IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

META PLATFORMS, INC.,
Petitioner

v.

JAWBONE INNOVATION LLC,
Patent Owner

IPR2024-00349
U.S. Patent No. 8,503,691

DECLARATION OF INGRID HSIEH-YEE, PH.D.

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I. Introduction

I, Ingrid Hsieh-Yee, Ph.D., do hereby declare as follows:

1. I have personal knowledge of all of the matters about which I testify in this declaration.

2. I have been retained as an independent expert witness on behalf of Meta Platforms, Inc. ("META") for proceedings before the United States Patent and Trademark Office ("PTO") regarding the authenticity and public availability of certain prior art references in relation to U.S. Patent No. 8,503,691.

3. I am being compensated for my work in this matter at my customary hourly rate. I am also being reimbursed for any reasonable expenses associated with my work and testimony in this investigation. My compensation is not contingent on the results of my study, the substance of my opinions, or the outcome of this matter.

II. Qualifications and Professional Experience

4. My complete qualifications and professional experience are described in my academic curriculum vitae (**Appendix A**). The following is a brief summary of my relevant qualifications and professional experience.

5. I was a professor in the Department of Library and Information Science at the Catholic University of America for 32 years and was conferred the title of Professor Emerita in December 2022. I have experience working in an academic

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library, a medical library, and a legislative library. I hold a Ph.D. and a Masters in Library and Information Studies from the University of Wisconsin-Madison.

6. I am an expert on library cataloging and classification and have published two editions on this subject, *Organizing Audiovisual and Electronic Resources for Access*: *A Cataloging Guide* (2000 and 2006). I have taught a variety of courses, including Cataloging and Classification, Advanced Cataloging and Classification, Organization of Internet Resources, Organization of Information, Metadata, Digital Content Creation and Management, Internet Searches and Web Design, Information Literacy Instruction, Advanced Information Retrieval and Analysis Strategies, and The Information Professions in Society. I am familiar with metadata schema design and implementation. In my teaching, I have covered the design and implementation of metadata in databases, search engines, digital repositories, digital libraries, and digital archives. I have also covered how information organization affects the discovery and access to digital resources on the Internet. My research interests cover cataloging and classification, information organization, metadata, information retrieval, information architecture, digital collections, scholarly communication, social media, user interaction with information systems, and others.

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III. Scope of This Declaration

7. I have been asked to offer an opinion on the authenticity and public accessibility date of the following documents:

(1) Brandstein, M., & Ward, D. (Eds.). (2001). *Microphone Arrays: Signal Processing Techniques and Applications*. Springer Science & Business Media, **Ex. 1005** ("*Brandstein*");

(2) McCowan, I. A., Moore, D. C., & Sridharan, S. (2002, January). Near-field adaptive beamformer for robust speech recognition. *Digital Signal Processing*, vol. 12, no. 1, pp. 87-106, **Ex. 1009** ("*McCowan*").

IV. Background on Library Cataloging

8. I am very familiar with a library cataloging encoding standard known as the "Machine-Readable Cataloging" standard, also known as "MARC," which became the national standard for sharing bibliographic data in the United States by 1971 and the international standard by 1973. MARC is the primary communications protocol for the transfer and storage of bibliographic metadata in libraries. It is widely accepted in my field to rely upon MARC records to determine whether and when a publication would have been publicly accessible.

9. A MARC record consists of several fields, each of which contains specific data about the work. Each field is identified by a standardized, unique, three-

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digit code corresponding to the type of data that follows. **Appendix B** is a true and correct copy of Parts VII to X of "Understanding MARC Bibliographic: Machine-Readable Cataloging" (http://www.loc.gov/marc/umb/um07to10.html) from the Library of Congress that explains commonly used MARC fields. For example, the personal author of a work is recorded in Field 100, the title is recorded in Field 245, publisher information is recorded in Field 260, the physical volume and characteristics of a publication are recorded in Field 300, and topical subjects are recorded in Fields 650.

