

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

GOOGLE LLC,

Petitioner

v.

JAWBONE INNOVATIONS, LLC,

Patent Owner

Case IPR2022-01059
U.S. Patent No. 10,779,080

Declaration of Shauna L. Wiest Regarding *Elko IEEE Workshop*

I, Shauna L. Wiest, state and declare as follows:

I. Introduction

1. I have prepared this Declaration in connection with Google LLC's ("Petitioner") Petition for *Inter Partes* Review of U.S. Patent No. 10,779,080, which I understand will be filed concurrently with this Declaration.

2. I am currently a senior research analyst with the Research & Information Services team at Finnegan, Henderson, Farabow, Garrett & Dunner, LLP, located at 901 New York Avenue, NW, Washington, DC 20001-4413.

3. I am over eighteen years of age, and I am competent to make this Declaration. I make this Declaration based upon my own personal knowledge, and my professional knowledge of library science practices.

4. I earned a Master of Science in Library Science degree from the University of North Carolina at Chapel Hill in 1999, and a Bachelor of Arts in Political Science degree from the University of California at San Diego in 1989. I have worked as a law librarian for over twenty years. I have been employed in the Research & Information Services Department at Finnegan, Henderson, Farabow, Garrett & Dunner, LLP since 2021. Before that, from 2000-2015, I was employed as a Law Librarian at Stoel Rives LLP, and from 2015-2016, I was employed as a Competitive Intelligence Specialist for Nossaman LLP.

II. Standard Library Practice for Receiving, Cataloging, and Making Materials, including Conference Publications, Publicly Available

5. I have knowledge of and experience with standard library practices regarding the receipt, cataloging, shelving, and making materials, including conference publications, available to the public. I am fully familiar with and have knowledge of and experience with the Machine-Readable Cataloging (MARC) system, an industry-wide standard that libraries use to catalog materials.

6. The MARC system was developed during the 1960s to standardize bibliographic catalog records so they could be read by computers and shared among libraries. By the mid-1970s, MARC had become the international standard for cataloging bibliographic materials and is still used today. Many libraries provide public access to their MARC records via the Internet and/or their electronic cataloging systems at the library. In a MARC record, each field provides specific information about the cataloged item, including how materials are held and made available to the public.

III. Monograph versus Serials Cataloguing of Conference Proceedings

7. Conference proceedings often consist of collections of papers presented at a meeting, symposium or conference held during a specific time-period in a designated geographic place. Although some ongoing conference publications are catalogued as serials (if a conference publication exhibits

evidence of seriality after the first issue and it does not contain a distinctive title or series number), cataloguing a conference publication as a monograph can provide a way for the public to easily access the conference resource, and its papers, by searching for a unique title and other data points based on descriptive elements of the resource. Monographs are efficient for public searching and offer the public an easier way to discover what an institution holds, while changes to the record are not an issue.

IV. MARC Records

8. The MARC record system uses a specific three-digit numeric code (“field tag”) (from 001-999) to identify each field in a catalog record. For example, field tag 008 provides the six-digit date the item was catalogued (*Date entered on file*). The first six characters of field tag 008 are always in the “YYMMDD” format. Descriptions and definitions of all of the character positions of field tag 008 are outlined here: <https://www.loc.gov/marc/bibliographic/bd008a.html>

9. MARC field tag 020 provides the International Standard Book Number (ISBN), a unique identification number assigned to a monographic publication. MARC field tag 245 identifies the full title statement for the work and field tag 264 identifies the place of publication, name of publisher, and copyright date of the publication.

10. Based on standard library practice, when a library receives an item, it stamps (and/or labels) the item with the library name, barcode, often with a date that is within a few days or weeks of receipt. Next, the library will catalog the item within a matter of a few days or weeks of receiving it.

11. Generally, after an item is cataloged, the public may access the item by searching a library catalog, browsing the library shelves, and either requesting or electronically accessing the item from the library. Standard library practice is to make the item available to the public within a few days or weeks of cataloging it.

V. Print Publication Availability and MARC Record for *Elko IEEE Workshop*

12. This Declaration relates to the dates of receipt and public availability of the following: Elko, Gary W. and Pong, Anh-Tho Nguyen, “*A Simple Adaptive First-Order Differential Microphone*,” presented during *Session 7: Transducers* as Paper 3 on October 17, 1995, between 4:20pm-6:00pm at the 1995 IEEE ASSP Workshop On Applications of Signal Processing to Audio and Acoustics, October 15-18, Mohonk Mountain House, New Paltz, New York (ISBN 0780330641) (“*Elko IEEE Workshop*”). Exhibit 1027 to the concurrently filed Petition is a true and accurate copy of *Elko IEEE Workshop*, which was downloaded from <https://ieeexplore.ieee.org/document/482983> on May 23, 2022.

13. **Appendix A** to this declaration is a true and accurate copy of the cover, verso, front matter, table of contents, and library stamp and call number from the University of Wisconsin-Madison Library for its copy of *Elko IEEE Workshop*. Appendix A also includes the pages comprising Paper 3, the *Elko IEEE Workshop* article. The library stamp on the verso page indicates that *Elko IEEE Workshop* is held at the Kurt F. Wendt Library at the University of Wisconsin-Madison. The call number on the front cover of Appendix A indicates that *Elko IEEE Workshop* is located at Call Number TK 7881.4 I34 1995.

14. **Appendix B** to this declaration is a true and accurate copy of The University of Wisconsin-Madison MARC record for its holdings of the 1995 IEEE ASSP Workshop On Applications of Signal Processing to Audio and Acoustics, including Paper 3 (*Elko IEEE Workshop*), which was downloaded from <https://search.library.wisc.edu/catalog/999780556202121/staff> on May 18, 2022.

15. The University of Wisconsin-Madison MARC record for *Elko IEEE Workshop* confirms the fixed data elements of MARC field tag 008 as 960319s1995 njua 100 0 eng d. As discussed above, the first six characters “960319” are in typical “YYMMDD” format and indicate that the 1995 IEEE ASSP Workshop On Applications of Signal Processing to Audio and Acoustics, including Paper 3 (*Elko IEEE Workshop*), was cataloged by the University of Wisconsin-Madison on March 19, 1996, with a single known publication date of 1995.

16. Based on standard library practices the 1995 IEEE ASSP Workshop On Applications of Signal Processing to Audio and Acoustics, including Paper 3 (*Elko IEEE Workshop*), would have been made publicly available within a few days or weeks of March 19, 1996. Members of the public would have had the opportunity to access *Elko IEEE Workshop* by browsing the University of Wisconsin-Madison Library shelves, requesting the item from a staff member, or by searching the Library's catalog within a few days or weeks of March 19, 1996.

