AES Audio Engineering Society - Pacific Northwest Section

Around the Puget Sound, Seattle, Washington, U.S.A.

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2004-2005 Officer & Committee Elections

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A Digitally Steered Line Array

With

Dave Gunness - Eastern Acoustic Works

Sponsored by

AES PNW Section

Wednesday, June 23, 2004 at 7:30pm

Meeting will be held at PNTA

615 S. Alaska Street, Seattle 98108

The line array loudspeaker has come back into vogue after a 30-odd year hiatus. If you're old enough, you'll recall the old ElectroVoice LR2 and LR4 arrays (marketing called them *Sound Columns* back then and (of course), the ubiquitous and oft maligned Shure Vocal Master. Leo Beranek described the line array in his text, *Acoustics* in 1954, followed by Harry F. Olson in his text, *Acoustical Engineering* in 1957. Today's line arrays apply improved transducers and directivity control devices to produce a composite array capable of the highest performance.

Since 1985 or so, researchers have been investigating the application of Digital Signal Processing to line arrays for the purpose of controlling their directional coverage. Today sufficiently powerful DSP chips exist to make a commercially viable product.

EAW’s new Digitally Steerable Array, or DSA™ Series, represents a radical step forward for small to mid-sized permanent installation applications. In essence, the DSA Series simplifies the KF900/PPST technology that permits digital steering and aiming of an array’s output and adapts it to applications where column loudspeakers would typically be specified.

Like a KF900 array, each driver in a DSA Series loudspeaker enjoys its own individual amplification and digital signal processing (DSP). Using EAW’s free DSA Pilot software program, users can vary the vertical coverage pattern from 15° to 120° as well as aim the coverage ±30°.

Unlike the KF900 system, however, all this power is housed in a compact, column speaker-type enclosure that requires no external amplification or processing. Users only need to connect AC power, audio signal, and network communications cables. DSA Pilot recognizes each loudspeaker connected to the network and allows users to control all of them from a single interface.

For additional information, visit EAW's website by clicking here. Click here for a bibliography of relevant papers that have been presented at past AES Conventions or published in JAES. These papers may be purchased from the AES via their website.

Directions to PNTA can be found by clicking here.

Join us at our June meeting when Dave Gunness describes and demonstrates the EAW Digitally Steerable Array.

Our Presenter

After graduating from the University of Wisconsin in 1984, Dave Gunness spent 11 years with Electro-Voice, before joining Eastern Acoustic Works in 1995. He spent his first year with EAW establishing a custom product design system; then undertook the development of the KF900 Series speaker system, as well as its Phased Point Source Technology (PPST) beam-forming technique - technologies which have been incorporated into the recently released DSA-Series digitally steerable arrays. He has designed over 100 unique horns and several-dozen speaker systems for EAW (among them: MH433, BH822, MQ series, LA400, etc.), holds six patents, and currently serves as EAW's Director of Research and Development. He is a member of the Audio Engineering Society.

Elections

We will conduct elections at the June Meeting. The following members are running for election:
Officers
Chair: Dan Mortensen
Vice Chair: David Christensen
Treasurer: Dave Franzwa
Secretary: Gary Louie

Committee
Dr. Melissa Harrison
Aurika Hays
James D. (JJ) Johnston
Mark Rogers
Dave Tosti-Lane
Steve Turnidge
Click here to find biographical sketches for the candidates and additional information about the election process.

Your section officers and the Section Committee members are all volunteers who share a common interest in audio and in making this section happen. According to AES HQ, we are one of the most active sections in the country.

If you are interested in taking your support of our Section's activities to the next level, give some thought to participating either as an Officer or via the Committee. You can find more information here. Please contact the section if you are interested in participating. If you are interested in running in this election, you can self-nominate at the June meeting or contact the section secretary.

Our meetings are open to anyone interested in Audio. AES membership is NOT required for you to attend our meetings.

Last modified 6/1/2004.

