VOLUME 9, ISSUE 2 APRIL 2013

Acoustics Today

Acoustics of Recording Studios
Control Room Design
Motion Picture Scoring Stages
Music, Rooms, and Listeners
And more

A publication of the Acoustical Society of America

Acoustics of the Recording Arts

EDITOR
Dick Stern (AcousticsToday@aip.org)

CONTRIBUTORS
George L. Augspurger, Marshall Long,
Elaine Moran, Shawn Murphy,
Richard Stern, Floyd E. Toole

ADDRESS
Acoustical Society of America
Publications Office
PO Box 274
1170 Main Street
W. Barnstable, MA 02668-1162

ASA EXECUTIVE DIRECTOR
Charles E. Schmid

ASA EDITOR-IN-CHIEF
Allan D. Pierce

ADVERTISING
Robert Finnegan, Director, Journal Advertising & Exhibits, AIP

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FUTURE ARTICLES

(Guest Editors)

Noise

(Scott Sommerfeldt)

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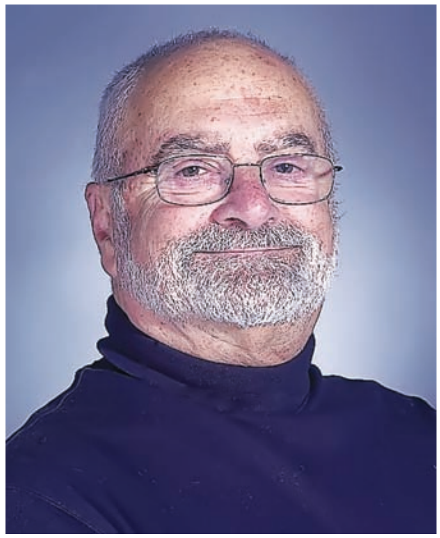
RICHARD STERN

1929–2013



The Acoustical Society of America recently learned that Richard (Dick) Stern, founding editor of *Acoustics Today*, died on June 19, 2013. No further details are available at the time of publication, but a full obituary is anticipated for the next issue.

Acoustics Today

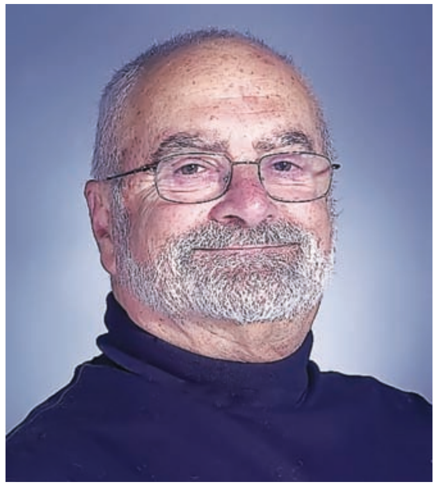
A Publication of the Acoustical Society of America

Volume 9, Issue 2 **April 2013**

Cover: HUM Studio A, Santa Monica, California, USA (Acoustician: Marshall Long, Acoustics), (Architect: Walter Meyer Associates)

FROM THE EDITOR

Dick Stern
Acoustical Society of America
Suite 1NO1 2 Huntington Quadrangle
Melville, New York 11747-4502



Audio recording is a fundamental link in the chain of the preservation of vocal, instrumental, and effects performances. As such it is a creative nexus of audio and acoustical engineering, involving both technical and artistic components. Each of these disciplines can learn from the other and it is in the hope of increased cross fertilization that our guest editor, Marshall Long, has invited a group of talented experts to talk about what they do. The Society is fortunate that they have taken the time to contribute to this issue. To obtain an overall perspective, Marshall has introduced each of the authors in *From the Guest Editor*.

Dick Stern

ACOUSTICS OF THE RECORDING ARTS

Marshall Long

To explore the interactions between recording, mixing, editing, and production as well as the acoustics of the spaces in which these activities occur, we have invited several authors to discuss each of the steps involved.

George L. Augspurger received a B.A. degree from Arizona State University at Tempe and an M.A. degree from the University of California, Los Angeles (UCLA), followed by postgraduate work at Northwestern University. After working in sound contracting and television production he joined James B. Lansing Sound, where he served as Technical Service Manager and later as Manager of the newly formed Professional Products Division. In 1970 Mr. Augspurger left JBL to devote full time to Perception Inc., a consulting office specializing in architectural acoustics and sound system design.

Mr. Augspurger is a fellow of the Acoustical Society of America, a fellow of the Audio Engineering Society, a member of the United States Institute for Theatre Technology, and a member of the National Council of Acoustical Consultants. His name is familiar as the author of numerous articles and technical papers, mostly dealing with loudspeaker design and application. His double-chamber speaker enclosure described in the December 1961 issue of *Electronics World* is still a favorite of amateur speaker builders. Today, there are more than 100 installations of custom monitor loudspeakers designed by Mr. Augspurger in professional recording studios throughout the world.

He contributes regularly to the Patent Reviews published by the *Journal of the Acoustical Society of America*.

Shawn Murphy is a well known sound engineer and mixer. He received a BA from San Francisco State University and a MFA from Stanford University. He worked extensively as a Technical Director and Theatrical Sound Designer for

American Conservatory Theater (San Francisco), and the Oregon Shakespeare Festival. As a free-lance mixer and audio supervisor he has worked for the Academy Awards, Boston Pops Television Series (PBS), Great Performances (PBS), and the original audio design and installation at NBC Saturday Night. He has recently worked as a consultant and mixer for the Boston Symphony Orchestra, Pacific Symphony Orchestra, Tanglewood Music Festival and the Hollywood Bowl. Since 1983 he has worked on over 340 feature films, primarily as a recording engineer and mixer. He won an Academy Award for Best Sound for *Jurassic Park* and was also nominated for *Indiana Jones and the Last Crusade*, and *Star Wars Episode I: The Phantom Menace*. Other credits include: *Lincoln, The Bourne Legacy, The Bourne Ultimatum, The Hunger Games, Men in Black (I and III), Indiana Jones and the Kingdom of the Crystal Skull, Star Wars Episodes I, II, and III, Harry Potter and the Prisoner of Azkaban, Titanic, Saving Private Ryan, Jurassic Park, Apollo 13, Braveheart, Schindler's List*, and *Dances with Wolves*. He is a Fellow of the Audio Engineering Society and a Member of the Academy of Motion Picture Arts and Sciences.

Floyd E. Toole studied electrical engineering at the University of New Brunswick, and at the Imperial College of Science and Technology, University of London, where he received a Ph.D. In 1965 he joined the National Research Council of Canada, where he reached the position of Senior Research Officer in the Acoustics and Signal Processing Group. In 1991, he joined Harman International Industries, Inc. as Corporate Vice President—Acoustical Engineering. In this position he worked with all Harman International companies, and directed the Harman Research and Development Group, a central resource for technology development and subjective measurements, retiring in 2007.

His research focused on the acoustics and psychoacoustics of sound reproduction in small rooms, directed to improving engineering measurements, objectives for loudspeaker design and evaluation, and techniques for reducing variability at the loudspeaker/room/listener interface. For papers on these subjects he has received two Audio Engineering Society (AES) Publications Awards and the AES Silver Medal. He is a Fellow and Past President of the AES, a Fellow of the Acoustical Society of America, and a Fellow of CEDIA (Custom Design and Installation Association). He has been awarded Lifetime Achievement awards by CEDIA and ALMA (Association of Loudspeaker Manufacturing & Acoustics International). He is the author of the book, *Sound Reproduction—Loudspeakers and Rooms*.

Marshall Long is the guest editor of this issue. He received a B.S.E. from Princeton University in 1965, attended the University of Grenoble in France (1965-66), and the University of Madrid in Spain (1966), and received M.S. and Ph.D. degrees (Distinguished Graduate) from the University of California, Los Angeles (UCLA) in 1971. He held a post doctorate position at UCLA in 1972. Since 1972, he has been engaged in acoustical and audio visual engineering consulting as principal of the firm he founded. His firm has established a national and international reputation, completing over 3,000 projects in architectural acoustics, noise and vibration control, environmental impact assessment, and audio visual design. He is a Fellow of the Acoustical Society of America and author of the engineering textbook, *Architectural Acoustics*.

ACOUSTICS OF RECORDING STUDIOS

Marshall Long
Marshall Long Acoustics
13636 Riverside Drive
Sherman Oaks, California 91423

Process and environment

The recording of music, either voice or instrumental, is a core industry supporting the entertainment arts. The process begins with the production of music in a studio. Here the audio generated by a musical instrument is captured by a transducer, either a microphone or an electronic pickup built into the instrument itself. The electronic signal from the microphone is transmitted to a control room where it is routed through various electronic devices and stored for future use. The signals may be in analog or digital format and may be transmitted via electrical or fiber optic cables or by means of a wireless transmitter. The signal can subsequently be played back and remixed, stored, or combined with other recorded signals until the final product has been produced. It is then packaged as part of a storage device for later presentation in commercial or home theaters, or distributed electronically to a playback or other receiver system. This issue of *Acoustics Today* will deal with aspects of the interaction between these processes and the acoustical spaces in which they occur.

"Foley stages... are often indistinguishable from junkyards, due to the general clutter."

Studios

A studio, in the most general sense of the word, is a space where music is played, recorded, and edited. In a narrower sense it is where music is played and captured by a microphone. A control room is a separate room where one or more mixers work and music is played back, edited, and stored.

Studios can range in size, from a closet to scoring stages accommodating a full symphony orchestra. Figure 1 shows sketches of several different types. In small home studios the mix board and other electronic equipment is located in the same space as the musicians. In larger facilities these functions are separated into different rooms, which may in turn be subdivided. All studios have common requirements irrespective of size. There may, in addition, be specialized requirements which are size dependent or function dependent. A summary of the common requirements is listed in Table 1 below. Some of these are acoustical in nature while others are purely functional.

Home studios

Home studios, sometimes known as project studios, are increasingly common as high quality recording equipment becomes smaller and more affordable. The sophistication of this electronic gear has had a direct influence on the proliferation of small studios since excellent recordings can now be made in a low-cost environment. The initial reaction from commercial studios was an effort to limit home studios through land-use regulations. A decade or two ago, under pressure from the commercial studio owners, Los Angeles prohibited people from using their homes to make commercial recordings. This led the City of Los Angeles, with the worst smog in the country, to require that people get into their cars and drive to a commercial studio to do their work, while raising the cost of the process. Under the cost pressures, rather than environmental enlightenment, the amount of time given to prepare the audio for a 45 minute television program has decreased from a week to about two days. Mixers now spend one day doing the bulk of the work in a home studio and a second day presenting it to the "suits" and transferring the results into the studio memory banks. The home studios can be equipped with the same equipment as the commercial studio, so after getting the executive input, changes can be made at the commercial site with no loss in quality.

The second problem confronting a home studio user is that regulations in residential neighborhoods restrict noise levels at neighboring properties. These property-line ordinances typically limit nighttime noise levels to 45 dBA or 5 dB

Table 1 – Common Studio Requirements

Acoustical	Functional
Quiet – below NC 20	Adequate ventilation and thermal control
Isolation from the surroundings	Access to bathrooms
Adequate reverberation	Visual contact with the control room
Freedom from acoustical defects	Storage areas
Reasonable diffusion	Equipment maintenance facilities
Isolation for different instruments	Break rooms and private phone areas
Control of bass reverberation	Communication areas (internet access)
Variable absorption	Offices and conference rooms
Moveable gobos (reflecting or absorbing acoustic panels)	Handicapped access
	Access to cabling

over the existing ambient, whichever is higher, within residential properties. Property line ordinances can limit the level at which a musician can play or require substantial construction to meet the local codes. Neighbors can also be sensitive to musicians arriving on Harleys in the middle of the night.

Picking a location that provides natural sound isolation is a good start. If a basement is available, it is probably the best choice; or a separate structure can be used. Probably the most common choice, and one of the most difficult, is a garage. A two-car garage is about 24 feet square, enough for a one-room studio. However garages are lightly constructed and must be heavily reinforced to achieve adequate sound isolation. Exterior surfaces of plaster, brick, or cement board, having a surface mass density of 10 lbs/sq ft (49 kg/sq m) or greater, can be used on the exterior with double drywall interior surfaces supported on a separate framing system or resilient isolators. The garage door must be removed, although it can be retained as an applied decorative element over an exterior wall.

A garage ceiling is too low and raising it requires added structural framing and review by a structural engineer. For sound control it should have a solid plywood roof and, at a minimum, a separately supported double-drywall interior. The fan coil unit can be located above the ceiling with an access panel for service or a package unit outside the building can be employed. Careful calculations are necessary to ensure isolation of the fan coil supply and return from the studio. Silencers or snaked flexible duct surrounded by batt insulation can help provide the necessary attenuation.

Figure 2 gives an example of a design for a personal studio, built into a freestanding garage. It illustrates some of the difficulties in making a successful conversion. The layout of a successful project studio is quite personal and reflects the working habits of the user. In this example, the operator can mix and compose on a keyboard, which doubles as a Musical Instrument Digital Interface (midi) controller linked to a computer. A small number of musicians can be accommodated for a recording session. Movable wall panels, hung on angled supports, provide absorption and can be replaced with diffusive elements or simply removed. Storage closets also double as bass traps. The floors are hardwood with throw rugs for variable absorption.

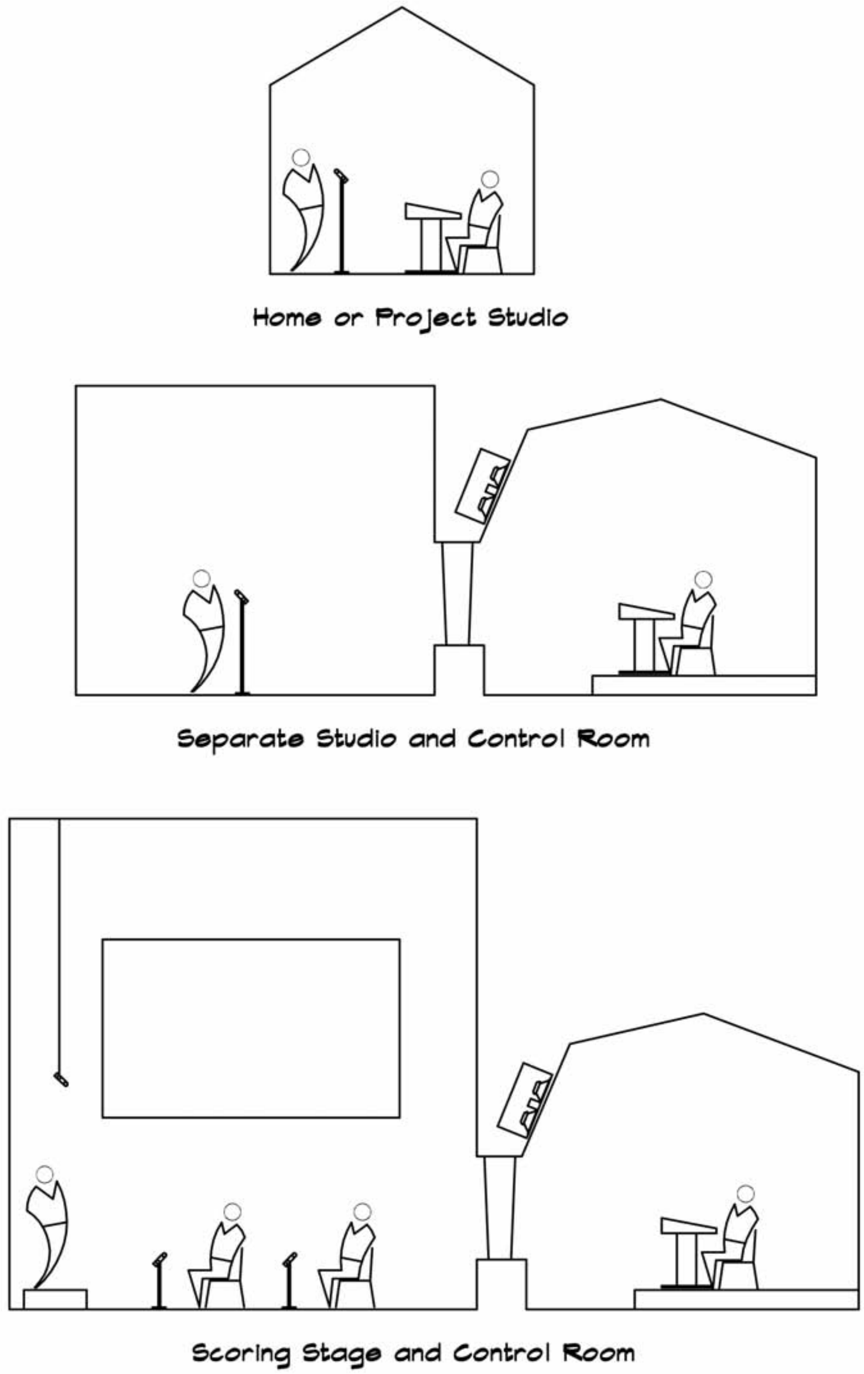


Fig. 1 Types of sound studios

Recording studios

Formal recording studios consist of one or more rooms, where music is played and recorded. The musicians may all be present at the same time or they may never see one another. With the ability to send recorded music from place to place electronically, musicians may perform in rooms a continent away and days or weeks apart. When musicians are playing simultaneously, separate rooms are desirable to isolate the instruments so that they do not bleed into other microphones. Isolation booths or simple baffles (gobos) also can be helpful in separating the studio into different acoustic environments.

