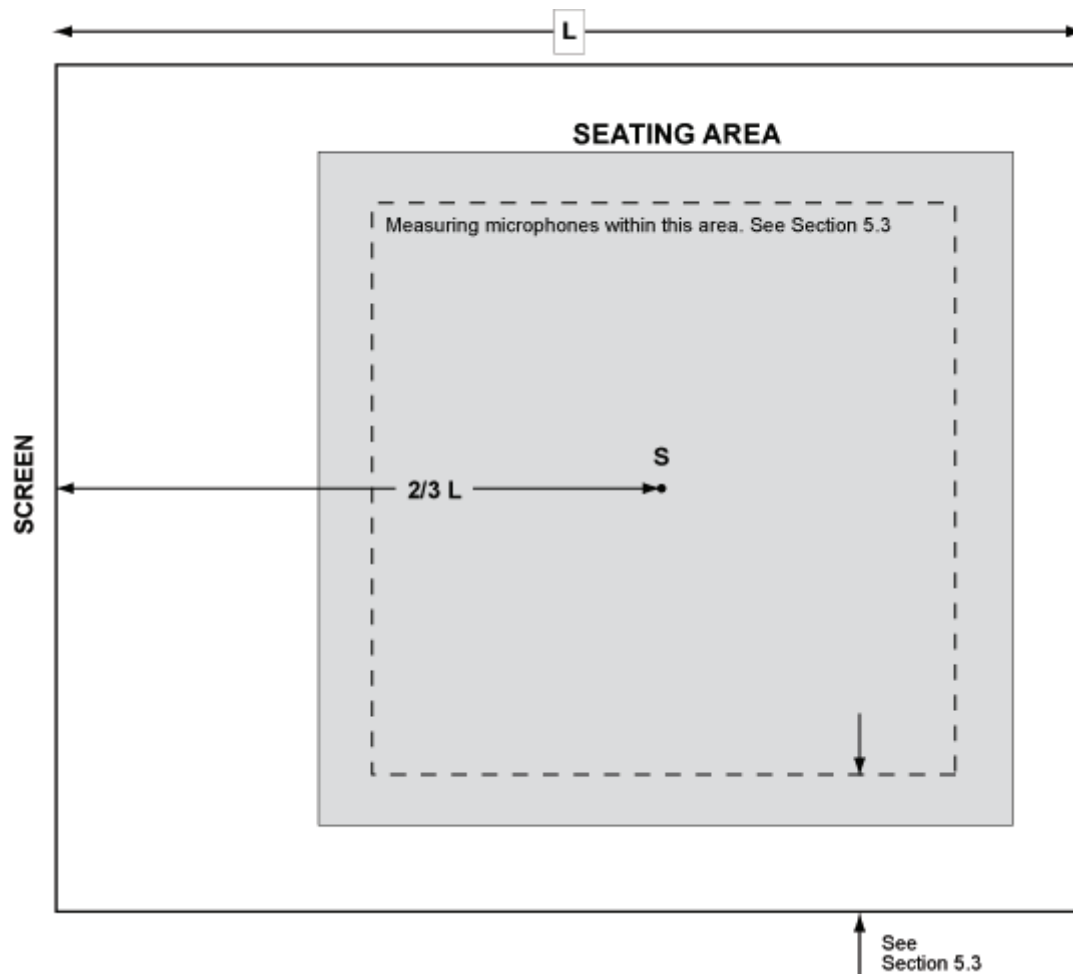**Figure 3 — Method of measurement of B-chain**

Figure 4 — Range of microphone placements

3.2   Sound pressure level (SPL) vs. Frequency measurements (see <u>Annex A</u>) shall be made as follows:

a)   On dubbing stages (in mixing 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. In rooms with a single primary listening position, care should be taken that this is not an aberrant location;

b)   In screening rooms, 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. This will typically be achieved with four positions;

c)   In indoor theatres, at a minimum with the position S as shown in <u>Figure 4</u>, and normally at a sufficient number of additional other positions to reduce the standard deviation of measured position-to-position response to less than 3 dB. This will typically be achieved with four positions (see <u>A.3.5</u>). An extra series of measurement positions will have to be added if the theatre has a balcony.

