SMPTE RP 2096-1:2017

SMPTE RECOMMENDED PRACTICE

Cinema Sound System Baseline Setup and Calibration

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3 Barker Avenue., White Plains, NY 10601
(914) 761-1100

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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. This SMPTE Engineering Document was prepared by Technology Committee 25CSS.

Intellectual Property

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

Introduction

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

The motion picture industry is unique in having a creation and distribution model whereby a movie is mixed and played back in potentially different environments that are calibrated to the same set of standards and are intended to reproduce similarly. The goal is to ensure that the audio heard in the playback theater/cinema matches that heard in the mix room/dubbing theater where the program material was created. Experience gained among its members has resulted in procedures to ensure that sonic match, many of which have never been documented.

The two primary criteria defining the audio environment are frequency response and level. Currently, the cinema sound system B-chain's electroacoustic frequency response is defined in ST 202 "Dubbing Stages (Mixing Rooms), Screening Rooms and Indoor Theaters — B-Chain Electroacoustic Response", with level defined in RP 200 "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".

The "Modern Digital Cinema Calibration" Recommend Practices, RP 2096-1 and RP 2096-2, expand on those two documents by providing a set of current 'best practices' methodology to better serve the above-stated goals and intentions, emphasizing contemporary fast Fourier transform (FFT) technology that is capable of transfer function analysis. The suite does not define new standards or supersede current ones, but offers current approaches and methodology using modern equipment to more accurately calibrate within the tolerances of the existing standards.

This "Baseline Calibration" Recommended Practice codifies a single set of procedures culled from current practices by experienced technicians to rigorously troubleshoot, calibrate and create detailed documentation for a dubbing theater or cinema. This provides a baseline for that theater that can be used for future "Maintenance Calibrations" and troubleshooting.

1 Scope

This document provides a single set of processes to complete baseline calibration and documentation of sound systems in dubbing stages (mixing rooms), screening rooms, and commercial cinemas to improve the consistency of reproduction, from room to room, of motion picture sound. This set of processes also provides a means to maintain sound system calibration in the future.

For normal repeat calibrations, please refer to SMPTE RP 2096-2, a substantially streamlined procedure which will call upon data collected during this baseline calibration procedure to ensure measurement consistency.

2 Conformance Notations

Normative text is text that describes elements of the design that are indispensable or contain 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 this Engineering Document’s requirements, specifications, procedures, or interoperability with other SMPTE Engineering Documents. Informative text does not contain conformance keywords.

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

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

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

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

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

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

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

3 Normative References

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

1. ANSI/ASA S1.11-2004 (R2009), “Octave-Band and Fractional-Octave-Band Analog and Digital Filters”.
2. ANSI/ASA S1.40-2006 (R2011), “Specifications and Verification Procedures for Sound Calibrators”.
3. ANSI/ASA S12.2-2008, “Criteria for Evaluating Room Noise”.
4. IEC 60942:2003, “Electroacoustics — Sound Calibrators”.
5. IEC 61260-1 Edition 1.0 B:2014, “Electroacoustics — Octave-band and fractional-octave-band Filters - Part 1: Specifications”.
6. IEC 61672-1, “Electroacoustics — Sound Level Meters — Part 1: Specifications”.
7. 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”.
8. SMPTE ST 202:2010, “Dubbing Stages (Mixing Rooms), Screening Rooms and Indoor Theaters — B-Chain Electroacoustic Response”.
9. SMPTE ST 2095-1:2015, “Calibration Reference Wideband Digital Pink Noise Signal”.

4 Terms and Definitions

B-chain — The portion of the sound system in a cinema space that reproduces the soundtrack in the theatre. It includes signal processing specific to the loudspeakers, and power amplifiers of the screen channels, surround channels, and low frequency effects (LFE) channel. It also includes the screen and the theatre acoustics.

dB — decibel.

DFT — discrete Fourier transform. The FFT is a subset of the DFT.

EQ — equalization.

FFT — fast Fourier transform.

Flat Response — A term that describes a system’s frequency response wherein the level across the frequency range of interest is essentially uniform and is within certain bounds, sometimes taken as ±0 dB, but in practice is within a defined tolerance.

Free-Field Microphone — Free-field microphones are pressure microphones that have been ‘corrected’ to give a flat response when pointed at the sound source (i.e., the sound arrives on-axis to the microphone). The ‘correction’ compensates for the microphone’s disturbing presence in the soundfield and associated high frequency buildup. With this orientation it measures the sound pressure at the diaphragm as it would appear if the microphone were not present. Sound arriving at different incidence angles will display a high frequency rolloff depending on angle and microphone diameter.

HF — high frequency(ies).

Hz — hertz, or cycles per second.

Impulse Response — The impulse response function of a system is the output that results when the system is presented with a brief input signal called an impulse. The impulse response describes the reaction of the system as a function of time. As the impulse, and therefore the impulse function contains all frequencies, the impulse response defines the response of a linear time-invariant system for all frequencies (see below for a definition of “linear”).

Linear System — An electronic system where the system output is linearly proportional, in some way, to the system input. In common terms, “linear” means that there is no nonlinear distortion in the signal, such as THD or IMD. That is to say, signals A+B at the input give rise to only signals A+B at the output, independent of amplitude. Note: even if a linear system is equalized, such that the frequency response is no longer flat, the system remains linear.

LF — low frequency(ies).

Measurement Channels:

- Single Channel — measurements where the signal fed to a single input (such as from a microphone) is analyzed without reference to any other signal.
- Dual Channel — measurement where a device compares a measurement signal against a reference source signal. For example the electrical output of signal processor or a microphone in the room might be compared against a single reference source signal.

ms — millisecond(s).

Near-field — The region from the radiating device that extends outward where the sound pressure and particle velocity are not in phase. As a result, the sound pressure level does not decrease by 6 dB for each doubling of the distance as occurs in the far-field. The near-field is limited to a distance from the source at least a wavelength or three times the largest dimension of the sound source, whichever is larger.

Pink Noise — Noise whose power spectral density ($\square/_{\text{Hz}}$) is inversely proportional to frequency and whose power per octave ($\square/_{\text{octave}}$) is constant. See [ST 2095-1] Calibration Reference Wide-Band Digital Pink Noise Signal.

PPO (Points Per Octave) — An expression of frequency resolution referring to the number of frequency divisions or analysis points within an octave bandwidth.

Random-Incidence Microphone — Random-incidence or diffuse-field microphones are pressure microphones designed to respond uniformly to a uniformly reverberant soundfield. However, since they exhibit the same directional properties as any other omnidirectional microphone of the same size, in order to achieve uniform response for all angles of incidence they must be moved to a series of different angles and the responses at all angles averaged to achieve this uniform response. Rather than use them in this manner, it is recommended to take advantage of their relatively uniform response for sounds arriving about 90º off-axis. Like all microphones, they will have more high frequency sensitivity for sounds arriving on-axis to the microphone. Response for sounds arriving at about 90º will be approximately flat and uniform throughout all angles of arrival in the horizontal plane. Therefore a random-incidence microphone should always be aimed about 90° to the sound source to capture direct sound as well as diffuse sound arriving at the microphone with flat response.

RT60: The time it takes for sound pressure to decay 60 dB from the moment a test signal is ended.

RTA (Real-Time Analyzer) — An instrument that measures and displays the spectral content of sound, typically through the use of filters or FFT. An RTA measures sound without regard to source or reference and cannot derive phase. Its measurements include direct sound, reflections, and noise. An RTA is a single channel measurement device.

Time Window — A mathematical function that is zero (0) outside a chosen interval (duration) and has a specific value inside the interval (duration). As an example, the equation below is a discrete equation, for an interval (duration) of N and number of samples n.

- Hann Window — A mathematical function that is zero outside a chosen time interval and has the value of a raised-cosine function inside the time interval, reaching the value 1 at the center of the time interval and zero at each end. It has the form:

$$\omega(n) = 0.5\left(1 - cos\left(\frac{2\pi n}{N-1}\right)\right)$$

Transfer Function — A mathematical function describing the response of a system to a defined input stimulus. The transfer function is the output of a system divided by the input to the system and includes both time and frequency components (think of it as a box with input(s) and output(s), and the transfer function is what goes on inside the box). In the frequency domain, the transfer function takes the form of the Frequency Response Function, which shows the amplitude and phase response of the system. In the time domain, the transfer function is expressed as the Impulse Response (see definition), which shows the time-pattern of arrivals of the output signal. Transfer function analyzers usually allow the user to examine the impulse response in order to differentiate the direct sound from reflections and reverberant sounds, which allows the acoustic measurement to more closely, but not perfectly, approximate human hearing.

5 Sound System Baseline Calibration: Purpose and Considerations (Informative)

5.1 Purpose

A baseline calibration is expected to be performed when a sound system is initially installed or when a major modification is made. This document provides a single set of processes to complete a detailed analysis, verification, and calibration of a cinema sound system.

5.2 Background Information

The objective of calibrating a sound reproducing system is to be able to deliver predictable sound quality to listeners. In the cinema context this means that mixers in dubbing stages and audiences in cinemas need to hear similar sounds. If we had the luxury of knowing the comprehensive acoustical performance of the loudspeaker and screen combinations, and some knowledge of the acoustical performance of the listening venues, we could approach the calibration from a significantly technical perspective using not only modern measurements but also predictive modeling capabilities.

In the absence of these kinds of data, we are left in a situation where we must rely on measurements inside existing, operating, facilities to “reverse engineer” the sound reproducing systems. This is not simple because it is essentially impossible, by measurements alone, to completely separate the performance of the loudspeakers from modifications added by the room. Consider that in live unamplified music and theatrical performances we are able to not only recognize voices and instruments, but we are also able to perceive subtleties in instrument timbre and musicianship. It is now known that humans quickly adapt to their acoustical surroundings and are able to substantially separate the acoustical distinctions of sound sources from the acoustical contributions of the venues. It is clear that the sound source is the dominant factor in what we hear, and that venue acoustics provide a setting for that sound. In other words, everything we *measure* inside a room is not reliably correlated with what we *hear*.

5.3 Recommendations

Technicians should utilize the calibration target ‘area’ as shown in [ST 202] (Figure 6.1 below), making the final assessment by listening rather than by only relying on an analyzer graph that strictly adheres to the central response line of the figure. It is recommended that calibrators avoid abrupt changes in applied equalization and the use of narrow high-Q filters, while focusing on general trends rather than narrow-band irregularities. Careful listening to several soundtracks is recommended, paying special attention to voice and music sound quality. Sound effects and dramatic LFE events are not reliable indicators of sound quality, however important they are to the overall artistic event.

6 Electroacoustic Calibration: Frequency Response Target Area and Tolerances

In this recommended practice, electroacoustic response is specified as being within a calibration target ‘area’ rather than matching a target curve or line. The sub-sections below are based on information in the current electroacoustic standard [ST 202], with additional LFE loudspeaker channel audio characteristics based on current best practices.

The calibration target ‘area’ in a 5.1/7.1-channel sound system is:

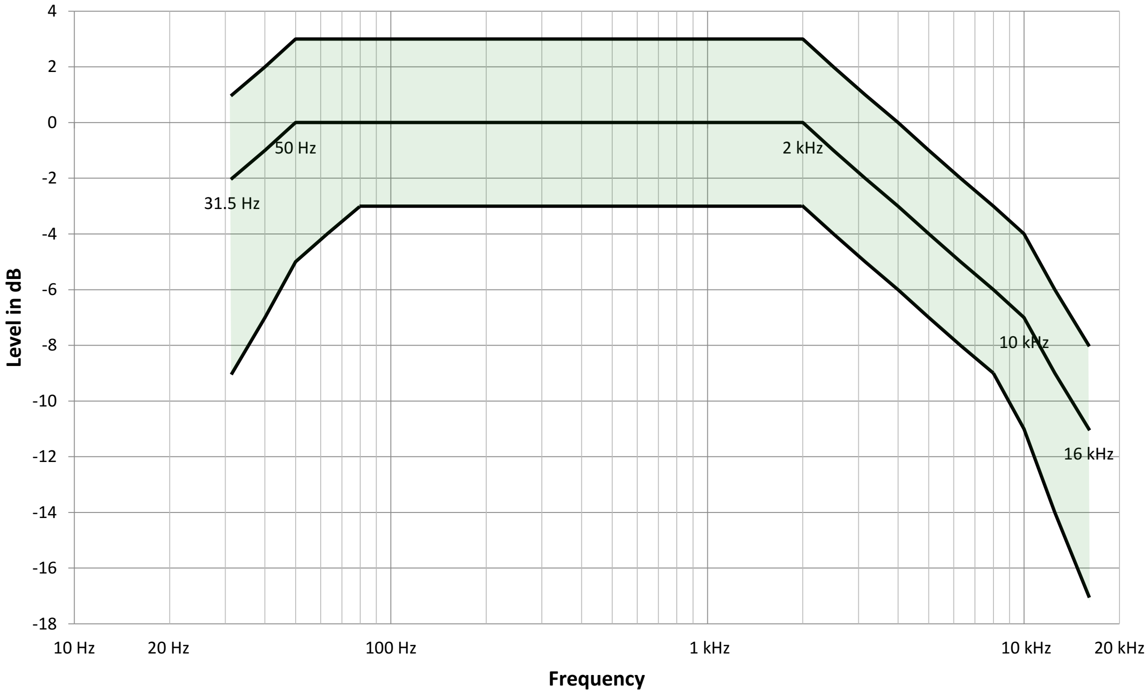


Figure 6.1. Frequency Response Target 'Area' for Screen Loudspeaker and Surround Loudspeaker Channels (from [ST 202] Table 1).

NOTES:
1. Green shading defines the target 'area'.
2. Tolerances in this figure are based 1/3-octave measurements.

Screen Loudspeaker Channels (see Figure 6.1):

The middle frequency range of the calibration target 'area' is 50 Hz – 2 kHz. Below 50 Hz is a relatively shallow-sloped low frequency rolloff with a widening tolerance at ever lower frequencies. Above 2 kHz is a relatively shallow-sloped high frequency rolloff up to 8 kHz. Above 8 kHz the slope of the high frequency rolloff increases, and the width of the tolerance increases with increasing frequency.