Studios can be generic or highly personal, based on the working preferences of an individual user. A good example of the latter is Hum Studio in Santa Monica, CA, designed for Jeff Koz, a well-known composer. Figure 3 shows the floor

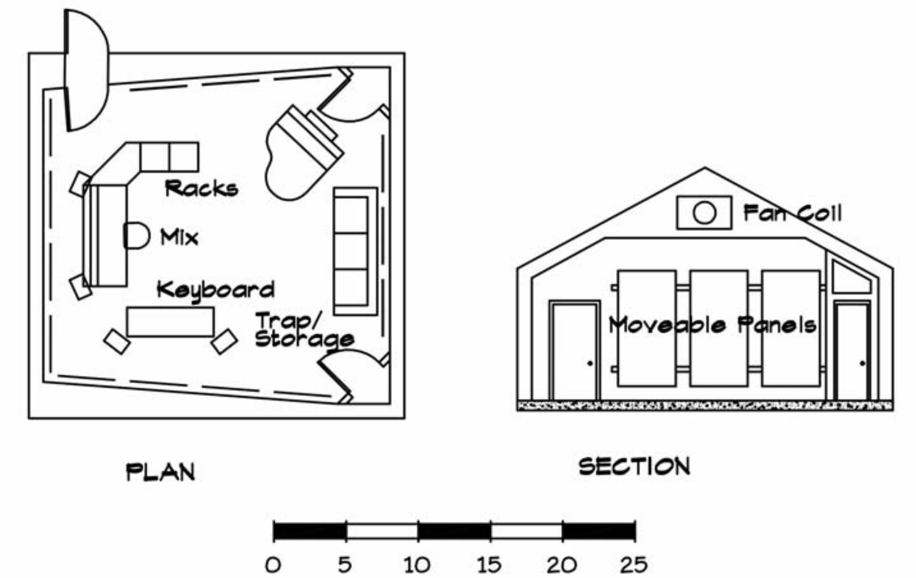


Fig. 2 An example of a garage studio

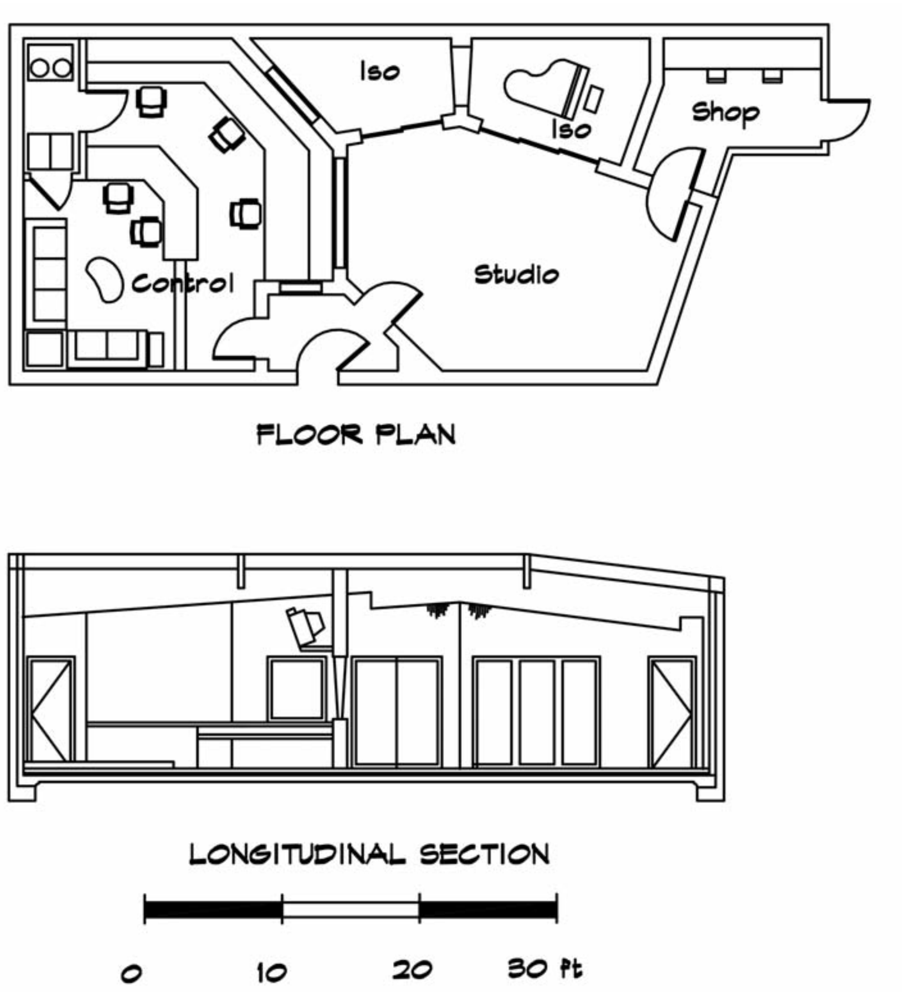


Fig. 3 Hum Studio A, Santa Monica, California, USA (Acoustician: Marshall Long Acoustics, Architect: Walter Meyer Associates)

plan for the main studio and control booth. Since most of the composition work is done at a keyboard with small digital mix boards, the traditional control room layout was not used. Instead, three work stations, each with a keyboard, a mix board and a computer, were arrayed along the front and side walls of the control room. Each could be used simultaneously during recording and mixing sessions. The main composing station was designed around the users' equipment. Since listening is done via small near-field loudspeakers, there was no need for large stereo monitors and no need for a large loudspeaker bridge above the main window. This arrangement freed up the center of the room for a client couch and social area instead of being dominated by a massive mixing console.

Hum studio consists of several rooms accessed from a small foyer separating the studio from the control room. Foyers can sometimes be used as isolation rooms particularly if there is a need for feedback such as with an electric guitar. Two isolation booths, with sliding glass doors, are available for individual instruments such as a piano or vocals. The walls and ceiling are constructed of multiple layers of drywall with a wood panel finish on the ceiling. Square quilted absorbers are hung from hooks on the walls and can be removed or folded to reduce their area. The mid-frequency reverberation time is about 1.2 sec and flat with frequency. Figure 4 shows the range of reverberation times appropriate for a sound studio.

Bass trapping is done using the return-air plenum built above the ceiling as illustrated in Fig. 5. Flexible ducts in this area make the space into a bass absorbing plenum. The area is filled with fiberglass insulation. Low frequency energy can enter via the diffusers and break out of the ducts into the treated volume. The segmented ceiling requires surface-applied wood diffusers to control flutter echo. The control room is designed to be much deader than the studio, about 0.5 sec at mid-frequencies. The walls are faced with 2" (52 mm) cloth-wrapped fiberglass panels. The ceiling is hard—two layers of 5/8” drywall hung from springs to provide noise isolation through the ceiling- roof. Bass traps are built into the space above the equipment closet and into the video monitor enclosure. Windows are arranged so that there is visual contact between the control room and any point in the studio, including the isolation booths.

Sound stages

Sound stages are large open rooms used for indoor movie production. Acoustically they are designed to be dead with all surfaces except the floors covered with 4 to 6 inch (100 to 150 mm) deep blankets of absorptive material. In the early 1950s many were built using recycled army mattresses hung on the walls. The floors are smooth and flat so that cameras can be dollied. The exposed wall surfaces can be faced with commercial quilted blankets covered with hardware cloth below an elevation of about 10 ft (3 m). The best rooms are built with isolated construction, floated floors, double-studded walls, and separately suspended drywall ceilings. Access is provided via sound rated doors, which can be quite large. Some facilities have control rooms adjacent to the stage for mixing and recording. Not infrequently, audio recording is done using directional microphones that transmit signals to wireless receivers located in racks in the same room but often 75-100 ft (23-30 m) away.

The most difficult aspect of sound stage design is noise control. Isolation from exterior noise is a challenge because many stages are built in converted warehouses with lightweight roofs and little thought to the isolation of traffic and aircraft noise. Large air conditioning units are required to cool the stage lighting fixtures and this equipment is often located on the roof, where it is difficult to control. It is preferable to separately support air handlers on grade or on an elevated steel platform dedicated to that purpose. Ductwork should be isolated from the structural framework either by resilient suspension or by a separate support system. Silencers located at a roof or wall penetration provide exterior as well as equipment noise control.

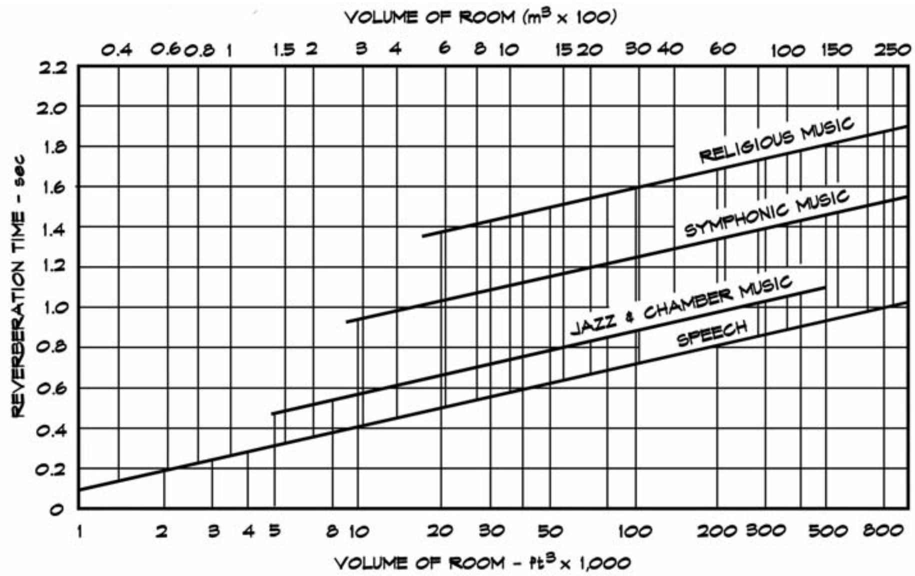


Fig. 4 Reverberation times for studos in the 500–1000 Hz. range (Doelle, 1972)

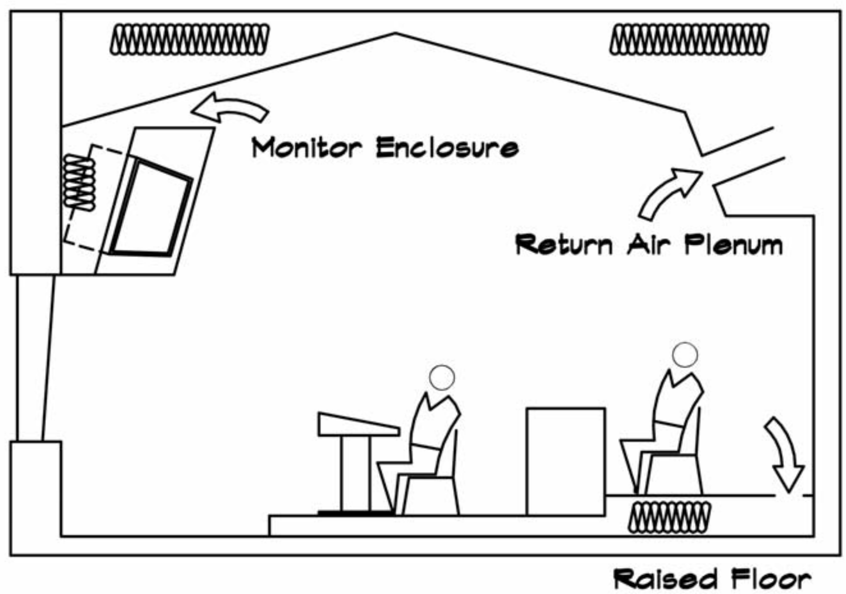


Fig. 5 Found space bass traps

Scoring stages

Scoring stages are rooms in which the music for a film is recorded. The orchestra conductor, who is often the composer, faces both the musicians and a large screen on which the film is projected. As he conducts, he may listen through a single headphone to a click track, which aids in synchronization of the film and the score. Visual cues are also projected onto the screen in the form of streamers that progress from left to right across the screen to mark the beginning of a transition or effect when they reach the right-hand side.

A scoring stage is large, almost the size of a concert hall. Like concert halls, the best ones are shoebox-shaped with high ceilings and irregularly shaped diffusers on the walls and ceiling. A very good one, Studio 1 at Abbey Road Studios in London, is shown in Fig. 6. Its dimensions are 92.6 ft × 59.7 ft × 39.4 ft high (28.2 m × 16.1 m × 12.2 m) and its total volume of 218,000 cu ft (6172 cu m) is about one-third that of Boston Symphony Hall. At one end there is a large (44 ft or 13.4 m wide) projection screen with the control room in an opposite corner.

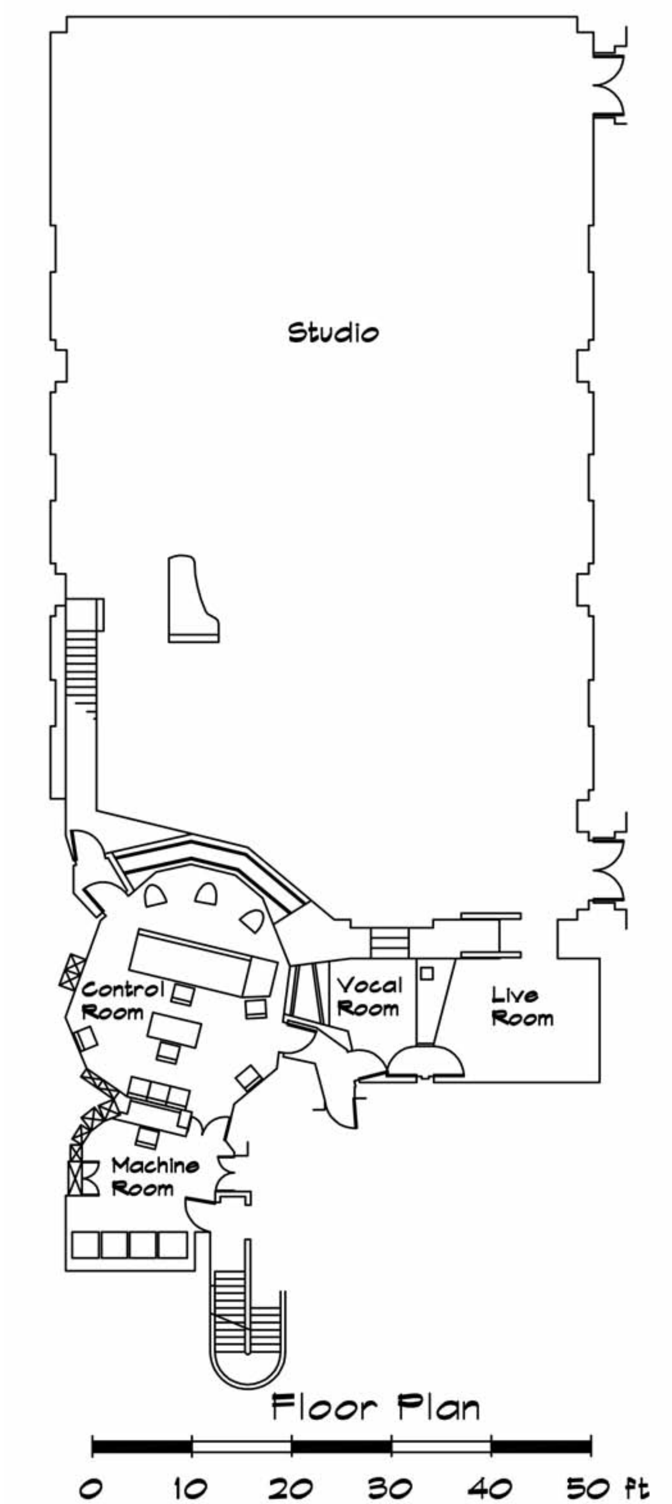


Fig. 6 Abbey Road Studio 1, London England (Abbey Road Studios, 2001)

Scoring stages are designed much like concert halls but without the requirements for an audience. The floors are flat and the walls and ceiling surfaces feature irregular shapes for diffusion. Reverberation times can be changed using moveable curtains or panels. For film, from 5 to 8 mics are used for the right-center-left and surround signals, and another 30 to 35 mics for individual instrument groups. The high ceilings sometimes make it difficult for the musicians to hear each other so 12 to 18 foldback channels are provided from the mix board to individual players through headphones. The orchestra can be seated on risers for visual cohesion and arranged to achieve a balanced sound.

Since sound stages are smaller than concert halls the orchestra cannot play quite as loudly as they would under performance conditions. When they do, the reverberation in the room, particularly the bass, can overwhelm the direct sound and yield a muddy recording. If the balance is correct and the control room is set up properly, the recording engineer can do a live mix including surrounds if necessary; however, the recorded tracks can be remixed at a later time, or used for sweetening.

The reverberation characteristics of a scoring stage are much the same as a concert hall. Abbey Road, in London, has a mid frequency reverberation time of about 2.2 seconds, rising slightly at the lower frequencies and remaining fairly constant at high frequencies. The lack of audience and seat absorption limits the falloff of the high frequencies to that due to curtains, musicians, and air absorption, so these rooms can be somewhat brighter than a performance hall. These stages have multiple hanging curtains, suspended on line sets from the ceiling, which can be lowered to reduce the reverberation time.

The recording of symphonic music can also be done in an empty concert hall. In these cases the room is often extensively modified to accommodate this use. For example, when Royce Hall at UCLA is used for recording, a wooden platform for the musicians is constructed over a portion of the seating area and the opera chairs in the orchestra section are covered with 3/4” (19 mm) plywood over visqueen sheets to decrease high-frequency absorption (Murphy, 2001).

Foley

Foley stages, where sound effects are generated by physical manipulation of devices, are often indistinguishable from landfills, due to the general clutter. They were named for Jack Foley, an early sound effects pioneer. A typical Foley stage consists of a dead room with walls and ceiling covered in broadband absorption and a hard-surface floor having multiple pits each 3 to 4 feet square, in which there are different walking surface materials. The Foley artists watch the film, projected on a screen against one wall, while making the sound effects with their hands and feet and an assortment of mechanical gadgets. For example, if the film requires the sound of running along a sidewalk, the artist runs in place on a concrete slab in time with the film actor’s steps, with a microphone suspended nearby. Gravel, wood, or sand may each have a separate pit. Water effects are created in a basin or large trough.

Since space is expensive, Foley pits sometimes are built into a traditional studio. This is less desirable than a dedicated space since recording studios are more reverberant than Foley stages and water is seldom available. Foley is messy and dirty and requires space around the pits for microphones and props. One approach is to build prop storage areas on the walls with absorptive panels mounted as

Fig. 7 Abbey Road Studio 1, London, England (Photo credit Shawn Murphy)

doors. The airspace behind the panels improves bass absorption and the props are close by. Figure 7 gives an example of a Foley stage design based on this concept. There are large libraries of sound recordings available along with those maintained by the studios. These are available for scratch tracks and for the less critical applications. Foley stages must be quieter than recording rooms since effects are mixed hotter than music. Thus the background noise is more apparent to the mixer.

ADR

ADR or automatic (sometimes automated) dialog replacement is a technique using voice over, or the recording of dialog after the film has been shot. Whenever possible, film makers like to use the original sound recorded during filming but background noise or technical problems can make this impossible. In ADR the actors rerecord their parts in sync with the film. ADR stages are small, sometimes no bigger than a bathroom, and relatively dead. Low-frequency reverberation is a concern. Most have at least 2” (52 mm) thick panels on the walls. Since dialog replacement includes singing, ADR artists prefer rooms that are not completely dead (Farmer, 2001) and have a bit of volume, on the order of 8’ × 12’ × 9’ (2.4 m × 3.7 m × 2.7

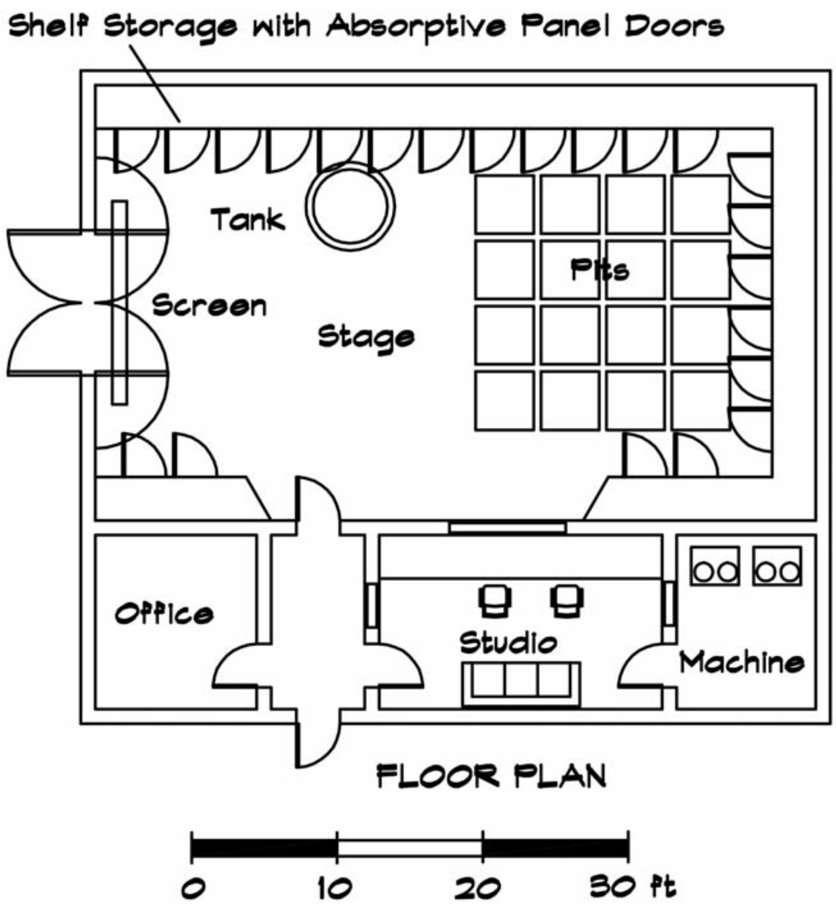


Fig. 8 Foley Stage

m). Larger stages can be used for recording voice over by adding absorptive panels around the actors. Flutter echo is particularly important to control so at least two nonparallel wall surfaces require treatment. ADR rooms should have a flat reverberation time versus frequency characteristic. Diffusion can be helpful and throw rugs are used to vary the room characteristics.