17. The University of Wisconsin-Madison MARC record field tag 020 denotes the unique International Standard Book Numbers (ISBN) assigned to *Elko IEEE Workshop* as 0780330641. MARC record field tag 245 denotes the title and statement of responsibility for the work as \$a1995 IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics :\$bOctober 15-18, Mohonk Mountain House, New Paltz, New York. MARC record field tag 264 identifies the place of publication, publisher, and copyright date as \$aPiscataway, NJ :\$bIEEE Service Center,\$c[1995].

18. **Appendix C** to this declaration is a true and accurate copy of the University of Wisconsin-Madison Library public catalog record for its copy of the 1995 IEEE ASSP Workshop On Applications of Signal Processing to Audio and Acoustics, including Paper 3 (*Elko IEEE Workshop*), which was downloaded from <https://search.library.wisc.edu/catalog/999780556202121> on May 18, 2022.

19. The University of Wisconsin-Madison Library's public catalog record (**Appendix C**) for its copy of the 1995 IEEE ASSP Workshop On Applications of Signal Processing to Audio and Acoustics, including Paper 3 (*Elko IEEE Workshop*), sets forth the public holdings and onsite location information for members of the interested public seeking a physical copy of the resource. The public catalog record indicates that *Elko IEEE Workshop* should be requested from the Verona Shelving Facility at Call Number TK7881.4 I34 1995.

20. Based on the evidence presented above, it is my professional opinion that Exhibit 1027 is an authentic document, which would have been made publicly available and publicly accessible within a few days or weeks of March 19, 1996.

VI. Conclusion

21. In signing this Declaration, I understand it will be filed as evidence in a contested case before the Patent Trial and Appeal Board of the United States Patent and Trademark Office. I understand I may be subject to cross-examination in this case and that cross-examination will take place within the United States. If cross-examination is required of me, I will appear for cross-examination within the United States during the time allotted for cross-examination.

22. I declare that all statements made herein of my knowledge are true, that all statements made on information and belief are believed to be true, and that these

Declaration of Shauna L. Wiest

statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

Executed on May 25, 2022.

A handwritten signature in black ink, appearing to read "Shauna L. Wiest", written over a horizontal blue line.

Shauna L. Wiest

Appendix A

**October 15-18, Mohonk Mountain House
New Paltz, New York**

1995 IEEE ASSP WORKSHOP ON APPLICATIONS OF SIGNAL PROCESSING TO AUDIO AND ACOUSTICS

**October 15-18, Mohonk Mountain House,
New Paltz, New York**

Kurt F. Wendt Library
University of Wisconsin - Madison
240 N. Randall Avenue
Madison, WI 53706-1688

1995 IEEE ASSP WORKSHOP
ON APPLICATIONS OF SIGNAL PROCESSING
TO AUDIO AND ACOUSTICS

October 15-18, Mohawk Mountain House
New Paltz, New York

Copyright and Reprint Permission: Abstracting is permitted with credit to the source. Libraries are permitted to photocopy beyond the limit of U.S. copyright law for private use of patrons those articles in this volume that carry a code at the bottom of the first page, provided the per-copy fee indicated in the code is paid through the Copyright Clearance Center, 222 Rosewood Drive, Danvers MA 01923. For other copying, reprint or republication permission, write to IEEE Copyright Manager, IEEE Service Center, 445 Hoes Lane, P.O. Box 1331, Piscataway, NJ 08855-1331. All rights reserved. Copyright © 1990 by the Institute of Electrical and Electronics Engineers, Inc.

IEEE Catalog Number:	95TH8144	
ISBN	0-7803-3064-1	(Softbound Edition)
ISBN	0-7803-3065-X	(Microfiche Edition)

WORKSHOP PROGRAM

SUNDAY

Invited Keynote Address

8:00 p.m. - 9:00 p.m., October 15

Use of Psychological Data in Building ASA Models, Albert S. Bregman, *McGill University, Department of Psychology, 1205 Docteur Penfield Avenue, Montreal, Quebec H3A 1B1, CANADA*

MONDAY

SESSION 1: HEARING AIDS

8:30 a.m. - 10:30 a.m., October 16

Session Chair: Jont B. Allen, *AT&T Bell Laboratories.*

1. Separation of Multiple Signals in Hearing Aids by Output Decorrelation and Time-Delay Estimation Pascal Bamford, Nishan Canagarajah, *University of Bristol, Centre for Communication Research, University Walk, Bristol, BS8 1TR, UNITED KINGDOM*

2. Microphone Array Calibration for Robust Adaptive Processing Michael W. Hoffman, *University of Nebraska-Lincoln, Electrical Engineering, 209N WSEC 0511, Lincoln, NE 68588-0511, USA*

3. An Evaluation of Hearing-Aid Array Processing James M. Kates, *City University of New York, Center for Research in Speech and Hearing Sciences, Graduate Center, Room 901, 33 West 42nd Street, New York, NY 10036, USA*

4. Wavelet Transform and Frequency Domain Kurtosis: Application to Assessment of Hearing Hazard from Noise Exposure Sheau-Fang Lei, Roger P. Hamernik, *State University of New York at Plattsburgh, Auditory Research Laboratory,*

Beaumont Hall 107, Plattsburgh, NY 12901, USA

5. Fixed Filter Implementation of Feedback Cancellation for In-the-Ear Hearing Aids Brian D. Woodruff, David A. Preeves, *Argosy Electronics, 10300 West 70th Street, Eden Prairie, MN 55344, USA*

6. Two Microphone Nonlinear Frequency Domain Beamformer for Hearing Aid Noise Reduction Eric Lindemann, *Audiologic, 6655 Lookout Road, Suite 200, Boulder, CO 80301, USA*

SESSION 2: AUDITORY SCENE ANALYSIS

10:45 a.m. - 12:25 p.m., October 16

Session Chair: Richard Duda, *San Jose State University.*

1. Binaural Modeling and Auditory Scene Analysis Markus Bodden, *Ruhr-Universitt Bochum, Lehrstuhl AEA, IC1/42, Bochum, D-44780, GERMANY*

2. A Neural Oscillator Model of Primitive Audio Grouping Guy J. Brown, Martin Cooke, *University of Sheffield, Department of Computer Science, Regent Court, 211 Portobello Street, Sheffield, S1 4DP, UNITED KINGDOM*