Surround Loudspeaker Channels (see Figure 6.1):

The criteria for surround loudspeaker channels is nominally the same as for screen loudspeaker channels (above). However, [ST 202] Appendix A.5.f., under certain conditions, allows for raising the high frequency rolloff point to 4 kHz or higher, or alternatively, reducing the slope of the high frequency rolloff, in order to improve timbre matching with the screen channels.

Low Frequency Effects (LFE) Loudspeaker Channel (see Figure 6.2.):

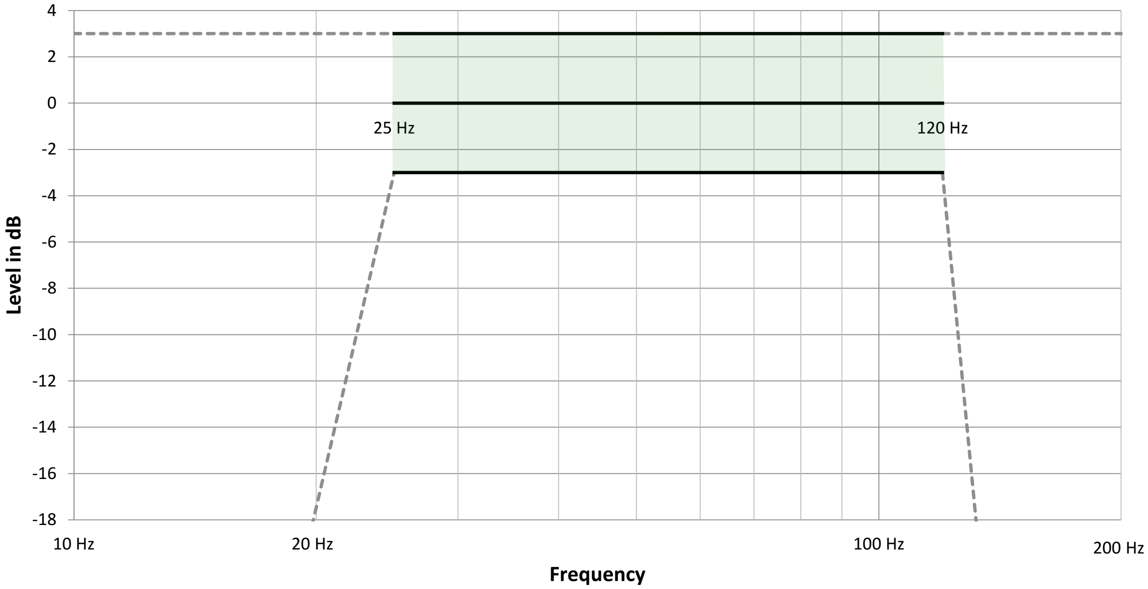


Figure 6.2. Frequency Response Target Area for Low Frequency Effects (LFE) Loudspeaker Channel (based on [ST 202] and current best practices).

NOTES:
1. Green shading defines the target area.
2. Tolerances in this figure are based on 1/3-octave measurements.

The calibration target 'area' of the LFE channel is ~25 Hz – ~120 Hz, with a tolerance width of 6 dB per [ST 202]. Best practices recommend a high frequency rolloff of at least 24 dB/octave. An optional low frequency rolloff of 12 dB/octave or more is permitted below 25 Hz.

7 Sound System Baseline Calibration Setup and System Check

7.1 Equipment Needed

All equipment shall be calibrated to its manufacturer's specifications prior to use.

- Analyzer: handheld or computer FFT-based analyzer capable of transfer function and real-time analyzer (RTA) measurements, and capable of computing C-weighted and A-weighted SPL over a defined subset of octave and fractional-octave frequency bands.
- Five omnidirectional calibration microphones with 1/4" capsules, whose frequency response satisfies requirements for a Class 1 (preferred) or Class 2 (minimum) sound level meter as specified in IEC 61672 are recommended, though a minimum of four may be used if necessary, with attention to moving positions for various measurements. All microphones shall have been calibrated by a recognized laboratory, with their responses available to the technician. In very large rooms, more microphones may be used. See Annex B.2 for more information on calibration microphone types.
- A matching number of high quality microphone preamplifiers and a high quality input device to connect microphones to analyzer. (Note that the microphone inputs of a computer are generally not of adequate quality.)
- A microphone calibrator, designed for the measurement microphones to be used. In some cases an adaptor might be needed to ensure proper fit. The calibrator should be certified to within a ±0.2 dB tolerance.
- Microphone stands.
- Microphone cabling.
- Signal generator capable of generating sine waves, programmable sweep tones, and [ST-2095-1] pink noise. If the generator cannot generate [ST 2095-1] pink noise, use the .wav files that are bundled with the [ST 2095-1] standard or the SMPTE pink noise DCP.
- Audio playback device capable of playing back uncompressed audio, if such a device is not integrated into the cinema sound system.
- Cabling to connect analyzer and signal generator to the cinema sound system. If an audio playback device is not integrated into the cinema sound system, appropriate cabling will be required.
- Recorded material for aural (listening) evaluation that is well known to the technician.
- A full-function multimeter.
- Thermometer.
- Hygrometer.
- Test material or a test system to verify picture/sound sync with flash and audible pop.
- Device to determine the offset between flash and pop, used to test system sync.
- Hearing protection as required by local Occupational Health regulations.
- Laboratory calibrated true-RMS sound level meter with C-weighting and A-weighting filters.
- Laser range finder to measure auditorium dimensions

7.2 Measurement Equipment Setup

1. Set up the analyzers in the theater seating area at least 10 feet to the side of and behind the area where the microphones will be placed, providing that the cinema sound processor and any other devices needing adjustment can be remotely controlled. This allows the effect of adjustments to be heard in real time rather than being observed only on the analyzer. If this is not possible, then the technician should devise a way to make an adjustment and quickly be able to listen to the results.
2. Set up the measurement microphones as detailed in section 7.4.
3. Connect the measurement microphones to respective microphone preamplifier inputs.
4. Connect the microphone-preamplifier outputs to appropriate audio analyzer inputs.
5. Connect the sine wave signal generator, compliant pink noise generator, and playback device to the cinema sound system and verify that they play through the system as expected. The signal injection point must be at the start of the playback chain, ahead of all processing.

   The pink noise source can be a [ST 2095-1] compliant pink noise generator, or it may be a DCP package on the server playing .wav files of [ST 2095-1] pink noise. Pink noise shall NOT be generated by the cinema processor, unless that processor's pink noise conforms to [ST 2095-1]. (Note that as of this writing most cinema processors' pink noise do not conform to the [ST 2095-1] standard.)

   In the case of a dubbing stage, a .wav sample of [ST 2095-1] pink noise may be played from a digital audio workstation into the sound system, at unity gain.

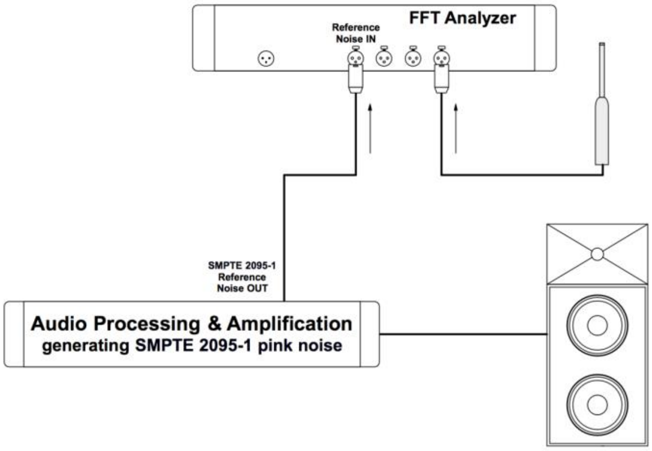


Figure 7.2.a. Example with external noise source.

Note: The SMPTE 2095-1 Reference Noise OUT channel from the processor should not have any equalization or dynamics enabled.

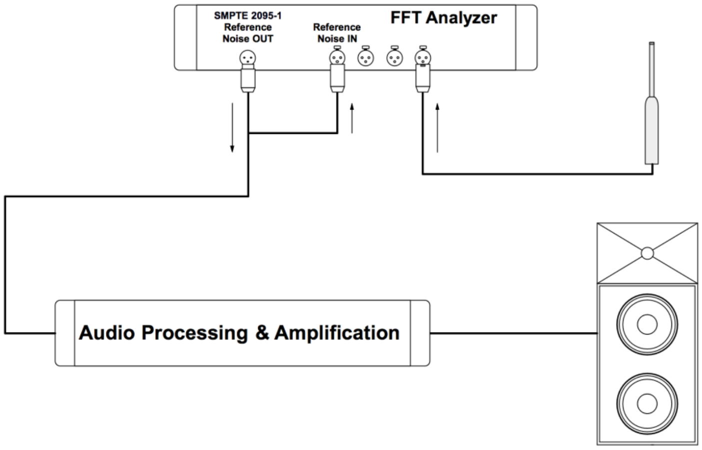


Figure 7.2.b. Example with internal noise source.

Note: In this case the reference noise is set internally in the analyzer to be the reference noise and no external reference connection is used.

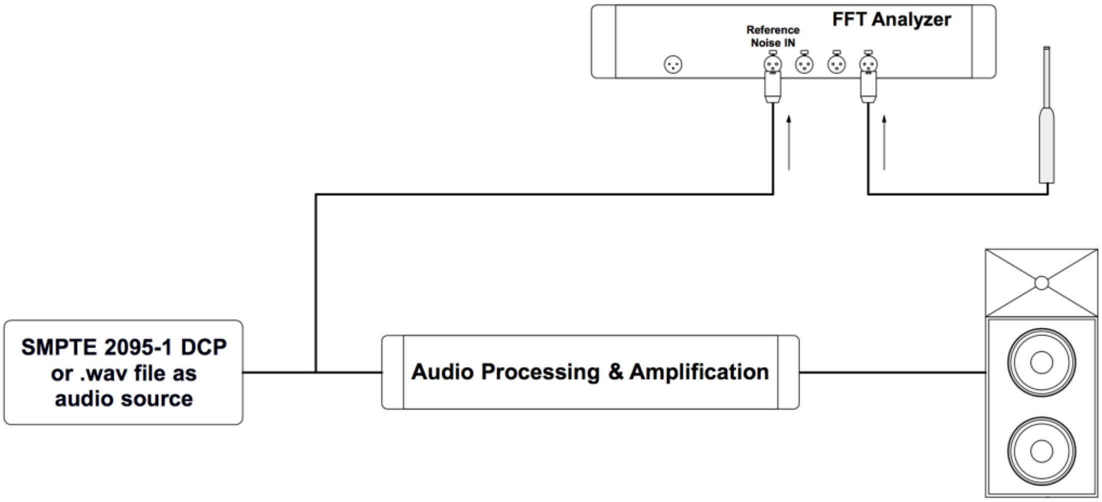


Figure 7.2.c. Example with DCP or .wav file source.

6. Configure the analyzer to perform transfer function measurements. For each playback channel or loudspeaker/array(full or subset) to be measured, set up a measurement in the analyzer.
7. If the analyzer has variable FFT size, such as fixed points per octave or time/frequency constant FFT size, use that setting. If it does not, then set the FFT size parameter to 32,000 points or greater for all channels.
8. Configure averaging measurements comprised of the microphone locations, while maintaining the ability to remove a microphone from the average if its response is radically different from the others for a particular measurement or in case of microphone malfunction. Set the analyzer to perform a power average (not arithmetic). If the device does not enable power averaging with a transfer function, refer to Annex B.1 regarding arithmetic averaging.
9. If the manufacturer of the microphones being used for calibration provides calibration charts for 90º incidence or diffuse sound correction, and the analyzer supports microphone response compensation, import the calibration chart into the analyzer. With this compensation activated the response will measure flat when the microphone is aimed straight up. Refer to Annexes B.2 and B.3 for details.
10. Calibrate each microphone’s level in the analyzer by using an appropriate laboratory calibrated microphone calibrator. Place the calibrator on each microphone and ensure that the corresponding SPL reading in the analyzer matches the calibrator output. This might require a coarse adjustment using the microphone preamp gain control and a fine adjustment in the analyzer.
11. Set transfer function delays for each microphone and loudspeaker per section 7.6. After the transfer function delays are set the system is ready to take measurements and record data.

7.3 Cinema Sound System Physical and Rudimentary Operational Inspection

An operational inspection and test of the cinema sound system shall be conducted to evaluate the system before any electroacoustic analysis is performed or changes are made. Any issues found during the inspection must be addressed prior to proceeding with additional calibration steps.

7.3.1 Loudspeakers Physical Inspection

- Check the condition of each loudspeaker cable and confirm that the cable is the correct gauge for the impedance of the loudspeaker and the length of the run per the loudspeaker and the amplifier manufacturers' specifications.
- Check that each loudspeaker is securely connected to the correct power amplifier channel.
- Check for loose bolts or other hardware problems.
- Check for physically damaged or missing drivers or other components.
- Check for proper polarity in wiring between the lower and higher frequency drivers, for each loudspeaker, and among all the channels. Note that the convention of which terminal of a driver — red or black — produces positive air-pressure into the auditorium has changed among some brands, so be certain to consult manufacturer documentation and specifications for the correct convention for the loudspeaker being inspected.
- Ensure that the loudspeakers are correctly aimed into the auditorium, and that they are not obstructed by the screen frame, struts, or other objects. Normally, screen channel loudspeaker systems likely will be aimed toward a listening position ~2/3 of the way from the screen to the back wall, with the left and right speakers angled toward the centerline for best audience coverage. However, this might vary depending on loudspeaker dispersion, seating area, and the philosophy of the sound system designer. Audibly even coverage from each playback channel over a large portion of the audience is the determining factor, rather than the simple aiming of the speaker to a specific position.
- If the sound system as found includes surround speakers with an X-Curve filter switched in, it is recommended for optimum performance that the switches be set to “flat” and a full calibration of the surrounds be performed per this recommended practice.

7.3.2 Amplifiers and Signal Processing Physical Inspection

- Ensure that all connectors are securely seated.
- Ensure that all amplifiers are connected to a suitable power source.
- Check for blown fuses or tripped circuit breakers.
- Check that all necessary components are powered on and free of fault lights/indicators, or otherwise indicate normal operational condition.
- Ensure that cooling fans (present in some devices, or separately mounted) are operational and unobstructed.
- Check and clean air filters and other parts of the airflow path.