3.3   Measurements shall 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 not closer than 5,0 m (16.4 ft) from the screen loudspeaker(s).

3.4   A suitable single loudspeaker auditorium sound pressure level with wide-band pink noise is 85 dB SPL C-weighted and slow reading (see <u>A.9</u>).

3.5   The measured level in any 1/3-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/ASA S1.13 (see Table 4).

3.6   A system for playing contemporary stereo films will generally employ a minimum of four wide-range channels: screen left, centre, and right loudspeaker systems, and a surround channel loudspeaker system employing a number of individual loudspeakers spaced around the left wall, rear wall and right wall of the room in such a way as to achieve uniform coverage. Most rooms where digital soundtracks are played have the surround channel separated into two or three separate channels, left rear and right rear, or left rear, centre rear and right rear. Such systems are frequently built up out of left wall, left rear, right rear and right wall banks of speakers. Most of these rooms also have a dedicated low-frequency channel using one or more sub-woofers. Some rooms may be equipped with two intermediate screen channels, one between left and centre, and one between centre and right. Regardless of the number of channels, each channel or bank shall be measured separately in turn and the equalization adjusted if necessary.

4   Characteristic amplitude responses with respect to frequency

4.1   The electroacoustic response of the B-chain for screen and surround channels shall be listed in Table 1 and shown in Figure 5 within the tolerances given. Note that this characteristic is for a medium-sized theatre (with between, say 200 and 500 seats) with average reverberation behaviour. See A.5 for a discussion of modifications required to this characteristic for larger and smaller spaces and for surround loudspeaker arrays.

4.2   It is recognized that there are a few older sound systems still in use in theatres which cannot meet the centreline 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 older systems in A.6.

Table 1 — B-chain screen and surround channel characteristics for medium-sized theatre (see A.5 e) and A.5 f) for modification factors and surround characteristics A.6)

Centre frequencies of 1/3-octave bands HZ	Level dB	Tolerances dB	
		+	-
31,5	-2	3	7
40	-1	3	6
50	0	3	3
63	0	3	3
80	0	3	3
100	0	3	3
125	0	3	3
160	0	3	3
200	0	3	3
250	0	3	3
315	0	3	3
400	0	3	3
500	0	3	3
630	0	3	3
800	0	3	3
1 000	0	3	3
1 250	0	3	3
1 600	0	3	3
2 000	0	3	3
2 500	−1	3	3
3 150	−2	3	3
4 000	−3	3	3
5 000	−4	3	3
6 300	−5	3	3
8 000	−6	3	3
10 000	−7	3	3
12 500	−9	3	3
16 000	−11	3	3
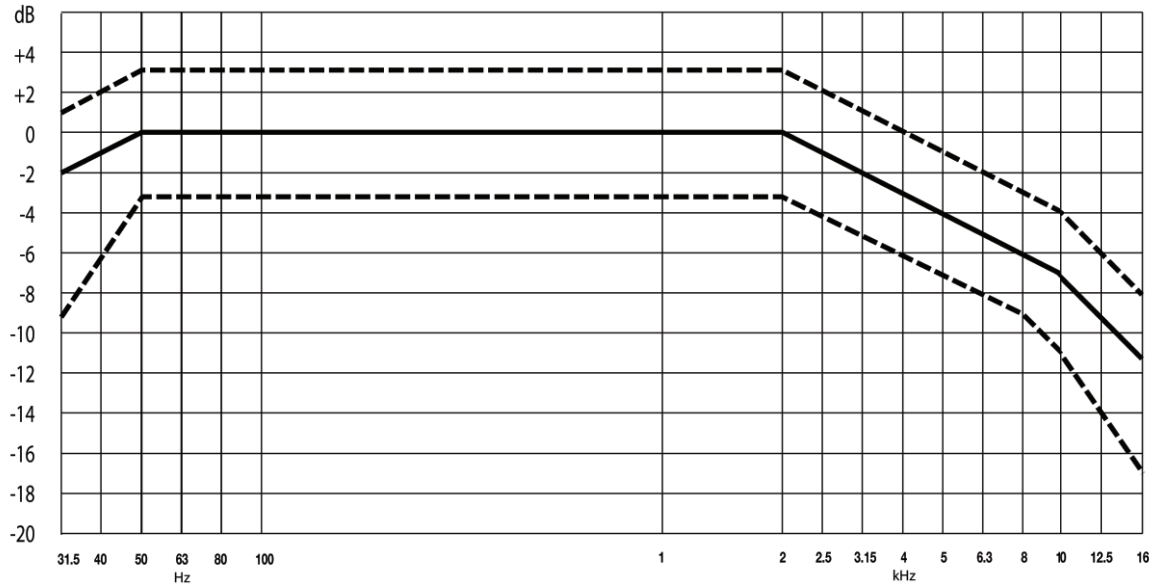
NOTE – Tolerances in this figure are based upon 1/3-octave measurements. If 1/1-octave measurements are used, reduce the tolerence by 1 dB.