3. Auditory Segregation of Concurrent Signals: An Operational Definition Pierre L. Divenyi, *Veterans Affairs Medical Center, Speech and Hearing Research, 150 Muir Road, Martinez, CA 94553, USA*

4. Underconstrained Stochastic Representations for Top-Down Computational Auditory Scene Analysis Daniel P. W. Ellis, *MIT Media Lab, Machine Listening Group, Room E15-401B, 20 Ames Street, Cambridge, MA 02139, USA*

5. Instantaneous and Frequency-Warped Techniques for Source Separation and Signal

Parameterization Avery L. Wang, *Chromatic Research, Inc., 800A East Middlefield Road, Mountain View, CA 94043-4030, USA*

SESSION 3: ACTIVE NOISE CONTROL

5:00 p.m. - 6:00 p.m., October 16

Session Chair: Steven J. Elliott, *University of Southampton, Institute of Sound and Vibration Research.*

1. Adaptive Minimisation of the Maximum Error Signal in an Active Control System A. Gonzalez, *Universidad Politecnica de Valencia, Dept. Comunicaciones, Camino de Vera s/n, Valencia, 46071, SPAIN, S. J. Elliott, University of Southampton, Institute of Sound and Vibration Research, Southampton, Hampshire, SO17 1BJ, UNITED KINGDOM,*

2. A Multidimensional Householder QR Algorithm for Transducer Placement in Active Noise Control Systems Larry P. Heck, *SRI International, 301-58, 333 Ravenswood Avenue, Menlo Park, CA 94025, USA*

3. Using A Fast-Recursive-Least-Squared Algorithm In a Feedback-Controller Thomas Auspitzer, S. J. Elliott, *University of Southampton, Institute of Sound and Vibration Research, Southampton, Hampshire, SO17 1BJ, UNITED KINGDOM D. Guicking, Universitaet Goettingen, Drittes Physikalisches Institut, Buergerstrasse 42-44, Goettingen, 37073, GERMANY*

SESSION 4: SPATIAL PERCEPTION AND PROCESSING

7:30 p.m. - 10:10 p.m., October 16

Session Chair: Fred Wightman, *Univ. of Wisconsin-Madison.*

1. Some Research Issues in Spatial Hearing Jack M. Loomis, *University of California Santa Barbara, Department of Psychology, Santa Barbara, CA 93106-9660, USA*

2. Some Considerations for the Design of Auditory Displays Robert H. Gilkey, *Wright State University, Department of Psychology, Dayton, OH 45435, USA*

3. On the Discriminability of Virtual and Real

Sound Sources Pavel Zahorik, Fred Wightman, Doris Kistler, *University of Wisconsin-Madison, Waisman Center, Room 323, 1500 Highland Avenue, Madison, WI 53705-2280, USA*

4. The Relative Contribution of Interaural Time and Magnitude Cues to Dynamic Sound Localization Elizabeth M. Wenzel, *NASA - Ames Research Center, Flight Management & Human Factor Division, Mail Stop 262-2, Moffett Field, CA 94035-1000, USA*

5. On the Minimum-Phase Approximation of Head-Related Transfer Functions A. Kulkarni, S. K. Isabelle, H. S. Colburn, *Boston University, Department of Biomedical Engineering, 44 Cummington Street, Boston, MA 02215, USA*

6. A Comparison of Head Related Transfer Function Interpolation Methods Paul R. Runkle, Michael A. Blommer, Gregory H. Wakefield, *University of Michigan, Department of Electrical Engineering & Computer Science, 1301 Beal, Ann Arbor, MI 48109-2122, USA*

7. A Spherical Basis Function Neural Network for Pole-Zero Modeling of Head-Related Transfer Functions Rick L. Jenison, *University of Wisconsin-Madison, Department of Psychology, 1202 West Johnson Street, Madison, WI 53706, USA*

8. Estimating Azimuth and Elevation from Interaural Difference Keith D. Martin, *MIT Media Lab, Perceptual Computing Section, Room E15-401, 20 Ames Street, Cambridge, MA 02139, USA*

TUESDAY

SESSION 5: ACOUSTIC ECHO CANCELLATION

8:00 a.m. - 10:40 a.m., October 17

Session Chair: Markus Rupp, *University of California - Santa Barbara.*

1. Limitations of Handsfree Acoustic Echo Cancellers Due to Nonlinear Loudspeaker Distortion and Enclosure Vibration Effects A.N. Birkett, R.A. Goubran, *Carleton University, Dept. of Systems and Computer Engineering, 1125*

Colonel By Drive, Ottawa, K1S 5B6, CANADA

2. Mean-Square Error Calculations for the Subband Adaptive Filter System Phillip De Leon II, Delores M. Etter, *University of Colorado at Boulder, Department of Electrical Engineering, Boulder, CO 80309-0425, USA*

3. Perceptually Optimum Adaptive Filter Tap Profiles for Subband Acoustic Echo Cancellers Eric J. Diethorn, *AT&T Bell Laboratories and AT&T Custom Electronic Systems, 67 Whippany Road, Room 15F-353, Whippany, NJ 07981, USA*

4. Parallel Processing of the Matched-Filter Array for Sound Capture Ea-Ee Jan, James L. Flanagan, *Rutgers University, CAIP Center, Busch Campus, Piscataway, NJ 08855-1390, USA*

5. Image Model for Computer Simulation of Sound Wave Behavior in an Enclosure Ea-Ee Jan, James L. Flanagan, *Rutgers University, CAIP Center, Busch Campus, Piscataway, NJ 08855-1390, USA*

6. Two Variants of the FxLMS Algorithm Markus Rupp, Ali H. Sayed, *University of California - Santa Barbara, Electrical and Computer Engineering, Santa Barbara, CA 93106-9560, USA*

7. Exponentially Decaying Time-Recursive Blind Deconvolution Algorithm for Speech Dereverberation Ping-Ya Zhao, James P. Reilly, *McMaster University, Communications Research Lab, 1280 Main Street West, Hamilton, Ontario L8S 4K1, CANADA*

8. Simulation of Teleconferencing Environments C. Thompson, K. Chandra, V. Mehta, *University of Massachusetts, Lowell, Center for Advanced Computation and Telecommunications, Lowell, MA 01854, USA*

SESSION 6: MUSIC RESTORATION AND ANALYSIS

10:55 a.m. - 12:35 p.m., October 17

Session Chair: Peter J.W. Rayner, *University of Cambridge.*

1. Some Methods for Detection and Interpolation of Impulsive Distortions in Old Audio Recordings Andrzej Czyzewski, *Technical University of Gdansk, School of Electronics,*