7.3.3 Equipment Climate Control

- Check that the temperature and humidity in areas where technical equipment is located are within the operating range as specified by the manufacturers of that equipment.
- Prior to making measurements, measure and log the temperature and humidity in the theater. It has been shown that these can affect measured responses, especially at higher frequencies.

7.3.4 Preliminary Listening Test Procedure with Pink Noise and Sweep Tones

This procedure is to be performed for each channel individually, cycling through all of the channels including the LFE channel.

1. Reduce the system playback level and play [ST 2095-1] pink noise through each playback channel individually to verify channel routing, or use the [ST 2095-1] pink noise DCP and verify routing by the slates.
2. After proper signal routing has been verified, raise the playback level of the pink noise to a comfortable level that is close to reference level per section 8.8.
3. For each playback channel, listen while walking around the seating area. Verify that there is no great change in loudness, timbre or tonal balance. A great change could be a significant drop in level, greater than 10 dB, in the high or low frequencies.

   If there is a great change in loudness or tonal balance investigate and correct the cause of any inconsistency, starting with the operation of all components and loudspeaker aiming.
4. For the following portion of the test the technician should use hearing protection as required by local regulations or personal preference. In the sweep tone generator, set the parameters for the sweep to the desired duration, sweep frequency endpoints, and playback level. The sweep duration for each channel should be approximately 20 seconds. Care must be taken in setting the frequency range of the sweep to prevent sound system damage due to operation outside of the typical frequency response of a cinema sound system. It is recommended that the frequency range of the sweep be restricted to 20 Hz – 16 kHz for screen and surround loudspeakers, and 20 Hz – 300 Hz for LFE/subwoofer loudspeakers. Alternatively, the technician can manually sweep the frequency generator, which allows for rapid homing in on problem areas.

   The sweep signal level should be sufficiently high, such as -3 dB FS for screen and LFE channels, and -6 dB FS for the surround channels, to reveal anomalies. If there is concern that such high levels might damage the system, a lower level such as -10 dB FS may be used.
5. Switch the analyzer to RTA display with a resolution of 1/24-octave. Engage the sweep. As the sweep runs, listen for mechanical rattles or distortion anomalies and also watch for them on the display, noting the frequency at which they occur. A distortion anomaly is a signal that appears at a harmonic of the test signal frequency. Figure 7.3.4 is an example of a sweep results display. Any harmonics displayed that are NOT more than 20 dB below the test signal should be investigated, as should any other suspect sounds.

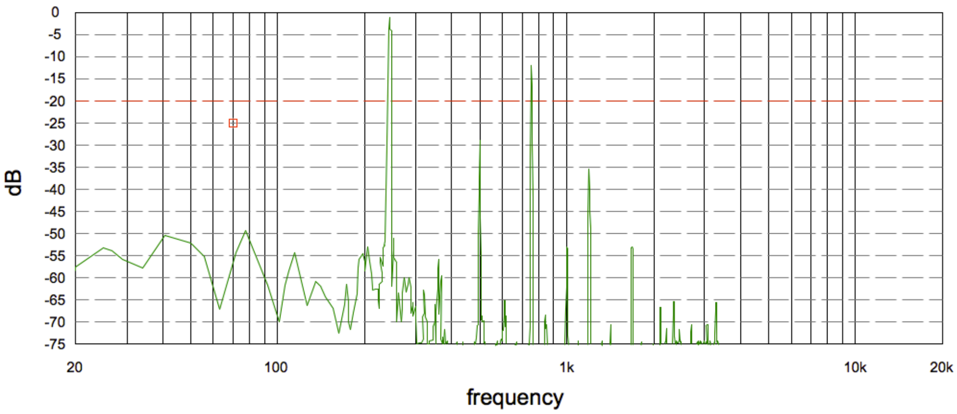


Figure 7.3.4. Example of a sweep results display.

Note the source signal peak at ~225 Hz, along with the likely unwanted peaks caused by distortion, rubbing or other causes at ~500 Hz, ~750 Hz, ~1 kHz, ~1.2 kHz, and ~1.7 kHz.

6. After noting any issues during the sweep, follow up by playing a sine wave at the frequency that excited the discovered issue, allowing for further investigation and problem correction in the physical domain, rather than attempting to implement an electrical solution.

7.4 Measurement Microphones Locations and Heights

7.4.1 Constraints for Microphones Locations in Cinemas

The microphones should be located

- not closer than ~2 m (~6.5’) to any wall
- not closer than ~5 m (~16’) to the screen
- if possible, not closer than ~2 m (~6.5’) to any other microphone of the microphone array

All microphone locations should be within the coverage pattern of the loudspeaker(s) or surround array zone(s) being measured. As measured in one-octave resolution over 500 Hz – 2 kHz, the level at each microphone location shall be within ±3 dB (maximum to minimum) of each other and as compared to the level at the primary microphone location. If the level at a microphone location is outside of this range, the microphone should be moved so that it is within the coverage pattern of the loudspeaker(s) or surround array zone(s) being measured and its measured level is within the ±3 dB (maximum to minimum) tolerance range in order to yield meaningful results when averaged with the other microphones in the array.

7.4.2 Measurement Microphones Locations

7.4.2.1 Microphones Locations Configuration for an Exhibition Theater

The screen to back wall distance L and the auditorium width W determine the microphone locations.

The primary microphone shall be located near the screen centerline, ~2/3L distance from the screen.

In the rare case that the seating area in the theater is significantly asymmetrical to the screen then consideration should be given to move the lateral position toward the centerline of the seating area.

The measurement area for the microphones scales with the size of the cinema. The additional four microphones shall be placed around the primary microphone in an offset diamond shape. This area should extend ~1/6L toward the front and back from the primary microphone and ~1/5W toward each side (see Figure 7.4.2.1.a.).

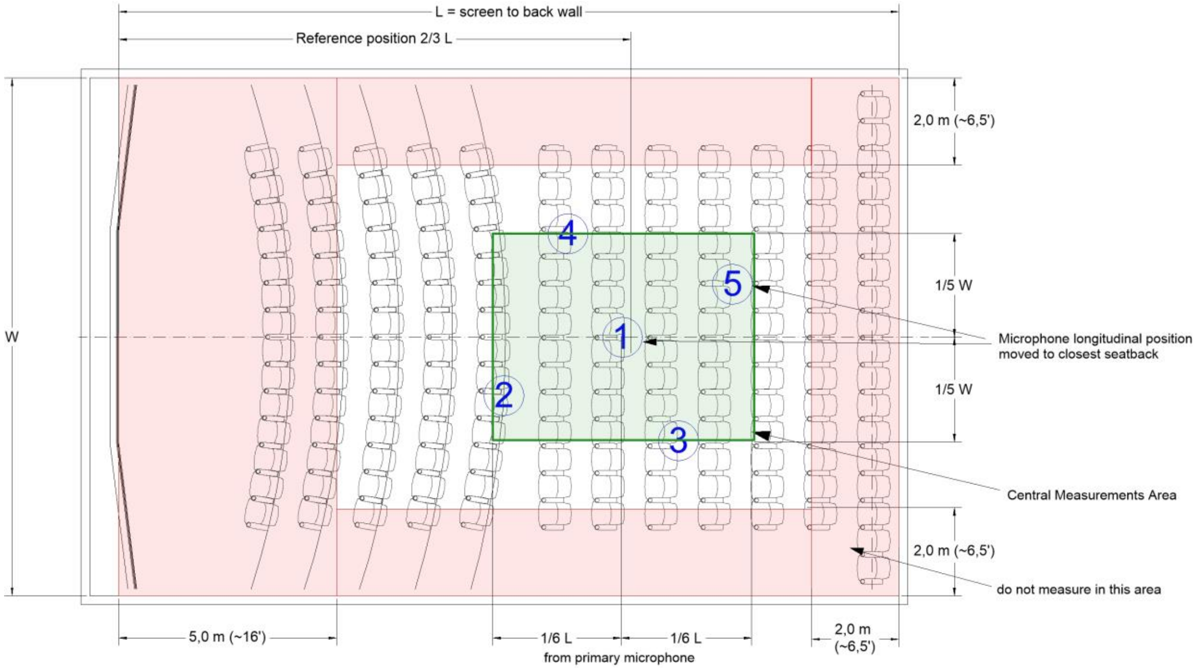


Figure 7.4.2.1.a. Suggested microphones locations if using five microphones in an exhibition theater.

If the measurement equipment supports only four microphones, the primary microphone (reference position 1) is still used, along with the offset diamond microphones 2 thru 4, to perform the equalization tasks (Figure 7.4.2.1.b.).

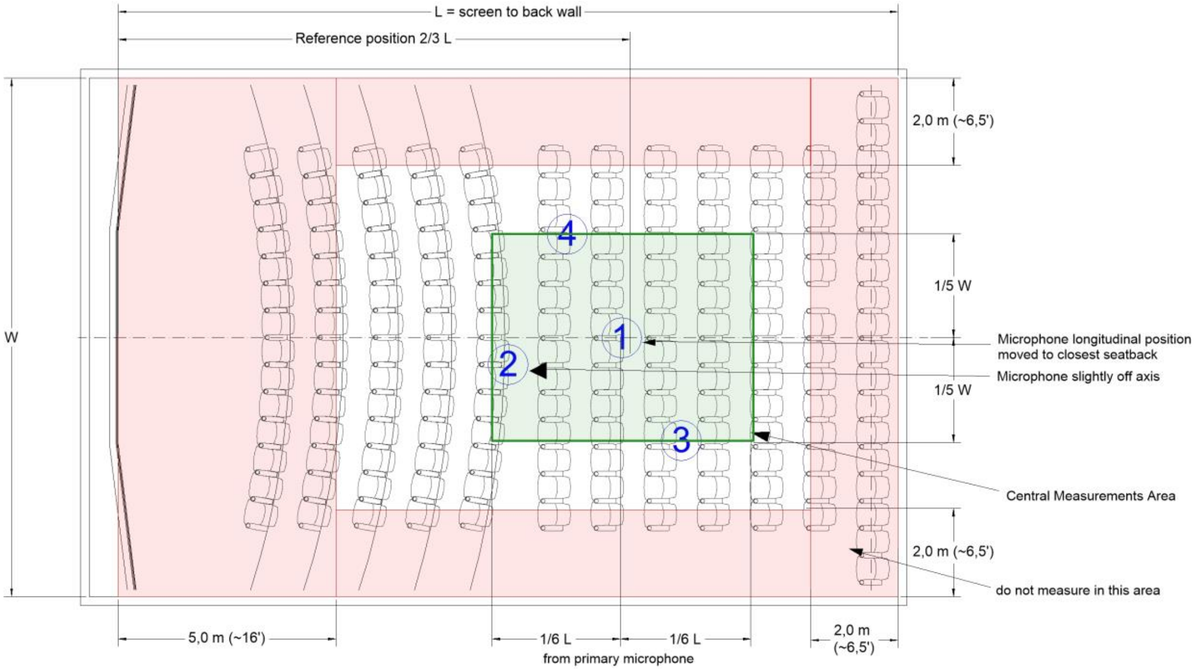


Figure 7.4.2.1.b. Suggested microphones locations if using four microphones.

Small theaters (such as in Figure 7.4.2.1.c.) with a screen to back wall distance less than ~12 m (~40') require some compromises, as small auditoriums cannot comply with all restrictions for microphone positioning. The priority is to comply with the minimal microphone to boundary distances, and to compromise the microphone to microphone distances. Once the microphones' positions are determined for a given theater, those exact positions shall be documented.

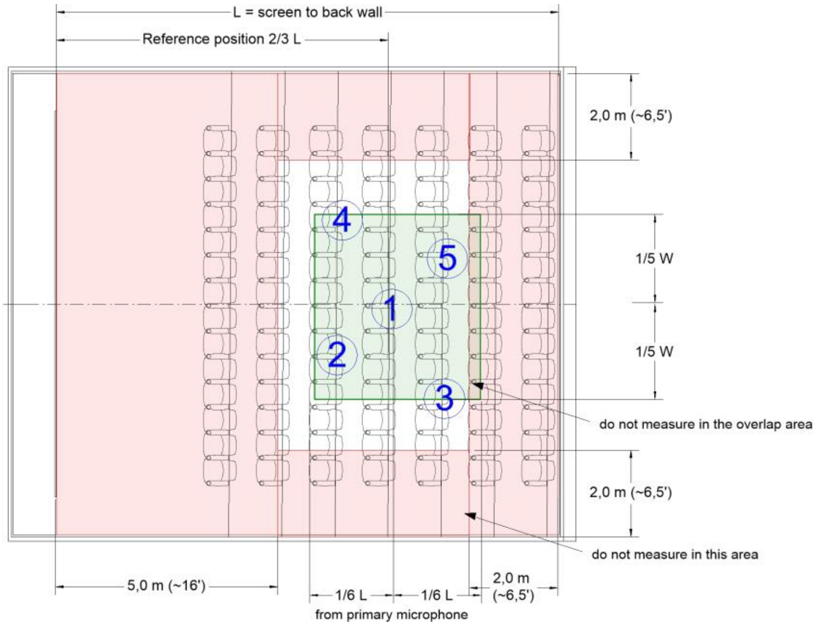


Figure 7.4.2.1.c. Suggested microphones locations if using five microphones in a small theater.

7.4.2.2 Microphones Locations Configuration for a Dubbing Theater

In a dubbing theater (Figure 7.4.2.2) the microphones will be in a much tighter cluster because the locations of the measurement microphones should reflect the critical listening positions for the mixer(s) and producer(s).

The microphone to microphone distances as described in 6.4.1 might not be feasible. However the microphone to microphone distance should not be less than ~0.6 m (~2').