10. I am very familiar with library cataloging and processing practices and the relationships between OCLC (Online Computer Library Center) and library catalogs, and how libraries create and share their records to help the public discover and access information.

11. The Online Computer Library Center (OCLC) is the largest bibliographic network of the world, and its WorldCat database contains more than 548 million records from thousands of member institutions (many of which are libraries of some type) in more than 100 countries. OCLC was founded in 1967 to promote and support library cooperation. According to the "Third Article, Amended Articles of Incorporation of OCLC Online Computer Library Center, Inc.," OCLC was created "to establish, maintain and operate a computerized library network and to promote the evolution of library use, of libraries themselves, and of librarianship,

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and to provide processes and products for the benefit of library users and libraries, including such objectives as increasing availability of library resources to individual library patrons and reducing the rate of rise of library per-unit costs, all for the fundamental public purpose of furthering ease of access to and use of the ever-expanding body of worldwide scientific, literary and educational knowledge and information" (source: https://www.oclc.org/content/dam/oclc/membership/articles-of-incorporation.pdf).

12. OCLC members can contribute original cataloging records in MARC to the system or derive cataloging records from existing records, an activity referred to as "copy cataloging." When an OCLC participating institution acquires a work, it can create an original MARC record for the work in OCLC's Connexion system (a system for catalogers to create and share MARC records), and the system will automatically generate a code for the date of record creation in the yymmdd format, and the creating library's OCLC symbol is recorded in subfield "a" of the 040 field. Once the MARC record is in Connexion, it becomes available to other OCLC members for adoption to their local online catalogs (*i.e.*, copy cataloging).

13. After a MARC record is created in Connexion, it also becomes searchable and viewable on WorldCat, which is a free web portal for users to explore more than 10,000 libraries worldwide. Records in WorldCat, however, are not presented in MARC fields. Instead, the data elements are labeled to help users

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interpret the records. Thus, the information stored in MARC records in Connexion is available to the interested public through the user-friendly WorldCat web portal.

14. Libraries create MARC records for works they acquire, including books, serials, motion pictures, and publications in other formats. MARC records are representations of materials held by libraries and are critical for the discovery and access to those materials. Monograph cataloging is fairly common in libraries, and most libraries make a newly cataloged monograph available to the public soon after the cataloging work is completed, usually within a week.

15. The cataloging of serials and the serial check-in process are discussed here to show how libraries usually provide access to newly received serial issues. According to the glossary of the *RDA: Resource Description and Access* cataloging standard, a serial is "a mode of issuance of a manifestation issued in successive parts, usually bearing numbering, that has no predetermined conclusion. A serial includes a periodical, monographic series, newspaper, etc." Because the publisher of a serial makes new issues of the serial available successively, a customary cataloging practice is to create one bibliographic record for the serial, and the serial record (encoded in MARC) typically provides information on the beginning date and frequency of the serial, not the dates of individual issues. In other words, libraries typically do not create MARC records for individual issues of a serial. Instead, they rely on a serial check-in system to track the receipt of new issues. A common

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check-in practice is to date stamp a new issue when it arrives. This practice has become automated since the late 1990s, and libraries now vary in how they share the receipt date of a new serial issue with the public. Some libraries use a date stamp, some affix a label to indicate the receipt date, some pencil in the receipt date, and some do not provide the information to the public.

16. The serial check-in process usually takes less than an hour, and one of the steps involves placing a date stamp on the new issue to document the date the issue is checked in. After that, the holdings information of the serial is updated in the library's catalog so that users know which issues are available for request or access. After serial check-in is completed, the new issue is placed on the shelf with the previous issues of the serial. Libraries with a public periodical room typically place new issues in the periodical room for easy user access. Because information presented in serials often reflects latest discovery, a general practice of libraries is to make new issues of serials available for user access soon after they are checked in, usually within a week.

17. I am rendering my expert opinion on the authenticity of the documents referenced herein and on whether it was a printed publication. I will provide my opinion on when and how these documents were publicly disseminated or otherwise made publicly accessible to the extent that persons interested and ordinarily skilled in the subject matter or art, exercising reasonable diligence, could have located and

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retrieved these documents.