Sound Engineering Department, Narutowicza 11/12, Gdansk, 80-952, POLAND

2. Robust Noise Modelling With Application to Audio Restoration Simon J. Godsill, Peter J.W. Rayner, *University of Cambridge, Signal Processing and Communications Laboratory, Trumpington Street, Cambridge, CB2 1PZ, UNITED KINGDOM*

3. Removing Preechos from Audio Recordings Jean Laroche, *Ecole Nationale Supérieure des Telecommunications, Departement Signal, 46 Rue Barrault, Paris Cedex 13, 75634, FRANCE*

4. A Multiresolution Audio Restoration Algorithm Hugh R. R. Scott, Roland Wilson, *Warwick University, Dept. of Computer Science, Coventry, CV4 7AL, UNITED KINGDOM*

5. Multi-channel Blind Signal Separation by Decorrelation Dominic C. B. Chan, Peter J. W. Rayner, Simon J. Godsill, *University of Cambridge, Signal Processing and Communications Laboratory, Trumpington Street, Cambridge, CB2 1PZ, UNITED KINGDOM*

SESSION 7: TRANSDUCERS

4:20 p.m. - 6:00 p.m., October 17

Session Chair: Gary Elko, *AT&T Bell Laboratories.*

1. Predicting the Quality of Sound Pickup Using Microphone Arrays James G. Ryan, *National Research Council, Acoustics and Signal Processing, Institute for Microstructural Sciences, Montreal Road, Ottawa, Ontario K1A 0R6, CANADA*

2. Second Order Volterra Inverses for Compensation of Loudspeaker Nonlinearity Hans Schurer, Cornelius H. Slump, Otto E. Herrmann, *University of Twente, Dept. of Electrical Engineering, Laboratory for Network Theory and VLSI Design, P.O. Box 217, 7500 AE Enschede, the NETHERLANDS*

3. A Simple Adaptive First-Order Differential Microphone Gary W. Elko, Anh-Tho Nguyen Pong, *AT&T Bell Laboratories, Acoustics Research Department, Room 2D-342, 600 Mountain Avenue, Murray Hill, NJ 07974, USA*

4. Multiple Beam Broadband Beamforming:

Filter Design and Real-Time Implementation
Stefaan Van Gerven, Dirk Van Compernelle, Piet Wauters, Wim Verstraeten, Koen Eneman, Kristof Delaet, K. U. Leuven, Dept. of Electrical Engineering - ESAT, K. Mercierlaan 94, Heverlee, 3001, BELGIUM

5. A New Method of Tracking Talker Location for Microphone Arrays in Near Field Ming Zhang, M. H. Er, Nanyang Technological University, Centre for Signal Processing, School of Electrical & Electronic Engineering, Nanyang Avenue, 2263, SINGAPORE

SESSION 8: HIGH-QUALITY AUDIO CODING, AUDIO PERCEPTION

7:30 p.m. - 9:50 p.m., October 17

Session Chair: Marina Bosi, *Dolby Laboratories*.

1. A New Orthonormal Wavelet Packet Decomposition for Audio Coding Using Frequency-Varying Modulated Lapped Transforms Marcus Purat, Peter Noll, *Technische Universitaet Berlin, Institut fur Fernmeldetechnik, Einsteinufer 25, Berlin, D-10587, GERMANY*

2. Subband Coding of Audio Using a Recursively Indexed Quantizer Yang-Jeng Chen, Robert C. Maher, *University of Nebraska-Lincoln, Department of Electrical Engineering, 209N WSEC, Lincoln, NE 68588-0511, USA*

3. Optimizing the Synthesis Filter Bank in Audio Coding for Minimum Distortion Using a Frequency Weighted Psychoacoustic Criterion Karine Gosse, Olivier Pothier, Pierre Duhamel, *ENST/SIG, 46 rue Barrault, Paris Cedex 13, 75634, FRANCE*

4. Evaluating a Model of Auditory Masking for Applications in Audio Coding Blas Espinoza-Varas, *University of Oklahoma Health Sciences Center, Department of Communications Sciences and Disorders, Oklahoma City, OK 73190, USA*, Surendra V. Cherukuri, *University of Oklahoma, Department of Electrical Engineering, Norman, OK 73019, USA*

5. An Optimal Auditory Filter Irino Toshio, *NTT Basic Research Laboratories, 3-1, Morinosato Wakamiya, Atsugi-shi, Kanagawa, 243-01, JAPAN*

6. The Bark Bilinear Transform Julius O. Smith

III, *Stanford University, CCRMA, Stanford, CA 94305, USA* Jonathan S. Abel, *Abel Innovations, 184 Tennyson Avenue, Palo Alto, CA 94301, USA*

7. Observations on Auditory Excitation and Masking Patterns Stephen Voran, *U.S. Department of Commerce, Institute for Telecommunication Sciences, NTIA/ITS.N3, 325 Broadway, Boulder, CO 80303, USA*

WEDNESDAY

SESSION 9a: COMPUTER MUSIC

8:00 a.m. - 10:20 a.m., October 18

Session Chair: Jean Laroche, *ENST*.

1. Regularized Estimation of Cepstrum Envelope from Discrete Frequency Points Olivier Cappe, Jean Laroche, Eric Moulines, *Ecole Nationale Supérieure des Telecommunications, Departement Signal, 46 Rue Barrault, Paris Cedex 13, 75634, FRANCE*

2. Filter Morphing for Audio Signal Processing Yinong Ding, *Texas Instruments Corporate R&D, P.O.Box 655474, MS 446, Dallas, TX 75265, USA*, Dave Rossum, *The Joint E-mu/Creative Tech Center, 1600 Green Hills Road, P.O. Box 660015, Scotts Valley, CA 95067-0015, USA*

3. Phase-locked Vocodor Miller Puckette, *University of California, San Diego, Department of Music (0326), La Jolla, CA 92093-0326, USA*

4. Overview of the Commuted Piano Synthesis Technique Julius O. Smith III, Scott A. Van Duyne, *Stanford University, CCRMA, Stanford, CA 94305, USA*

5. Real-time Musical Applications using Frequency Domain Signal Processing Zack Settel, *IRCAM, Lysis, 1 place Igor Stravinsky, Paris, 75004, FRANCE*, Cort Lippe, *University of Buffalo, Music Department, 222 Baird Hall, Box 604700, Buffalo, NY 14260-4700, USA*