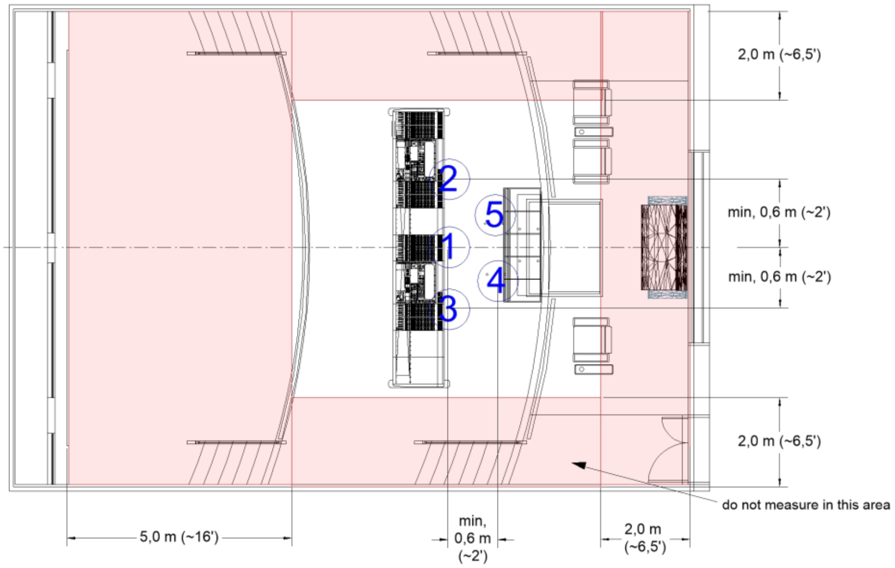


Figure 7.4.2.2. Suggested microphones locations for a tight mixer listening area.

The primary microphone (position 1 in Figure 7.4.2.2) shall be located at the primary mixing position, slightly behind (~0.2 m (~8")) the armrest of the console.

Two additional microphones (positions 2 and 3 in Figure 7.4.2.2) are placed on either side of the primary microphone at the same height, corresponding to a typical three-mixer configuration, at least 0.6 m (~2') distance from the primary microphone. The exact placement and distances should be judged by the arrangement of the mixing area — the idea is to get a good sampling of where the mixers are working.

Note: All equalization decisions should be determined primarily by the response of these three microphones.

Two additional microphones (positions 4 and 5 in Figure 7.4.2.2) should be set up to sample the typical producer's listening area(s). Depending on the dubbing theater and how its clients generally work, these microphones can be either behind the mixer's working area, or one in front of the console and one behind the mixers.

Note: As those microphones contribute to the power average, their responses as compared to the response of the microphones at the mixers' positions must be checked to ensure they are within ±3 dB of those microphones so that the average is not compromised.

Once the microphones' positions are determined for a given dubbing theater, those exact positions shall be documented. In addition, the response differences among mixers' positions and producers' positions measurements shall be documented.

7.4.2.3 Microphones Locations Configuration for a Dubbing Studio Also Used as a Screening Room

For a multipurpose room used as a dubbing studio and as a screening room, two switchable system setups are recommended:

1. Mixing setup with microphone configuration as described in section 7.4.2.2;
2. Screening setup with offset diamond microphone configuration as described in the exhibition theater section (7.4.2.1).

7.4.3 Measurement Microphones Heights

In a dubbing stage the microphone height shall be ~1.4 m (~56”) above the floor to help mitigate console reflections. The mixer chairs should be removed from behind the console; any fixed chairs or couches, such as those for producers, should remain. Any other displays or computer screens that are in the path between the loudspeakers and the measurement microphones should be removed if possible, or moved to minimize impact on the measured response.

In a cinema auditorium, the microphones shall be mounted at a height of 45 cm (18”) above the seat backs in order to minimize interference from the seat. Irregular heights between the limits of 30 cm (1’) and 60 cm (2’) are acceptable, but, in such cases, the height of each microphone should be carefully recorded for future reference. At no time shall the microphones be less than ~1.2 m (~48”) above the floor.

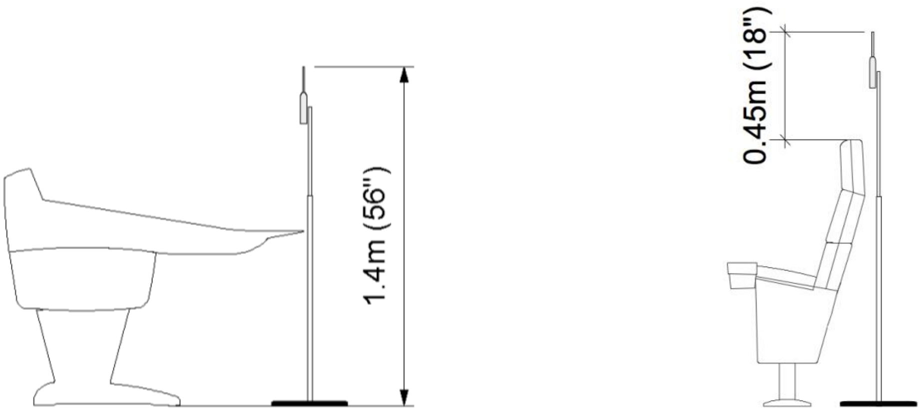


Figure 7.4.3. Recommended microphones elevations.

7.4.4 Measurement Microphones Aiming

Measurement microphones per section 7.1shall be aimed straight up at the ceiling, with the analyzer compensated for the 90°/diffuse field calibration chart per section 7.2 (9). For other setups refer to Annex B.2 for alternative aiming.

7.5 Procedure for Setting Surround Channel Signal Delays

This procedure must be completed before any equalization adjustments are made. Adjusting the delay after equalization likely will result in changes to the frequency response and sound of the surround field, depending on where and how many delay zones are in the system.

The objective of having surround channels is to provide a sense of space and envelopment, but without pulling the dialog from the screen channels. To accomplish this effect, the sound from the surrounds should arrive at a time equal to or later than sound from the screen channels in the majority of seats. Cinema processors offer control over this delay parameter to account for the variety of different-sized spaces. Some processors offer only a single global delay, while others offer delays per surround channel or even per individual speaker in the surround field.

Ideally, surround channels in a cinema would be aligned to achieve coincident arrivals for all listeners, but that goal is not possible due to the expanse of the seating area. Depending on the size of the theater, sounds can reach listeners seated at one end of the room ~25 – 75 ms later than they arrive at listeners seated at the opposite end of the room.

While the surround speakers are usually placed on the side and rear walls near the audience, typically the front speakers (behind the screen) are farther from the audience. Therefore the first objective for surround delays is to compensate for the extra distance the screen sounds must travel relative to the microphone 1 reference listening location. A second objective for surround delays is to ensure that time related sounds (such as dialog or a gunshot) from the screen channels are heard before the associated reverb sounds coming from the surround channels. An additional delay offset is thus added to address the sound arrivals at the back rows of seats. While adding delay to the surround channels can ensure the desired result for front-originating sounds, the same remedy does not apply to sounds originating from the side or back surround channels when their reverbs come from an opposite direction. However, this type of problem is usually addressed when mixing the soundtracks.

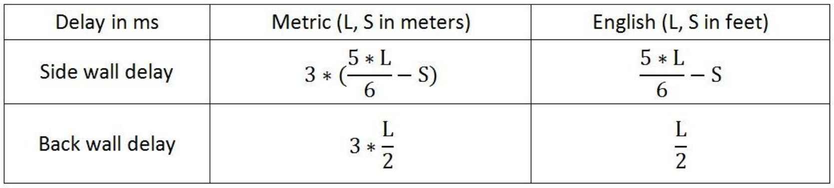
Surround delays might be based on acoustic measurements or on physical dimensions, but should still be checked by listening to a soundtrack to confirm that nothing had been incorrectly entered.

7.5.1 Installations with Individual Delays per Surround Channel

Measuring surround delays:

1. Set all surround delays to 0 ms.
2. With a microphone at position 1 (the reference listening position), measure the time difference between the first arrival from the surround channel and the first arrival from the center channel.
3. Add a room-length-dependent offset in milliseconds of L/2 ms (L in meters) or L/6 ms (L in feet) to the difference.

Alternatively, the delay can be calculated by these distance measurements:



Delay in ms	Metric (L, S in meters)	English (L, S in feet)
Side wall delay	$3*(\frac{5*\mathrm{L}}{6}-\mathrm{S})$	$\frac{5*\mathrm{L}}{6}-\mathrm{S}$
Back wall delay	$3*\frac{\mathrm{L}}{2}$	$\frac{\mathrm{L}}{2}$

”L” is the distance from the screen to the back wall surround loudspeakers.
“S” is the distance from the theater’s centerline to the side wall surround loudspeakers, (typically, “S” is half the room’s width).

4. Set the surround delay in the processor to these values.

7.5.2 Installations with One (Global) Surround Delay

For installations offering only one global surround delay, set the delay to the maximum of the side-wall and back-wall surround delays as determined in section 7.5.1.

7.5.3 Installations with Multiple Surround Delays within Channels

For an installation offering individual delays per surround group or individual loudspeaker, the delays may be set uniformly, thus achieving the same performance as the global delay setting.

However, the time relationship between the screen sounds and the surrounds might be improved by reducing the delays for speakers closer to the screen, or increasing the delays for the rear surrounds to ensure that the surrounds furthest from the screens are the last ones heard by audience members

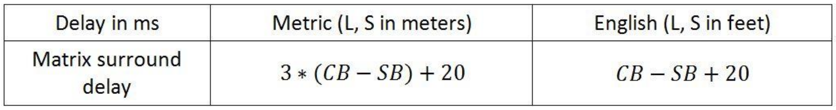
Using different delays within a surround array can cause false localization of surround sources at various seating areas, which would degrade the intended surround perception. After setting the surround delays, the result should be verified by playing voices from each surround channel, checking for localization errors while walking around the seating area.

7.5.4 Surround Delay for Analog Matrix-based Sound Formats

Processors supporting an analog matrix-based sound format (such as 35 mm optical audio) typically provide a separate surround delay setting for these formats. This requires longer delay settings to accommodate the Haas (precedence) effect, which helps avoid false localization of dialog caused by crosstalk into the surrounds.

Set this delay for a seat in the back row, using the distance from the center speaker to the back row, and the distance from the closest rear located surround speaker to the back row, for the calculations as delineated in section 7.5.1.

Calculating the surround delay:



Delay in ms	Metric (L, S in meters)	English (L, S in feet)
Matrix surround delay	$3*(CB-SB)+20$	$CB-SB+20$

CB is the distance from the center-channel speaker to the back row.
SB is the distance from the closest surround speaker to the back row.

7.6 Setting Transfer Function Delays

The transfer function delay is the delay between the source/reference signal and the signal as received by a measurement microphone. It will vary for each microphone position and for each loudspeaker being tested. The transfer function delay for each microphone must be correct for each loudspeaker under test prior to conducting transfer function measurements on that loudspeaker.

In some analyzers this might be an automatic procedure and might occur without any direct action from the technician. If the delay setting process is not fully automated, refer to the analyzer's user manual to determine the procedure to set the transfer function delays across each measurement channel. Store the delay values in the analyzer and enter them into the calibration log for each microphone and each loudspeaker.

After the delays have been set, verify that the delayed impulses (see Figure 7.6.a.) have first impulse arrivals that go to the same side of the Y-axis on each measurement channel, that the coherence has high values, that there is no excessive or unexpected dense wrapping in the phase display (Figure 7.6.b. vs. Figure 7.6.c.), and that the delay time found by the process is realistic based on current settings of the measurement system. For example, a sound system delay might be as high as 200 ms in a digital cinema system, as the delay would have been set to align the sound system to the projector latency.

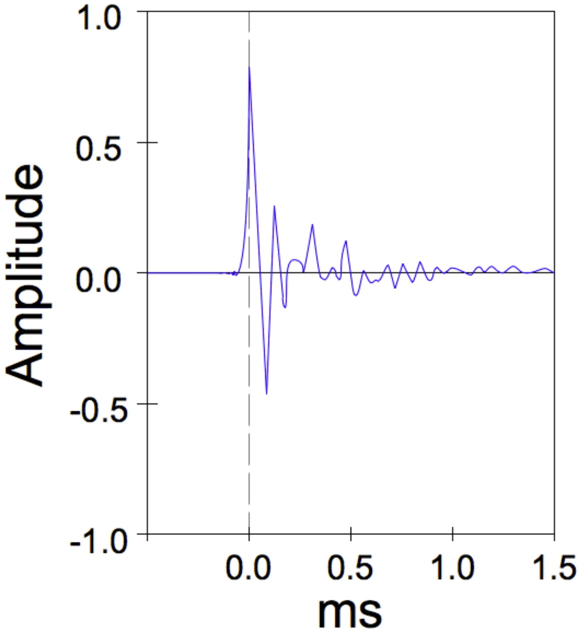


Figure 7.6.a. Impulse response with positive first arrival.

Coherence should retain high values across the loudspeaker crossover points and into the high frequencies (Figure 7.6.b). Figure 7.6.c. shows an improperly set transfer function delay that results in a loss of coherence in the higher frequencies and a phase trace that shows significant errors. Note the excessive wrapping in the phase response along with the drop in coherence value above 10 kHz. These two issues are good indicators that the delay has not been set properly or at all.

NOTE: In order to set a delay across the measurement channel that involves only a subwoofer in a transfer-function measurement, it might be necessary to temporarily remove any low pass filters on the subwoofer(s). Be certain to restore the filters once the delay measurements and adjustments are concluded. The characteristics of the subwoofer used will determine whether the analyzer is able to obtain an accurate delay. Alternately, one can manually calculate the delay.