With animated films the dialog is often recorded first and the animation created later to fit the sound. With film or video, the actors must watch the picture and synchronize their voices to it. Video monitors are built into voice-over booths for this purpose. A communication system including a window between the booth and the studio and an intercom is a necessary part of the design.

Portions of this article have previously been published in *Architectural Acoustics*, 2006, Academic Press.**AT**

Marshall Long lives in Sherman Oaks, California with his wife of thirty-four years (She was very young when they were married). He enjoys reading, writing, designing home additions, and concert halls. His website is www.mlacoustics.com.

CONTROL ROOM DESIGN: THE MONITORING ENVIRONMENT

George L. Augspurger
Perception Inc.
1537 Cerro Gordo Street
Los Angeles, California 90026

Introduction

The term "control room" originated in radio broadcasting and was later adopted by the music recording industry. In the early days of disk recording, the control room was little more than a booth, and it is still called a "cabina" in Latin-American countries. Today, the room is much larger, perhaps 18 by 25 feet. It is used both to control the actual recording process ("tracking") and to assemble the multiple tracks into a final 2-channel or surround sound product ("mixing"). In a large music recording facility, mixing may be performed in a separate, dedicated room. If the product is to be released as a music album it will be sent to a mastering facility for a final check. The mastering engineer usually adds minor electronic processing, but may recommend re-mixing certain portions before the recording is released. A third type of monitoring environment is a production studio (or "composer's room") in which music is both created and edited.

"The bulk of music produced in the U.S. today probably comes from residential studios."

Historical Perspective

Before the introduction of audio tape recording in the late 1940s, commercial music was recorded direct-to-disk. In the U.S., the control booth was usually a small, utilitarian space adjacent to the recording studio. A window, perhaps 3 feet high by 5 feet wide, provided visual contact between the two rooms. The cutting lathe and a small audio control panel were located close to the window, giving the operator a good view of the studio. An ordinary radio-quality loudspeaker in a box was mounted on the wall above or beside the window. The only acoustic treatment consisted of perforated fiber tiles on the ceiling and upper wall areas.

The Capitol Records Tower in Hollywood, containing two state-of-the-art recording studios, was constructed in 1956. Each control room was a triangular structure built across one corner of the studio. At that time most recordings at Capitol were made on 3-channel tape, with one or two microphones routed to each channel. Three-channel master tapes made it easy to release albums as mono or (after 1958) stereo LP disks. The audio control board was placed against a wide studio window, and three loudspeakers (one for each channel) were set in an alcove above the window. Recording engineers had access to a patch bay and a fair amount of audio processing gear—mostly equalizers and limiters. Tape machines were located in the rear corner. Although a great deal of attention was given to the acoustics of the studios, the control rooms were not intended to be critical listening spaces.

The idea that the control room should function as a reference listening environment dates from about 1966. By then, 8-track recording was common, but the process of combining all 8 tracks into a 2-channel stereo master was still in the experimental stage. RCA Records had opened large, new studios in several U.S. cities. These were designed by John Volkman to meet the special requirements of multitrack recording, and he established acoustical goals for the control rooms as well as the studio spaces.

Also in 1966, the British Broadcasting Corporation standardized basic acoustical requirements for broadcast control rooms, based on the belief that: "listening rooms and control rooms should not be very dissimilar from the average conditions encountered in private houses." BBC control rooms were therefore designed to have reverberation times of 0.4 second up to 250 Hz, gradually decreasing to 0.3 second at 8 kHz.[1]

By 1969 the number of tracks for music recording had increased to 16 and it became apparent that monitor loudspeakers should serve as a reference for the final 2-channel product rather than providing dedicated sound sources for individual tracks. Instead of the BBC's simulated living room, the mixing environment became a kind of acoustical magnifying glass. As an example, the Los Angeles Record Plant opened in 1969 as one of the world's first 16-track recording facilities. Control rooms, designed by Tom Hidley, looked more like space ship cockpits than conventional listening rooms.

Each control room had a pair of high-power, custom-designed monitor loudspeakers flushed into a tilted wall above the studio window. The speakers abutted a hard, sloping ceiling that descended to a height of about 7 feet above the work area. In the rear third of the room the ceiling leveled off and was covered with absorptive treatment. The large 16-track console was located near the center of the room, providing a fairly wide area in which stereo playback could be judged. Hidley's goal was to provide an accurate stereo image in the console working area and not worry about the remainder of the room. As a bonus, the "compression ceiling" delivered powerful, gut-punching bass that was a new experience for recording engineers.

The next ten years saw a further increase in the number of recording channels from 16 to 32. It also saw a proliferation of new control room design philosophies, each characterized by its own technical jargon, such as compression ceiling, live-end-dead-end, Haas fushion process, bass trap, quadratic residue diffusor, and reflection-free zone. F. Alton

Everest gave a well-researched comparison of several design approaches in the 1987 *Handbook for Sound Engineers.* [2] One of them generated enough interest and controversy to deserve a brief discussion here.

In 1980, Don Davis and Chips Davis (the two authors are not related) published a paper titled, "The LEDE Concept for the Control of Acoustic and Psychoacoustic Parameters in Recording Control Rooms."[3] The LEDE (live-end-dead-end) concept suppresses first-order reflections in the range from 0 to at least 5 milliseconds. Later reflections are made as dense and diffuse as possible. To achieve this goal in a control room of practical dimensions, the front half of the room must be almost completely absorptive, and the rear half must consist of reflective scattering surfaces. The design was said to provide two important advantages. First, it tried to avoid comb filtering generated by early reflections. Second, the dense, later arriving reflections were intended to create the subjective effect of a much larger room.

Regardless of the pros and cons of LEDE theory, it is difficult to implement in practice because the "dead" surfaces should be fully effective down into the 200 Hz region. A number of very small LEDE rooms were built, and they sounded just as bad as other too-small control rooms. In any case, although LEDE control rooms were actively promoted for several years, they disappeared almost overnight, as did other radical designs. In the following 30 years, much additional research was done regarding the role of early reflections and other aspects of listening room acoustics. The subject is fully covered in Toole's *Sound Reproduction*, first published in 2008.[4]

In the 1980s and 1990s the music recording industry grew rapidly, stimulated by the introduction of the digital CD as a universal playback medium. At the same time there was a steady shift away from large, multi-studio facilities owned by the major record labels. More and more albums were recorded in smaller independent studios. It became common practice to cut individual tracks in various venues, and then assemble the final product in a dedicated mix room. During that period the independent mastering engineer became an important figure in the production process, serving as a final retouch artist before an album was released.

After an unfortunate detour for quadraphonic monitoring, new control rooms in the U.S. gradually began to fit a common template, one that emphasized 2-channel playback but allowed for surround sound mixing as well. At the turn of the century the recording industry mistakenly assumed that consumers would rush to buy surround sound albums of their favorite artists. Things didn't work out that way.

In the 2002 edition of the *Handbook of Recording Engineering*[5] Eargle gives a description of a generic, high-quality control room. Its design will be taken up a little later, but one feature should be noted here. Eargle explains, "A center loudspeaker is often soffit mounted in the front along with the traditional large stereo loudspeakers, and this is to facilitate film work." In other words, only three years after Sony's introduction of the Super Audio Compact Disc, surround sound was not considered to be a successful format for music recordings.

After 1990 or so, although professional recording engineers had reached a consensus regarding the characteristics of a good mixing room, fewer and fewer such rooms were being built. The Pro Tools digital work station had become the accepted standard for tracking, processing, and mixing recorded music. Almost every music composer and producer acquired a Pro Tools setup and proceeded to use it in the nearest convenient location. A spare bedroom became a professional music production room. If we include music composed for television and movies, the bulk of music produced in the U.S. today probably comes from residential studios. A major challenge for studio designers is how to make a small room acoustically acceptable for stereo monitoring and mixing.

Control/Mix Rooms

As noted above, there is substantial agreement as to what constitutes a good mix room. It is a fairly large room because it must accomodate production personnel (or keyboard players) in addition to the recording engineer. Two or three large loudspeakers are usually flushed into the front wall. The edge of the mixing console is about 7 feet from the wall, such that the distance from the engineer's ears to the speakers is around 8 feet. A low cabinet behind the engineer's chair holds a variety of electronic processing gear and also serves as a producer's desk, with space for chairs at the rear. All these functional requirements add up to a room length of about 24 feet.

The room will be used at very high sound levels, and leakage into adjoining spaces is difficult to control, especially at low frequencies. Background noise should be held to NC-25 or less, which may require placing computers and other noisy equipment in an adjacent closet or machine room.

The generic mix room shown by Eargle[5] is 17 feet wide at the front, 22 feet wide at the rear, and 24 feet front-to-back. The ceiling height rises from 9 feet at the front to 11 or 12 feet at the rear. The room is acoustically neutral, with a scattered mix of absorptive and reflective surfaces. Eargle does not specify a preferred reverberation time, but expects the engineer to hear an equal mix of direct and reflected sound from the main loudspeakers. Working backward from that requirement, the corresponding reverberation time is about 0.3 second, and roughly half of the interior surface area must be absorptive. Dolby and THX standards for mixing cinema or TV sound in a room of this volume require a reverberation time of 0.25 to 0.3 second, so it seems that a room optimized for 2-channel stereo mixing should also be acceptable for surround sound mixing. An informal survey of West Coast recording engineers supports that conclusion.

In fact, bilateral symmetry and the control of early lateral reflections are more important for 2-channel stereo than surround sound. Good stereo imaging requires a pair of well-behaved, closely matched loudspeakers, but if early reflections are suppressed then the listener must be exactly centered between the two speakers. Moreover, because each ear

Fig. 1. Interscope Records control room, showing flush-mounted loudspeakers, sloping ceiling, and side soffits.

hears both speakers (interaural crosstalk), a phantom center image is not the same as that heard on headphones. The direct path length from either loudspeaker to one ear is different from that to the other, producing a comb filter with its first dip around 2 kHz. In contrast, a good ensemble of symmetrical lateral reflections spreads out the sweet spot, adds depth to the stereo image, and helps fill in the 2 kHz dip.

Loudspeaker/listener geometry is controlled to some extent by the need for visual contact with the recording studio. If the main loudspeakers are located above a wide window then they are a little too high for optimum mixdown. In some control rooms, especially those designed for surround sound mixing, the studio window may be located to one side, allowing the front wall to be used for loudspeakers and a viewing screen. Another common alternative omits the center speaker and places two stereo speakers on either side of a fairly narrow studio window.

For the past ten years or so, mixing engineers have relied more on small, nearfield loudspeakers than the main monitors. The big speakers are still important, but they are used for periodic checks and for playback to the producer's area. Therefore, the room must provide good stereo listening under three conditions: (1) main speakers to console, (2) nearfield speakers to console, and (3) main speakers to producer's desk. Good correspondence between the two sets of speakers is important, and the main monitors are sometimes equalized to match a particular pair of console-top speakers.

Achieving acceptable low frequency response is much easier in a comfortably large mix room than a small production room, but audible peaks and dips below 100 Hz or so can be expected, and this is true for the nearfield speakers as well as the large monitors.[6] Because of the requirement for good sound isolation, room boundaries do not dissipate very much low frequency energy. As a result, a substantial amount of interior volume must be used for broadband low frequency absorption.

The most common broadband absorber is a cavity loosely filled with fibrous absorptive material and faced with porous fabric. It is called a "trap" or "bass trap" by studio designers. The cavity must be more than two feet deep to be effective down to the 50 Hz region. Since the main goal is to absorb low frequencies, a substantial reduction in depth can be realized by facing the cavity with wood slats or pegboard, making it a low-Q Helmholtz resonator. A pegboard-faced wainscot, perhaps two inches deep, was a familiar feature of many older recording studios and control rooms. The writer favors somewhat deeper "bunker traps" that can be conveniently located under windows or behind seating. The same basic construction can be built from floor to ceiling to create an effective corner trap.

Deep soffits on the side and rear walls can serve as bass traps. These may be augmented by vertical traps in the rear corners. In older rooms it was common to create a two-foot deep broadband trap across the entire rear wall, effectively placing the seating area in an acoustical black hole. Some designers later replaced the rear trap with very deep diffusers, hoping to scramble low frequencies rather than absorbing them, but the subjective results were equally unsatisfactory. As with the side walls, a reasonable mixture of reflective and absorptive surfaces seems to work best.

The mixing console itself is an important but often overlooked element in optimizing low frequency reproduction. The console is the biggest piece of furniture in the room, and its exact location can have a surprising effect on audible bass response. Even though the console position is specified as part of the original room design, a six-inch shift forward or back will sometimes result in worthwhile subjective improvement.

Figure 1 is a control/mix room designed by Vincent Van Haaf for Interscope Records. The photo clearly shows the

Fig. 2. Marcussen Mastering, Hollywood, California.

spatial relationships between the loudspeakers, the sloping ceiling, the console, and the outboard equipment cabinet. The studio window affords full visibility between the loudspeakers, but the glass dips down and extends under the loudspeakers as well. The rear ceiling and the undersides of the soffits are fully trapped.

Mastering Rooms

Music was played on disc recordings from the very first phonograph records until the late 1980s. In recording studios, "mastering" was the process of cutting a master disc from a master tape. The procedure could be tricky, and involved additional audio processing to keep from overcutting the spiral groove. Some engineers became known for their ability to turn out high quality masters, and their prestige matched that of top-ranking recording engineers.

One might have expected the mastering room to disappear during the changeover to digital audio playback, but the opposite occurred. The mastering engineer became a digital guru who made sure that a digital master tape took full advantage of the medium and met all formatting standards before it was sent to a CD production facility. The mastering room became larger, quieter, and was fitted with expensive playback loudspeakers.

By 2005 most of the large U.S. recording studios had closed. Mastering engineers began to set up their own practices, following the lead of successful independent mastering facilities such as The Mastering Lab in Hollywood and Sterling Sound in New York. In today's world of digital audio files, the location of a mastering room is not all that important, and many mastering engineers prefer to work at home. Residential mastering rooms usually require some compromises, but building a mastering facility in a rented commercial space may be equally difficult.

Today, a mastering room is used as a critical listening space in which the smallest details must be audible. Mastering is done at relatively low sound levels, and sometimes at very low levels, so background noise should preferably be no higher than NC-20. The geometry of the room should be favorable for 2-channel stereo listening and also for 5.1 monitoring since the final product may be released in a variety of formats. The room requires very little furniture—a control desk, a client couch, and a few storage cabinets. Computers and other noisy electronic equipment can be located in an adjoining closet.

Acoustical goals are usually quite similar to those for a mix room, and the design of a mastering room may also be similar to a good mix room, but not always. Unlike a commercial recording studio, a mastering room is required to meet the desires of only one person, and the design may deviate substantially from the norm. A few mastering engineers like to work in an acoustically dead environment. A few prefer fairly lively acoustics, something closer to a good home listening room. In most cases, high quality freestanding loudspeakers will be used, but some engineers prefer flush-mounted monitors.

In residential mastering rooms size is usually the biggest limitation. The smallest mastering room encountered by the writer was about 11 by 13 feet, and the ceiling height was a little less than 8 feet. Fortunately, the client was aware of the room's shortcomings and was satisfied to make it merely workable.

Most mastering engineers would like to work in a large room, perhaps 20 by 28 feet, but a somewhat smaller space is considered acceptable. A good example of current design practice is Marcussen Mastering in Hollywood. Figure 2 shows Stephen Marcussen's original mastering room, which was closely duplicated at a new location in 2009. The new room is a fully isolated structure inside a concrete block commercial building. Interior dimensions of the rectangular shell are about 18 by 26 by 10 feet. Stephen originally requested a 12-foot ceiling, but it would have been too costly to modify the existing structure. (The theoretical distribution of room modes is actually a little better with the lower ceiling.)

The room's distinctive appearance was designed by architect Frank Glynn. The horizontal wood slats on the side walls are quite narrow and the gaps are relatively large, so the screens become acoustically transparent below 2 kHz or so. Varied "checkerboard" acoustic treatment is hidden behind the screens and on the end walls. Wall treatment is augmented by 5-inch deep bunker traps below the wood screens. The floor is carpeted except for a central hardwood work area. Five large B&W loudspeaker systems are arranged in a standard 5.1 configuration. The final locations of the loudspeakers and the work station were established subjectively through extensive listening tests.

Music Production Rooms

Commercial recording facilities often include include small production spaces rented to independent producers or music composers. It is even more common for composers of film and television music to set up work spaces in their homes. These tend to be fairly small rooms - perhaps 12 by 15 feet—designed primarily for efficient work flow. In almost all cases, the room will be set up for 2-channel stereo monitoring using small, nearfield speakers.

If possible, such a production room should be laid out symmetrically as if it were a smallish mix room, with a separate computer closet and possibly a small vocal booth. In many cases however, there is barely enough space for the equipment, which includes a digital audio workstation, a computer, loudspeakers, outboard processing gear, and keyboards. Existing doors and windows are additional constraints. Acoustic treatment may be limited to plant-on absorptive panels and perhaps a bookcase or a few throw pillows.