Dave Gunness' Papers	
Improved Loudspeaker Array Modeling Author(s): Gunness, David W.; Hoy, William R. Publication: Preprint 5020; Convention 107; August 1999	Abstract: Measurements of directional response are often used to predict interactions in arrays. Implicit in this approach is a simplistic source model with demonstrable limitations. The source models can be greatly improved by incorporating the known physical attributes of the horns. Example models of horn directionality are presented which agree closely with measured data, and which accurately predict array performance.
Improved Loudspeaker Array Modeling·Part 2 Author(s): Gunness, David W.; Hoy, William R. Publication: Preprint 5211; Convention 109; August 2000	Abstract: The modeling technique presented in Part 1 of the work is extended to three-dimensional space through the use of a flat tessellation of the horn mouth. This is made possible by a more complete version of the Kirchhoff·Helmholtz integral, which is applicable to a surface of arbitrary shape. The three-dimensional technique is effective with asymmetrical devices and produces better agreement with measurements at low frequencies and at angles near and beyond 90 degrees off axis.
Loudspeaker Acoustical Field Calculations with Application to Directional Response Measurement Author(s): Gunness, David W.; Mihelich, Ryan J. Publication: Preprint 5210; Convention 109; August 2000	Abstract: The traditional method of predicting the acoustical field produced by an arbitrarily shaped source is a high-frequency, angle-limited reduction of the Kirchhoff·Helmholtz equation. The broadband, broad-angle version of the Kirchhoff·Helmholtz equation is derived and implemented as a numerical method. Acoustical field predictions of real sources developed with this method agree closely with measured data. This agreement even extends to low frequencies and angles near and beyond 90 degrees off of the primary axis. Applications of the technique are described, including a powerful and efficient directional response characterization method.
Loudspeaker Directional Response Measurement Author(s): Gunness, David W. Publication: Preprint 2987; Convention 89; August 1990	Abstract: For a loudspeaker directional response measurement system to qualify as rigorous, several requirements must be met. The system must have adequate acuity in both frequency and space, and the directional information must be properly filtered before sampling. A fully automated pink-noise based system is described which satisfies all of these criteria, and some of the pitfalls of less rigorous approaches are illustrated.

Loudspeaker Manifolds for High-Level Concert Sound Reinforcement Author(s): Carlson, David; Gunness, David Publication: Preprint 2387; Convention 81; October 1986	Abstract: The acoustical manifold is presented as a device for improving the performance of high-level concert sound reinforcement speaker systems. Recently developed bass, midbass, and high-frequency devices are discussed, each of which effectively sums the output of four loudspeakers, producing from the four a single coherent source. The advantages of manifolds over multiple sources are improved audience coverage, reduced polar response lobing, small-frontal area arrays, small transportable enclosures, and in certain cases reduced distortion and increased efficiency.
Loudspeaker Transfer Function Averaging and Interpolation Author(s): Gunness, David W. Publication: Preprint 5440; Convention 111; November 2001	Abstract: Transfer functions of acoustical systems usually include significant phase lag due to propagation delay. When this delay varies from one transfer function to another, basic mathematical operations such as averaging and interpolation produce unusable results. A calculation method is presented which produces much better results, using well-known mathematical operations. Applications of the technique include loudspeaker complex directional response characterization, complex averaging, and DSP filter design for loudspeaker steering.
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Loudspeaker Transfer Function Averaging and Interpolation 753341 bytes (CD aes18) Author(s): Gunness, David W. Publication: Preprint 5440; Convention 111; November 2001	Abstract: Transfer functions of acoustical systems usually include significant phase lag due to propagation delay. When this delay varies from one transfer function to another, basic mathematical operations such as averaging and interpolation produce unusable results. A calculation method is presented which produces much better results, using well-known mathematical operations. Applications of the technique include loudspeaker complex directional response characterization, complex averaging, and DSP filter design for loudspeaker steering.
A Technique for Improved Performance of Multi-Driver Loudspeaker Arrays Using New Measurement, Modeling and Optimization Processes Author(s): Gunness, Dave; Siegel, Stephen Publication: Paper MAL-16; Conference: AES UK Conference: Microphones & Loudspeakers, The Ins & Outs of Audio (MAL); February 1998	Abstract: Large, multi-driver loudspeaker arrays typically suffer from coverage and response anomalies as a direct result of driver interaction. A new technique has been developed to accurately predict and manage these interactions by tailoring the electrical signal to each driver, thereby optimizing the response of the entire array as a single unit.