Figure 5 — B-chain screen and surround characteristics for medium-sized theatre (see A.5 e) and A.5 f) for modification factors and surround characteristics)

4.3   The bandwidth of the low-frequency effects channel on a modern soundtrack extends from 5 Hz to 120 Hz. A linear sub-woofer acoustic response is desirable from approximately 25 Hz to 120 Hz. The 120 Hz soundtrack cut-off is extremely steep, so a suitable sub-woofer need have little response above 125 Hz. For information on subwoofer adjustment refer to ISO 22234:2005.

Many rooms have one or more dominant resonant frequencies within the low-frequency effects channel bandwidth. If not damped this can lead to a characteristic low-frequency "ringing" every time the soundtrack contains low-frequency information. Most cinema B-chain processors have at least one parametric equalizer for use within the sub-woofer bandpass. After adjustment, the response between 25 Hz and 120 Hz shall be flat to within ±3 dB.

7

Annex A
(informative)

Factors outside the scope of this International Standard

A.1   General

This International Standard was prepared in the belief that an extended and uniform frequency response is a fundamental component of good sound quality in review rooms and theatres.

However, compliance with this International Standard is a necessary but not sufficient condition for the complete achievement of high-quality sound reproduction. Subjective judgments of sound quality are influenced not only by the frequency response of the B-chain (which is the subject of this International Standard), but also by such factors including but not limited to

a)   A-chain performance. For analogue soundtracks, it is important 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 de-emphasis be applied where applicable (see Bibliography). Appropriate alignment includes such parameters as frequency response, signal-to-noise ratio, wow and flutter, track-to-track crosstalk, and the like. While it may be assumed that an A-chain for playback of digital film soundtracks or for digital cinema files has an inherently flat frequency response, quality can still be limited by digital to analogue converters, etc.;

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

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

d)   placement of loudspeaker sources vs. picture; and

e)   loudspeaker distortion, and many others.

A.2   Preliminary checks

It is important that preliminary checks for gross acoustic errors be made prior to measuring the electroacoustic response as described in this International 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 (for example, 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 greatest 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 analysing the point-to-point response as measured in A.3.5.

A.3   Qualifying the accuracy of measurements

A.3.1   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 using

pink noise with a constant-percentage bandwidth real-time spectrum analyser, such as a 1/3-octave real-time analyser, on the other hand, includes the influence of room acoustics and has been found to be most useful and convenient in day-to-day alignment of sound systems. Traditional real-time analysis has been improved in reliability by the method outlined in this International 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.3.2   Background noise

See 3.5.

A.3.3   Maximum sound pressure level caution

Excessive sound pressure levels may risk damage to loudspeakers.

A.3.4   Microphone response, directivity, and mounting

The microphones used for theatre measurements are subjected to three sound fields, all of which will be taken account of in the measuring process. 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 is different. Therefore, small diameter calibrated microphones are strongly preferred for accuracy over large diameter types, but large diameter ones can be used for an approximate analysis 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. [see A.5 f)]).

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 rise 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° with 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 is strongly discouraged, as 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.

It is important that the frequency response of the measurement microphone be known through calibration under conditions similar to its use. In addition it is important that the measurement microphone 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 known from the calibration procedure and employed in making measurements.

A.3.5   Spatial averaging

A spatial average of different positions within the room, yet falling within the placements given in 3.2 and 3.3, 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.