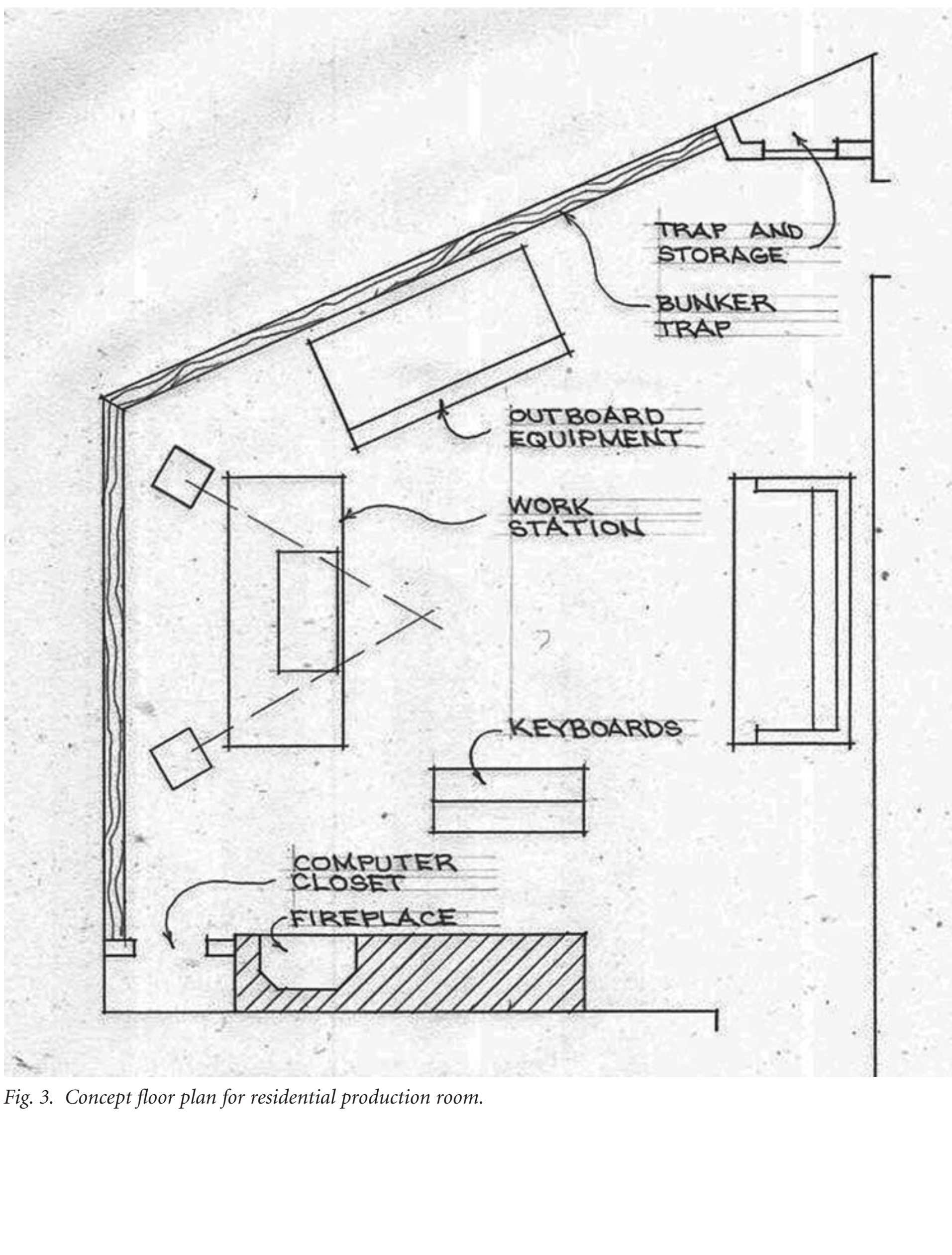


Fig. 3. Concept floor plan for residential production room.

In such a situation, it is a big mistake to jam the work station and loudspeakers against one wall. The only practical method of low frequency "tuning" is to experiment with the placement of the work station and loudspeakers, even if it requires two or three different arrangements of equipment and furniture. If the best sounding arrangement turns out to be awkward, the client can compare the trade-offs and make an informed decision.

A conceptual floor plan for a home production room is shown in Figure 3. (Two additional layouts were presented to the client.) The proposed design includes a new corner trap that doubles as a storage area. A 26-inch high bunker trap extends along two walls. The remaining acoustic treatment consists of plant-on wall panels and a large, suspended panel made of fabric over pegboard—an "acoustic cloud".

Conclusion

Even though a few Super Audio Compact Disc albums are released every month, surround sound has failed as a medium for home music listening. Rooms designed for music composing, mixing, and mastering must be optimized for 2-channel stereo playback.

For the next few years at least, the trend is expected to continue, and 2-channel stereo will remain the standard format for music production. Most popular music producers and recording engineers hate working with a center channel in spite of its obvious advantages. They have learned how to transform deficiencies into benefits, and the situation is not likely to change soon. Films, TV, and computer games all benefit from surround sound, but so far as music is concerned, the only viable consumer market seems to be luxury automobiles.AT

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George Augspurger is an audio and acoustical engineer who has been active in studio design for many years through his firm, Perception Inc. He spent over a decade with JBL starting in 1958 as JBL's Technical Service Manager and later established and managed the Professional Products Division. In 1968 he became Technical Director for JBL, a position he held for two years before moving on to independent consulting. He now teaches at the Viterbi School of Engineering at USC. He is a Fellow of the Audio Engineering Society and the Acoustical Society of America and reviews audio and acoustic patents for both journals.

MOTION PICTURE SCORING STAGES AN OVERVIEW

Shawn Murphy
Murphy Balance Engineering
3800 La Crescenta Ave. #205
La Crescenta, California 91214

Introduction

The Motion Picture Scoring Stage holds a unique place in the universe of recording and performance venues. Originally a necessity, the Scoring Stage has become the center of recording versatility and technical innovation over the past three decades.

This piece intends to describe a short history of the stages and the included technology. The current state of the scoring/recording art and utilized technology will be examined with some examples from current projects. The approach is not scientific; it is a personal description and history from a music mixer/engineer who has resided in this business for over thirty-five years.

"Scoring Stage monitor environment must be both accurate and dramatically viable, several choices of monitoring elements are provided."

The composer

The art of film scoring originates with the form itself. Live music accompaniment to moving images began with the introduction of film storytelling. With the marriage of sound and picture in synchronization, the film composer now had the ability to permanently enhance the drama, describe the action and enrich the emotion of the story.

The composer reads the script, screens the edited picture and commits to a discussion of style and form with the director. Often a process of thematic demonstration follows and further refinement of musical approach is accomplished. The composer then "spots" the picture for music with the director, film editor and sometimes the producer. Specific cue timings and functions are discussed. The music editor then creates "spotting/timing notes" and a cue-by-cue "breakdown" of the music in the film. This breakdown includes start and stop times for cues, all dialog timings, and important action timing within the cues. The timings were originally noted in feet and film frames and are now indicated by the Society of Motion Picture and Television Engineers (SMPTE) time code numbers in minutes-seconds and frames. Currently, timings within frames are common due to the resolution of film composition software.

The composer then begins writing the music for each indicated cue. He includes timings of important action and dialog, which align with the editors' timing notes. The tempo, bar structure and musical notation is included in a "sketch" which is developed over a period of composition and refinement. The sketch may be made by hand or via computerized music notation software and usually includes at least eight musical staves and often numerous "midi" lines recorded in the notation software. The composer constantly checks and re-checks his composition for picture sync and includes changes and enhancements required by the director.

After the cues are approved, his sketch is turned over to an orchestrator, who fills out the parts needed for full performance on the recording stage. Often, the orchestrator will create "cued" parts to enhance the composition if needed in the recording process. The orchestrator will assign parts to instruments bearing in mind the range of each instrument and it's ability to play the desired part. The orchestrator is by far the key person after the composer in the recording process. The qualities of the orchestration determine the speed and ease of the performance and recording process on the scoring stage.

The orchestrator then submits his full score to the music copyist/library to make transposed parts for each individual musician to be used on the scoring session. Given a standard scoring orchestra of between 65-90 players, this can result in the manufacture of more than 50 parts per cue, for the specified number of bars in each cue. The music library also manufactures full scores for the composer, the booth score reader (often the orchestrator), the music editor, the engineer and the assistant engineer. The library also produces a breakdown of which musicians perform on each cue, the cue timing/number of bars, as well as a breakdown specification of percussion, keyboards and solo instruments. It also indicates doubles in the winds and brass departments and any score-indicated overdubs.

The next step in the process is scheduling the music scoring sessions into a scoring stage by the production company. Professional musicians are then engaged for the sessions by a contractor, who includes the input of the composer and the orchestrator in regards to specific players and soloists. The scoring mixer is engaged, and he schedules the equipment needed, the stage setup and the required crew. A comprehensive recording schedule is developed based on the complexity of the score and the number of minutes to be recorded for the score. A typical film scoring session would schedule six to seven minutes of completed music per three-hour session. Most film scores include between forty and one hundred minutes of finished music. However, the inclusion of alternates and on-session fixes can increase the minute count by twenty percent.

On the scoring stage, specific picture sync is maintained either by mechanical means (sprockets), or by electronic methods involving SMPTE time code and video picture synchronization. Picture cues, based on the composer's notes and scores, are determined by the music editor and are marked on the film picture or programmed into an editorial computer that runs in sync with the picture via SMPTE time code. Picture cues include streamers (vertical lines crossing the picture from left to right in timed intervals of 3 to 5 feet to prepare for a picture "hit"), punches (to specifically indicate a "hit") or flutters (to indicate a passing bar line). Additionally, the music editor will often prepare a "click track" which will coincide with the composer's indicated tempo for all or part of each cue. The click tracks were originally a punched optical track and evolved to edited magnetic click tracks, click loops, analog metronomes, digital metronomes and finally computerized click tracks. Click tracks can be steady state or variable in the above forms. The current computerized click can adjust in minute increments to place music exactly on hits, even when a steady tempo would slightly miss.

Click tracks are played back to musicians via headphones (single or double sided) during the performance of each cue. Often, timing of a cue will involve clicked measures and free timed or conducted measures. The picture marks that are described above are used to support the click bars and to manually time the conducted bars.

The composer supervises the recording process in terms of performance and sound quality. Often, multiple elements will be recorded to form a complete cue. Separation of elements is necessary for the final film mix process, described later in this presentation. The composer oversees the final music mix and production of the music track and sometimes supervises the music mix in the final dub as well.

The stages

Shortly following the introduction of sound for picture, the requirement to record an accompanying music score to picture came to the foreground. Quickly realizing this requirement, Film Studios converted the most appropriate and underused facility on their production lots to the purpose of music recording. Invariably this was an unused shooting stage. Examples are Stage 1 at MGM/Sony, Stage 1 at Disney and Stage 10 at Universal. These shooting stages were not equipped for sound recording or musical performance. They were a large enough space to house the studio orchestra and usually had an adjacent insert stage space, which could be used as a control room.

Slowly, over the next decade, these spaces were adapted for better noise isolation and acoustic character. Sometimes, they were replaced by purpose-built scoring facilities. While orchestral scores were initially the norm, the size of ensemble varied from a few players to a "large" studio orchestral of 40-60 players. Often, there were soloists and rhythm players involved as well which required special isolation, baffling or separate recording treatments. The current large orchestra of 85-105 players did not come in to play until the late 1970's, and clearly overworks the available volume of the earlier dedicated spaces.

Recording formats initially were single track optical and soon evolved to multiple single channel recorders operated in sync to facilitate separation of instruments, vocals and musical effects. Progressively, multi-track formats flourished following the introduction of magnetic recording. The recording formats have followed or lead the artistic ideals on magnetic film carrying up to six separate tracks, then to 2" 24 track analog, to 48-track digital and now to an unlimited track count.

Control room monitoring was based on theatrical playback systems and generally included only a single channel. Currently, multichannel monitoring systems, both permanent and portable, are utilized in scoring venues worldwide. Musician headphone monitoring for synchronization (click) performances, was provided by single-ear carbon headsets, and is still in use today.

Microphone technology advanced swiftly during the 1930-1950 period. Designs introduced during that period are still in use today. Similarly, tube and later solid-state technology quickly responded to artistic requirement and became state of the art.

Scoring Stage electronic and acoustic control sophistication has increased during the recent past, and the development of digital and network technologies have driven the technical implementation of these spaces dramatically.

Venues other than Scoring Stages have been adapted for use in Film Score Recording. Multi-track record recording facilities, concert halls, radio and television production facilities have all fallen into this category.

Acoustic scoring requirements

The initial requirement of "a place to record the score" has given way to some very organized acoustic environments. The scoring stages at MGM, Fox, and Warner's have all received treatment to normalize the acoustics for film score recording. The Disney Stage A and the Republic/CBS/Todd-AO Stage were both purpose built Scoring Stages constructed in the 1940s. Having said that, none of the three remaining stages present an ideal orchestral environment. The versatility of the recording stage takes precedence to the single-purpose orchestral treatment.

Typical usage of Scoring Stages could encompass a full symphony orchestra during the day to a small rhythm section in the evenings. Even within the same session, one can encounter diverse ensemble configurations. To accomplish this, stages were designed with less active acoustics, that ability for musicians to hear each other properly without adversely injecting an acoustical signature to the recording. Treatments to existing stages varied, some resulting in "happy accidents." The MGM Scoring Stage is probably the best example of this; with dimensions of 66'x93' and a volume of approximately 160,000 cu ft., it maintains a mid-band reverb characteristic of 1.0-1.3 seconds and can support large ensembles without apparent acoustic overload.

The Warner's and Fox Scoring Stages exhibit a lower reverb time but still provide adequate musician-to-musician feedback. The CBS/Todd-AO Stage, now closed, was re-designed acoustically in the early 1990's to provide a more symphonic environment. While somewhat variable, the basic reverberation signature of the 72'x108' 250,000 cubic foot scoring stage was approximately 1.8 seconds mid-band.

Fig. 1 Sony Pictures Scoring Stage.4

The volumes indicated above do not approach concert hall sizes, yet the reverberation times are sometimes in the concert hall range. This factor alone can result in a very active acoustic environment, which can overwhelm the direct microphone pickup. Even purpose-built classical studios, such as Abbey Road Studio 1, do not possess adequate volume to support their 2.2+ second reverb time with a large symphony orchestra. Most classical engineers will prefer concert halls, assembly halls or churches to recording studios for their largest ensemble projects.

Because Scoring Stages are required to support diverse ensembles, there has been an increased use of isolation rooms in these facilities. All of the four existing US Stages (MCM/Fox/Warner's/Skywalker) and both of the UK Stages (Air Lyndhurst/Abbey Road) have multiple isolation rooms capable of housing drums, percussion, grand pianos and small vocal groups. This permits simultaneous recording of performances by ensembles, which might not balance acoustically in the same space. The requirements of music mix delivery also often specify separation of solo elements for use in the final mix.

Sometimes, separate sections of the orchestra or ensemble are recorded in isolation. Often, the entire ensemble is rehearsed and then each section is recorded separately. This could include but not be limited to: Strings, Winds, Brass, Percussion, Harp and Keyboard. Individual section microphones are recorded while the room or overall pickup is duplicated for each pass. This provides a recording with the same overall room/hall sound/ambience and allows the musicians to rehearse together. Again, delivery or editorial requirements often drive this technique.

Dramatic requirements often shape the technique used to record on the Scoring Stage. While the overall approach may be symphonic, there could be instances where individual instruments need to be featured, sometimes in an unmusical fashion. The use of portable baffles and microphone technique often substitute for isolation rooms when the featured instrument is also playing along with the full ensemble. Additionally, the perspective of the recording often needs to change with the dramatics. A chase scene would require a closer, more aggressive orchestral pickup than a love scene, even though the cues may be recorded back-to-back.

Recent experimentation with active electronic modification has been prevalent. The ability to extend reverberation times through electronic enhancement has proven workable, as well as slight modification in the overall characteristic of the room (via equalization). A more inert environment can be adjusted to a larger hall acoustic for a symphonic approach, while shorter reverb times with increased early reflections can aid in musician monitoring across the room.

The key word is versatility—any ensemble, any recording technique and a quick change between sizes and techniques are mandatory. (See Figs. 1 and 2)

Recording and mixdown techniques and technology

Even with the introduction of sound-for-film, there has been a requirement for multi-track or quasi-multi-track techniques. The earliest disc and optical recording often used multiple recorders either as backup or to receive separate and complementary material. Although the earliest mechanical synchronization methods did not allow for a cohesive stereophonic recording, the implementation of multi-track recording in notable productions such as "Fantasia," allowed for more depth and dramatic effect when synchronized with picture.

Progressively, the introduction of magnetic recording in multi-track (3/4/6 track) and large format magnetic analog recording (8/16/24 track) have allowed for a diverse stereo and multi-channel sound field for music presentation. Beginning with multi optical recorders and continuing through multiple magnetic film recorders and multiple large format multi track recorders, more and more tracks have been utilized in the production of film scores. One hopes that this track utilization is all in the interest of best sound quality. However, in some cases, it merely represents the adage that one will always fill the maximum available number of tracks.

Fast forward to current day, we have witnessed a revolution in recording technology while still utilizing similar musical and dramatic techniques to obtain the dramatic and musical effect.

Digital recording, via reel-to-reel recorders and current workstations, provides a virtually unlimited track count and state-of-the-art audio quality in the scoring industry.

Our current record format for orchestral/acoustic music is often 192kHz/32bit in a multi-track format of multiple 96-track workstations. One current production utilized two 96-track 192kHz workstations plus a 128 channel 44.1kHz/24 bit workstation to build multiple stems for presentation to the final dub of 64-track at 96kHz/24bit.

As an example of recording formats used during a recent project (See Fig. 3):

Elements of the live score included:

- 95-piece orchestra
- Ethnic percussion
- 40-voice choir
- Ethnic winds
- Early instrument Consort (12 piece)
- Cello solo
- Piano Solo (1 and 2 Piano)

For the Orchestra, the entire 95-piece ensemble was assembled and recorded both as a single unit and as sectional stems. Often, the cue was rehearsed extensively and recorded as a single unit, and then it was broken into record-

Fig. 2 Fox Scoring Stage.