V. Legal Principles

18. I am informed by counsel that an item is considered authentic if there is sufficient evidence to support a finding that the item is what it is claimed to be. I am also informed that authenticity can be established based on the contents of the documents themselves, such as the appearance, content, substance, internal patterns, protocols followed or other distinctive characteristics of the item.

19. I am informed by counsel that a given reference is a "printed publication" if it is publicly accessible upon a satisfactory showing that such a document has been disseminated or otherwise made publicly available to the extent that persons interested and ordinarily skilled in the subject matter or art could locate it exercising reasonable diligence and obtain the document. I have also been informed by counsel that materials available in a library constitute "printed publications" if they are cataloged and indexed according to general library practices and protocols that make the references available and accessible to members of the interested public.

20. I have been informed by counsel that the U.S. Patent No. 8,503,691 ("the '691 patent") was filed on June 13, 2008, in the United States, and the earliest priority date of the '691 patent is June 13, 2007. I have assessed public accessibility of the documents referenced below as of this claimed priority date.

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VI. Materials Considered in Forming My Opinions

21. I am personally familiar with many online catalogs, databases, and search engines. In preparing for this declaration, I used the following authoritative information systems to search for records:

- Google Scholar (https://scholar.google.com)
- Online catalog of the Library of Congress (https://catalog.loc.gov/)
- Online catalog of the library of Iowa State University (https://www.lib.iastate.edu/)
- Online catalog of the University of Michigan Library (https://search.lib.umich.edu/catalog?library=All+libraries)
- WorldCat (https://search.worldcat.org/)

22. In the preparation of this declaration, I have reviewed the documents referenced below and any other documents I reference herein, and each of these is a type of material that experts in my field have widely accepted and reasonably rely upon when determining publicly accessibility of a publication:

(1) The documents referenced above in Section III;

(2) Copy of portions of *Brandstein* produced from a used copy, **Appendix 1005A**;

(3) Copy of portions of *Brandstein* obtained from Library of Congress, **Appendix 1005B**;

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(4) Bibliographic and MARC records for *Brandstein*, available at https://lccn.loc.gov/2001031428 from the online catalog of the Library of Congress, accessed and obtained on December 8, 2023, **Appendix 1005C**;

(5) Bibliographic and MARC records for *Brandstein*, available at https://search.lib.umich.edu/catalog/record/990041953010106381 from the online catalog of the University of Michigan Library, accessed and obtained on December 11, 2023, **Appendix 1005D**;

(6) Citations to *Brandstein*, obtained from Google Scholar, **Appendix 1005E**;

(7) Publisher copy of *McCowan*, obtained from Elsevier, **Appendix 1009A**;

(8) Bibliographic and MARC records for *Digital Signal Processing* that contains *McCowan*, available at https://lccn.loc.gov/91650983 from the online catalog of the Library of Congress, accessed and obtained on December 8, 2023, **Appendix 1009B**;

(9) Library copy of *McCowan*, obtained from the Iowan State University Library, **Appendix 1009C**;

(10) Bibliographic and MARC records for *Digital Signal Processing* that

contains *McCowan*, available at https://quicksearch.lib.iastate.edu/permalink/01IASU_INST/174tg9m/alma990008738310102756 from the online catalog of the Iowa State University Library, accessed and obtained on December 11, 2023, **Appendix 1009D**;

(11) Citations to *McCowan*, obtained from Google Scholar, **Appendix 1009E**.