6. The Tetrahedral Digital Waveguide Mesh Scott A. Van Duyne, Julius O. Smith III, *Stanford University, CCRMA, Stanford, CA 94305, USA*

7. Automated Analysis and Computationally Efficient Synthesis of Acoustic Guitar Strings and Body Kevin Bradley, Mu-huo Cheng, Virginia L. Stonick, *Carnegie Mellon University, Electrical and Computer Engineering, 5000 Forbes Avenue, Pittsburgh, PA 15213, USA*

Atlantic University, Department of Electrical Engineering, Boca Raton, FL 33431, USA

8. The Recreation of a Castrato Voice, Farinelli's Voice Phillipe Depalle, Guillermo Garcia, Xavier Rodet, *IRCAM, 31 rue St. Merri, Paris, 75004, FRANCE*

SESSION 9b: ROOM ACOUSTICS FOR MUSIC PROCESSING

10:35 a.m. - 11:15 a.m., October 18
Session Chair: Jean Laroche, *ENST*

1. A Numerical Investigation of the Invertibility of Room Transfer Functions William Putnam, Julius O. Smith III, *Stanford University, CCRMA, Stanford, CA 94305, USA*, Davide Rocchesso, *Centro di Sonologia Computazionale, Universita' degli Studi di Padova, Padova, 35123, ITALY*

2. Simulation of Room Transmission Functions Using a Triangular Beam Tracing Computer Model Emmanuel De Geest, *Numerical Integration Technologies N.V., Ambachtenlaan 11A, Leuven, B-3001, BELGIUM*, Ralph Garcea, *LMS N. America, 5455 Corporate Drive, Suite 303, Troy, MI 48098, USA*

SESSION 10: TIME AND FREQUENCY ANALYSIS

11:30 a.m. - 12:30 p.m., October 18
Session Chair: John Princen, *Silicon Graphics.*

1. Time Frequency Analysis with Low Numerical Complexity Using a t-f-Mapping of the DFT Magnitude Robert R. Hoeldrich, *Institute of Electronic Music Graz, Jakominiinstrasse 3-5, A-8010, Graz, AUSTRIA*

2. Time-Scale Modification with Inconsistent Constraints T. F. Quatieri, *MIT Lincoln Laboratory, 244 Wood Street, Lexington, MA 02173, USA*, T. E. Hanna, *NSMRL, Box 900, Naval Submarine Base New London, Groton, CT, USA*

3. Adaptive Filters Based on the Best Matched Wavelet Tree Z. Chen, N. Erdol, F. Bao, *Florida*

A simple adaptive first-order differential microphone

Gary W. Elko, Anh-Tho Nguyen Pong

Acoustics Research Department
AT&T Bell Laboratories
Murray Hill NJ 07974

1. Introduction

As communication devices become more portable and used in any environment, the acoustic pick-up by electroacoustic transducers will require the combination of small compact transducers and signal-processing to allow high quality communication. This paper covers the design and implementation of a novel adaptive first-order differential microphone. The self-optimization is based on minimizing the microphone output under the constraint that the solitary null for first-order systems is located in the rear-half plane. The constraint is simply realized by the judicious subtraction of time-delayed outputs from two closely-spaced omnidirectional microphones. Although the solution presented does not maximize the signal-to-noise ratio, it can significantly improve the signal-to-noise ratio in certain acoustic fields.

2. Derivation of the adaptive first-order array

For a plane-wave signal $s(t)$ with spectrum $S(\omega)$ and wavevector $\mathbf{k}$ incident on a two-element array with displacement vector $\mathbf{d}$ as shown in Fig. 1, the output can be written as,

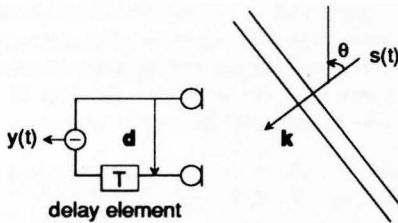


Figure 1: Diagram of first-order sensor composed of two zero-order sensors and a delay.

$$\begin{aligned} Y(\omega, \theta) &= S(\omega) (1 - e^{-j(\omega T + \mathbf{k} \cdot \mathbf{d})}) \\ &= S(\omega) (1 - e^{-j\omega [T + (d \cos \theta)/c]}) \end{aligned} \quad (1)$$

where d is the interelement spacing, T is equal to the delay applied to the signal from one sensor and we have made the substitution $|\mathbf{k}| = k = \omega/c$ where c is the speed of sound. Taking the magnitude of Eq. 1 yields,

$$|Y(\omega, \theta)| = 2 \left| S(\omega) \sin \frac{\omega [T + (d \cos \theta)/c]}{2} \right|. \quad (2)$$

If we assume a small spacing and delay ($kd \ll \pi$ and $\omega T \ll \pi$),

$$|Y(\omega, \theta)| \approx \omega |S(\omega) [T + (d \cos \theta)/c]|. \quad (3)$$

As expected, the first-order differential array has a monopole term and a first-order dipole term $\cos \theta$ which resolves the component of the acoustic particle velocity along the sensor axis. One thing to notice from Eq. 3 is that the first-order array has a first-order differentiator frequency dependence (increases linearly with frequency). This frequency dependence is easily compensated in practice by a first-order lowpass filter. The term in the brackets in Eq. 3 contains the array directional response. In the design of differential arrays the array directivity is of major interest. To simplify the analysis for the directivity of the first-order array let us define β such that,

$$\beta = \frac{T}{T + d/c} \quad (4)$$

and,

$$1 - \beta = \frac{d/c}{T + d/c}. \quad (5)$$

The normalized directional response is,

$$Y_N(\theta) = \beta + (1 - \beta) \cos \theta \quad (6)$$

where the subscript N denotes the normalized response of a first-order system, i.e., $Y_N(0) = 1$. If β is constrained to values between $0 \leq \beta \leq 0.5$, then the directional response of the first-order array shown in Eq. 6 contains a single zero (or null) at an angle between $90^\circ \leq \theta \leq 180^\circ$. The null location for values of β between 0 and 1/2 is at:

$$\begin{aligned} \theta_{null} &= \arccos \left(\frac{\beta}{\beta - 1} \right) \\ &= \arccos \left(\frac{-Tc}{d} \right) \end{aligned} \quad (7)$$

An obvious implementation to realize a first-order adaptive differential sensor would be to have a variable time delay T as shown in Fig. 1. However, this solution is not an attractive one as we now require the ability to generate *any* time delay less than or equal to d/c . The computational requirements to realize the general delay and the necessary adaptive algorithm are unattractive for a real-time implementation. Fortunately, there is a much simpler way to implement an adaptive differential array as discussed below.