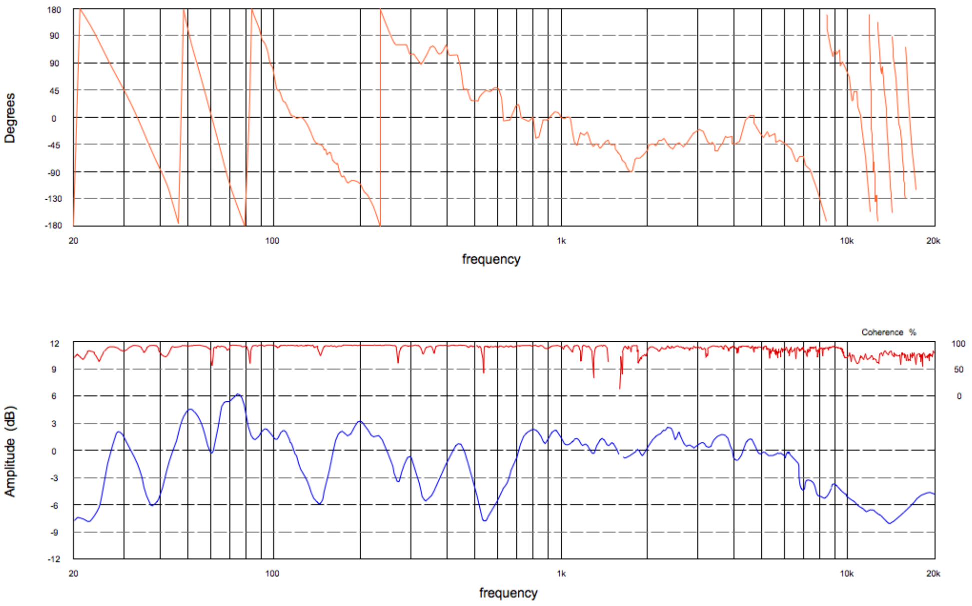


Figure 7.6.b. Additional transfer function data:
Upper graph — phase data showing acceptable wrapping.
Lower graph — Coherence values are acceptable, even above 10 kHz.

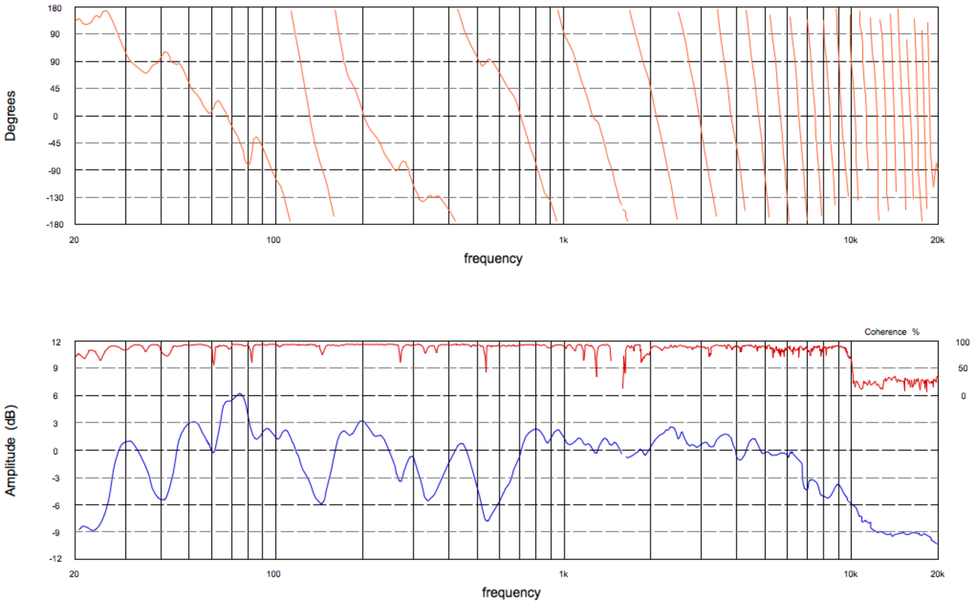


Figure 7.6.c. Additional transfer function data:
Upper graph — phase data showing unexpected and dense wrapping, especially at higher frequencies.
Lower graph — Coherence values are well below acceptable values at many points along the frequency spectrum, especially and continuously above 10 kHz.

7.7 Background Noise Check

Preparation

It is important to measure the background noise in the venue prior to taking sound system measurements. If there is a significantly high noise floor (especially in the lower frequencies, such as might be caused by a malfunctioning HVAC system), it could be mistaken for a system performance characteristic, resulting in unnecessary equalization being applied in an attempt to solve a problem that is not caused by the sound system. The noise floor must be ≥10 dB below the level of the test signal in the room in order to ensure adequate measurement accuracy.

Background noise measurements shall be made under conditions that closely approximate normal venue operating conditions (including noise from operating HVAC systems, neighboring businesses, and adjacent cinema rooms).

The background noise check will use data from the primary microphone and all additional microphone locations. 1/4”microphones may be used although they will be at the edge of their performance range. If additional detail is needed, 1/2” or 1” microphones should be used to provide the best data.

There are multiple ways to measure Noise Criterion. When using a discrete Fourier transform (DFT, or its FFT subset) based analyzer, a frame size of sufficient length (duration) to provide a time constant of ≥170 ms shall be selected to ensure sufficient frequency resolution in the 16 Hz-octave band to closely approximate the precision of a standard band-pass filter. A raised-cosine (such as Hann) or comparable data-window function shall be applied to the time-record for each DFT/FFT frame before transforming, to minimize leakage between neighboring frequency data points. Time records for each successive DFT/FFT frame shall overlap by a sufficient percentage (for example: 67% for a Hann window). An equal-weighted arithmetic average of the squared-amplitudes for each frequency data point in all DFT/FFT frames collected during the measurement period shall be calculated. The resulting time-averaged spectrum for each measurement location shall be aggregated into IEC-standard one-octave bands [Normative References: 5 and 6].

After the measurement is complete, store the spectrum analysis data from the primary microphone, to be used in future maintenance checks.

Procedure

With all equipment powered on and ventilation systems running at normal levels, collect data for 30 seconds from all of the microphone locations and compute an average. These results will be viewed in octave bands and compared to ANSI-standard Noise Criterion (NC). The lowest NC curve in the criterion table that is not exceeded by any of the octave-band measurements is the resulting NC rating for the room (see Figure 7.7.). Note that some analyzers have the NC curves built in, as reference curves.

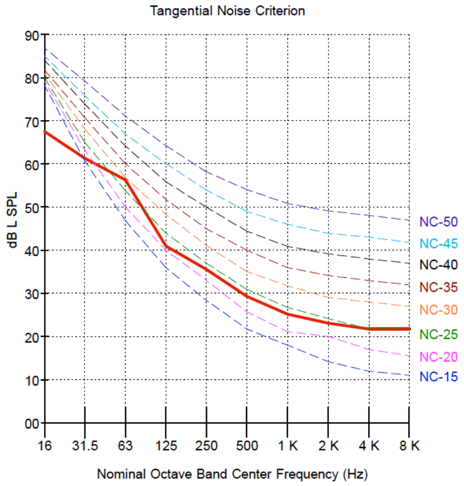


Figure 7.7. ANSI Noise Criterion graph, with simulated measurement qualifying for an NC-30 rating.

As an alternative, some Fourier transform based analyzers include a built in background noise test preset. If it meets the above criteria, use it.

The optimum NC rating for cinema rooms is NC-25; in no case should the noise level exceed NC-35. If the measurements show an octave-band that is significantly different (standing out from the measured noise criteria curve more than the other octave-band measurements), the room should be investigated and the cause remedied.

To determine the noise contribution of the sound equipment, repeat the NC measurement with the power amplifiers off.

7.8 Reverberation Decay Time

Knowledge of a room's reverberation decay time can provide the basis for an explanation for certain sound characteristics that cannot be fixed with equalization, such as upper-bass "boominess" or a lack of intelligibility in the middle to upper frequencies. A typical 200 – 300 seat theater should have a mid-frequency RT60 of ~0.5 to 0.6 seconds. It is recommended that the reverberation decay time be checked as part of:

- The first tuning of a newly constructed space if the reverberation decay time has not been verified as part of the construction process; or
- If physical changes have been made to the space or its contents.

Refer to ISO standard 3382-2 Part 2 for methods and procedures to conduct a reverberation decay time measurement.

7.9 Procedure for Verifying and Adjusting System Gain Structure

To verify gain structure across the entire sound system in a digital cinema, a 1 kHz sine wave reference tone at -20 dB FS shall be played back through each channel of the theater's sound system. Set all level controls to unity gain so that the 1 kHz tone may pass through the system at unity. In addition to the main playback level this may require temporarily bypassing equalizer and other level controls. Once necessary controls are set to unity or temporarily in bypass mode, play the reference tone through each channel individually.

Observe at each point (device) in the digital playback chain that the meters, set to peak mode in each device, match the reference level of the signal being played.

At each point where the digital signal converts to analog, such as the output of a signal processor before the input to an amplifier, verify with a multimeter (capable of measuring a frequency range greater than 1000 Hz) that the electrical signal's voltage level is as expected.

Verify that the input sensitivity of any device that is receiving an analog signal is set to or matches the electrical level that is being sent to it, and that adequate headroom is maintained throughout the signal chain.

7.10 Loudspeaker Component Crossover and Delay Verification Procedure

In a loudspeaker with multiple drivers, where each driver operates over a portion of the loudspeaker's full operating range, the arrival of sound from the drivers should be matched in time, generally to the driver with the most delay. Driver delay is a combination of the physical location of the driver and the group delay in the crossover filters. The drivers' output levels must also be matched at the crossover frequency to achieve the loudspeaker's desired operating frequency response. In the baseline calibration, each loudspeaker with multiple drivers must be checked to ensure that the crossovers and delay are set according to the manufacturer's specifications, and the results logged for future calibrations.

In older systems, the crossovers often are passive and have no user adjustment, although some passive crossovers have adjustments for the level of HF drivers, and might have switches to compensate for different horn directivities. In the latter case it is important to check that the correct horn is selected.

The following procedure presumes a system with active crossovers and separate amplification for each driver, and that the component crossover and delay parameters are adjustable. These parameters are often handled in a processor that is externally controllable by a laptop computer or other means.

Procedure

The initial listening test in section 7.3.4 aurally verified that the loudspeaker was emitting sound across its entire operating frequency-range, leading to the assumption that all components are working. If this was not the case, then this must be investigated and all relevant system components verified to be in good working order prior to proceeding.

Having set the delay across the transfer function measurement in section 7.6, select the appropriate transfer function measurement channel to analyze a loudspeaker.

Play the [ST 2095-1] pink noise signal through each loudspeaker.

At the primary microphone location, examine the loudspeaker's frequency and phase response and verify that it matches expectations based on the manufacturer's specifications and documentation.

A properly set crossover will result in a loudspeaker that exhibits minimal frequency response irregularity through the crossover frequency region as measured both on and off the axis of the loudspeaker's centerline. Figure 7.10 shows an example of an expected result if the crossovers are properly set.

A useful technique to verify this is to temporarily invert the polarity of one driver section and to look for a deep notch at the crossover frequency for the inverted driver section. E.g. for a 3-way system temporarily invert the mid frequency drivers. If the physical alignment and the section delay time settings in a digital crossover are correct the frequency response should show deep notches at the crossover frequencies to the high and to the low frequency sections instead of the smooth response in Figure 7.10.

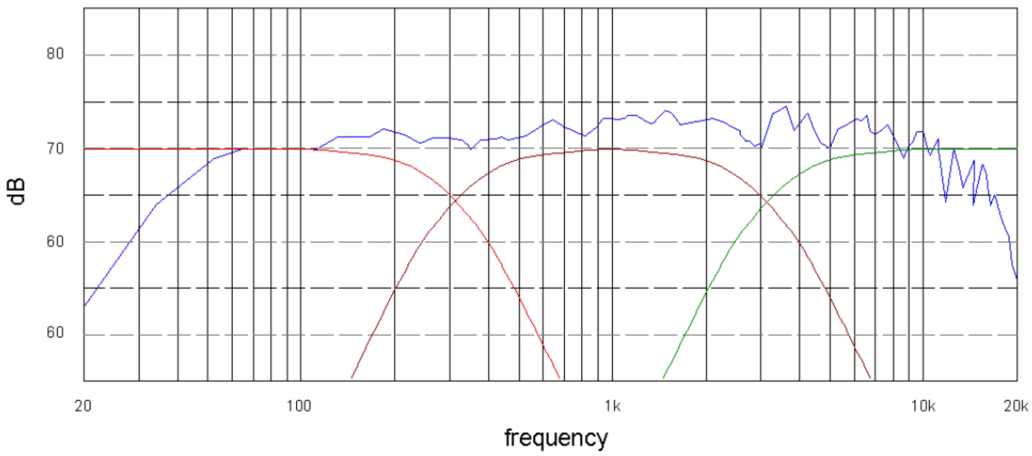


Figure 7.10. Loudspeaker frequency response showing a three-way speaker with proper crossover behavior.

The behavior of the phase across the crossover point will vary by manufacturer and model, and by measurement microphone location. Some will have a consistent slope and some might reveal a 180º phase change (a polarity reversal; which can be by design, and is not necessarily an error). Consult the manufacturer's specifications to verify crossover characteristics. Completing this verification will also confirm that polarity relationships of drivers within a loudspeaker are as intended.

If the loudspeaker crossover response is not within manufacturer's specifications, the loudspeaker must be checked for issues. If the crossovers have been tampered with, or a driver replaced, the crossovers will require adjustment. This should be undertaken as a separate set of steps using the manufacturer's specifications prior to performing a baseline calibration of that loudspeaker.

7.11 Polarity and Phase Alignment Procedure

Verifying a consistent polarity is required, as a polarity reversal can compromise performance, especially in regard to sonic imaging or frequency response at low frequencies. It is good practice to follow a consistent absolute polarity convention such that a positive signal into the loudspeaker channel results in positive pressure in the listening area.

Using the microphone at the primary location, verify that the measurement channel is selected for the loudspeaker to be checked. Verify that the delay has been set across that measurement channel transfer function per section 7.6.

Take an impulse response measurement of each loudspeaker including the subwoofers in the LFE channel. Consult the analyzer's documentation for device specific impulse response measurement parameters and procedures. Store each impulse response, then repeat the measurement for the next loudspeaker channel. In the case of a surround array, to get an accurate impulse response there are two possible approaches:

- If the loudspeakers are set up so the array can be broken down into smaller segments, turn off loudspeakers so that the impulse response of each smallest array-segment (possibly an individual loudspeaker) can be taken.

- If the array cannot be reduced to smaller segments, position the microphone on-axis for each loudspeaker, at a distance of one-quarter the distance to the nearest adjacent loudspeaker, in order to isolate it from other loudspeakers and get an accurate impulse response that can be easily examined.

Once the impulse responses (IRs) have been taken, display the trace of the behind screen center channel loudspeaker IR. Compare the polarities of the rest of the impulse responses against that reference.