VII. Authenticity and Public Accessibility of *Brandstein* (Exhibit 1005)

A. Authentication

23. **Ex. 1005** is a true and correct copy of *Microphone Arrays: Signal Processing Techniques and Applications*, ("*Brandstein*"), edited by M. Brandstein and D. Ward, Springer Science & Business Media, that I received from counsel. I understand from counsel that this document is being submitted as Meta Platforms, Inc. Exhibit 1005 in the above-captioned *inter partes* review. Page 1 is the cover that shows the title, editors, series title, and publisher. Pag 2 shows "Richard Stern" is the owner of this copy. Page 3 shows Springer, the publisher, has offices around the world, and page 4 is the copyright page that shows this book has a "2001" copyright date and Springer-Verlag holds the copyright. It also shows the ISBN (International Standard Book Number) of *Brandstein* is "3-540-41953-5." Pages 5 to 8 are the preface, pages 9 to 16 are the table of contents, pages 17 to 18 show a list of

contributors, and page 18 (internal page XVIII) is the last page of the front matter. The rest of **Ex. 1005** includes Chapters 1, 2, 3, 5, 8, 13 and the index, and the last numbered page of *Brandstein* is page 398. **Ex. 1005** shows *Brandstein* has a total of 18 chapters, each with its references, and includes equations, figures, and an index.

24. To authenticate **Ex. 1005**, I obtained a used copy of *Brandstein* from Amazon.com. I have closely compared **Ex. 1005** with the used copy and found the pages in **Ex. 1005** and their corresponding pages in the used copy are the same. As support, I include with this declaration **Appendix 1005A**, which is a true and correct copy of portions of *Brandstein* from the used copy. **Appendix 1005A** includes the cover, title page, publisher page, copyright page, table of contents, the list of contributors, the first and last pages of each chapter, the index, and the back cover. It shows that *Brandstein* includes 18 chapters, each with its own references, the front matter ends on internal page XVIII, the last numbered page is page 398, and the book includes equations, figures, and an index. Based on my comparison of **Ex. 1005** and the used copy of *Brandstein*, it is my opinion that these two copies are the same and *Brandstein* is a published book that is authentic and publicly available.

25. To further authenticate *Brandstein*, I obtained a copy of portions of this book from the Library of Congress. **Appendix 1005B** is a true and correct copy that includes the front and back covers, title page, publisher copy, copyright page, table

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of contents, preface and list of contributors, and the first page of each chapter. The copyright page (page 5) shows a "2001" copyright date and the publisher as the copyright holder. This page carries a library date stamp of "Library of Congress JUL – 31 2001 CIP." I have closely compared this LC copy (**Appendix 1005B**) with **Ex. 1005** and **Appendix 1005A**, and it is my opinion that all the pages in **Appendix 1005B** are the same as the corresponding pages in **Appendix 1005A** and **Ex. 1005**.

26. **Ex. 1005** is in a condition that creates no suspicion about its authenticity. Specifically, this copy is not missing any intermediate pages, the text on each page appears to flow seamlessly from one page to the next, and there are no visible alterations to the document. In addition, **Ex. 1005** has the same content as the used copy I obtained. Furthermore, **Ex. 1005** and **Appendix 1005A** (portions of the used copy) and portions of *Brandstein* from the Library of Congress copy (**Appendix 1005B**) show they have the same front matter, and the same beginning page for each chapter. It is therefore my opinion that, for the purpose of my analysis, **Ex. 1005** is an authentic copy of excepts from the published *Brandstein* book.

B. Library of Congress Records

27. To determine whether *Brandstein* is publicly accessible at libraries, I searched WorldCat by the ISBN of *Brandstein* to identify libraries that hold this book, and a WorldCat record (https://search.worldcat.org/title/46791090) showed that Library of Congress held this book. **Appendix 1005C** is a true and correct copy

of the bibliographic and MARC records for *Brandstein* that I obtained from the online catalog of the Library of Congress. I personally located, identified, and obtained these records. They are the type of record experts in my field would reasonably rely upon when forming their opinions.