One very simple possibility for the realization of a general first-order sensor can be seen by looking at the two terms in

Eq. 6. These terms represent a monopole component and a dipole component. The adaptive algorithm to minimize the array output would hinge on finding the appropriate combination of an omnidirectional and dipole sensor such that the mean-square output would be minimized. The minimization could incorporate a constraint on β (or equivalently on T) so that the null is limited to some range of angles.

The dipole directivity pattern can be realized by subtracting two closely-spaced omnidirectional microphones. The omnidirectional pattern can be realized by adding two closely-spaced omnidirectional microphones. Early directional microphones were implemented using the combination of pressure and velocity (or pressure-differential) microphones [1, 2]. Although the combination of the omnidirectional and dipole microphones is simple to implement, there is a problem with the need for a lowpass filter in the path of the dipole output. The filter introduces an interchannel phase shift. This phase shift can result in the inability of the adaptive algorithm to properly steer a null in a noise source direction.

The back-to-back cardioid arrangement can easily be implemented as shown in Fig. 2. The lowpass filter shown following the output $y(t)$ in Fig. 2 is used to compensate the differentiator response of the differential microphone. By setting the sampling period equal to d/c we can form the back-to-back cardioid microphone outputs directly. The back-to-back

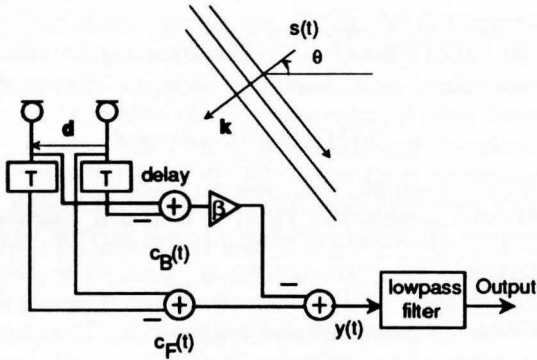


Figure 2: Schematic implementation of an adaptive first-order differential microphone using the combination of a forward and backward facing cardioids.

cardioid configuration is used in some commercial first-order directional microphones [3]. The manufacturer usually gives the user the ability to select one of a few types of preset patterns.

By examining Fig. 2 and Eq. 1 and noting that the sampling period $T = d/c$, we can write the expressions for the forward facing cardioid C_F , and the backwards facing cardioid C_B , (note that to simplify the following expressions the spatial origin is placed at the array center):

$$C_F(\omega, \theta) = 2jS(\omega)e^{-j\omega T/2} \sin \frac{kd(1 + \cos \theta)}{2} \quad (8)$$

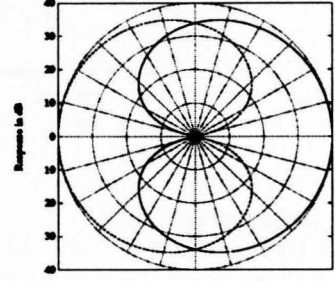


Figure 3: Directional responses of the back-to-back cardioid transducer as shown in Fig. 3.

and,

$$C_B(\omega, \theta) = 2jS(\omega)e^{-j\omega T/2} \sin \frac{kd(1 - \cos \theta)}{2} \quad (9)$$

Normalizing the output signal by the input spectrum $S(\omega)$ results in,

$$\left| \frac{Y(\omega, \theta)}{S(\omega)} \right| = 2 \left| \sin \frac{kd(1 + \cos \theta)}{2} - \beta \sin \frac{kd(1 - \cos \theta)}{2} \right| \quad (10)$$

3. Frequency response of the array

One of the paramount issues in the design of the adaptive array is the variation of the frequency response for a front-half plane source when the null location varies in the rear. If the frequency response of the resulting adaptive microphone varies greatly with rear null location, the user will hear a modulation distortion of the desired on-axis signal as the rear null varies in position. We therefore develop the analytic expressions for the transducer frequency response.

From Eq. 10, the transfer function from a desired source to the array output can be written as,

$$\frac{Y(\omega, \theta)}{S(\omega)} = (1 + \beta) \sin \left(\frac{kd \cos \theta}{2} \right) \cos \left(\frac{kd}{2} \right) + \frac{1 - \beta}{\cos \theta} \left[\sin \left(\frac{kd}{2} \right) \cos \left(\frac{kd \cos \theta}{2} \right) \cos \theta \right] \quad (11)$$

For small kd the quantity inside the brackets in Eq. 11 is proportional to the first term. Using this approximation yields,

$$\left| \frac{Y(\omega, \theta)}{S(\omega)} \right| \approx \left| \frac{2 - (\beta + 1)(1 - \cos \theta)}{\cos \theta} \sin \left(\frac{kd \cos \theta}{2} \right) \cos \left(\frac{kd}{2} \right) \right| \quad (12)$$

It has been found that the above approximation is very good for array spacings up to $\lambda/4$. One thing to note in Eq. 12 is that the frequency response shape does not vary with β . The only effect of a change in β is to change the amplitude of the response to an off-axis source. For normal incidence ($\theta = 0$), the frequency response is independent of β . Figure 4

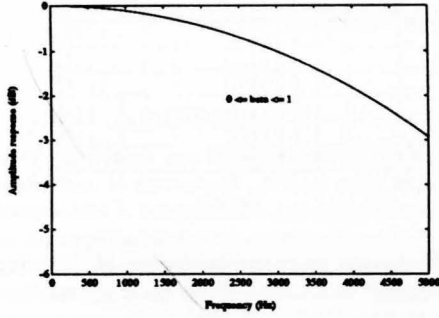


Figure 4: Frequency response of 1.5 cm spaced array with a signal arriving on-axis ($\theta = 0^\circ$), for $0 \leq \beta \leq 1$.