	AFTER EARTH	192/32 X 64 TRACKS		
	JAMES NEWTON HOWARD	AVID HD I/O@-20		
	SONY SCORING			
1	WW OHL	CMC3/MK8	8'	G1
2	WW OHC	CMC3/MK8	8'	G2
3	WW OHR	CMC3/MK8	8'	G3
4	SOLO WW	TLM170		G4
5	FLUTES	DPA4011		G5
6	OBOES	DPA4011		G6
7	CLARINETS	DPA4011		G7
8	BASSOONS	DPA4011		G8
9	WIDE LEFT	EHRLUND	12'	P1
10	TREE LEFT	M50BRIGHT	11'	P2
11	TREE CENTER	M50BRIGHT	11'	P3
12	TREE RIGHT	M50BRIGHT	11'	P4
13	WIDE RIGHT	EHRLUND	12'	P5
14	BASS 1	LUCAS CS4-CARDOID		P6
15	SURROUND LEFT	CMC3/MK3	20'	P7
16	SURROUND RIGHT	CMC3/MK3	20'	P8
17	HARP HIGH	LUCAS CS4-CARDOID		P9
18	HARP LOW	U67		P10
19	PIANO LEFT	C222/MK21H		P11
20	PIANO RIGHT	C222/MK21H		P12
21	STRINGS LEFT	KM56-FIG8	9'	G9
22	STRINGS CENTER	KM86-FIG8	9'	G10
23	STRINGS RIGHT	KM56-FIG8	9'	G11
24	BASSES	KORBY C12-WIDE CARD	8'	G12
25	PERC OHL	CMC6/MK2S	9'	G13
26	PERC OHC/TYMP	CMC6/MK21	8'	G14
27	PERC OHR	CMC6/MK2S	9'	G15
28	GC	CMC3/MK2		G16
29	PERC LEFT	2XMKH40		N17/18
30	PERC CENTER	2XMKH40		N19/20
31	PERC RIGHT	2XMKH40		N21/22
32	TUBA	KORBY U67		G32
33	HORNS FRONT LEFT	C24-XY-CARDOID		P13
34	HORNS FRONT RIGHT	C24-XY-CARDOID		P14
35	HORNS REAR LEFT	KM140		G17
36	HORNS REAR RIGHT	KM140		G18
37	TRUMPETS LEFT	SF12		G19R
38	TRUMPETS RIGHT	SF12		G20R
39	TROMBONES LEFT	ARABELLA-CARDOID		P15
40	TROMBONES RIGHT	ARABELLA-CARDOID		P16

Fig. 3 After Earth microphone type and placement

ed stems for editorial and dramatic purposes. Approximately 10% of the score was recorded as Orchestral stems and 90% was recorded as a Complete Orchestra. The Orchestral stems included:

- Strings (33Tracks) (Violins/Violas/Celli/Basses)/
- Winds (Flutes/Oboes/Clarinets/Bassoons)/
- Harp/Keyboard (Piano/Celeste)
- Additional String Overlay (20 tracks)
- Brass (18 Tracks) (Horns/Trumpets/Trombones/Tuba)
- Orchestral Percussion (14 Tracks)
- (Tympani/Gran Casa/Orchestral Toms/Tam Tam/ Suspended Cymbals)

The Ethnic Percussion was recorded during separate sessions and included The use of numerous large and small drums, shakers, metal effects, etc. (Multiple of 11 tracks)

The Choir was recorded during a separate session and was multi-tracked with two or more passes to increase size and for editorial isolation. (Multiple of 8 tracks)

The Consort was recorded during a separate session as a single unit as well as the Consort with Cello Solo. (Multiple of 24 tracks)

The Cello solo was recorded during the Consort session, separately from the Consort. (Multiple of 16 tracks)

The Solo Piano(s) were recorded during a separate session. (Multiple of 11 tracks)

The recorded tracks were spread across two digital workstations, each with a total of 96 tracks available at 192kHz/32bit. (Workstation Systems #1 & #2).

The live material was combined with various pre-recorded tracks from the composer's production studio. This material included orchestral samples (not used), guitars, sampled percussion, choir, ethnic flutes and various synthesized effects. Although this material was delivered to the mix down at 44.1kHz/24bit, it was sample rate converted to 96kHz/24bit as stem outputs via a Multichannel Audio to Digital Interface (MADI) converter/router. (Workstation System #3)

The output of the 192kHz/32bit mix stems was converted to 96kHz/24bit via Digital to Analog (D/A) and Analog to Digital (A/D) converters. During the analog stage of conversion, overall bus equalization was applied to the orchestral material. All of the 96kHz/24bit material, as stems, was returned to a digital mixing console to finalize the balance between stems for the final music mix.

The stem outputs included:

Left/Center/Right/Left Surround/Right Surround/Low Frequency Effects (L/C/R/LS/RS/LFE)

- Orchestra A (Composite Orchestra or String Stem) 5.1
- Orchestra B (Orchestral Solos or Brass) 5.1
- Low Percussion (Orchestral Percussion and/or Low Ethnic Drums) 5.1
- Mid Percussion (Ethnic Drums) 5.0
- High Percussion (Ethnic Percussion/Metals) 3.0
- Synthesizer Pads 5.1
- Synthesizer Pulses/Rhythm (5.1
- Solos #1 (Cello Solo, Piano Solo, Ethnic Winds) 5.0
- Solos #2 (Ethnic Winds)
- Choir (Live Choir and Vocal Samples) 5.0
- Consort (Live Early Instrument Consort & String Effects) 5.1
- Extra (Anything not covered above: Guitars, Effects, etc.) 5.1

The console output was routed to a 64 channel digital workstation at 96kHz/24bit/-20. (Workstation System #4) This material is then conformed to picture and finally mixed with dialog and sound effects to complete the soundtrack.

The picture, along with temporary music tracks, dialog tracks and sound effect pre-mix tracks, was played back on another digital workstation. (Workstation System #5)

Because various outboard effects are not able to operate at the 192kHz/32bit sample rate, an additional workstation was needed to run these plug-in effects via sample rate conversion or analog converters. This workstation ran at 96kHz/24bit resolution. (Workstation System #6)

Complexity notwithstanding, the number of tracks typically utilized for soundtrack production has soared with the introduction of digital workstations. The most current iteration of the workstation delivers an exceedingly high technical quality potential of recording. High bit and sample rates have reduced the difference between bus and playback to nearly negligible. However, more often than not, these technical advances are not fully utilized in lieu of obtaining the maximum number of tracks.

Picture synchronization

Synchronization of picture and music is what film scoring is all about. The dramatic impact of a film score coupled with the storytelling and pacing available with musical accompaniment defines the traditional art of the film soundtrack. Only in the past three decades have sound effects become a significant contributor to the film soundtrack. Reviewing the composition of soundtracks from the late 1920s through the mid 1980s, one may observe the preponderance of music contribution versus that of the sound effects. For the initial history of the film soundtrack, effects were used primarily to support the visuals and not often used as a separate dramatic element.

Methods of picture synchronization have progressed from the mechanical through various electronic techniques. Currently, all picture sync methods are electronic, and this can introduce all sorts of variables, which must be considered when producing the music tracks.

Early picture synchronization was purely mechanical. A belt or chain drive attaching the projector or picture source to a sound recording device. This assured repeatable synchronization within some variation and allowed for the music, and later dialog and effects, to be kept in sync with the picture.

Technology quickly advanced to support electro-mechanical synchronization via line driven selsyn systems. A central drive motor supplied power to a bus on to which various projectors and sound recording devices were connected. Driven in sync with the line frequency (60Hz), the system locked all elements into tight sync with one another. Typically, one perforation (10ms) was the window of sync accuracy. This system was used until the 1980s when electronic motor drives allowed synchronization with high speed forward and reverse. Still typically using only magnetic film as a recording medium, multi-track analog playback machines were slowly integrated starting in the late 1970s using SMPTE time code printed on magnetic film and locked to the sync system. Later, shaft encoders and finally master time code generators locked to the electronic drive system were used to lock multi-track machines to picture. Later, multi-track digital machines were introduced and synchronized (beginning in 1981) with difficulty, due to the speed vs. sample rate variable inherent in these devices. Lock-up time

Fig. 4 Sony Pictures Control Room.

varied, and tight synchronization, especially with digital machines, was not assured due to synchronizer programming and the typical "lock and drop" mode of operation. This necessitated running the film master sync system on a master clock, which matched the digital machines, so speed control could be assured.

Finally, in the mid-1990s, the workstation provided a fully synchronized platform between picture and sound/music. Variables in lock-up time prevented perfect sync for a few iterations, but finally, the sync can be held to within one or two milliseconds. Film picture was still the norm, but within the next ten years, it would all but disappear from the post-production scene. High definition video has replaced film in all areas of scoring and sound post-production. Typically we now run high definition video at 24fps. Sometimes, scoring copies are made to run 30fps (29.97) for ease of stage operation.

Control room/mix room specifications and monitoring

While all recording studios require a control room and monitoring environment, the Scoring Stage differs in several significant respects:

- The number of participants in the recording process can exceed twenty, and many must be in a position to judge the recording and performance within the monitoring direct field.
- Film playback formats must be mimicked in the scoring monitor environment. Therefore, both versatility and easy changeability must be designed into the monitor systems.
- There are several standard playback response curves that must be built into the speaker tuning.
- Several different formatted recording sessions may be scheduled within one day. Therefore, size of support areas and ease of changeover are mandatory. Most scoring stages have extensive wiring and routing capabilities to handle any required format.
- Extensive communication systems are required to enable easy and fast contact between the composer, conductor, musicians, editors and technical crew.
- Picture cueing systems differ between editors and composers. The control room technical capabilities must encompass all current and future cueing and picture playback systems.

The current standard monitoring format for Scoring Stages is the 5.1 surround system. Speaker systems appropriate to the room size and dimensions are included in each stage design. Typical monitor system manufacturers include ATC, PMC, Genelec, JBL, Dynaudio, and B&W. While only theatrical playback systems may differ in size, the response of the scoring monitor system is typically tailored to follow closely the theatrical playback curve. Near field and midfield monitoring is typical as an alternate to the main monitor systems provided by the stage.

Quite often, the Scoring Mixer will bring his own preferred system for use as a midfield or near field monitor.

Utilizing a preferred portable speaker system enables a quick translation between various stages and mixdown rooms using the same or similar speaker systems. Also quite often, the mixer will bring only the Left/Center/Right front speakers and utilize the built-in surrounds and sub-woofer system. All scoring consoles have multiple speaker outputs with trims available for "guest" speaker systems. This allows for the fine tuning of levels between the main house monitor system and the guest or scoring mixer portable system.

Film monitor systems utilize similar front Left/Center/Right systems and smaller diffuse surround speaker systems typically used in theaters. The bass response of the control room versus theaters makes tuning and specification of sub-woofers problematic, but the specification is still adhered to in terms of having an low frequency effects (LFE) channel and monitor system available.

Main monitor systems are leveled front (Left/ Center/Right) similarly to theaters at 85dBC at the mix position based on standard electronic level (-20 digital). Surround speakers are often leveled at 79dBC each and 82dBC combined. This is due to the proximity of the speakers in control rooms versus theaters; there is less air or volume to move in order to hear the surrounds in the closer environment. This slightly lower monitor calibration allows the mixer to increase his electronic level sufficiently to overcome the extra volume of the theater, which is leveled at 85dBC, the same as the front. Sub-woofers are leveled at 91dBC and again, depending on room response, may or may not be used in the scoring environment.

There has been some discussion and a bit of controversy regarding the restricted response and dynamic range of the "standard" theatrical monitoring and playback systems. The Scoring community utilizes typically higher quality monitoring due to the demands of purely music recording. Response of dialog (restricted) and sound effects (wide) are less considered on the Scoring Stage than in the re-recording environment, where a close match to theatrical playback qualities must be maintained. While the move toward higher resolution monitoring in theatrical playback systems is commendable, one must always bear in mind the restrictions of original dialog recording (storytelling element) versus the potentially higher resolution demands of music and effects. The playback systems must be supportive of the storytelling, not an end unto itself.

Because the Scoring Stage monitor environment must be both accurate and dramatically viable, several choices of monitoring elements are provided. Dialog and Sound Effects faders are available to make sample mixes with music playback, so the film Director and the Composer can hear the music in context.

The monitor speaker system is then removed from cali-

Fig. 5 Fox Scoring Stage Control Room.

brate and the music played at a level which can satisfactorily give the best impression of cue. This is sometimes necessary to "sell" the cue, even though the playback may be out of scale and unrealistically loud. (See Figs. 4 and 5)

Musician support

More than in any other recording environment, the Scoring Stages must provide extensive audio and visual monitoring support for the musicians and performers. Utilizing extensive headphone/cue systems and picture monitors, most players receive multiple levels of performance support. These include:

- Multiple and extensive headphone mixes, often more than twelve mono mixes (divided among Orchestra sections) and six stereo mixes (for conductors, editors, rhythm section and soloists). The mono headphones are single or dual sided and carry a mix of cue/click tracks and other musical elements performing live or prerecorded. The stereo mixes also carry click tracks plus a more extensive mix of associated musical elements.
- Video monitors carrying the film picture (with or without streamers/cue marks).
- Video monitors of the conductor for players in isolation or without a clear view of the conductor.
- Video monitors with the bar/beat count. This assists players in locating/counting during complex cues that have had little or no rehearsal.
- Video monitors with pictures of other soloists for visual cueing. Rhythm sections often use multiple personal video monitors to see each other for performance cueing.
- Electronic music stands. While this is a new technique, used more often in live performances, the future of a unique video driven electronic music stand for each player or group of players is certainly in our future. The ability to turn pages remotely (foot switch) and for each player to mark his/her music uniquely would be included. This would facilitate fast composition changes and re-assignment of parts to other musicians. The Music Library would typically integrate the electronic stands into their copy-composition software.
- Studio Loudspeakers. Often used for general playback, these loudspeakers are also often used to send tuning notes from pre-recorded tracks to the entire orchestra.
- Local Loudspeakers are often used for soloists in isolation who do not wear headphones. Sometimes, the speakers are wired out-of-phase and placed on either side of the performer/microphone to reduce leakage.

Technical facility

One may ask, following all of this description, what is the difference between a Scoring Stage and a Recording Studio?

Fig. 6 Fox Scoring Stage. Neve 88RS. Input Section.

Fig. 7 Fox Scoring Stage. Neve 88RS. Monitor Section.

Aside from the size of the recording space, and the ability to seat over one hundred players, the Scoring Stage technical capability encompasses the following:

- Numerous isolated recording spaces adjacent to the larger recording space.
- The ability to provide picture projection in any format to numerous locations.
- The ability to provide extensive cue/headphone monitor feeds (often more than 32) to the performers.
- The ability to monitor in the Control Room various and diverse formats. This
- includes multiple surround formats on several alternate speaker systems.
- Availability of baffling and platforms for various stage setups.
- Often, the availability of variable acoustical treatment in the main recording room. This would be attained by mechanical and/or electronic means. (Meyer Sound/Lexicon)
- The immediate flexibility of setup and configuration of recording systems.
- An experienced crew to accomplish all of the above, and more.

The Scoring Stage technical support systems have grown from the basic requirement for monophonic optical recording to complicated and extensive multi-track surround production in many formats.

During the later analog years, it was not uncommon for a stage to be wired for sixty to seventy microphones and up to five multi-track (24 track) recorders. Currently, with the proliferation of digital workstations, the stages are wired for one hundred twenty to two hundred microphones and at least four ninety-six channel record rigs. This recording capability is in addition to master clock systems, networking of audio and control, system backups and diagnostics.

Large multi-track analog recording consoles are still the norm. In fact, all current stages in the US and UK utilize the Neve 88RS Scoring console with between one hundred twenty and one hundred ninety-two inputs. This console is custom designed to afford great flexibility in monitoring and configuration. It includes forty-eight times two multi-track busses, thirty-six stem mix busses (patchable to many more), and up to twenty-four auxiliary sends. While this is the latest and probably last generation of analog desk, it does provide the extensive capabilities required by current scoring sessions. (See Figs. 6 and 7)

Many mix downs are accomplished in the digital domain and here we find both console and controllers a common ingredient. The use of higher sample rates in recording normally defines the use of controllers (mix in the box) rather

Fig. 8 Fox Scoring Stage Machine Room (left side).

editors on site to assemble and composite the pieces of the cue. This assembly is then sent back to the playback rig for future use as review and /or overdub material. This network requirement for audio, video and control now exist on all stages and via large rental systems in the scoring industry. Master clock systems, as well as clock, time code and control system distribution, is also a normal function in the technical infrastructure.

Many, if not all, Scoring Mixers utilize some personal equipment for recording and monitoring. The interface of this gear must be anticipated in the construction and configuration of the stage and control room. (See Figs. 8 and 9)

Considerations include:

- Microphone powering and mounting hardware.
- Microphone preamp wiring and remote control.
- Remote A/D Converter clocking and wiring (Digital and Analog).
- Portable monitor system configuration. (Leveling and integration with in-house multichannel monitoring systems.)
- Capability to integrate rental and outside engineer workstation to in-house wiring infrastructure.
- Stereo fold down and headphone integration.
- Portable clock wiring, triggering and remote integration.
- Portable communication systems, such as two-way radios, cue radios, etc.

than consoles with the heavy demand of DSP in the higher sample rates.

Digital mix consoles are configurable, with up to one hundred eighty inputs available at 96kHz/24bit. Additionally, up to sixty-four mix busses can be configured along with twenty-four auxiliary sends. This capability, along with hybrid technology enabling workstation control from the mix desk, allows the mix engineer great latitude in assigning effects and mixdown stems.

Plug-in technology, especially at high sample rates, has taken the place of much of the analog mix gear previously used for signal processing. However, many engineers still utilize vintage and/or highest quality devices for equalization (Manley/Avalon/ Massenburg) and reverberation effects (Lexicon/ Bricasti). This requires digital-to-analog conversion into the device and a reconversion to digital for final printing of mix stems. Often, this conversion process is used to accomplish sample rate conversion to the final mix format, normally 96kHz/24bit.

The need to provide nearly instantaneously edited material for review mandates complex and extensive network capabilities on the stages and mix rooms. When a cue or portion of cue is recorded, it is immediately saved and backed up on several (usually three) drive arrays. At least one of these arrays is available to the

Future of scoring stages

The opening of this article stated the necessity for a “place to record the music” as the initial requirements for a Scoring Stage. Over the nearly ninety years since, there have been numerous stages and facilities built for film scoring. At the height of the “Studio System” and TV production years, there were a minimum of ten Scoring Stages on studio lots (depending on how one counts) and at least ten other stages at independent facilities adequately equipped and staffed for a scoring session. This group of facilities existed in Los Angeles alone. Coupled with New York, London, etc., there were over thirty facilities world-

Fig. 9 Fox Scoring Stage Machine Room (right side).

wide under the umbrella of Scoring Stage.

Since the heyday of film and television production, the industry has become more diffuse and much of the previous studio production has shifted to smaller independent facilities and to in-house composers' studios.

The economics of the Scoring Stage has never been attractive to the financial arm of the film studios. When the simple real estate equation is used, the return on investment for a 100,000 square foot facility for music recording never makes sense. Consequently, the major studios have, over the past twenty-five years, slowly divested themselves of music recording facilities while assuming there would always be someone else who would continue to offer this service. At this date, there are three remaining large Scoring Stages in Los Angeles (Sony/MGM, Fox and Warner Brothers), two in London (Abbey Road and Air/Lyndhurst), none in New York and one each in Sydney, Australia and San Francisco. There are a few large venues around the world, which can be and are used for scoring and numerous smaller studios and concert halls, which are used as needed. Unfortunately, the likelihood of any new Scoring Stages being constructed in the future is slim. Nearly all of the mix downs are now accomplished at smaller studios or dub stages (with Theatrical monitoring). Much of the recording of smaller ensembles is done as overdubs at small studios or composer facilities. The performing ensemble is rarely recorded as one and live in the studio.

The drive for these techniques has arisen from the popular notion that film music must be "produced" rather than only composed and recorded. Pop record production and performers now populate the Film Scoring world. The generation of classically trained composers is fading rapidly. While there is a younger generation of highly skilled musician/composers utilizing the orchestral palette, they also rely on overdubs, sectional recording and samples as part of their "sound". The classical orchestra utilized for Film Music is becoming a thing of the past, hence decreasing the need or requirements for appropriate large recording facilities.

The industry requirement to record the music for films will always exist. However, it remains to be seen whether this will encompass the large and formalized environment of the Scoring Stage or some hybridization of Concert Hall, Church, Small Studio, Overdub Room and Mix Room/DubTheater.

Appendix/crew titles and duties

Scoring Mixer: Administrative and artistic head of crew. Specifies recording formats, stage layout, microphone choices, mix layout and monitoring system. Directs setup and coordinates with editorial and music library regarding recording order, stage positioning and technical lash up. Confers with composers and orchestrators regarding sonic approach and recording quality. Balances musical elements and directs playbacks and editorial approach to score. The Scoring Mixer is responsible for the ultimate audio quality of the score, all technical and stage scheduling, and for the on-time delivery of mixes. The mixer is sometimes a trained musician and often utilizes a full score to reference musical elements in the original composition.