Papers by other Authors

An Array Filtering Implementation of a Constant-Beam-Width Acoustic Source Volume 38 Number 4 pp. 221-230; March 1990 Authors: Harrell, Jefferson A.; Hixson, Elmer L.	For acoustic arrays (microphones or loudspeakers) the main beam width of the radiation pattern is determined by the spacing between the elements in terms of the wavelength. Thus the beam pattern depends on the frequency. It is demonstrated that by controlling the spacing and the frequency response of a pair of superposed arrays, constant directivity of the main beam lobe over a wide frequency range can be achieved. Large powers can be achieved with relatively low energy densities at each element. A procedure is described by which line arrays of loudspeakers can be superposed with appropriate signal conditioning to control the main beam coverage with frequency. This methodology is extended to an X-shaped planar array for three-dimensional beam control with frequency. Computer models and experiments verify the process for one octave, and models are used to extend the method to wider frequency ranges and more optimal beam synthesis.
Multiple-Beam, Electronically Steered Line-Source Arrays for Sound-Reinforcement Applications Volume 38 Number 4 pp. 237-249; March 1990 Author: Meyer, David G.	Long, narrow rooms with high ceilings are particularly difficult spaces for which to design sound-reinforcement systems. Rooms such as these typically require a combination of short-throw (low Q) loudspeaker components configured as a large, central cluster to provide uniform coverage over the seating space. Unfortunately many such rooms, in particular those of A-frame design, do not have adequate provisions for mounting a large cluster of loudspeaker components in an aesthetically pleasing fashion. A possible alternative for such a venue is described. In place of a central cluster with a large vertical dimension, we propose using an ensemble of horizontally mounted, electronically (phase-delay) steered line-source arrays. Each line source has a Q commensurate with the distance it must throw and the area it must cover. Shading is employed to minimize side-lobe energy as well as to help stabilize coverage patterns. Many desirable properties associated with a central cluster system, such as minimum energy directed toward the speaking position, stable coverage patterns, and preservation of locality of reference, apply to the proposed alternative described. Advances in signal-processing hardware technology help make a

	sound-reinforcement system design based on this approach both technically and economically feasible.
Constant-Beamwidth One-Octave Bandwidth End-Fire Line Array of Loudspeakers Volume 43 Number 7/8 pp. 581-591; June 1995 Authors: Harrell, Jefferson A.; Hixson, Elmer L.	Broad-band acoustic signals, such as voice and music, intended for entertainment require uniform coverage over an audience such that the spectral distribution in the area remains constant. Thus the effective beamwidth of the acoustic source needs to be constant with frequency. The ideal solution to array-beamwidth control would be to create a coverage pattern that has one main lobe with small or no sidelobes, with the entire pattern remaining constant with frequency. Such a method for two-dimensional beamwidth control over an octave is presented. The methodology is most flexible and useful when utilizing digital signal processing, since the element responses could be derived with loudspeaker response corrections as part of the synthesis task. The relatively simple algebra of line array superposition yields constant beamwidths very close to theory (when the driver size is accounted for). Considering the strong match with predicted beam pattern, with 20-dB sidelobe suppression for weightings designed for such, this case is presented as a compelling validation of the basic technique.
Digital Control of Loudspeaker Array Directivity Volume 32 Number 10 pp. 747-754; September 1984 Author: Meyer, David G.	In an earlier paper a mathematical model was described which allows computer simulation of the directivity characteristic of sound reinforcement loudspeaker arrays constructed using "real" drivers, that is, drivers possessing defined amplitude, group delay, and horizontal/vertical polar directivity characteristics. In the present paper a loudspeaker array implementation is presented which allows digital control of the parameters suggested by the mathematical model. Simulation study results illustrate the effects of the various control parameters to which this implementation allows access. Plans for a prototype digitally controlled loudspeaker array are outlined.

Linear Loudspeaker Array for Sound Reinforcement System Using Delay Device Preprint Number: 1522 Convention: 64 (October 1979) Authors: Kido, Ken'iti; Abe, Masato; Ishigame, Masaaki	This paper presents a new loudspeaker system which is useful for sound reinforcement systems. The loudspeaker system is composed of a linear array of loudspeakers driven by the electric current having adequate delay. A convenient method is proposed for the determination of the delay time to be given to the driving current of every loudspeaker of the array. The proposed method is proved to be useful by the numerical computations and experiments. The restrictions imposed on the architectural design due to the setting of the loudspeakers for good sound reinforcement have been removed, as the direction of the array can be arbitrarily chosen.
Multiple Beam, Electronically Steered Line Source Arrays for Sound Reinforcement Applications Preprint Number: 2826 Convention: 87 (September 1989) Author: Meyer, David G.	The fundamental basis of the analysis presented in this section appears in [2], [3], [4], [7], [8], [9], [12], [13], [14], [15], and [16]. Here we extend these results to accommodate multiple, electronically-steered arrays consisting of elements possessing frequency-dependent amplitude, phase, and polar directivity characteristics. Also shown is how the array ensemble directivity pattern is mapped onto the "ear plane" of an audience scaling area. The resulting mathematical model is incorporated into a computer-aided design tool, described in Section 2 of this paper.
Dynamic Amplitude Shading of Electronically Steered Line Source Arrays Preprint Number: 3272 Convention: 92 (February 1992) Authors: Schmidmaier, Richard; Meyer, David G.	Long, narrow rooms with high ceilings typically require loudspeaker components configured as a large, central cluster. The aesthetically most pleasing configuration consists of horizontally mounted, electronically (phase delay) steered line source arrays. A difficult problem, however, is the need to dynamically shade the array such that the sidelobe energy is minimized and the directivity patterns remain stable for all frequencies. A family of optimum shading functions is derived and illustrated with a case study.