If the impulse response polarity from each other loudspeaker essentially matches that of the center channel loudspeaker reference, the polarity matches. If they are opposite (one being positive, the other negative), then the polarity is also opposite, and the problem must be fixed. Figure 7.11. shows a positive and a negative first arrival impulse.

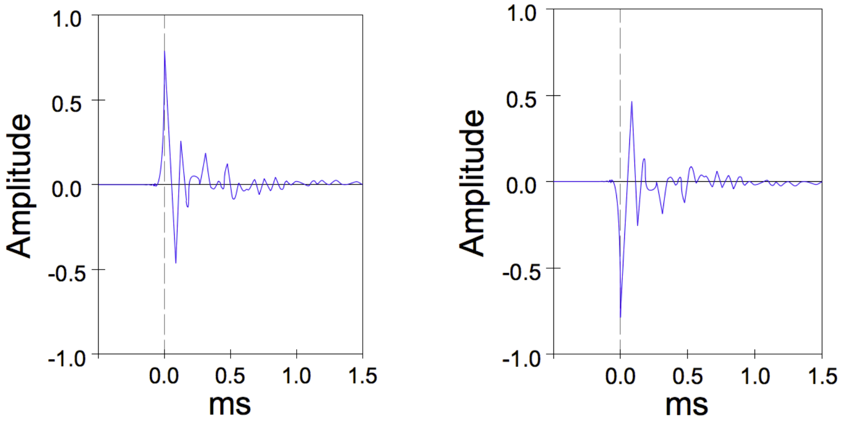


Figure 7.11. Two impulse responses:
Left graph — Positive polarity of first arrival impulse.
Right graph — Negative polarity of first arrival impulse.

Once the impulse responses have been verified and the delay set across the transfer function, the phase should be observed.

Again using the microphone at the primary location, and starting with the center channel, select the appropriate measurement channel in the analyzer. As the delay for the measurement channel has been set previously, play pink noise. Observe the measurement and store a copy. (Note that in some analyzers this data might have already been stored as part of previous measurements. If this is the case with the analyzer in use then a simple visual comparison of the stored result data traces is all that is needed.) Take measurement traces of the output from each of the other loudspeakers. Compare the traces against the center channel loudspeaker reference. Loudspeakers of the same type should have nearly the same phase response. If the phase response across the operating range is consistently different by 180° it is an indication of a polarity reversal in the signal chain.

The subwoofers in the LFE channel should also have a phase measurement taken and stored. The phase measurement of the LFE channel should also be compared to the center channel. If necessary, delay should be added to the LFE channel where appropriate to match the phase between it and the center channel. Matching polarity and phase response between the center channel and LFE channel will create a smooth response should an effect or voice in the center channel be doubled into the LFE channel.

8 Electroacoustic Measurement Methodology

8.1 Introduction (Informative)

Modern cinema processors may offer equalization with greater capability than that of traditional fixed 1/3-octave wide filters. Examples may include parametric, asymmetric, raised cosine and broadband tone controls. It is strongly recommended that the technician use the capabilities in the cinema processor efficiently.

Equalization is recommended to be used only if the electroacoustic response anomaly is determined to be caused by issues that:

- Are of a magnitude that causes the average response to be well outside of target-response tolerances.
- Are determined, by listening, to negatively impact perceived sound quality, and are deficiencies that equalization can legitimately solve.
- Are not measurement artifacts, such as narrow band dips or peaks that are isolated to single microphone locations, especially below 200 Hz.

Guideline: If applying equalization does not alter the response in a 1:1 manner (such as: applying +3 dB at frequency X does not change the response at frequency X by close to +3 dB), then equalization cannot solve that problem and cannot be used; a different approach needs to be taken. If equalization cannot solve that problem and the problem cannot be fixed otherwise then the tolerance area per section 6 may be exceeded as narrow peak or dip at the offending frequency. Dips caused by phase cancellations e.g. caused by strong reflections or seat rows in front of the microphones (seat dip) or from overlapping coverage of surround speakers cannot be equalized. Never try to fill dips caused by cancellations.

If the sound system is behaving normally in its overall response, the role of equalization is to correct, with as few filters as possible, the overall low frequency response or the overall treble response.

8.2 Initial Sound Pressure Level Measurement and Adjustment

Prior to taking electroacoustic measurements, the sound system must be verified to be close to reference sound pressure level. This is a coarse level verification to ensure that there is sufficient signal above the noise floor to get accurate measurements. A final SPL-setting procedure will be performed after equalization adjustment.

To perform a coarse level check, first place the microphones in the locations per section 7.4. Adjust the main fader to its reference position, which is 7.0 on most cinema processors. Play [ST 2095-1] pink noise through each channel in turn. Switch the analyzer to the 1/3-octave RTA view while averaging all microphone locations together. In a 5.1 or 7.1 channel sound system, verify or adjust the level of each playback channel so that the screen channels' unweighted (dBZ) SPL in-band values are ~71.5 dB SPL in each 1/3-octave band from 500 Hz to 2 kHz center frequencies, and the surround channels' SPL values are ~68.5 dB SPL in each 1/3-octave band from 500 Hz to 2 kHz center frequencies. Verify that the LFE channel generates a level of ~81.5 dB SPL in each 1/3-octave band from 25 Hz to 100 Hz center frequencies (which effectively covers the 20 Hz – 120 Hz bandwidth of the LFE channel). There might be some variation band-to-band, which is acceptable because this is a coarse setting.

8.3 Electroacoustic Measurements

The screen and LFE channels are to be measured using transfer-function mode. However, the surround array channels comprising more than one loudspeaker are to be measured using RTA mode.

The recommended channel sequence for taking measurements in a 7.1 channel system is: Center, Left, Right, LFE, Left Side Surround, Left Rear Surround, Right Side Surround, Right Rear Surround.

For each channel in turn follow these nine steps before proceeding to the next section (there are additional considerations for measurement of LFE/subwoofers in section 8.6.1 and of surround speakers in section 8.7.1):

1. Switch the analyzer to the proper mode (multi-time window transfer function for screen (Center, Left, Right) and LFE channels; RTA for surround array(s)).
2. When using transfer function mode, verify that the channel delay is properly set per section 7.5.
3. Play the [2095-1] pink noise signal through each loudspeaker channel to be measured, taking data simultaneously through all of the individual microphone measurement channels while employing data averaging per section 7.2 (8).
4. Observe the response from each microphone, and the average from all of the microphones. Compare individual microphone responses to each other and to the average. If the response from an individual microphone appears substantially different from the others (greater than ±3 dB at 1-octave resolution), that microphone and location should be investigated for the cause, which must be corrected if it is determined to be solvable.
5. Allow the signal to be averaged for at least 20 seconds. Store and log the measurement as "<channel name> as-found-response <YYYYMMDD>". Allowing the signal to be averaged for at least 20 seconds helps to ensure adequate immunity from incidental extraneous noises. Use a consistent time for all averages.
6. Log the system gain, equalization settings, and resulting SPL for each measurement channel.
7. If the "as found" measured response is out of the calibration target area of [ST 202], adjust the equalization of the loudspeaker channel under test so that the electroacoustic response is within the target area for that loudspeaker channel per section 6.
8. Listen to the results and compare the sound of the adjusted equalization to the "as found" equalization. If the new equalization sounds better than the "as found" equalization, was not excessive (<6 dB), and uses filters in an efficient manner, then this indicates good tuning and may remain. If not, readjust the equalization settings until these criteria are met.
9. Once satisfied with the equalization adjustments, measure the electroacoustic response. Store and log the frequency and phase response data as "<channel-name> <mic-#> final-response <YYYYMMDD>". It is advisable to take a screen-shot of the results.
10. If using a digital processor, save the filter settings. If an analog equalization filter set is used, take a photo of the settings and write down the approximate values.

8.4 Considerations in Measuring and Adjusting Screen Channel Loudspeaker Equalization (informative)

With reference to figure 6.1, the electroacoustic response of the screen channels between 50 Hz and 2 kHz should be essentially flat within the target area using as little equalization as possible, with a smooth transition centered at 2 kHz that follows the rolloff within the target area to a nominal -3 dB at 4 kHz. Avoid boosting at 2 kHz to create a sharp knee or break point. On graphic equalizers, follow the trend of the 1.6 kHz and

2.5 kHz band gains when setting the gain of the 2 kHz band. Above 4 kHz, the response will follow a gentle slope of approximately 3 dB/octave to 10 kHz, and a steeper rolloff above.

If using parametric equalizers, avoid the use of narrow band or high-Q filters. If using graphic equalizers, avoid abrupt changes between adjacent frequency bands. In general for any type of equalizer, attenuating frequency bands is preferred to boosting. Avoid equalization of uncorrectable acoustic anomalies per section 8.1. When in doubt about how to set an individual filter, listen to program material and verify that applying this filter improves the result.

Some systems will require a fair amount of high frequency equalization to attain screen channel responses within the target area, depending on the capability of the high frequency driver, screen transparency, air attenuation and other factors. The technician is cautioned that use of large amounts of high frequency equalization can compromise headroom, increase distortion, contribute to component failure, and cause listener fatigue. If the system is not reaching the target area on the high end during the equalization process, and previous checks show that the system is functioning correctly, follow the guideline in section 8.1. If it is found that a certain degree of equalization adjustment fails to achieve a similar degree of acoustic effect, that is evidence that the problem cannot be effectively addressed by equalization and no more should be applied. If the system requires a massive high frequency boost to attain response within the target area, pay particular attention while conducting the listening step in section 8.10 to listen to program material that includes dialogue with strong sibilant “s” at loud levels to verify whether the boost sounds good and does not cause sibilance or stridency. It is better to allow additional high frequency rolloff than to over-equalize and strain the system.

Below 50 Hz, though the target area increases to accommodate the variability of low frequency capability of playback systems and the nature of different rooms, the response would nonetheless ideally follow a well-behaved rolloff to its low-end response capability. In general it is not a good idea to apply a lot of boost equalization to compensate for systems with poor low end response as this will sound unnatural and could compromise headroom and damage components.

When the equalization has been completed, listen to the results and adjust the final treble and bass response using known listening material per section 8.10.

8.5 Check Screen Loudspeakers Timbre Matching

After performing the electroacoustic measurements and EQ adjustments above, the technician shall verify timbre matching among the screen loudspeaker channels.

- Sit at the microphone 1 position. Play [ST 2095-1] pink noise through each speaker individually while looking directly at the active speaker. Verify that the screen channels sound similar.
- Next, listen to known reference content. A good test is to listen to a single channel of known material in the same manner as in the pink noise test above.

If the screen speakers’ timbre do not audibly match, investigate the cause. The first step is to bypass equalization on the playback channel in question, leaving in the crossover settings. Then check the timbre matching per the above steps.

With the equalization bypassed, if the differences among the playback channels are audibly small, and the differences are judged to be largely based on issues such as proximity to walls, then the channel-to-channel equalization should be revisited and choices in filters used should be adjusted to correct the differences in timbre, with the caveat that the channel responses as measured are still within the calibration target area, and the in-band responses and equalization settings are within the ranges specified in section 6.

If there are significant audible differences with the equalization bypassed, then equalization cannot be used to correct the problem and the technician must investigate the cause. Possible causes might be different drivers, different crossover components, issues with power amplifiers, or issues with routers or in-line processors. Once the issue causing the significant difference is resolved, the timbre matching and equalization process must be repeated. Be aware that if this comparison test is done while looking straight ahead and not directly at each speaker there will be minor differences in the timbre of the sound.

8.6 Low Frequency Effects Channel Measurement and Adjustment Considerations

8.6.1 Introduction (Informative)

The nominal (and typical) operating range of a cinema low frequency effects channel (LFE) is flat over 25 Hz – 120 Hz ±3 dB, with a relatively steep rolloff above the passband. Some subwoofers used in LFE channels are installed with high pass filters to restrict the low frequency extension of the signal reaching the subwoofers. Also, differing performance among subwoofers might mean that not all subwoofers will have identical response to 120 Hz, and the rolloff of an individual subwoofer above 120 Hz might vary subwoofer-to-subwoofer, based on the design.

The technician needs to be aware of the filter sets employed in the subwoofer system that comprise the LFE channel when making measurements and adjusting equalization. Filters used to create a defined band-pass response are to be treated as fixed, and any other equalization that might be employed to address anomalies can be treated as adjustable.

8.6.2 LFE Equalization Adjustment

If equalization adjustments are indicated, verify that the fixed high pass and low pass filters are engaged. If these filters do not already exist for the given LFE system, the technician should create them if the equalization processor allows for it.

Take measurements per section 8.3.

If the response within the pass-band varies more than ±3 dB from a nominal flat response, and listening tests indicate that equalization is required, apply additional filters to correct for any major deviations in response so the result is within ±3 dB response tolerance in the pass-band 25 Hz – 120 Hz.

In some cases, the measured LFE response might have a definite peak rather than flat response across the pass-band. In this case, it is best to apply attenuation EQ to flatten the response, and adjust the overall channel gain to be within the target area tolerance, rather than applying boost to what appears to be a response deficiency. Boost-EQ on LFE/subwoofer systems is rarely effective, and can compromise headroom. To prevent over equalization of the subwoofer channel, some processors limit the available bands of equalization to as few as one parametric filter.

8.7 Surround Channel Measurement and Adjustment Considerations

8.7.1 Introduction (Informative)

The traditional use of surround loudspeakers was to envelop the listener, and, as such, the arrays of loudspeakers were intended to provide a more diffuse soundfield. In order to obtain fairly uniform coverage of the auditorium's seating area, several loudspeakers are employed for each surround channel, the collection of which is called the "surround array". The number and spacing of loudspeakers in a surround array will vary based on the dimensions and seating layout of the auditorium.

Modern soundtracks also place specific sounds in individual surround channels, in addition to ambient sounds; thus surround channels serve a dual purpose. In older surround sound systems, the frequency response of the surround channels was bandwidth-limited at the low and high end, but the current calibration target area for surround channels indicates that they need to deliver a frequency response similar to that of the screen channels.

Some older systems had all of the loudspeakers for each surround channel connected to one amplifier channel, but many modern systems allow for addressable surround zones, or even individual loudspeakers. If the surround sound system allows this level of control, it is best to first verify the performance of individual segments (zones or loudspeakers) of the surround array, and then set the final calibration by testing the array as a whole.