Stage Engineer: Chief Technical Engineer for the session and Stage. Otherwise known as the Maintenance Engineer, he/she is responsible for the actual technical wiring and hookup of all gear. This would include microphones, preamps, analog and digital consoles, recorders/workstations, clocking, communications, networking of control and audio, cue systems, monitor systems, projection systems, etc. The engineer solves all system problems and assists in the setup of rental and composer gear on the recording session. The engineer manages all outside setup of remote feeds via satellite or Internet, involving remote performers, directors or studio executives. The Stage Engineer is on hand to assist during sessions and to prevent any potential down time.

Digital Recordist: This is the recordist responsible for the main record/workstation systems. He/she sets up the sessions in the workstation, pre-programs the input/output configuration and operates the workstation on the session. Often, this person performs preliminary edits and organizes the tracks for overdubs and later mixing. This recordist interfaces with the Stage Recordist regarding patching, clocking of digital gear, routing, etc. He/she also coordinates with the Music Editor and composer staff to load tempo maps and other session documents. She/he is responsible for data management, backups and distribution of recorded material. This recordist normally follows the project through preparation, recording and mix down at numerous facilities.

Stage Recordist: The Stage Recordist manages the control room portion of the Scoring Stage. This recordist is familiar with the in-house wiring, consoles, communications and playback systems. He/she works with the Digital Recordist and Music Editor to connect/patch all recording systems. This recordist often operates the backup workstation and manages the picture playback workstation. He/she also coordinates with the stage crew to route all inputs and outputs as needed to the headphone console and main recording console. He/she also coordinates audio and video feeds to the outside world, such as telephone communication feeds to remote locations and Integrated Services Digital Network (ISDN) or synchronous feeds to overdub or record performers at other locations (Satellite or Internet feeds.)

Digital Editor/Conforming: The session and mix conforming editor receives material from the session (via local network) and immediately edits this material for use in overdubs or temporary mix situations. This editor also moves material musically to provide best sync and performance when the orchestral sections are recorded separately. Additionally, this editor checks sync with the prerecorded material from the composer's studio to assure best musical performance when combined with the various live elements. This editor works from the score and is usually a trained musician. He/she also follows the project through scoring and mix down.

Stage Manager: The Stage Manager handles the studio side of the glass. He/she is responsible for executing the setup of chairs, stands, headphones and microphones in the studio. Most importantly, this position deals directly with the musicians and their endless litany of requests and problems. ("It's too hot/It's too cold/Too much space/Too little space" etc.) The combination of setting up the stage for best recording results along with keeping the players happy is a real art. The Stage Manager also checks microphone positions—and repositions microphones as needed (particularly percussion and keyboards) for each cue. The best Stage Managers can assist the Scoring Mixer in the choice of microphone type, seating position and stage layout with regards to their particular venue. If risers and baffles are needed, he/she places and/or supervises the placement of these elements.

Stage Assistant/Cue Mixer: The Stage Assistant also acts as the cue/headphone mixer on the session. On most Scoring Stages, the Cue Mixer is placed on stage, allowing easy communication between the players and mixer. The mixing consoles have between 56-72 inputs and upwards of 32-48 outputs, all for cue mixes and headphones. The sources are derived from the workstations and console in the control room and distributed to the cue mixer by the Stage Recordist. The Stage Assistant helps with microphone and headphone setup prior to the session. During the session he/she manages the multiple mixes going to the headphones and "rides" the click level to the headphones. The dynamics of the music determine the available headphone level allowed before leakage into the microphone would occur. It is the duty of the Cue Mixer to set and constantly adjust this level to avoid click leakage, which could ruin the take.

Click/Auricle Operator: Most scoring sessions utilize both visual and audible cues to synchronize the music track being recorded with the picture. Computerization of this cueing has been systemized through the use of onboard generation of clicks and streamers in the workstation and via outboard computer programs such as Auricle. Many sessions utilize both onboard and outboard systems. The Auricle operator has pre-programmed the cue marks and clicks prior to the session to correspond with the bars and beats of the recorded cue. The clicks can be subdivided to allow for easier performance of difficult time sequences. Picture cues are often determined both by the composer and conductor to assist in accurately placing musical emphasis with picture. The Auricle Operator can change any of these parameters on the spot to allow for instant music-to-picture adjustments. Music can be speeded up or slowed down, and bars and beats can be added or subtracted, all with corresponding click and picture cueing changes. This operator normally sits on the stage in proximity to the conductor. Changes in click construction are often ongoing in the recording process. Additionally, it is often desirable to free-time (conduct) a section of a cue (that is, to eliminate the click track and allow the conductor to perform the cue to picture only). In many cases, the cue will include sections of click and non-click bars. In these cases, the Auricle operator will build warning clicks to prepare the musicians for the incoming section of music.

Session/Composer Engineer: In the current era of hybrid scores, and the need to demo each cue for the director and studio prior to recording, nearly all composers now employ assistants and in-house engineers to facilitate this process. The composer's engineer typically prepares the pre-record playback machine, lays out the demo tracks to picture, and builds the tempo map for each cue. This engineer often attends the session to operate the pre-lay/playback workstation in order to edit "on the fly" any material subject to change, or to include any alternates included in the session. Since this engineer has been associated with the initial composition and construction of the cue, she/he is often the best reference for questions regarding sync and musical content of the pre-orchestrated cue.

Music Editor: Described in some detail earlier, the Music Editor takes the score from the earliest conceptual stages through composition, demos, recording, editing and final mix. The Music Editor is usually the final word involving sync and picture changes/adjustments. Many Music Editors are composers themselves, and are capable of sophisticated sync and musical adjustments if required to do so by picture changes. The Music Editor confers with the Scoring Mixers regarding stem layouts for dubbing, and arranges for final print takes and alternates to be included in the final mix session. The Music Editor usually is the designator of take numbers and session notation regarding best takes, director and producer requests and desires regarding music, and dubbing notes.

Assistant Music Editor: The Assistant Music Editor manages the data from the recording and mix sessions and sets up the final music mix material in picture sync sessions to be played back in dubbing. The assistant typically keeps track of updated picture and change notes and manages the input of updated mixes into the dub sessions. The Assistant Music Editor often handles the technical interface with the Digital Recordist and the final dub stage. This position requires a very detail-oriented approach to the project involving multiple mixes, stems and alternate takes and choices requested by the director.

While often commuting between the Pacific Northwest, Los Angeles, Boston, and Chicago, Shawn Murphy actually does enjoy listening to music (all types) as well as cycling and snow skiing. He resides in Seattle with his wife and two canine children.

MUSIC, ROOMS AND LISTENERS
SCIENCE IN THE CREATION AND DELIVERY OF AUDIO ART

Floyd E. Toole
1301 King James Court
Oak Park, California

> *Could the "best" loudspeaker simply be the one perceived to be the "least bad?"*

Sound sources and rooms are interactive systems. Concert halls and auditoriums are integral parts of live performances. Because architects thrive on distinctive designs, the venues are all different, making each combination of conductor, orchestra and hall a unique auditory event, never, perhaps, to be repeated again. Audiences expect and embrace the spatial and timbral idiosyncrasies and music is enjoyed. Generations of trial and error, and scientific research, have provided guidance about how to design halls that maximize pleasure while not exceeding the limits of listener adaptation. With care, the art —the performance—is satisfactorily delivered to audiences. The music may be relatively constant, but the auditory experience is not. This is sound *production*. It is what it is at the time, and it may never be again.

Elaborately illuminated and sound reinforced, large-venue popular music performances begin with microphones that sample the extreme near field of individual voices and instruments. Gigantic loudspeaker arrays make no effort to place the music into a natural acoustical context; in fact, they are designed to address the audience, avoiding the room boundaries. Much of the artistry is the responsibility of the "front-of-house" mixer, who sits at a console determining how much we hear from each of the musicians on stage, while manipulating signal-processing parameters that affect perceptions of timbre, space and dynamic range. This person can make or break a performance, regardless of how well the musicians perform, how excellent is the inherent design of the loudspeaker system, or the quality of the acoustical environment. This also is sound *production*. It is what it is at the time, and it may never be again.

As enjoyable as live performances are, the bulk of our music is mundanely delivered through loudspeakers in our homes, cars, in cinemas, or through headphones or earbuds as we walk the dog or travel. The music itself has been captured through microphones that sample portions of the near and far fields of voices and musical instruments, with or without additional information from acoustical settings. These streams of data are manipulated in control rooms by recording engineers who decide precisely what we, the audience, will hear of those sounds. Voices and instruments are modified using any of the nearly countless electronic processing algorithms. This is done while monitoring the experience through specific loudspeakers in a specific room. Normally this is done in two channels—stereo. This is the creation of the art, the original performance; it is sound *production*. Unless the audience has playback—i.e. sound *reproduction*—capabilities that precisely duplicate this situation, this is the only time it will be heard. It is what it is at the time, and it may never be again. (Figure 1)

There are no standards for loudspeakers or rooms used in the music industry. Individual studio designers, owners and recording engineers have expectations of what they want to hear in control rooms. There are large differences among them, especially with the advent of home studios. Recording engineers attempt to anticipate what consumers are hearing, trying out their mixes in cars and over inexpensive systems in vogue at the time. Some choose to use monitor loudspeakers that they think portray the characteristics of "average" consumer playback systems. The problem with this approach is that it is not possible to standardize "bad sound." In reality, most playback systems, at all prices, aspire to be neutral. For a variety of reasons they may fail, and when they do they fail in infinite different ways. After nearly 40 years of examining

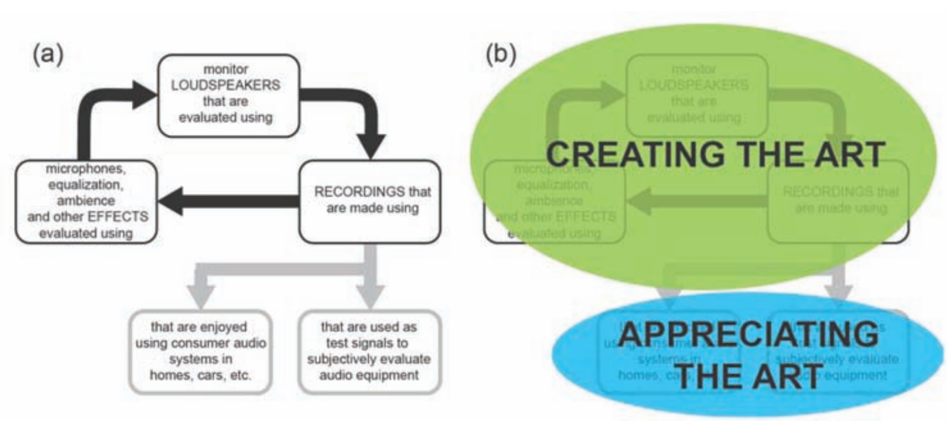

Fig.1 (a) The "circle of confusion" at the core of the audio industry and (b) the two domains that must exhibit fundamental similarities if listeners are to hear the art that was created. From Toole, 2008.

consumer playback devices, I will assert that the only *consistent* factors identifying small and/or inexpensive loudspeakers are a lack of low bass and an inability to play loud.

So, in our everyday music listening—sound reproduction—what can we expect? At the beginning of the process, microphones sampled the sound field radiating from voices and instruments. All of the sound that would reach our ears in a live performance is not captured. Therefore a perfect reproduction of a "live" event is precluded at the outset.

Storage and playback through two channels has been the industry norm for decades, and it may be convenient, but it is incapable of delivering the timbral nuances, directional effects and spatial envelopment of live performances. Instead we get two "real" sources of sound, the left and right loudspeakers, and some number of panned phantom images between the loudspeakers, assuming that we have the discipline to sit in the symmetrical sweet spot. The phantom images suffer from acoustic crosstalk—the sound from both loudspeakers reaches both ears—and both the timbre and spatial representations are unnatural (Toole, 2008, Figure 8.4 and Section 9.1.3). The spectral corruption of the important phantom center image—often the featured artist—is such that even speech intelligibility is degraded (Shirley et al., 2007). With the best of intentions, and unlimited financial investment, when listening to stereo recordings what we hear cannot be the same as a live acoustical experience.

Playing stereo recordings through headphones generates a totally different experience, and one not anticipated by a production process using loudspeakers. It is what it is, and whatever it is, it is not what was intended by the creators of the art. Multichannel audio moves us significantly closer to a desirable objective, but sadly, other than for movies, it has not been commercially viable.

Therefore, in sound *reproduction*, just as in concert hall situations, the "music" may be relatively constant, but our auditory experiences are not. It is what it is at the time, but because it is reproduced sound, we can play recordings again, and again. However, only if our personal playback equipment shares important qualities with that used to create the art, can we be assured of who or what takes the credit or blame for what we hear. We need to disrupt the "circle of confusion" by making the two domains shown on the right in Fig. 1 as similar as possible.

How is it that we find ourselves deriving pleasure from this grossly flawed system? It is because human listeners are remarkably adaptable, and not a little bit susceptible. Over 100 years ago Edison, in his "tone tests," was able to persuade normally intelligent people that his first generation phonograph was indistinguishable from real voices and instruments. He and others mounted live vs. reproduced tests in concert halls. They were all successful (Toole, 2008, Section 2.1). But wait, perhaps listeners were responding to the excellent acoustics of the halls (the recordings were "dead," without reverberation). As several studies have shown, envelopment is a critical quality of a good hall, and therefore of anything produced—or reproduced—within it. If this is not a factor, we are forced to consider that there has been no consequential improvement in reproduced sound in the past century.

Apparently it is not necessary to deliver sounds to the ears that are identical to the "real thing" for listeners to think that they are hearing something resembling, even closely resembling, the real thing. If the basic clues are there, the brain can fill in a lot of blanks. The boundary between reality and perception is a blurry one. Perhaps the most perfect sound reproduction systems are those that provide the most, and the most persuasive, perceptual "hooks" without exhibiting flaws that go beyond the limits of human adaptation.

But expectation also plays into this. There are examples of people hearing things that simply cannot be there. In high-end audio there have been numerous examples of tweaks and gadgets that defy both common sense and physical laws, all of which found a following. If you believe something, there is a chance that you will hear it. All of this can be entertaining so long as it does not encroach on the basics of a family budget.

And then there is the scientific approach.

The literature on concert hall and large-space acoustics and psychoacoustics is extensive, and it has contributed much to understanding sound reproduction in small rooms (Toole, 2006, and 2008, Chapters 4 – 11). However, recording control rooms, mastering rooms, domestic homes and cars are all small spaces. They are subject to enormous variations due to room modes that add low-frequency coloration, and the associated standing waves dictating that no two people in a room will hear exactly the same bass. At higher frequencies, the small dimensions would seem to be contrary to attempts at creating impressions of being in large spaces. But these are understandable phenomena, responsive to mathematical analysis and psychoacoustic experimentation. The problem is that relatively little scientific effort has been invested in trying to understand the acoustical factors underlying the recorded music and film industries. Is this scientific elitism? As a result, folklore, misinformation and simple ignorance compromise what is achieved in these industries. Without some trustworthy technical and acoustical guidance, the circle of confusion will never be broken. At some time, measurements of the right kind need to be trusted to describe what could be considered to be a "reference" sound quality, one that could be the target performance for both production and reproduction. The question is: what are those measurements?

Identifying the right quantitative measures

The familiar claim that "we cannot measure what we hear" stems from observations that curves may look the same but the sound is different. In the early years of audio this was certainly true. In 2013 it mostly relates to situations where the measured data are inadequate in quantity and quality, or are of the wrong kind, or that post processing has not been applied for more effective interpretation.

An omnidirectional microphone at head height at listening locations has long been employed as a basic method of evaluating sound systems in rooms. Traditionally these have been 1/3-octave filtered steady-state amplitude responses. There is a superficial logic to this, but it is not reasonable to assume that a simple omnidirectional microphone, however technically excellent, coupled to a real-time or other analyz-

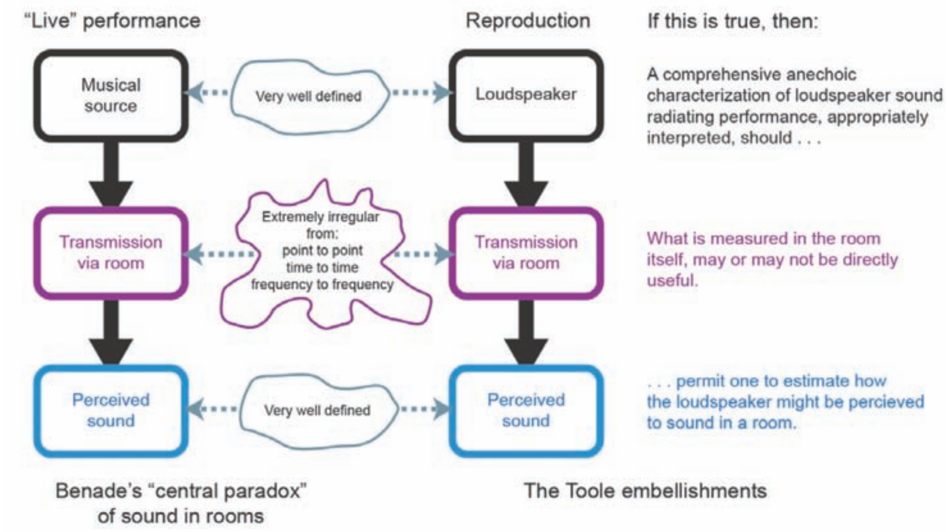

Fig. 2. A concept presented by Benade, (1984) extended by Toole.

er, is a substitute for two ears and a brain.

- To begin with, even though 1/3-octave filters approximate the critical bands/equivalent rectangular bandwidth (ERBs) over some of the frequency range, timbral cues in the form of beats and roughness originate within each of those bands—we need higher resolution if we are to have an adequate predictor of perceived timbre from sound reproducing devices (Toole, 2008 pp. 450-451).
- The common ± 3 dB tolerance is extremely generous, especially because there is no bandwidth associated with it. Humans respond to localized spectral variations at much lower amplitudes (Toole and Olive, 1988).
- The measurements include the room, and the associated non-minimum-phase reflections. Humans treat these very differently than measuring devices, because they arrive at different times and amplitudes, and from directions different from that of the direct sound. What may be perceived as innocent, indeed pleasurable, spaciousness to a human may be interpreted as a bump or dip in a measured curve that suggests a need for equalization. Evidence of non-minimum-phase phenomena should not be equalized, or what is thought of as a remedial measure has the potential to create audible problems. I suspect that this misuse of equalization is responsible for much of the criticism of it.
- The "room curve" may fluctuate because of amplitude response flaws in the loudspeaker or because of frequency-dependent directivity. Equalization can compensate for the former, but not the latter. Neither can it compensate for all fluctuations caused by frequency selective absorption at room boundaries. Significant understanding of underlying causes is required before deciding on remedial actions.
- Finally, there is indecision about the target curve to which a sound system is equalized. There is broad agreement that a flat steady-state "room curve" sounds too bright. So, depending on the venue and the program, different installers/consultants/industries employ different forms of high-frequency rolloff and/or downward spectral tilts. This is usually done with no knowledge of, or requirements for, the loudspeakers or the rooms, and yet what is measured embraces both. Such practices cannot be generalized.