Development of a Shaded, Beam Steered Line Array Loudspeaker with Integral Amplification and DSP Processing Preprint Number: 4835 Convention: 105 (August 1998) Authors: Leembruggen, Glenn; Packer, Neil; Goldburg, Bruce; Backstrom, Donald	The principles of shading, electrical tapering and beam-steering of arrayed transducers are now more accessible and economical through application of Digital Signal Processing (DSP) for tapering, delays, and equalization. The authors describe how DSP techniques were applied in a very brief development period to produce a loudspeaker of modest dimensions and reasonable cost incorporating transducers, processing, and amplification electronics within one enclosure, and a PC-based GUI for site adjustment of speaker parameters.
Line Arrays: Theory and Applications Preprint Number: 5304 Convention: 110 (April 2001) Author: Ureda, Mark	Line arrays of loudspeakers are often employed to provide increased directivity, generally in the vertical plane. For improved performance, contemporary line arrays employ specially designed loudspeaker elements to provide a nearly continuous line source. However, even these may have "imperfections" relative to a perfect line source. This paper provides mathematical models to evaluate the directivity response of line sources and to quantify the effects of certain imperfections.
Toward VLSI Implementation of Loudspeaker Array Directivity Control Signal Processing Electronics Preprint Number: 2225 Convention: 78 (April 1985) Author: Meyer, David G.	The work described in this paper concerns development of a specialized signal processing architecture, targeted for VLSI implementation, which would make a loudspeaker array with digital directively control practicable. The solution presented here utilizes a specialized array element signal processor to provide the signal processing capabilities required at each loudspeaker array element. Block diagrams of the array element signal processor and the user control interface are presented along with the overall system organization.
A Directivity Controlled Loudspeaker System with Low Distortion Preprint Number: 4122 Convention: 99 (September 1995)	This paper describes a monitoring loudspeaker system composed of four low-distortion woofers, each disposed in the vertices of a rhombic form, with a midrange horn located at the center of each woofer and a tweeter horn at the center of the midrange horn. This newly developed loudspeaker system has controlled directivity over a wide frequency range, low distortion, and a sharp sound image.

Beam Dithering: Acoustic Feedback Control Using a Modulated-Directivity Loudspeaker Array Preprint Number: 3384 Convention: 93 (September 1992) Authors: Elko, Gary; Goodwin, Michael	The problem of acoustic feedback in public address systems has been with us since the advent of the electronic amplifier (and, of course, the loudspeaker and microphone). Various schemes to ameliorate the problem have been proposed: filters, frequency shifting, directional transducers, etc. One possible technique that has not been explored in the past is to allow the loudspeaker directivity to be modulated in time. The modal excitation of the room can be time-varied, thereby averaging the response at a point. Such directivity dithering reduces the peak response in the system transfer function (between loudspeaker and microphone). We have constructed a three-octave digital beamforming loudspeaker array whose directivity can easily be modified as a function of time. The presentation will discuss the theory of the use of the modulated directivity and report on the design and construction of the beamforming loudspeaker array.
A Digital Control Unit for Loudspeaker Arrays Preprint Number: 3836 Convention: 96 (January 1994) Authors: de Vries, G.; van Beuningen, G.	Amplitude shading and fase shading are readily used methods to gain control over the directivity of a loudspeaker array. The results are limited by the analogue techniques that are employed. Implementation of digital signal processing will overcome these disadvantages resulting in a nearly compromiseless control. The use of FIR-filters for this purpose will be discussed, as well as techniques to compensate for the effect of temperature and airflow on the alignment of the individual array elements. Application of this approach has lead to the design of a DSP-based modular array control unit.
Electronically controlled loudspeaker arrays without side lobes. Preprint Number: 5322 Convention: 110 (April 2001) Author: van der Werff, Johan	It is possible to arrange loudspeakers in such a way that only one lobe emits from the array. This lobe can have an arbitrary beam width and to a certain extend an arbitrary beam shape. Because of this control over the beam, narrow beam widths can be made where wave fronts travel coherently 200 meters or more. It is possible now to cover an area below and in front of the array from almost zero to 200 meters with even direct-sound distribution of ± 3 dB, where the frequency response is only dependant on the transducer used and the air absorption. This eliminates the coloration-effects due to side or grating lobes.