4. Optimum β

The optimum value of β is defined as the value of β which minimizes the mean-square value of the sensor output. Starting with the time-domain representation of the back-to-back cardioid as seen in Fig. 2, we can write,

$$y(t) = c_F(t) - \beta c_B(t). \quad (13)$$

Squaring and taking the expected value yields,

$$E[y^2(t)] = R_{c_F c_F}(0) - 2\beta R_{c_F c_B}(0) + \beta^2 R_{c_B c_B}(0) \quad (14)$$

where, $R_{c_F c_F}(0)$ and $R_{c_B c_B}(0)$ are the powers of the front and back cardioid signals and, $R_{c_F c_B}(0)$ is the cross-power between the front and back cardioid signals. Since the output power $E[y^2(t)]$ is a quadratic function of the variable β , we can be certain that there is only one minimum. The minimum value can be found by taking the derivative of Eq. 14 with respect to β and setting the result to zero. The result is,

$$\beta_{opt} = \frac{R_{c_F c_B}(0)}{R_{c_B c_B}(0)}. \quad (15)$$

The value of β_{opt} is a minimum since the second derivative with respect to β is positive. The second derivative is positive as it has the same sign as an autocorrelation function at zero lag. The autocorrelation function must be positive (or zero) for zero-lag and therefore the error surface $E[y^2(t)]$ is concave with one minimum. The result shown in Eq. 15 is the optimum Wiener filter for a filter of length one.

In an actual DSP implementation we must use estimates of the power and cross-power. This restriction is not as problematic as one may think. In fact, it is desirable to use short-time estimates of the correlation functions since we know a priori that the acoustic fields in which we intend to operate the adaptive microphone are nonstationary.

5. LMS version

The LMS or Stochastic Gradient algorithm is a commonly used adaptive algorithm due to its simplicity and ease of implementation [4]. We therefore develop the LMS algorithm

for the back-to-back cardioid adaptive first-order differential array. To begin, recall that

$$y(t) = c_F(t) - \beta c_B(t). \quad (16)$$

Squaring Eq. 16 results in,

$$y^2(t) = c_F^2(t) - 2\beta c_F(t)c_B(t) + \beta^2 c_B^2(t). \quad (17)$$

The steepest descent algorithm finds a minimum of the error surface $E[y^2(t)]$ by stepping in the direction opposite to the gradient of the surface with respect to the weight parameter β . The steepest descent update equation can be written as,

$$\beta_{t+1} = \beta_t - \mu \frac{d E[y^2(t)]}{d \beta} \quad (18)$$

where, μ is the update step-size and the differential gives the gradient of the error surface $E[y^2(t)]$ with respect to β . The quantity that we want to minimize is the mean of $y^2(t)$ but the LMS algorithm uses the instantaneous estimate of the gradient, i.e., the expectation operation in Eq. 18 is not applied and the instantaneous estimate is used. Performing the differentiation yields,

$$\begin{aligned} \frac{d y^2(t)}{d \beta} &= -2c_F(t)c_B(t) + 2\beta c_B^2(t) \\ &= -2y(t)c_B(t). \end{aligned} \quad (19)$$

Thus, we can write the LMS update equation as,

$$\beta_{t+1} = \beta_t + 2\mu y(t)c_B(t). \quad (20)$$

Typically the LMS algorithm is slightly modified by normalizing the update size so that explicit convergence bounds for μ can be stated that are independent of the input power. The LMS version with a normalized μ is therefore,

$$\beta_{t+1} = \beta_t + 2\mu y(t) \frac{c_B(t)}{\langle c_B^2(t) \rangle} \quad (21)$$

where the brackets indicate a time (or block) average.

6. Implementations and measurements

An experimental adaptive array was realized using two Bruel&Kjaer 4183 microphones spaced at 1.5 cm. Each microphone was fed into a preamp with a gain of 40 dB and then fed into an Ariel DSP-32C PC-DSP board.

Although we have implemented most of the variations that have been discussed in the preceding, we will only discuss the correlation-based optimum β algorithm described in section 5 since this realization was the most robust and simplest implementation. The implementation presently takes less than 40% of an AT&T DSP32C digital signal processing chip. The sampling rate was set to 22 kHz which allows the easy formation of the back-to-back cardioids by appropriately subtracting the present samples from the previous samples. A 5 kHz lowpass filter was utilized to eliminate signals that are above the highest design frequency for the array. One possible way to significantly reduce the amount of real-time DSP cycles would be to sample the input channels close to the Nyquist rate. For the array that we have constructed this would correspond to an 11 kHz sampling rate: one-half of

the present rate. The interpolation for upsampling could be done by the 5 kHz lowpass filter that is presently filtering the input sequences. If the sampling rate is reduced to 11 kHz, we could expect the real-time utilization to drop below 20% of an AT&T DSP 32C.

By constraining the value of β to $0 \leq \beta \leq 1$, we are guaranteed that the null is in the rear-half plane. Of course we are assuming that the sensors are ideal and the source to be nulled is in the farfield. Since the adaptive array presented in this paper is designed to track in nonstationary acoustic sound fields, it is difficult to document the dynamic performance. We can, however, measure the directional responses of the array for various stationary noise fields as well as frequency responses. Figure 5 shows directivity plots measured in the anechoic chamber at Murray Hill. The plots are shown for a 2 kHz sinusoidal signal and for values of β which resulted in nulls being placed in approximately 10° increments.

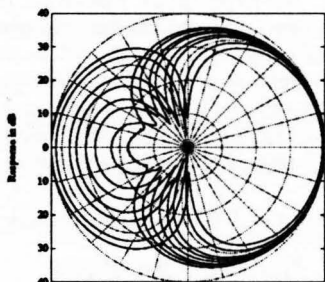


Figure 5: Measured directional responses for the adaptive array for $0 \leq \beta \leq 1$ and β chosen to give nulls in approximately 10° increments. Measured directional responses for the adaptive array for $0 \leq \beta \leq 1$ and β chosen to give nulls in approximately 10° increments.

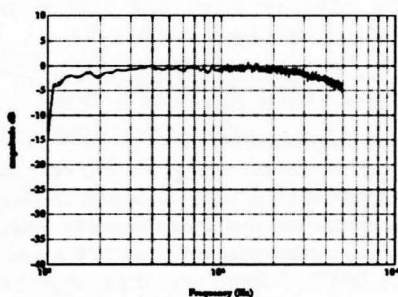


Figure 6: Measured frequency response of the adaptive first-order differential array for on-axis signals ($\theta = 0^\circ$) with null angles at 90° , 135° , and 180° .