8.7.2 Measuring and Adjusting Surround Channel Electroacoustic Response

If the surround speakers have an “X-Curve filter”, it should be set to flat.

When equalizing surround channels, avoid the use of narrow band or high-Q filters to create a sharp knee or break point at 2 kHz, it should be a smooth rolloff. Note also that [ST 202] Appendix A.5.f. allows for raising the high frequency rolloff point to 4 kHz and altering the slope of the high frequency rolloff under certain conditions. Ultimately, the response of the surround speakers should be audibly timbre-matched to the screen loudspeakers.

In individual surround channel loudspeaker arrays in many cinema sound systems, each small group of loudspeakers, and in some cases each loudspeaker, is individually addressable. In such instances, each small group, or where possible each loudspeaker, first should be measured and calibrated using transfer function measurements prior to checking the performance of the entire array. If the system being tested does not have the ability to address smaller zones of speakers within the larger array, proceed per paragraph 8.7.3. below.

For systems that allow each surround loudspeaker to be addressed, a transfer function analysis may be used. Measure each individual speaker with system-EQ bypassed, and determine if it falls within the calibration target area for surround speakers per section 6. Engage the system-EQ and observe the differences. If this EQ puts the response into the target area and is not excessive, then it may remain. If not, it might be best to flatten the EQ and start from scratch. Make minimal adjustments only, because when the loudspeakers are summed within the array, the individual responses will change due to the overlapping sound from other loudspeakers in that surround array. Store and log the measurements.

8.7.3 Check Full Surround Array

Once the individual speakers are set, measure each array, Left Side Surround, Right Side Surround, Left Rear Surround (if applicable), and Right Rear Surround (if applicable). Select the analyzer’s RTA mode, with a measurement-view smoothing resolution of 1/3-octave. Check the measured response of each surround array. Most likely the response of each surround array will appear to be less smooth, but if it is within the target area tolerances it is best to not perform additional compensation EQ. If it is outside the target area tolerances, then additional equalization may be judiciously applied. It is best if this equalization is very broad rather than over narrow frequency bands.[1]

8.7.4 Check Surround Channel Timbre Matching to Screen Channel Sound

The technician shall verify acceptable timbre matching of the surrounds against the screen channels as described below. Depending on the size and frequency response of the surround speakers, the frequency extremes might not sound the same as the screen channels, but the main “body” of the sound (approximately 250 Hz – 2 kHz) should match.

1. While looking directly at the center channel speaker listen to [ST 2095-1] pink noise through the center channel screen loudspeaker. Then look directly at a surround channel and play [ST 2095-1] pink noise through the surround channel. Listen and see if the surround channel reasonably matches the timbre of the center channel.
2. Listen to known reference program content. A good test is to listen to a single channel of known content using the same procedure as for the above pink-noise test.

If the timbre match of the surround channel to the screen channel is not acceptable, the technician shall investigate the cause. The first step is to bypass equalization on the surround channel. Then check the following items:

- Verify that any X-Curve switches on surrounds are in the off position.
- Verify that all drivers in all loudspeakers are in good condition.
- Verify that all surround speakers match in phase and polarity.

Re-evaluate the equalization of the surround channel and adjust the filters to correct the differences in timbre, with the caveat that the surround channel responses as measured are still within the calibration target area tolerance and the in-band responses and equalization settings are within the range specified in section 6. Note that if the comparison listening is done while facing the center channel and all channels are listened to and the head is not moved to look directly at the channel playing material the surrounds will be perceived as having slightly less high end as compared to the center channel due to the nature of how human hearing works.

8.8 Setting Final Reference Sound Pressure Level

Once the electroacoustic responses of all channels have been set and documented, the final sound pressure level of each channel is set.

Setting the final reference sound pressure level will require proper tools and techniques. It is recommended that an A-weighted and a C-weighted measurement be taken, as well as a spectral measurement.

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[1] In case of certain room geometry and surround array loudspeaker pitch, it may be useful to check the response from a pink noise signal [ST 2095-1] sent to all surround arrays at once. Using a 1/2” random incidence microphone at the reference position pointed straight up, and the analyzer in RTA mode, the best match to the screen channels can appear relatively flat to 8 kHz.

In addition to the FFT-based analyzer with RTA capability, it is good to have an accurate wideband true-RMS SPL meter with C-weighting and A-weighting capability in order to perform simultaneous comparisons. If the FFT-based analyzer already has these capabilities and a true-RMS SPL meter is not available, then multiple readings can be taken in different measurement modes.

Sound pressure level must be measured over a defined bandwidth. If there is any variation in the frequency response over that range, it will be reflected in the SPL values. For the screen and surround channels, the measured bandwidth is ideally the two octaves from 500 Hz – 2 kHz, and for the LFE channel it is ideally the two octaves from 25 Hz – 100 Hz. These frequency ranges are in the flattest part of the bandwidth response, and are more likely to avoid the effects of uncertain loudspeaker responses.

To set the reference playback level, with the main fader set to reference level position, which is 7.0 on most cinema processors, [ST 2095-1] pink noise is injected into the sound system at the start of the playback chain, using the setup detailed in section 7.2.

Each overall SPL measurement shall be taken using the microphone locations already in place, with the analyzer set to compute the spatial average.

Set the analyzer to its 1/3-octave RTA mode and configure it to also compute an overall SPL (C-weighted, slow-reading: ≥1 second). If the analyzer allows simultaneous monitoring of two SPL values, set it to measure a second level: an overall A-weighted, slow-reading (≥1 second) SPL measurement. If the analyzer cannot take both measurements simultaneously, perform them sequentially (Alternatively, if available, set up the portable SPL meter for A-weighting, slow-reading: ≥1 second).

It is best to start with the center screen-loudspeaker channel, then the other screen channels. Next, test the surround channels and finally the LFE channel.

Procedure

Play the [ST 2095-1] pink noise through the channel-under-test for a long-enough duration that the measurement settles on the analyzer. At the same time, take an averaged A-weighted measurement with the portable SPL-meter in-hand while walking across the listening area. Set the SPL levels for each loudspeaker channel based on the best correlation of the spectral 1/3-octave bands, A-weighted, and C-weighted readings for each loudspeaker as noted in Table 8.8.

Measurement conditions	Screen channel	LFE	Surround channel
Sound Level Meter			
C-weighted SPL	85.0 dB	n/a	82.0 dB
A-weighted SPL	82.0 dB	n/a	79.0 dB
RTA (1/3-octave resolution)			
Average in-band SPL per each 1/3-octave	71.5 dB between 500 Hz and 2000 Hz center-frequencies	81.5 dB between 25 Hz and 100 Hz center-frequencies	68.5 dB between 500 Hz and 2000 Hz center-frequencies

Table 8.8. Sound Pressure Levels.

- Note that the in-band SPL levels in Table 8.8. vary from [RP 200]. The values in this table take into account C-weighting and the X-Curve response, whereas the value of 70 dB in [RP 200] is reflective only of flat pink noise.
- Compare the A-weighted level against the C-weighted level. If they differ by <2 dB or >4 dB then the equalization of the system should be checked to verify its conformance to the frequency response characteristics of section 6.If the A-weighted and C-weighted levels do not correlate per Table 8.8, the final level setting should be based on the A-weighted levels as the A-weighted levels better correlate to perceived dialog level.
- LAeq and LCeq measurements may be used for the screen channels and surrounds using a single microphone moved through a listening area. LAeq and LCeq levels should match the levels in Table 8.8.When setting levels in a 7.1 channel format it is necessary to verify that in 5.1 channel format the surround channels also adhere to the target levels. If the level of the surrounds in the 5.1 channel format is different than in the 7.1 channel format, check the settings of the 5.1 channel format and adjust the surround-offset per the manufacturer's installation manual for proper surround level in the 5.1 channel format.
- A broadband SPL meter is not recommended for setting the LFE/subwoofer channel reference level, as the measured level might vary widely depending on the meter.

Storing the data

After setting the level of each channel, store the frequency-response data and the computed C-weighted SPL. Also store the computed A-weighted SPL, whether simultaneously or sequentially acquired, or the A-weighted SPL from the portable meter.

After the reference SPL has been set, it is a good idea to take a reading using just the primary microphone position. Take and store the 1/3-octave frequency response plus the A-weighted and C-weighted SPL values at the primary microphone location. This will enable future calibrations to be accurately compared against this set with only one microphone, but be sure to make note of the microphone type, position (in all three axes) and orientation.

8.9 Picture and Sound Synchronization

To verify synchronization (sync), a playback file will be needed that has a white frame flash aligned with a -20 dB FS 1 kHz sine wave that is one frame in duration on the center channel playback track. This should be in a digital cinema package (DCP) on the server in cinemas using digital cinema projection. In a dubbing theater it should be played back from the recorder that hosts the final tracks.

Also, a device will be required that is capable of responding to the light and sound arrival at a location within the theater and can correlate the two, providing a numerically displayed offset value in milliseconds (ms).

- In a dubbing stage place the measuring device at the primary microphone location. In a cinema or screening room place the measuring device in the middle row of the audience seating.
- Play the test track.
- Observe the offset value and adjust the processing delay by this amount. In an exhibition theater this is adjusted in the D-Cinema server. In a dubbing theater it is typically adjusted using an offset in the master synchronizer for the dubbing stage.
- Test again and verify that the offset value between the flash and the sound is zero. If not, make fine adjustments to the processing delay to obtain zero offset. Once this is confirmed measure at 1 m in front of the first location and at 1 m behind the first location and note that the measured value changes as expected.

Alternative method:

- If a playback track with the flash and pop, and a measuring device, are not available, a countdown leader with a “2 pop” (a 1 kHz pop that is recorded on the 2-frame flash) can be used to judge sync offset by eye. The processing delay is then adjusted until the “2 pop” is judged to be in sync when viewed at the position used for setting sync as described above.

8.10 Listening to Program Material and Subjective Adjustments

After conducting measurements and setting the equalization based on data, and setting final levels for each channel, the final and most important step is to listen to the result.

Program content that is familiar to the technician shall be used. This program content should come from material that was mixed and prepared for a commercial cinema environment in a room calibrated to [ST 202]. Home theater content (Blu-ray Disc, DVD, etc) or music content that was mixed in a room calibrated to a different target curve than [ST 202] should not be used to judge system balance.

Some of the program material for subjective listening shall be selected to highlight dialog. Listen to different kinds of dialog: male, female, and males with deep voices. It might be desired that certain mid-bass 1/3-octave bands, or other adjustable filters, be adjusted to achieve the most natural sounding voices.

Some of the program material for subjective listening shall be selected to highlight treble-balance as well as any sibilance problem that might be present, and not because it might be a favorite mix (in fact, favorite mixes can easily distract the listener from critical evaluation). This means that program material used for treble evaluation shall be known to contain considerable wider-band high frequency program material such as wind instruments and orchestral strings, rather than just voices. It is recommended that each listening content segment be of no more than a few minutes duration. While listening, the technician shall adjust the broadband treble controls (if available) or a shelving filter of the screen channels equally, and by small increments, and only as needed, so that the sound is normally bright for the given program, but not so bright as to be irritating. Adjustments of the broadband treble controls should be very small: no more than ±2 dB; if more is needed, the equalization should be reevaluated.

If there is a sibilance problem, determine if it is a frequency response issue or a result of frequency based system distortion. The former can be corrected, the latter can only be subjectively minimized. Adjust the 1/3-octave or other narrow-band filters to address the offending frequency bands to subjectively compensate. Avoid the temptation to correct bandwidth-limited sibilance issues with wideband treble controls. After adjusting the screen channel treble response, adjust the treble controls equally for all surround channels. Listen again to the surround channel timbre against the screen channels per the procedure (section 8.7.4) to check that their treble balance matches that of the screen channels.

Some of the program material for subjective listening shall be selected to highlight the subwoofer balance and tone. This material should contain familiar programs with light, medium and heavy LFE channel content that is both music and effects. This allows the technician to determine if the audible subwoofer response and character is as it should be. As some processors offer limited adjustment parameters of the LFE channel, minor adjustments within +/-1 dB may be made to the overall level should equalization not be available to address any tonal issues. If equalization adjustments of more then ±2 dB are needed, the equalization should be re-evaluated.

If any changes are made to the equalization during the subjective listening process final reference measurements for each channel must be taken and saved.

8.11 Headroom Test/Distortion Sweep

Perform a final headroom and distortion check after the electroacoustic responses and sound pressure levels have been calibrated in order to verify that the headroom has not been compromised by equalization or other system adjustments during the calibration.

If the system passes this final test, verify that the resulting traces and data reflecting the final settings for each playback channel with each microphone have been taken. If this data has not been recorded be sure to record and store the final settings data for all playback channels. If analog, a photo of the controls might be preferable.

If the playback system does not pass this final distortion and headroom test, step back through the process to identify where in the playback chain the issue is being caused.

If any changes are made to the equalization during the subjective listening process final reference measurements for each channel must be taken and saved.

9 Record Keeping and Stored Data

Keeping physical and data records of a calibration in as detailed a manner as possible is required to maintain consistency of performance. This is especially true if different technicians might service the same facility or playback room. In order to maintain consistency in calibration and prepare for a subsequent maintenance calibration per [RP 2096-2], a record and a back-up record must created and kept consisting of:

- Date (in the format YYYYMMDD; which will automatically sort chronologically) and time (using a four-digit 24-hour format: HHMM, which also will sort chronologically within a day) of each calibration.
- Temperature and humidity during each calibration.
- Type of microphones used (including brand, model and serial number).
- Precise location (seat, and behind/in-front of it) and mounting position (height, and angle to the center channel speaker) of the microphones, with notation of which is the primary reference microphone for the measurement set, and notation of which is the measurement location that is to be used for subsequent maintenance calibrations using [RP2096-2].
- Types of equipment used (including brand, model and serial number).
- Variable settings within all equipment used.
- Stored data traces and screen shots of analyzer results of final settings from previous calibrations that can be referenced by all technicians calibrating the theater, with traces or data sets specifically designated to be the reference traces for future maintenance calibrations using [RP 2096-2].
- Notations of any special conditions, characteristics (including anomalies), and other considerations that can be informative to the technician.