This incomplete list of issues refers to common practice within the audio industry. But, more seriously, some or all of them are embodied in international standards purporting to set objectives for sound quality within the broadcast and film industries.

Can we do better? Almost certainly. I can think of no better way to introduce the viewpoint than with Fig. 2, beginning on the left with the basic observation from Arthur Benade, including my embellishments to bring it into the present context.

This is a significant change in perspective, yet it aligns with everyday experience. We can track a voice as a conversation moves from one room, down a corridor to another room. There are huge, complex, changes to the sounds arriving at our ears, and yet subconsciously we know that the sound of that voice remains essentially constant. It is perceived as a voice in changing acoustical contexts. Some of us have experienced moving around within a space, listening to a repeated passage of music, noting that what we hear is more stable than the varying details in "room curves" measured where we are located. When we stop moving, and adapt to the acoustical circumstances, rooms tend to become contexts. To a physicist, a room adds an impossibly complicated distortion of the transfer function between a sound source and a listener. To a listener, a good room embellishes the music. Understanding how reflections are perceived is important (Olive and Toole, 1989).

Adaptation—adjusting to life in an ever-changing (acoustical) world

Chapter 9 in my book discusses adaptation, beginning with:

> "In the contexts of precedence effect (angular localization), distance perception and spectral compensation

> (timbre), humans can track complex reflective patterns in rooms and adjust our processes to compensate for much that they might otherwise disrupt in our perceptions of where sounds come from, and of the true timbral signature of sound sources. In fact, out of the complexity of reflected sounds we extract useful information about the listening space, and apply it to sounds we will hear in the future. We are able, it seems, to separate acoustical aspects of a reproduced musical or theatrical performance from those of the room within which the reproduction takes place. This appears to be achieved at the cognitive level of perception – the result of data acquisition, processing and decision making, involving notions of what is or is not plausible. All of it indicates a longstanding human familiarity with listening in reflective spaces and a natural predisposition to adjusting to the changing patterns of reflections we live in and with. The inevitable conclusion is that all aspects of room acoustics are not targets for "treatment". It would seem to be a case of identifying those aspects that we can, even should, leave alone, and focusing our attention on those aspects that most directly interact with important aspects of sound reproduction—reducing unwanted interference on the one hand or, on the other hand, enhancing desirable aspects of the spatial and timbral panoramas." (Toole, 2008, p.171).

There is a caution to be noted here. It is that adaptation takes time. When we are moving around we hear things that may gradually disappear when we sit down, or which may not be identified at all if one is seated when the sound begins

A dramatic example of the power of this adaptation is described in Section 11.3.1 (ibid), where three very good loudspeakers were subjectively compared to each other in four different rooms. In addition to live (listener in the room) double-blind, randomized, comparisons, binaural recordings were made for subsequent comparisons using insert earphones. It turned out that when the comparisons were organized in the manner of the live tests, one room at a time, the binaural test results were essentially the same as the live results. Statistically, the variable "loudspeaker" was highly significant ($p = 0.05$) and "room" was not a significant factor. Then those same binaural recordings were presented in a different sequence, allowing each loudspeaker in each room to be compared to each other. The results were very different: "room" was a highly significant variable ($p = 0.001$) and loudspeaker was not a significant factor. The sound of the room had merged with the sound of the loudspeaker and could not be separated because listeners had no opportunity to adapt. In this version of the test, the sounds of the different rooms were more distinctive than the sounds of the different loudspeakers. Among other things this is a caution to observe when performing binaurally recorded subjective comparisons.

Characterizing the sound source: collecting the data

Describing the three-dimensional sound fields emanating from voices and musical instruments could be one of those endless tasks because they exist in infinite variations. However, describing loudspeakers intended to *reproduce* voices and musical instruments is entirely feasible, indeed desirable, if one expects to reproduce those sounds without degradation. Ideally, we would look for indications of transparency, or "neutrality". Because we listen in reflective rooms, it is necessary to make many measurements.

Beginning in the early 1980s I collected data on loudspeakers over full horizontal and vertical orbits. It was very revealing of what listeners were responding to when judging sound quality (Toole, 1985, 1986). A smooth and flat on-axis frequency response was a starting point. As loudspeakers improved, it became clear that the loudspeakers awarded the highest subjective ratings also had relatively smooth sound power—i.e. relatively constant directivity vs. frequency. It was also shown that these anechoic data were capable of closely predicting steady-state room curves measured at the listening positions in a small room (Toole, 1986, Figures 18 – 20). This provided the basis for taking the technique to a higher level by combining measurements made at different angles to estimate the sounds arriving at a listener's ears in more generalized listening rooms (Devantier, 2002).

What we now call the "spinorama" consists of 70 anechoic measurements made at 2 m at 10° increments on horizontal and vertical orbits, frequency resolution 2 Hz (1/20-octave smoothed). These data are then processed to reveal:

- The on-axis curve: important to design engineers and solo listeners.

- Listening window: the average frequency response within a ± 30° horizontal and ± 10° vertical window. This describes the direct sound for an audience.
- Early reflected sound: the average frequency response at angles estimated for first reflections in typical domestic listening rooms.
- Sound power: average of all 70 frequency responses, each one weighted according to the proportional area of the sphere it represents.
- Total sound power directivity index: the difference between the listening window and sound power curves (a unique definition).
- First reflections directivity index: the difference between the listening window and early-reflected sound curves (a unique definition). This is included because first reflections account for much of what is measured, and heard, in rooms.

See Fig. 3. The increasing spatial averaging that occurs in the progression from on-axis through to sound power allows for the separation of acoustical interference effects (not very audible) from resonances (easily audible). The example loudspeaker is exemplary in all respects: flat and smooth axial frequency response, well-behaved, relatively constant, directivity, no evidence of audible resonances. This professional monitor loudspeaker should ensure that the recording engineer is making artistic decisions while listening to sound that is about as good as it gets. However, as noted earlier, it also represents the performance target for the majority of loudspeakers at any price. The important consistent factor, the limited low-frequency extension of small inexpensive loudspeakers, can be imitated with a variable high-pass filter in the signal path.

But what about consumers? To disrupt the circle of confusion shown in Fig. 1, consumer loudspeakers must be similar in performance to professional loudspeakers. Figure 4 shows that this is possible, even at moderate prices.

Consumers listening to these loudspeakers will hear spatial and timbral aspects of the art very much as the creators did. Most of the irregularities in the curves are close to or below the thresholds of detectability, and are not likely to seriously detract from the experience (Toole and Olive, 1988). Very low bass output is somewhat lacking, and this small cone/dome system will not play as loud as the monitor. Bass management and subwoofers would address both problems. However, larger, more expensive, domestic loudspeakers can perform in a manner that closely emulates the monitor. Sadly, price is not a reliable indicator of sound quality, and most manufacturers are reluctant to reveal useful specifications on their products, leaving consumers in an unfortunate circumstance. At present there is a standards group working on implementing the spinorama as a basis for loudspeaker specifications. However, it will be a voluntary standard.

The intent of this measurement scheme was to be able to anticipate how loudspeakers would behave in rooms both subjectively and objectively. As was shown in Toole, 1986, sound power is the dominant factor at low-to-middle frequencies, and the direct sound is the dominant factor at the highest frequencies. Taking this simple approach, Fig. 5 shows a comparison for a loudspeaker having both frequency response and directivity problems, measured in a typical domestic listening room.

Obviously there is a close relationship among these three curves. The sound power includes effects of both the axial frequency response and directivity, the inverted Directivity Index (DI) relates only to directivity. The fact that a major cause of the unfortunate shape of the room curve is off-axis radiated sound, i.e. reflections, means that equalization may not be an appropriate corrective measure. Replacement of this $10K loudspeaker seems like a better option. The shortfall at the highest frequencies indicates that the direct sound is the dominant factor in that range. Disagreement at fre-

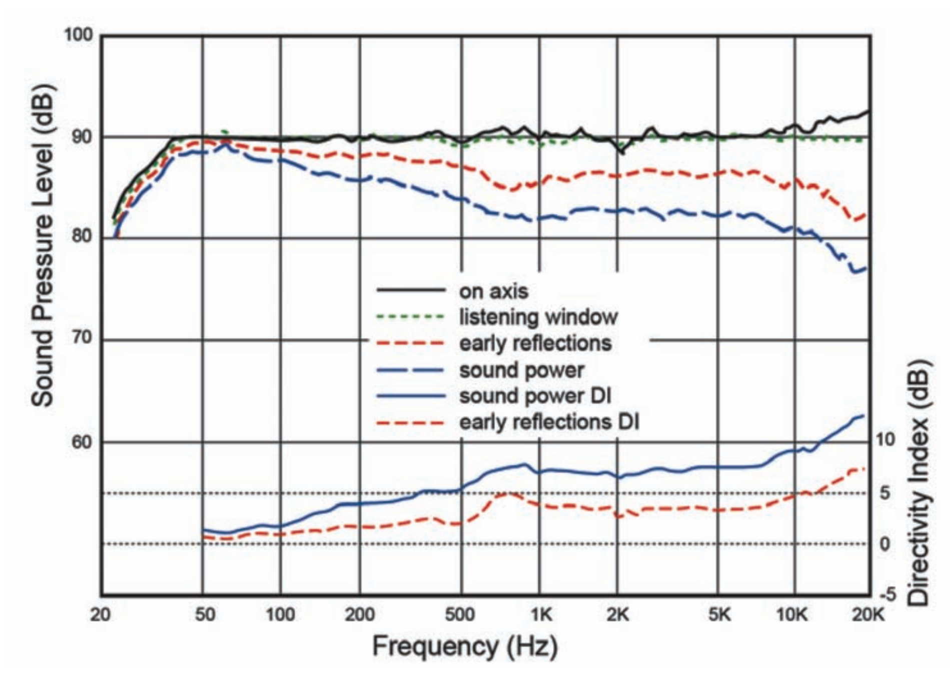

Fig. 3 The spinorama for a high power cone/horn professional monitor loudspeaker. Data: Harman International.

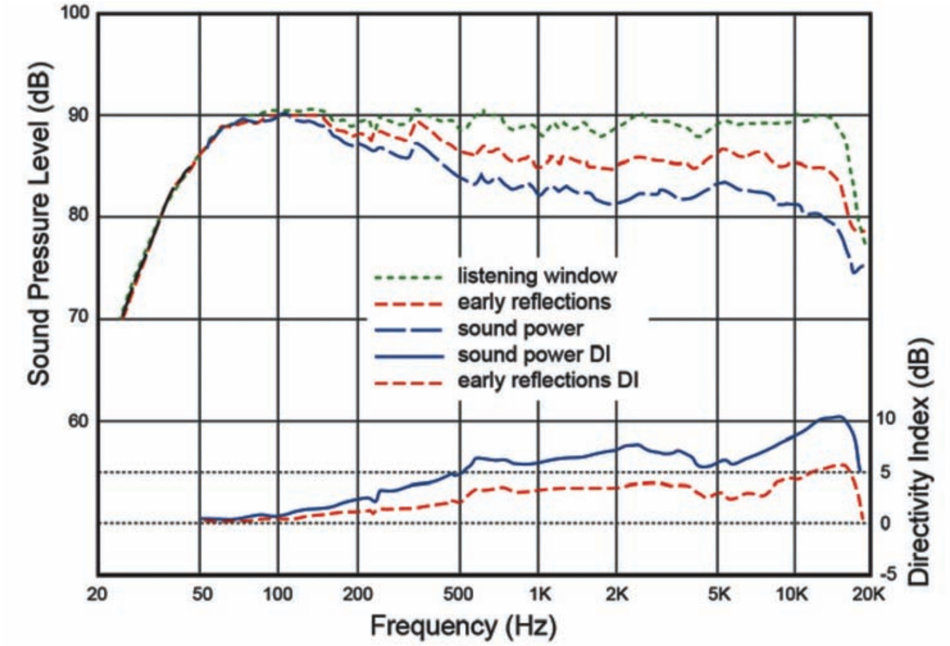

Fig. 4 The spinorama for a well-designed floor standing domestic loudspeaker with a retail price of $329 at time of writing. Data: Harman International.

quencies below about 200 Hz is the result of standing waves in the small room. Clearly an on-axis measurement alone is insufficient to describe events in small listening rooms.

Recently I presented evidence that this basic relationship holds for loudspeakers in rooms ranging in volume from small domestic rooms to large auditoriums (a volume ratio exceeding 100:1). In large venues the effects of air absorption must be incorporated and, of course, there are the inescapable effects of low-frequency modes in small rooms (Toole, 2012). Notwithstanding the assurance that comprehensive anechoic data on loudspeakers can be used to estimate acoustic measurements in rooms, the real question is: how does all of this relate to subjective evaluations of sound quality? Can we look at a set of anechoic measurements and anticipate how a loudspeaker will sound in a room?

In the 1970s, when I began my research into this topic, it was clear to me that without good subjective data progress would be impossible. I conducted well-controlled, double-blind, multiple-comparison (four at a time) subjective evaluations of loudspeakers, using many listeners, and many musical selections. It was all randomized, not automated, and it took a lot of time. The results showed that certain features of loudspeaker performance, as evidenced in the anechoic data, appeared to be strongly related to subjective ratings of sound quality. Others were not (Toole, 1982, 1985, 1986a, 1986b, most of this is summarized in Toole, 2008). It was found that listeners with even relatively small hearing losses exhibited measurable degradation in rating consistency. Nowadays, listeners are selected and trained (Olive, 1994, 2001), resulting in more efficient tests, while not affecting the ability of those ratings to reflect opinions of consumers in general (Olive 2003).

It has been a consistent observation that the relative merits of loudspeakers are basically stable across different rooms. It may have been advantageous that these were multiple-comparison tests, as opposed to single stimulus or simple A vs. B tests. Having more comparison sounds could have assisted listeners in separating, or streaming (Bregman, 1990), the sounds of the three or four loudspeakers as distinct from the relatively constant timbre contributed by the room. As Bregman says, "It seems likely that the auditory system… has developed principles for "betting" on which parts of a

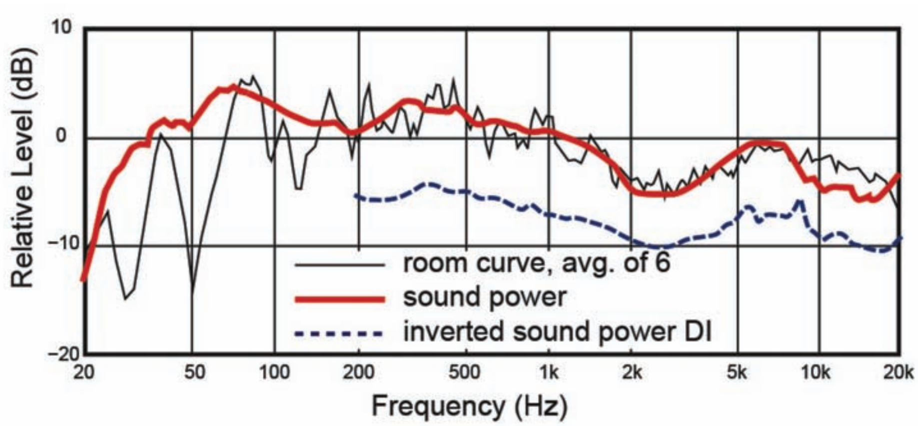

Fig. 5 The average of six in-room measurements is compared to the sound power, and the inverted sound power directivity index (DI) for the same loudspeaker. Acoustically the room is a typical, furnished, domestic space. Data: Harman International.

sequence of sensory inputs have arisen from the same source. Such betting principles could take advantage of properties of sounds that had a reasonably high probability of indicating that the sounds had a common origin." (ibid, p.24).

Of comparable interest is the fact that listeners formed remarkably similar and consistent sound quality ratings when listening to widely different kinds of music, most of which was created in recording studios. Listeners in the tests never heard it in the control room and thereby had no true reference of excellence—there was only one's internal generic sense of what it might have sounded like. But, without a certain mental image of perfection, one may be able to recognize imperfections—aberrations, colorations or distortions that are not part of *any* natural sounds. Evidence that this might be so could be seen in the descriptive reports prepared by the listeners. Detailed essays incorporating sometimes colorful language described unpleasant attributes associated with low scores, while high scores were justified with few words of flattery. Could the "best" loudspeaker simply be the one perceived to be "least bad"?

Summarizing what these listeners seem to have done, they first separated the timbral contributions of the loudspeakers from those of the rooms in which they were evaluated. This was done while listening to program material for which they could not have had a "live" reference experience. And, finally, they were able to identify, and rate, loudspeakers according to the degree of imperfection. All highly rated loudspeakers conformed to very simple "motherhood" objectives: smooth flat axial frequency response, relatively constant directivity, and low distortion. This is remarkable, yet my colleagues and I have conducted hundreds of such tests over about 35 years, and there have been no surprises.

Closing the loop

Until recently the observed relationships between measured data and listening test results have been entirely subjective. That we had no numerical correlations didn't mean that we couldn't see what good loudspeakers "looked like" in spinoramas. Nevertheless, ultimately, the objective was to devise an algorithm for processing measured anechoic data that yielded predictions of subjective ratings in rooms. In 2004, Olive (Olive 2004a, 2004b) assembled data on 70 loudspeakers that had been used in competitive analysis of products at Harman International. They ran the gamut from large expensive floor standing units to small bookshelf units. For each of them he had the results of double-blind listening tests in a room, and anechoic spinorama data. Based on years of observation, and psychoacoustic research data, he created metrics and exercised them in a multiple regression model. The result was a correlation coefficient of 0.86 between a rating predicted from anechoic data, and the results of listening tests conducted in a normal listening room. Clearly this is not guesswork. This is benchmark research, but it is not a complete answer. In these tests the listening room was a constant factor, meaning that all room mode and adjacent boundary issues were fixed. And, this was a domestic/control-room-size room. Comparably competent listening tests and correlations have yet to be done in large venues.

About 30% of the factor weightings leading to sound quality ratings related to bass performance. Therefore, in calibrating systems in different venues, in addition to spinorama data, we will need some in-room measurements, and possibly some room-specific adjustments at low to mid frequencies.

The roles of room acoustics, acousticians and psychoacousticians

In auditoriums for sound production, the room is part of the performance, and therefore it matters greatly. The science applicable to this is very well documented, and research is ongoing. However, the small rooms in which we are entertained at home, and control rooms in which music recordings and many components of film soundtracks are created, are very different matters.

The room is the dominant factor at low frequencies—standing waves, and the manner in which sources and listeners interact with them are the central issues. Room dimensions, acoustical absorption and its placement, locations of sound sources and listeners, are prime determinants of the spectral and temporal quality of bass that is heard. Massive amounts of low frequency damping helps, but is costly and/or bulky—not compatible with common notions of interior décor. Because all modes are not equally energized by woofers and not equally heard by listeners, the traditional "ideal room" investigations do not yield generalizable solutions. The supposedly advantageous dimensional ratios apply only to predetermined arrangements of sound sources and listeners within the room boundaries. With multiple sound sources operating independently (i.e. connected to separate channels) the acoustical coupling to the room modes is simply not predictable. However, if the multiple sources of low frequency energy are driven by the same signal (bass management in surround processors), it is possible to employ strategies of constructive and destructive interference among the low-frequency room modes to control the modes in a perfectly rectangular space that are and are not energized. This allows the placement of multiple listeners in regions where the bass may be more uniform and more similar.