It is important that none of the microphone placements used in calculating the spatial average evidence extraordinary acoustic behaviour. Positions which could cause problems include those exactly on lateral or transverse theatre centrelines, or under the lip of a balcony. Microphone positions employed in a spatial average will 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 3.2 and 3.3. The minimum spacing of the microphones in an average will typically be 1,0 m (3.3 ft).

The calculation of a spatial average can be done by the sum of the squares of the sound pressure levels as in Formula (A.1):

$$L = 10\log_{10}\left[\frac{1}{N}\sum_{k=1}^{N} anti\log_{10}\left(\frac{L_k}{10}\right)\right] \tag{A.1}$$

where N is the number of positions and L_k is the sound pressure level at each position. For four positions, the 1/3-octave by 1/3-octave average would be computed as Formula (A.2):

$$SPL = 10\log_{10}\left[\left(\frac{1}{4}\right)\left(10^{\frac{L_1}{10}} + 10^{\frac{L_2}{10}} + 10^{\frac{L_3}{10}} + 10^{\frac{L_4}{10}}\right)\right] \tag{A.2}$$

where L_1 equals the sound pressure level in a 1/3-octave band at position 1, L_2 equals the sound pressure level in the same 1/3-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.3.6   Temporal Averaging

Stochastic signals such as pink noise produce a fluctuating sound pressure level. The level fluctuations become more severe as the bandwidth of measurement is decreased and as the centre frequency of the measurement is lowered. In order to obtain high accuracy with such a fluctuating-level test signal, it is useful to perform temporal averaging on the data obtained from a 1/3-octave band spectrum analyser. At least two methods are widely used for temporal averaging:

RC-type averaging in the detector circuit of the analyser, and calculated averaging in an integrating real-time analyser. 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 analyser will typically 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 to within ±1 dB.

A.4   Methods of Measurement

There are two preferred methods of measurement for evaluating the electroacoustic response of the B-chain which utilize pink noise as a test signal. For each of the methods, generate wide-band pink noise, and apply to each screen loudspeaker channel, left, centre, right, and each surround channel in turn. The methods of measurement are:

a) Measure the electroacoustic response with a set of four omnidirectional and calibrated microphones connected to a microphone multiplexer switch (not a mixer), the output of which is connected to an audio-frequency 1/3-octave band spectrum analyser. Position the set of calibrated microphones according to 3.2 and 3.3. Temporally average the data for a sufficient amount of time to produce a standard deviation less than 1 dB;

b) Measure the electroacoustic response with a calibrated microphone and an audio-frequency 1/3-octave band spectrum analyser at each of a number of locations and compute the spatial average, as specified in 3.2 and 3.3.

Other methods which conform to the accuracy of the given methods may be employed, such as use of a 1/3-octave band filter set and a voltmeter, measuring each 1/3-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 1/3-octave analysis equipment.

Note that the pink noise source is 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 theatres, the active or passive A-chain de-emphasis cannot easily be disabled.

A.5 Acoustical and psychoacoustical effects

The electroacoustic 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 theatre loudspeakers and normally perforated screens, the attenuation on-axis is typically 3 dB between 5 kHz and 8 kHz, with a 6 dB/octave characteristic roll-off 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 will 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 typical reverberation characteristic falls with increasing frequency above about 2 kHz. As a result, with all other factors held constant, for steady-state signals in more reverberant spaces, the sound will be duller, with rolled off high frequencies. The effect is not present with short-duration transient sounds, where the direct energy has stopped prior to the arrival of any reverberant signal, inevitably with a longer path-length to the listener.

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. This is less likely to occur with modern stadium seating, as the slope results in a more oblique angle of incidence.

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 an apparent high-frequency roll-off when assessed with steady-state measurements. The source of the need for this measured roll-off 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 roll-off with steady-state signals is shown, this International Standard documents the response for best interchangeability of product across many auditoriums. It is important to note that this apparent roll-off relates to the steady-state measurement — depending on the loudspeaker directivity and the measurement microphone, the perceived direct sound from a far-field loudspeaker may well have a flatter response that that measured.