Smart directional and diffuse digital loudspeaker arrays Preprint Number: 5362 Convention: 110 (April 2001) Author: Hawksford, Malcolm	A theory of smart loudspeaker arrays is described where a modified Fourier technique yields complex filter coefficients to determine the broadband radiation characteristics of a uniform array of micro drive units. Beam width and direction are individually programmable over a 180-degree arc, where multiple agile and steerable beams carrying dissimilar signals can be accommodated. A novel method of diffuse filter design is also presented that endows the directional array with diffuse radiation properties.

Audio Engineering Society - Pacific Northwest Section

AES PNW Section

2004 Election of Officers

Once per year, our section is directed by its bylaws to elect a new set of officers for the coming meeting year (September thru June). There are four section officers, each serves for a term of one year. There are no term limits. The Section Executive Committee has twelve seats; committee members serve for a two-year term. Six of the seats elects in odd numbered years; six of the seats elects in even numbered years. Elections take place at the June section meeting. A quorum of the members in the region must be present before the election can be held. This year, a quorum is 15 members.

Accordingly then, it is time to conduct our yearly election. Any member of the society may participate in this process by voting and/or by running for office. Typically the current officers and committee members draw up the ballot, however any member may 'throw their hat into the ring' by announcing their candidacy to one of the section officers or committee members. Furthermore, any member may self-nominate at any time after the election has been announced up until the moment of the election.

Those persons running for one of the committee positions will be elected on the basis of total vote count; the open seats going to the six largest number of votes.

Biographical sketches can be found after the List of Candidates.

Officers
Chair: Dan Mortensen
Vice Chair: David Christensen
Treasurer: Dave Franzwa
Secretary: Gary Louie

Committee
Dr. Melissa Harrison
Aurika Hays
James D. (JJ) Johnston
Mark Rogers
Dave Tosti-Lane
Steve Turnidge

Biographical Sketches

Chair - Dan Mortensen

> Dan is President of Dansound Inc., which specializes in live sound reinforcement and is Washington State dealer for Meyer Sound Laboratories, among other dealerships. He is the current AES PNW Treasurer, past Section Chair, Vice-Chair, and Committee member. He is

the current Executive Director of the Washington Association of Production Services. After more than 10 years, Dan continues to find that serving on the AES PNW Committee is one of his favorite things.

Vice Chair - David Christiansen

is on the faculty of the Audio Production Department at the Art Institute of Seattle. He is the advisor for the AIS student section. as well as an independent producer/engineer, focusing on classical music. Svensson Plays Bach is his most recent release. David graduated from USC with a BA in Film Production. David is currently a member of the PNW Section Committee.

Secretary - Gary Louie

Gary has been the recording technician for the University of Washington School of Music since 1979, previously earning his BSEE at the UW. He has served as AES PNW Section Chair, Vice Chair, Committee, and most recently, Secretary since 1993. He is the co-author of *The Audio Dictionary* (3rd edition).

Treasurer - Dave Franzwa

Dave has been employed in the audio manufacturing industry in the Seattle area since 1979, having worked at TAPCO, Carver, Spectral (briefly), and currently at Mackie Designs where he is Technical Documentation Manager. He graduated from Cogswell College North in 1995 with a Bachelor of Science degree in Electronics Engineering Technology. He enjoys playing music and working with audio and sound reinforcement equipment in his spare time. Dave is currently a member of the PNW Section Committee.

Committee Positions

Dr. Melissa Harrison PhD.

Dr. Melissa Harrison is an independent technical consultant. She has a PhD in applied mathematics from the University of Maryland and is interested in signal processing, particularly for audio signals.