Figures 6 and 6 show the frequency response for signals arriving on-axis ($\theta = 0^\circ$) and at $\theta = 45^\circ$. The frequency responses

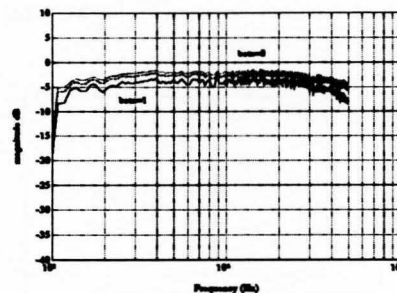


Figure 7: Measured frequency response of the adaptive first-order differential array for signals incident at $\theta = 45^\circ$ with null angles at 90° , 135° , and 180° .

are shown for three different values of β which correspond to the null angles of 90° , 135° , and 180° . The deviation between these responses is barely discernible except for a small amplitude shift for signals impinging from $\theta = 45^\circ$. The highpass characteristic is due to the design of the first-order lowpass filter that is used to correct the differentiator response of the first-order differential sensor. The lowpass response that occurs above 2 kHz is due to the finite difference of two closely-spaced sensors and can also be seen in Fig. 4 which shows the theoretical frequency responses.

7. Conclusions

A novel first-order adaptive differential microphone with a null angle constrained to the rear-half plane has been presented. The back-to-back cardioid was shown to be the best implementation of the simple arrangements presented. An array was constructed and the performance was shown to closely match the theoretical predictions. Two adaptation algorithms were presented: the Wiener optimal solution, and the LMS algorithm.

References

1. H.F. Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton NJ, (1957).
2. R.N. Marshall, and W.R. Harry, *A new microphone providing uniform directivity over an extended frequency range*, J. Acoust. Soc. Am. Vol 12, (1941), pp.481-497.
3. *Handbook for Sound Engineers*, Second Edition, Howard W. Sams, Carmel, Indiana, (1991), pp. 445-447.
4. B. Widrow, S. D. Stearns, *Adaptive Signal Processing*, Prentice-Hall, (1985).
5. R. A. Monzingo, and T. W. Miller, *Introduction to Adaptive Arrays* Wiley, New York, (1980).
6. *Special Issue on Adaptive Antennas*, IEEE Trans. Antennas and Propagat., vol. AP-24, Sept. (1976).
7. R. T. Compton Jr., *Adaptive Antennas*, Prentice-Hall, Englewood Cliffs, NJ, (1988).

Appendix B

MARC Bibliographic Record

LEADER 01785mam a2200409 a 4500

001 9935404493602122

005 20020927092559.0

008 960319s1995 njua 100 0 eng d

020 \$a0780330641 (softbound ed.)

020 \$a078033065X (microfiche ed.)

035 \$a(OCOLC)34407029

035 \$9AUS1152UW

035 \$a(WU)3540449-uwmadisondb

035 \$a(EXLNZ-01UWI_NETWORK)999780556202121

040 \$aEYP\$beng\$cEYP\$dGZM

049 \$aGZMA

090 \$aTK7881.4\$b.I34 1995

111 2_ \$aIEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics\$d(1995 :\$cNew Paltz, N.Y.)

245 10 \$a1995 IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics :\$bOctober 15-18, Mohonk Mountain House, New Paltz, New York.

246 30 \$aWorkshop on Applications of Signal Processing to Audio and Acoustics

246 Page 24 of 31 \$aApplications of Signal Processing to Audio and Acoustics

264 _1 \$aPiscataway, NJ :\$bIEEE Service Center,\$c[1995]

336 \$a\$text\$btxt\$2rdacontent

337 \$aunmediated\$bnc\$2rdamedia

338 \$avolume\$bnc\$2rdacarrier

500 \$aSponsored by the Signal Processing Society of the IEEE.

504 \$aIncludes bibliographical references.

500 \$a"IEEE catalog number 95TH8144"--T.p. verso.

530 \$aAlso available in electronic format on the World Wide Web.

650 _0 \$aSignal processing\$vCongresses.

650 _0 \$aAcoustical engineering\$vCongresses.

650 _0 \$aElectro-acoustics\$vCongresses.

650 _0 \$aMusic\$xAcoustics and physics\$vCongresses.

710 2_ \$aIEEE Signal Processing Society.

710 2_ \$aInstitute of Electrical and Electronics Engineers.

730 0_ \$aIEEE Xplore conferences.

856 41 \$zAlso available through IEEE
Xplore\$uhttp://ieeexplore.ieee.org/servlet/opac?punumber=3495

997 \$aMARCIVE

MMS IDs

Document ID:

999780556202121

Network Electronic IDs:

999780556202121

Network Physical IDs:

999780556202121

mms_mad_ids:

9935404493602122

Appendix C



1995 IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics : October 15-18, Mohonk Mountain House, New Paltz, New York

AUTHOR / CREATOR

[IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics \(1995 : New Paltz, N.Y.\)](#)

CONFERENCES

IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics (1995 : New Paltz, N.Y.)

AVAILABLE AS

[View Online](#)

[Request a Copy](#)

[♥ Save to Lists](#)

[” Citation](#)

☐ Physical Locations

Verona Shelving Facility

☒ **by request only**

TK7881.4 I34 1995

○ Publication Details

FORMAT

Books

CONTRIBUTORS

[IEEE Signal Processing Society](#)

[Institute of Electrical and Electronics Engineers](#)

PUBLICATION

Piscataway, NJ : IEEE Service Center, [1995]

ISBNs

9780780330641, 0780330641, 9780780330658, 078033065X

OCLC

ocm34407029

○ Notes

Sponsored by the Signal Processing Society of the IEEE.

"IEEE catalog number 95TH8144"--T.p. verso.

Includes bibliographical references.

Also available in electronic format on the World Wide Web.

- Signal processing -- Congresses.
- Acoustical engineering -- Congresses.
- Electro-acoustics -- Congresses.
- Music -- Acoustics and physics -- Congresses.

Items Related By Call Number

Shelf Browse

<

Modern recording techniques

Huber, David Miles., author

TK7881.4 .H783 2005

Modern recording techniques

Huber, David Miles, author

TK7881.4 H783 2010

Modern recording techniques

Huber, David Miles., author

TK7881.4 .H783 2014

Modern recording techniques

Huber, David Miles, author

TK7881.4 .H834 2018

IEEE ASSP Workshop on Applications of Signal Processing to Speech and Audio

IEEE ASSP Workshop on Applications...

TK7881.4 .I34

1997 IEEE ASSP Workshop on Applications of Signal Processing to Speech and Audio

IEEE ASSP Workshop on Applications...

TK7881.4 I34 1997

1999 IEEE ASSP Workshop on Applications of Signal Processing to Speech and Audio

IEEE ASSP Workshop on Appl

TK7881.4 I34 1999

>

Information from the Web

▼ Library Staff Details