To account for b), c), and e) above, the measured characteristic to maintain subjectively identical response will differ slightly according to auditorium size and possibly reverberation characteristic. It has also been conjectured that a further factor affecting the ideal characteristic is psychoacoustic in nature, relating to the distance from a typical member of the audience to the screen. In consequence of all the above, the measured response by the method of this International Standard will have a slightly attenuated high-frequency characteristic in a large theatre when compared with Table 1 and

Figure 5 (for a medium-sized room), when comparing auditoria using the same absorption/frequency characteristics. In the same way, there will be a slightly elevated response in a small theatre. Table A.1 gives approximate suitable correction factors which are added numerically to the characteristic given in Table 1 and Figure 5, and additionally the range of adjustments is shown in Figure A.1

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 and review of the reverberation time vs. frequency characteristic.

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

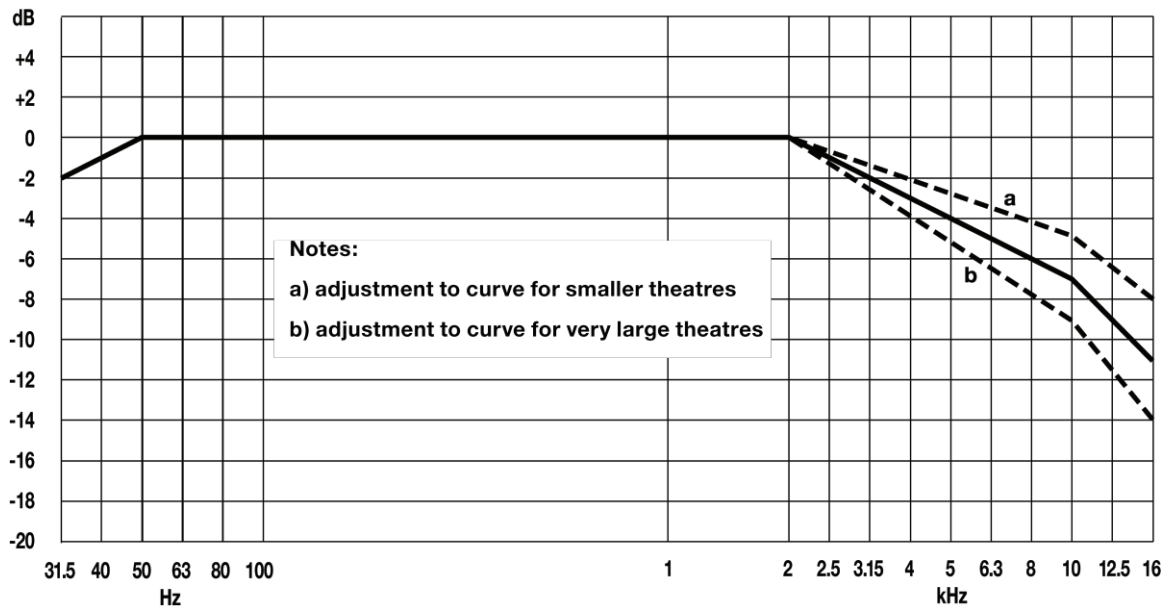| Frequency	Number of seats					
kHz	30	150	500	1 000	1 500	2 000
2,0	0	0	0	0	0	0
4,0	1,0	0,5	0	−0,5	−1,0	−1,5
8,0	2,0	1,0	0	−1,0	−2,0	−3,0

Figure A.1 — Modification of characteristic for auditorium size

f) In a normally shaped quasi-rectangular theatre, most of the audience will be significantly closer to surround loudspeakers than the screen loudspeakers. As a result there will be an increase in the ratio of direct-to-reverberant sound fields. In addition, the radiation patterns of the loudspeakers are different between screen and surround loudspeakers: the screen loudspeakers approximate a point source, whereas the surround channel uses an array of loudspeakers. As a result it is possible that a better match between surrounds and screen can be achieved by equalizing the surrounds with a slightly brighter apparent high-frequency characteristic, akin to the recommended screen channel characteristic for a smaller room discussed in e) above. Alternately, the same slope can be maintained as for the screen channels, but with the "hinge" point being raised from 2 kHz out to 4 kHz or even higher when the average listener is much closer to surround speakers than to the screen in a highly reverberant theatre.

g) It is important that whenever possible, the electroacoustic response 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.6 Troubleshooting

With good acoustic design and modern loudspeakers with uniform coverage, it is important that not only the overall spatially averaged electroacoustic response fall within the tolerances specified in Table 1 and Figure 5, but also that the response for each position fall within those tolerances.