Dr. Harrison has worked in various areas of computing and mathematics for over 20 years including Fourier analysis, irregular sampling of bandlimited signals, robotic vision & route programming, system administration, and technical typesetting. She has over 10 years experience teaching mathematics at the university level. Dr. Harrison was a presenter at the 2004 AES Section meeting, "From Hear to Eternity."

Aurika Hays

Aurika is an independent audio software engineer. She currently serves on the Section Committee as Queen of the Database. In this capacity, she maintains the database used for monthly mailings to members and other interested parties. Aurika became involved with the AES while attending Stuyvesant High School in New York City. She graduated from Stanford University with an individually designed Bachelor of Science degree in Audio and Acoustical Engineering. At the University of Miami, she was the first woman to earn a Master of Science in Music Engineering Technology. Her master's thesis was titled "Pitch Recognition and Intonation Error Quantification of Discrete Violin Tones." She spent 3 years at RealNetworks in Seattle, WA where she worked as an Audio Software Engineer, an Audio Quality Specialist, and as a Program Manager in the Tools and Authoring Products Group.

James D. (JJ) Johnston

James received the BSEE and MSEE degrees from Carnegie-Mellon University, Pittsburgh, PA in 1975 and 1976 respectively.

JJ worked 26 years for AT&T Bell Labs and its successor AT&T Labs Research. He temporarily retired in 2002. He was one of the first investigators in the field of perceptual audio coding, one of the inventors and standardizers of MPEG 1/2 audio Layer 3 and MPEG-2 AAC, as well as the At&T bell Labs or AT&T Labs-Research PXFM (perceptual transform coding0 and PAC (perceptual audio coding0 and the ASPEC algorithm that provided the best audio quality in the MPEG-1 audio tests.

Most recently he has been working in the area of auditory perception of soundfields, ways to capture soundfield cues and represent them, and ways to expand the limited sense of realism available in standard audio playback for both captured and synthetic performances. He is currently employed by Microsoft.

Mr. Johnston is an IEEE Fellow, an AES Fellow, a NJ Inventor of the Year, an AT&T Technical Medalist and Standards Awardee, and a co-recipient of the IEEE Donald Fink Paper Award. Mr. Johnston was a presenter at the 2004 AES Section Meeting, "From Hear to Eternity."

Mark Rogers

Mark is Director of the AV Department at the Greenbusch Group, a Seattle engineering consulting firm. He is a designer of audio/visual systems, including sound reinforcement, audio reproduction, video projection and displays, videoconferencing and audioconferencing, and related control systems. Typical projects include corporate boardrooms, convention centers, universities and hospitals. He has designed and installed AV for 30 years, and also teaches classes and seminars on AV technology. He is a registered Professional Engineer (Idaho) and earned his BSEE at the University of Idaho. He is the current Vice Chair of this section as well as a past member of the PNW AES Section Committee. Mark has presented several topics to the section.

Dave Tosti-Lane

Dave is a theatrical sound designer, lighting designer and technical director, and is a founding faculty member and Chair of the Performance Production Department at Cornish College of the Arts in Seattle. Dave holds a BS degree in management and an MFA degree in Lighting Design and Technical Direction from Virginia Tech, but his real audio education began when handed his first tape recorder at age 10. He is Chair of AES Standards Committee Working Group SC-05-03 on Audio Connectors, Vice-Commissioner for Education in the Sound Design Commission of the US Institute for Theatre Technology, and the associate editor for sound of TD&T, the journal of the USITT. He is currently serving as the Section Chair. He is a past Vice Chair and Committee member of the PNW AES Section.

Steve Turnidge

Steve is a past presenter from our September 2003 meeting, "What's All This Mastering Stuff, Anyway?"

Steve is a mastering engineer at Ultraviolet Studios. His specialties are there are CD Mastering, Vinyl (phonograph record) restoration, Noise Reduction and Audio Processing. In addition to mastering, Steve is an accomplished printed circuit board designer. Prior to starting Ultraviolet Studios Steve held positions at Rane Corporation, Pavo Corporation, and Digital Harmony. He was recently appointed to the Advisory Committee for the music technology program at Shoreline Community College where he also taught a Music Technology/Audio Recording class.

In the early 1980's, he formed the Ultraviolet Catastrophe, a pre-midi synth-pop band. He is a member of the AES and a full voting member of the Recording Acadamy.

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