Maintaining an archive of the calibration records for a particular room or facility is critical for consistency in calibrations over time. It is recommended that an archive be maintained of the calibration records for a particular room or facility in both analog and digital formats so that records are available easily and locally in the facility as well as being maintained at a central location should local copies be compromised.

NOTE: Sample record-keeping forms are in Annex A.

Annex A Supplemental Forms

The following forms can be used for Record Keeping and Data Storage

A.1 Detailed Calibration Checklist Forms

Theater

Date

Engineer

Test report number

Test Equipment - *make and model*

Operator should be able to demonstrate competent operation

RTA

FFT

RT

Multimeter

Polarity

Calibrator

Mic Cal Performed

Random Incidence / Direct

Mic type

No of mics

Survey - Measurements

Room Dimension L

W

H

Flat / Raked

Screen Type *1

Size H

Size W

Dist from left wall

**1 Screen type example; Perforated, Micro Perforated, Woven, or other type.*

If different from general room height

2/3 Ref Position Distance from screen

Ceiling Height (at REF position)

Survey - Loudspeakers - *Insert values in the relative boxes*

Cable values either metric or AWG / Ft

	Left	Left C	Center	Right C	Right	LFE	Side Srrnds	Rear Srrnds
Make								
Model								
Amp Make LOW								
Amp Model LOW								
Amp Make MID								
Amp Model MID								
Amp Make HI								
Amp Model HI								
Cable Length								
Cable Gauge LOW								
Cable Gauge MID								
Cable Gauge HI								
Amp Location								
*1 Power feed from								
Power feed size (A)								
*2 Polarity Check							*3 All tested	
LOW								
MID								
Hi								

**1 Power feed should be recorded as to which distribution board feeds the circuit for that channel*

**2 Polarity should be checked from Amplifier input to Loudspeaker output by use of a "popper" type tester (Insert value + or -)*

**3 All surround polarities should be tested individually and ensured to be the same, then the overall + or - value recorded here.*

Preliminary checks

Make / Model

System Processor

Room / Rack

Processor Location

Pass / Fail

Cabling Verified

Yes / No

Correct Cable Identification

Yes / No - check manufacturers spec

Check Safe X-Over Freq

Yes / No

Initial Amplifier Levels Set

Playback Channel Routing Check *Pass / Fail*

Pass / Fail

Room Vibration test (Freq sweep)

Pass / Fail

Fault Check

Pass / Fail

Check for Noise and Distortion

Where Items fail tests at this stage, cease all further tests and refer for maintenance before continuing.

Room RT time

Nominal - Attach graph of octave band RT

NC

Page 1 of 6

Theater		**Date**	
		Engineer	
		Test report number	

Acoustic tests

Full System Polarity Check - *system Input to driver output*

Record measured
polarity on this table
as value + or -

		LEFT		LEFT C		CENTER
	LOW		LOW		LOW	
3-way	MID		MID		MID	
	HI		HI		HI	

		RIGHT C		RIGHT		LFE
	LOW		LOW		1	
3-way	MID		MID		2	
	HI		HI			

Surround array components should be individually verified as having the same polarity as the rest of the array before recording the array polarity value

	Surround Array		*Surround Array*		*Surround Array*
Ls		Lr		Rs	
	Surround Array				
Rr					

Wrong results should be immediately rectified before continuing this test any further.

Crossover Frequency

Ensure measured response follows filter settings as expected

Failure to do so may indicate wrong settings or poor component choice

Record set frequencies
on this table
Record HPF to LPF values
Example [10Hz - 200Hz]

		LEFT		LEFT C		CENTER
	LOW		LOW		LOW	
3-way	MID		MID		MID	
	HI		HI		HI	

		RIGHT C		RIGHT		LFE
	LOW		LOW		1	
3-way	MID		MID		2	
	HI		HI			

Ensure values are within manufacturers design limits for the application

Check for Correct Summation at Crossover - *Check phase and amplitude response*

Confirm optimum results
on this table

		LEFT		LEFT C		section
2-way	L-M		L-M		L-M	
	M-H		M-H		M-H	

Record SUM / CANCEL

		RIGHT C		RIGHT
2-way	L-M		L-M	
	M-H		M-H	

CANCEL may indicate poor driver time alignment to the REF position, poor phase correction at crossover, or poor slope choice.

Apply surround loudspeaker delay (if required)

Record delay times of each array

Sur L Side Delay	Sur L Back Delay	Sur R Back Delay
Sur R Side Delay		

Theater

Date

Engineer

Test report number

Acoustic tests

Check Loudspeaker response (FLAT)

Prior to starting any application of reference curves check the system responds correctly to default settings and crossover points - rectify any problems

Store analyser plot and record file name here
Print plots and attach to
This document

Record plot for each surround array

LEFT	LEFT C	CENTER
RIGHT C	RIGHT	LFE 1 2
SURROUNDS		

Check Loudspeaker response (REF curve)

Apply SMPTE 202:2010 X-Curve where required

Store analyser plot and record file name here
Print plots and attach to
This document

Record plot for each surround array

LEFT	LEFT C	CENTER
RIGHT C	RIGHT	LFE 1 2
SURROUNDS		

Check Loudspeaker REF Output Level

Apply reference signal from program source playback device

Record measured Levels on this table

Any failure to meet the required standards should be recorded in the advisory section

LEFT	LEFT C	CENTER
RIGHT C	RIGHT	LFE 1 2
SURROUNDS		

Check for even coverage of audience areas

Make an evaluative aural assessment of channel coverage

Ensure that there is an even coverage of the room from each channel without dead spots.

Record observed results on this table

Record PASS / FAIL
Record results of FAIL in Advisory Section

LEFT	LEFT C	CENTER
RIGHT C	RIGHT	LFE 1 2
SURROUNDS		

Signal Clip and Limiter Check

This check ensures that applied processing has not compromised system headroom
Carefully apply full scale level program material and check for system clip at all stages

Record observed results on this table

Record PASS / FAIL
Record results of FAIL in Advisory Section

LEFT	LEFT C	CENTER
RIGHT C	RIGHT	LFE 1 2
SURROUNDS		

Ensure all system limiters are NOT activating in normal use.

Theater ☐

Date ☐

Engineer ☐

Test report number ☐

Acoustic tests Verification and final tests

General *Any test failed should be reported in the advisory comment section*

Test for system stress under high load PASS ☐ FAIL ☐ *Run wide band program material to full scale and listen*

Test for background noise

Room NC without system powered ☐	Room NC with system powered ☐
Is noise audible at seating positions? ☐	Is noise intrusive? ☐

If the system noise is too high, system gain structure may need resetting.

Listening to Content and Subjective Assessments

Test for acceptable general quality by playing known dynamic program material with picture and watching for a reasonable period

Any test failed should be reported in the advisory comment section

Does the system perform as expected?	Yes ☐	No ☐
Is the system free of distortion?	Yes ☐	No ☐
Is the volume level correct for the room?	Yes ☐	No ☐
Is the sound in sync with the picture?	Yes ☐	No ☐
Does the frequency balance sound correct?	Yes ☐	No ☐
Is intelligibility acceptable	Yes ☐	No ☐

Final Checks

Outside Influences
(Electrical)

Run all room systems including ventilation, projectors and lighting dimmers and listen for any excessive noise

These systems may not have been running in theater down time or during testing

Any test failed should be reported in the advisory comment section

Interference from ventilation?	Yes ☐	No ☐
Interference from Dimmers?	Yes ☐	No ☐
Interference from other systems?	Yes ☐	No ☐

Check against all applicable standards *Check against all theater certification requirements that apply to this theater*

Does the theater comply to the required standards	Yes ☐	No ☐

Housekeeping

Have all adjustable controls been sealed against tampering and their values recorded?	Yes ☐	No ☐
Have all test systems been removed and all controls left in operating condition? - *including sound check*	Yes ☐	No ☐

System processor file saved as ☐

DATE ☐

System test periodicity

Due to the specific factors affecting this installation and type of equipment used, it is recommended that this system is periodically inspected by experts to ensure Safe and correct functioning at the following intervals for the duration of the life of the system

The system will be inspected NO LESS THAN EVERY ☐ MONTHS

Reference mic position

Draw mic layout with dimensions

Screen

LOUDSPEAKERS

LEFT CENTER RIGHT

Back of room

REF mic position	Distance from screen	Distance from right wall	Seat number	Height
Mic position 1				
Mic position 2				
Mic position 3				
Mic position 4				
Mic position 5				
Mic position 6				
Mic position 7				

Advisory Comments

TEST	COMMENT

Test report number ☐

CERTIFICATION

Sign and date the appropriate section and cross out the non-applicable sections

1 **The system has passed** all tests in a satisfactory manner

Date		Engineer (Signature)		Company	

2 **The system has passed** the majority of tests and complies to applicable standards however
There are areas of concern outlined in the advisory section which compromise performance

Date		Engineer (Signature)		Company	

3 **The system has failed** to achieve an acceptable standard of operation and will not meet the required standards because of the following reasons

Date		Engineer (Signature)		Company	

Reason

Re-test date *See recommended test periodicity on page 4 of this document*

NEXT PERIODIC INSPECTION WILL BE DUE NO LATER THAN / / DATE

Test expiration date sticker should be affixed to the main equipment rack

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Annex B Microphone Considerations

B.1 Multi-Microphone Data Arithmetic Averaging

The calculation of a spatial average can be performed by the sum of the squares of the sound pressure levels:

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

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

$$\mathrm{SPL} = 10\ \log_{10}\left[\frac{1}{4}\left(10^{\frac{\mathrm{L1}}{10}} + 10^{\frac{\mathrm{L2}}{10}} + 10^{\frac{\mathrm{L3}}{10}} + 10^{\frac{\mathrm{L4}}{10}}\right)\right]$$

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 might indicate **significant acoustic** or **loudspeaker coverage** problems.

B.2 Microphone Aiming For Various Microphone Types (Informative)

Measurement microphones are pressure devices and are therefore omnidirectional at frequencies up to those that have wavelengths comparable to the diaphragm diameter. Above that they become progressively more directional. The impact of the directionality increases with microphone diameter. However all microphones have incidence angles that exhibit flat frequency response over a useful frequency range.

Microphones calibrated as “free field” are flat on the axis that is perpendicular to the diaphragm i.e. 0° straight ahead. Such microphones are intended for aiming at the loudspeaker being measured.

In this recommended practice, we utilize the fact that very good and consistent results can be obtained by aiming 1/4” free-field microphones at 90º incidence (also known as the “random incidence position”), and thus this is the stated method. The errors are small: a small high frequency rolloff above 8 kHz and an error of 2 – 3 dB at 12.5 kHz for typical auditoriums. Applying diffuse field correction the error will be reduced to less than 1 dB at 12.5 kHz error if no ceiling speakers are measured.

Microphones calibrated as “diffuse field” a.k.a. “random-incidence” are calibrated so that the average response is flat for sound arriving from all directions. These microphones are also flat in the plane parallel to the diaphragm i.e. 90° to the microphone. Such microphones must be angled at 90° to each loudspeaker being measured. With this aiming, direct and diffuse sound components are measured approximately flat.

However, incorrect sound source incidence causes measurement errors at high frequencies. Direct sound arriving sidewise at a free field corrected microphone measures with a high frequency rolloff. A diffuse field compensated microphone measures direct sound arriving on axis with a high frequency boost.

The magnitude of these errors increases with the microphone diameter.

Using 1/2” free field microphones requires that the microphone is always exactly pointing to the loudspeaker to be measured. While this can give very accurate results, this method requires keen attention to detail, constant interaction with the microphones, and extremely detailed records. Misaiming of a 1/2” free field microphone will result in significant measurement errors of the high frequency response.

Therefore it is strongly recommended to use 1/4” measurement microphones compensated in the analyzer as described in Annex B.3 for the equalization work because these allow the microphones to be aimed pointing straight up to the ceiling for all channels regardless of the source incidence as detailed in this RP. This method is easy to use and gives reproducible results.

For other setups the microphones should be aimed according to the following decision table for best results.

Microphone diameter and type	Analyzer compensation	Screen channel aiming	Surround channel aiming	Requires re-aiming
7 mm (1/4”) and 9 mm (3/8”)	90°/diffuse field calibration chart	Straight up at the ceiling	Straight up at the ceiling	No
7 mm (1/4”) and 9 mm (3/8”)	0°/free field calibration chart	Pointing to the center speaker	Pointing to the middle speaker of each surround array	Yes
Diffuse field 7 mm (1/4”) and 9 mm (3/8”)	No	Straight up at the ceiling	Straight up at the ceiling	No
Free field 7 mm (1/4”) and 9 mm (3/8”)	No	Pointing to the center speaker	Pointing to the middle speaker of each surround array	Yes
Partially compensated 9 mm (3/8”) with high frequency boost but less than 2.5 dB at 12.5 kHz in the 0° response curve	No	Pointing to center speaker with 45° vertical angle	Straight up at the ceiling	Yes

The aiming has no impact on the measurement of the LFE channel. No re-aiming is required for the LFE channel.

B.3 Microphone compensation in the analyzer (Informative)

Many analyzers provide the option to load microphone calibration data to compensate the microphone for improved measurement accuracy.

For the recommended straight up microphone aiming, the individual calibration charts for 90° incidence a.k.a. diffuse field response should be used for best accuracy.

Some measurement microphones only come with calibration charts for free field/0° incidence response. In this case the manufacturer should be asked for the individual calibration charts for 90°/diffuse field response.

Typically there are some calibration differences between the five microphones. Therefore, the average of the individual calibration charts should be loaded into the analyzer. When averaging the five microphone responses, the errors are essentially nulled out.

Follow the analyzer manufacturer instructions for loading the compensation data. It should be carefully verified whether the analyzer expects the microphone response to be loaded or the inverse of it.

Bibliography (informative)

The following citations provide helpful information.

A. ISO 3382-1, “Acoustics — Measurement of Room Acoustic Parameters — Part 1: Performance Spaces”.

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E. TC-25CSS B-Chain Frequency and Temporal Response Analysis of Theaters and Dubbing Stages; 1 October 2014; https://www.smpte.org/sites/default/files/SMPTE TC-25CSS-B CHAIN FREQUENCY AND TEMPORAL RESPONSE ANALYSIS OF THEATRES AND DUBBING STAGES 1 Oct 2014.pdf