Provided that the B-chain meets the tolerances specified, the electroacoustic frequency response for sound reproduction will normally be satisfactory for all types of photographic and magnetic analogue recordings and digital media, provided the additional items outlined in A.1 are given attention.

It is important to take care 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 among those included in the following list:

a) faulty power amplifier;

b) incorrect or faulty loudspeaker performance, caused by malfunction or faulty assembly;

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) obsolete loudspeaker design, unable to perform according to current specifications.

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 International 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 theatre 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 standard.

A.7 Equalization

Adjustment of the electroacoustic response to this International Standard may well require some electrical equalization, typically 1/3-octave, which for most bands approximately corresponds to the critical bands of human hearing. The following points should be noted.

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

b) Equalization above 8 kHz will not normally be attempted with older loudspeaker designs which exhibit rapid roll-off beyond 8 kHz.

c) Extreme equalization for low-frequency modes with the screen channels should be avoided, since the low-frequency modes, particularly below 100 Hz, are often very narrow response anomalies which normally cannot be corrected with the relatively broader 1/3-octave band equalizer without audible consequences at frequencies away from the modal one. For sub-woofer equalization, see 4.3.

A.8   Uniformity of sound pressure level from different loudspeakers

It is important that the same voltage of wide-band pink noise at the input of the B-chain for each screen loudspeaker system in turn yields 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, and in the same way, that the level of each surround channel be adjusted to within ±1 dB. For a discussion of channel level setting see ISO 22234:2005.

A.9   Spectral level

Although 85 dBC is commonly used as a wide-band signal level reference (see 2.4), on some analysers the display may be such that it may be convenient to use as a preferred sound pressure a decade level such as 70 dB SPL in each 1/3-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 1/3-octave bands will add to approximately 83 dBC. Note that the selected level for checking and if necessary equalizing the B-chain is purely of convenience, taking into account likelihood of clipping, background noise, etc. Again, for a discussion of channel level setting see ISO 22234:2005.

A.10 Curve-N

Correct playback of historical material may require a B-chain characteristic known as Curve-N, used in conjunction with pre-emphasized analogue photographic and magnetic soundtracks. This characteristic is referenced in some A-chain documents.

Table A.2 — B-chain characteristics — Curve-N

Centre frequencies of 1/3-octave bands	Level
Hz	dB
40	−8
50	−5
63	−3
80	−1
100	0
125	0
160	0
200	0
250	0
315	0
400	0
500	0
630	0
800	0
1 000	0
1 250	0
1 600	0
2 000	0
2 500	−1
3 150	−2
4 000	−3
5 000	−5
6 300	−8
8 000	−11
10 000	−14
12 500	—
16 000	—

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.5 f).

As stated in the scope, 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 International

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 discussed in a), b), and c) above, as equipment becomes more widespread which is capable of easily making reverberation time measurements, the results of such measurements could be combined with information on the directivity against frequency characteristics of the loudspeakers in use, and typical ear/brain response times against frequency. This data could be employed at some point in the future to add further correction factors to the basic curve.

Bibliography

[1]   ISO 266, *Acoustics — Preferred frequencies*

[2]   ISO 22234:2005, *Cinematography — Relative and absolute sound pressure levels for motion-picture multi-channel sound systems — Measurement methods and levels applicable to analogue photographic film audio, digital photographic film audio and D-cinema audio*

[3]   IEC 60268-17:1990, *Sound system equipment — Part 17: Standard volume indicators*

[4]   IEC 60942: 2003, *Electroacoustics — sound calibrators*

[5]   IEC 61672: 2003, *Electroacoustics — sound level meters*

[6]   IEC 61260, *Electroacoustics–Octave-Band and Fractional-Octave-Band Filters*

[7]   IEC 61672, *Electroacoustics–Sound level meters*

[8]   Ioan Allen The X–Curve: Its Origins and History, SMPTE Motion Imaging Journal, July/August 2006

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