Taking this to a higher level, one that includes rooms of arbitrary shape and allowing for more flexible arrangements of listeners and subwoofers, it is possible to process the signals supplied to each subwoofer, manipulating the room modes so that the result is a more uniform bass performance at several listening locations, and a superior bass performance in all locations. Interestingly this can be very successful with no low-frequency absorption other than that naturally occurring in the room boundaries. Adding absorption simply makes it easier. All of this is discussed in detail in Toole, 2008, Chapter 13, and references therein.

Above what I call the transition frequency (called the Schroeder crossover frequency in large auditoriums), around 200-300 Hz in domestic-size rooms, the direct sound and first reflections dominate what is measured and heard, meaning that loudspeaker directivity is a major factor, as well as the frequency-dependent absorption at the reflection points. Evidence suggests that listeners prefer loudspeakers radiating similar spectra in all directions (i.e. relatively constant directivity). Consequently, it is logical that reflections of those sounds should not be spectrally altered by reflecting or scattering surfaces of room boundaries. In practical terms, this argues for areas of either full reflection or full absorption. At present I know of no spectrally neutral sound-attenuating device, although scattering/diffusing devices can approach this, but with other consequences. The widely used (and recommended in some standards) 1-inch absorbing panels are ill advised, certainly at first-reflection locations. At very high frequencies direct sound dominates, simplifying acoustic concerns.

The reverberation time (RT) target for home entertainment spaces, based primarily on speech intelligibility, is easy to hit: ≤ 0.5 s. This number applies also to cinemas and film production facilities (dialogue again), but music recording control rooms tend to aim for lower RTs, sometimes much lower. Even at 0.5 s, with relatively directional sound sources, there is nothing resembling a diffuse sound field, meaning that random incidence absorption coefficients are of limited use. The importance of first-reflected sounds suggests that it might be advantageous to know the angle-specific frequency-dependent absorption and scattering/diffusing properties of acoustical materials and devices.

There is evidence that the precedence effect deteriorates when the spectra of the direct and delayed sounds differ. It is plausible to think that similar effects extend to other aspects of perception, including spaciousness and timbre. Chapters 5 thru 10 in my book provide an overview of some of the factors, but it is clear that we need more data elaborating the progression of perceptual effects for level and spectral variations within isolated and multiple reflections. These data would ideally come from psychoacoustic experiments incorporating delay and directional variables associated with realistic listening circumstances. The result would be solid evidence supporting performance targets and tolerances for the off-axis performance of loudspeakers and the reflecting/absorbing/diffusing surfaces at which first reflections occur in room. This could be an interesting collaboration between scientists with acoustical and psychoacoustical expertise. A global industry awaits guidance.

Subjectively it has been found that the effect of the room is greatest with a single loudspeaker (channel) with its effects diminishing as the active channel count increases. However, with a very high proportion of movie and TV sound emerging from the front-center channel (a mono signal) the room cannot be ignored.

The inevitable question is: What constitutes an "ideal" listening room? Right now we don't know, and given the ability of humans to adapt to differing rooms, it may matter less than some people would like us to think. However, there is a limit to what we can adapt to, and adaptation very likely utilizes a portion of our neural "horsepower" (causing fatigue?). So, perhaps that feeling of exquisite relaxation I get when I listen to a superb sound system is real, not a figment of my imagination. If so, there is motivation for research by acoustical scientists, and work to do by competent acoustical engineers.

Looking Ahead

Because of the science we have, and the abundance of affordable measurement tools, the standards of sound reproduction in general have been elevated in homes and recording facilities. However, problems remain, in the form of loudspeakers that are less good than they could have been, flawed acoustical treatment practices and misguided attempts to "equalize" rooms. Right now, there is no assurance that reproduced sound closely resembles what was heard at the time the art was created. This is a pity, because it is not possible to confidently attribute credit or blame for what we hear.

In the end, consumers, audio professionals and acoustical consultants need to be able to anticipate whether a playback facility is likely to deliver a reasonable facsimile of an original performance, without exceeding the tolerances of normal adaptation. We certainly need more and better specifications on loudspeakers, and manufacturers with the courage to publish them. Between here and there are many opportunities for challenging applied research projects, generating new knowledge, and interminable committee meetings for those willing to undertake the standards work.

The technology to do much better exists. In the meantime, there are countless personal opinions to sort through, and a lot of adapting to get on with so that we can enjoy the abundance of music out there. Fortunately music is what it is, in spite of the acoustical variations and abuses we heap upon it.

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Floyd E. Toole studied electrical engineering at the University of New Brunswick, and at the Imperial College of Science and Technology, University of London, where he received a Ph.D. In 1965 he joined the National Research Council of Canada, where he reached the position of Senior Research Officer in the Acoustics and Signal Processing Group. In 1991, he joined Harman International Industries, Inc. as Corporate Vice President—Acoustical Engineering. In this position he worked with all Harman International companies, and directed the Harman Research and Development Group, a central resource for technology development and subjective measurements, retiring in 2007.

His research focused on the acoustics and psychoacoustics of sound reproduction in small rooms, directed to improving engineering measurements, objectives for loudspeaker design and evaluation, and techniques for reducing variability at the loudspeaker / room / listener interface. For papers on these subjects he has received two Audio Engineering Society (AES) Publications Awards and the AES Silver Medal. He is a Fellow and Past President of the AES, a Fellow of the Acoustical Society of America, and a Fellow of CEDIA (Custom Design and Installation Association). He has been awarded Lifetime Achievement awards by CEDIA and ALMA (Association of Loudspeaker Manufacturing & Acoustics International).

Acoustical News

Elaine Moran
Acoustical Society of America
Melville, New York 11747

Theodore M. Farabee

Theodore Farabee awarded ASME's Per Bruel Gold Medal

Theodore M. Farabee, a chief scientist in the Signatures Department of the Naval Surface Warfare Center, Carderock Division (NSWCCD) in West Bethesda, MD, was awarded the Per Bruel Gold Medal for Noise Control and Acoustics by the American Society for Mechanical Engineers (ASME). Established in honor of Dr. Per Bruel, who pioneered the development of sophisticated noise and vibration measuring and processing equipment, the medal recognizes eminent achievement and extraordinary merit in the field of noise control and acoustics.

Dr. Farabee was recognized for significant accomplishments in fluid mechanics, aero-hydrodynamics, complex propulsor fluid mechanics flow interactions and platform structure-elastic interactions; particularly leadership in the understanding and control of induced sound, and work on ship and submarine flow-noise reduction.

Dr. Farabee is the U.S. Navy's senior research scientist/technical consultant (ST) for radiated flow noise signature control. In this position, he is responsible for the conduct of broad-based, multidisciplinary research, integrating all aspects of acoustic signatures and related mitigation technologies for application to ships and submarines. He earned his bachelor's degree in ocean engineering at Florida Atlantic University in 1973. He earned his master's in engineering and his Ph.D. in mechanical engineering at The Catholic University of America, Washington, D.C., in 1976 and 1986, respectively.

He is a Fellow of the ASME and a Member of the Acoustical Society of America.

Michael R. Moldover

Michael R. Moldover honored by ASME

Michael R. Moldover, NIST Fellow and leader of the Fluid Metrology Group at the National Institute of Standards and Technology, was awarded the Yeram S. Touloukian Award by the ASME in 2012. He was recognized for outstanding contributions as an experimentalist who, over the last 35 years, has had a tremendous fundamental impact in both the equilibrium and transport areas of the thermophysical properties field. The award recognizes outstanding technical contributions in the field of thermophysical properties.

Dr. Moldover joined the National Bureau of Standards, now the National Institute of Standards and Technology, in 1972. He has been a NIST Fellow since 1995 and leader of the Fluid Metrology Group, now in the Sensor Science Division, since 2004. Under his leadership, NIST's Fluid Metrology Group measured the properties of replacements for ozone-layer-damaging refrigerants and the properties of reactive gases used in semiconductor processing. Now, the group is studying nozzles to improve flow standards, measuring the CO2 in flue gases emitted by coal-burning power plants, and developing methods to measure rapidly changing fluid flows.

Dr. Moldover received his bachelor's degree in physics at Rensselaer Polytechnic Institute in 1961. He earned his master's degree and Ph.D. in physics at Stanford University, California, in 1962 and 1966, respectively. He is a Fellow of the Acoustical Society of America and has authored more than 150 peer-reviewed papers.

Dr. Moldover's awards include the U.S. Department of Commerce's Bronze (1980), Silver (1982), Gold (1987) and Silver (2011) medals; NIST's Samuel Wesley Stratton Award for Research Excellence (1988) and Chemical Science and Technology Laboratory Technical Achievement Award (2005); and the U.S. Office of Personnel Management's Presidential Rank Award (2009).

Victor Zue honored by IEEE

Victor Zue, the Delta Electronics Professor of Electrical Engineering and Computer Science at MIT and former Director of the Institute's Computer Science and Artificial Intelligence Laboratory (CSAIL) from 2007 - 2011 (and co-director of CSAIL since its inception in 2004), is the recipient of

Victor Zue

the 2013 IEEE James L. Flanagan Speech and Audio Processing Award. He is cited "for pioneering contributions to acoustic phonetics and conversational spoken-language systems." The IEEE James L. Flanagan Speech and Audio Processing Award was established in 2002 as a Technical Field Award given for outstanding contribution to the advancement of speech and/or audio signal processing. Victor will be presented the award at the May, 2013 International Conference on Acoustic, Speech, and Signal Processing in Vancouver.

His work with the MIT Laboratory for Computer Science's Spoken Language Systems Group, which he headed from 1989 to 2001, focused on the development of many systems that enable a user to interact with computers using multiple spoken languages (English, Japanese, Mandarin, and Spanish). His current research interests are in the area of applying human language technologies to enable easy access of structured and unstructured information from the web, especially in applications such as education and healthcare.

Dr. Zue has served on the technical advisory board of many multinational corporations, and on numerous committees for the US government. From 1996-1998, he chaired the Information Science and Technology, or ISAT, study group for the Defense Advanced Research Projects Agency of the U.S. Department of Defense, helping the DoD formulate new directions for information technology research. In 1999, he received the DARPA Sustained Excellence Award. Dr. Zue is a Fellow of the Acoustical Society of America, and a Fellow of the International Speech Communication Association. He is also a member of the U.S. National Academy of Engineering, and an Academician of the Academia Sinica.

Jan D. Achenbach awarded the ASME Medal

Jan D. Achenbach, distinguished McCormick School professor emeritus in service at Northwestern University, was awarded the ASME Medal of the American Society of Mechanical Engineers (ASME). He was recognized for groundbreaking contributions to the theory and applications of waves in solids, particularly in the ultrasonic range,

Jan D. Achenbach

applied to acoustic microscopy, dynamic fracture, and laser-based ultrasonics; and for pioneering ultrasonic methods for quantitative nondestructive evaluation and structural health monitoring. He received the prestigious ASME Medal. The medal, established in 1920, is awarded for eminently distinguished engineering achievement.

Achenbach earned his Ph.D. in aeronautics and astronautics, with a minor in mathematics, from Stanford University, California, in 1962. He received an honorary doctorate from Zhejiang University, China, in 2011.

Dr. Achenbach has been a member of the faculty at Northwestern University since 1963. In 1985 he founded the Center for Quality Engineering and Failure Prevention at Northwestern. He is known for his work on the propagation of waves in solids, with present emphasis on the theory and applications of ultrasonic methods to quantitative nondestructive evaluation, particularly the measurement of elastic properties of thin films by acoustic microscopy, and the detection of cracks and corrosion in safety-critical structures. He is the author of *Wave Propagation in Elastic Solids* (North-Holland, 1973) and *Reciprocity in Elastodynamics* (Cambridge University Press, 2003), as well as numerous papers in technical journals.

Dr. Achenbach an Honorary Member of the ASME and a Fellow of the Acoustical Society of America, and the ASME. He is a member of the National Academy of Sciences, the National Academy of Engineering and five other academies. In 2003 he was awarded the National Medal of Technology for engineering research and education in the use of ultrasonic methods, and in 2005 he received the National Medal of Science for pioneering the field of quantitative nondestructive evaluation. His awards include the 2012 ASME Medal, the Timoshenko Medal, the William Prager Medal, and the Theodore von Karman Medal. In 2011, he was awarded a rare honorary doctorate degree from China's Zhejiang University.

NIOSH and NHCA present 2013 Safe-in-Sound Excellence in Hearing Loss Prevention Awards™

The National Institute for Occupational Safety and Health (NIOSH), in partnership with the National Hearing Conservation Association (NHCA), is pleased to announce the winners of the 2013 Safe-in-Sound Excellence in Hearing Loss Prevention Awards™. The awards honor organizations that have shown dedication to excellence in hearing loss prevention practices in the work environment and beyond.

The recipient of the 2013 Safe-in-Sound Award for Excellence is the Vulcan Materials Company (VMC), a major producer of construction aggregates. Recognized for their commitment and implementation of a quality data-driven hearing loss prevention program, VMC has embraced innovative and cost–effective noise measurement and control strategies. They provide extensive noise measurement and control

training for select employees to function as industrial hygiene support staff. VMC is also leading the advancements in noise monitoring strategies for mobile workers by integrating sophisticated technologies such as GPS, and video into their noise measurement protocols.

This year there are two innovation awardees. One is presented to Johns Manville (JM), a Berkshire Hathaway company and a leading manufacturer of premium-quality building insulation, commercial roofing, roof insulation, and specialty products for commercial, industrial and residential applications. They are recognized for their implementation of an innovative hearing loss prevention program that uses metrics to track noise exposure levels along with noise control engineering training.

The second award for Innovation goes to Dangerous Decibels®, a multi-faceted, evidence-based intervention program dedicated to the prevention of noise-induced hearing loss and tinnitus. It is being recognized for the development, widespread dissemination and cultural adaptation of innovative training strategies shown to positively change knowledge, attitudes and behaviors in youth and adults, including occupational settings (www.dangerousdecibels.org), and applying solid scientific and theoretical basis into all program aspects.

Nominations for the next awards will be accepted until September 6, 2013. For further information please visit www.safeinsound.us.

Calendar of Meetings and Congresses

Compiled by the Information Service of the International Commission for Acoustics

2013

23 - 25 April	Marrakech, Morocco **1st Euro-Mediterranean Conference on Structural Dynamic and Vibroacoustics** (**MEDYNA 2013** http://www.medyna2013.com
1 - 4 May	Singapore **3rd International Congress on Ultrasonics** (**ICU 2013**) concurrently organized with the 32nd International Symposium on Acoustical Imaging (AI 2013) http://www.epc.com.sg/ICU 2010.pdf
20 - 23 May	Hong Kong **2nd Symposium on Fluid-Structure-Sound Interactions and Control** http://www.fssic2013.com/
26 - 29 May	Krakow-Rytro, Poland **11th International Conference on Active Noise and Vibration Control Methods** (**MARDiH-2013**) http://www.vibrationcontrol.pl
26 - 31 May	Vancouver, Canada **2013 IEEE International Conference on Acoustics, Speech and Signal Processing** (**ICASSP**) http://www.icassp2013.com
02 - 07 June	Montreal, Canada **21st International Congress on Acoustics** (**ICA 2013**), **165th Meeting of the Acoustical Society of America, and 52th Annual Meeting of the Canadian Acoustical Association** http://www.ica2013montreal.org
09 - 11 June	Toronto, Canada **International Symposium on Room Acoustics** (**ISRA 2013**) http://www.isra2013.com
23 - 28 June	Corfu, Greece **1st International Conference and Exhibition on Underwater Acoustics** http://www.uam-conferences.org
07 - 11 July	Bangkok, Thailand **20th International Congress on Sound and Vibration** (**ICSV20**) http://www.icsv20.org
09 - 09 July	Glasgow, UK **Invertebrate Sound and Vibration** (**ISV 2013**) http://www.isv2013.org
30 July 03 Aug.	Stockholm, Sweden **Stockholm Music Acoustics Conference** (**SMAC 2013**) http://www.european-acoustics.org/smac-2013
26 - 28 Aug.	Denver, USA **NOISE-CON 13** http://www.inceusa.org
27 - 30 Aug.	Denver, USA **Wind Turbine Noise 2013** http://www.inceusa.org
04 - 06 Sept.	Portoroz, Slovenia **12th International Conference "Application of Contemporary Non-destructive testing in Engineering"** (**ICNDT 2013**) http://lab.fs.uni-lj.si
08 - 13 Sept.	Pirenopolis, Brazil **International Bioacoustics Congress** (**IBAC 2013**) http://www.ibacbrazil.com/Home.html
15 - 18 Sept.	Innsbruck, Austria **Internoise 2013** http://www.internoise2013.com
23–27 Sept.	San Diego, **USA IEEE Oceans 2013** http://www.oceans13mtsieesandiego.org/
9 - 11 Oct.	Hangzhou, China **4th Pacific Rim Underwater Acoustics Conference** (**PRUAC 2013**) http://pruac.zju.edu.cn/index.htm
23 - 25 Oct.	Kerala, India **12th biennial Symposium on Ocean Electronics** (**SYMPOL 2013** http://sympol.cusat.ac.in
10 - 15 Nov.	New Delhi, India **Acoustics 2013 New Delhi** http://www.acoustics2013newdelhi.org
2 - 6 Dec.	San Francisco, USA **166th Meeting of the Acoustical Society of America** http://www.acousticalsociety.org

2014

7–10 April	Taipei, Taiwan **OCEANS'14** http://www.oceansconference.org/
5 - 9 May	Providence, USA **167th Meeting of the Acoustical Society of America** http://www.acousticalsociety.org
06 - 10 July	Beijing, China **21th International Congress on Sound and Vibration** (**ICSV21**)

07 -12 Sept. Krakow, Poland **Forum Acusticum 2014**
http://www.fa2014.pl/

14–19 Sept. St. John's, Newfoundland, Canada, **OCEANS'14**
http://www.oceansconference.org/

06 - 10 Oct. Prague, Czech Republic **11th European Conference on Non Destructive Testing**
http://www.ecndt2014.com/

27 - 31 Oct. Indianapolis, USA **168th Meeting of the Acoustical Society of America**
http://www.acousticalsociety.org

16 - 19 Nov. Melbourne, Australia **Internoise 2014**
http://www.internoise2014.org

2015

11 - 15 May Metz, France **4th International Congress on Ultrasonics (ICU 2015)**
http://www.me.gatech.edu/2015-ICU-Metz/

18 - 22 May Pittsburgh, USA **169th Meeting of the Acoustical Society of America**
http://www.acousticalsociety.org

19–21 May Genoa, Italy **OCEANS'15**
http://www.oceansconference.org/

2 - 6 Nov. Jacksonville, USA **170th Meeting of the Acoustical Society of America**
http://www.acousticalsociety.org

2016

05 - 09 Sept. Buenos Aires, Argentina **22nd International Congress on Acoustics (ICA 2016)**
http://www.ica2016.org.ar/

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