28. Pages 1 and 2 of the Library of Congress records for *Brandstein* (https://lccn.loc.gov/2001031428) show the title, publisher, copyright date, and the physical description that shows the book's front matter ends on pages xviii, its last numbered page is page 398, and it contains illustrations. The records also show the book's ISBN is "3540419535." These bibliographic details match the information presented in **Ex. 1005, Appendix 1005A** (portions of *Brandstein* from the used copy), and **Appendix 1005B** (portions of *Brandstein* from the LC copy), confirming this record represents *Brandstein*. The "Item Available" area shows the book's call number is "TK5986 .M53 2001 Copy 1" and it can be requested at the "Jefferson or Adams Building Reading Rooms." The status of this copy is "Not Charged" (meaning available for user request).

29. Page 3 and 4 are the MARC record that shows the date of record creation and revision. The first six digits of Field 008 show the MARC record was created on "010417" (*i.e.*, April 17, 2001), and the "s2001" code following this creation date indicates this work has a single publication date of 2001. Field 955 records local cataloging activities, and a note of "2001-07-16 bk rec'd, to CIP ver."

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indicates the library received a copy of *Brandstein* on July 16, 2001, and sent it to CIP (the Cataloging in Publication Program) for record verification. Another note of "2001-09-04 to BCCD" shows the book was sent to BCCD (*i.e.* the Binding and Collections Care Division) for processing. Subfield "a" Field 040 shows "DLC" created the original record. According to the *Directory of OCLC Members*, (https://www.oclc.org/en/contacts/libraries.html), "DLC" is the OCLC library symbol for the Library of Congress. Fields 008, 955 and 040 inform my opinion that the Library of Congress created a Cataloging-in-Publication MARC record for *Brandstein* on April 17, 2001, and completed the record on September 4, 2001, then sent the book to processing on the same day. Based on my experience with the Library of Congress collection, it can take one to three months for a newly cataloged book to become publicly accessible. This MARC record informs my opinion that *Brandstein* became discoverable in the online catalog of the Library of Congress on April 17, 2001, when the CIP record was created. After the cataloging work was completed and the book was processed, the library's copy of *Brandstein* would have become publicly accessible as early as October 4, 2001, and no later than December 4, 2001.

30. Field 010 of the MARC record (**Appendix 1005C**) shows the Library of Congress Control Number assigned to *Brandstein* and Field 020 shows the book's ISBN. Field 245 shows the title and editors, Field 260 shows the publisher

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information and copyright date, Field 300 shows the physical description, Fields 490 and 830 show the series title, Field 504 shows a reference and index note, and Fields 700 show the editors are included in the record to help users discover this book.

31. Subjects are represented in Field 050 by a Library of Congress Classification (LCC) number of "TK5986" that represents "Microphones" and in Field 082 by a Dewey Decimal Classification (DDC) number of "621.382/84" that represents "Communications Engineering – Specific devices." Subject is also represented in Field 650 by a Library of Congress subject heading "Microphone." These subject representations support interested users to conduct subject searches to discover this book.

32. This MARC record (**Appendix 1005C**) shows that *Brandstein* was published in 2001 and became discoverable on April 17, 2001, at the Library of Congress. Users would have been able to conduct subject searches by the LCC number, DDC number and the Library of Congress subject heading in the MARC record to discover this book. They would also have been able to search for this book by the LCCN, ISBN, title, editors, and series title.

33. Page 5 of the Library of Congress copy of *Brandstein* (**Appendix 1005B**) shows a date stamp of "JUL – 31 2001 CIP" indicating the book was received by the CIP Program on July 31, 2001. Field 955 of the MARC record (**Appendix 1005C**) shows the record was completed on September 4, 2001, and the

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book was sent to processing that day. A customary library practice is to process a newly cataloged book promptly to make it publicly accessible. In most academic libraries, the processing of such a book takes no more than a week after the MARC record is completed. Based on my personal experience with the Library of Congress collections, the processing of a newly cataloged book can take one to three months. It is therefore my opinion that, after processing, the Library of Congress copy of *Brandstein* would have become publicly accessible as early as October 4, 2001, and no later than December 4, 2001.

C. University of Michigan Library Records

34. To further demonstrate the public accessibility of *Brandstein*, I obtained another library record. **Appendix 1005D** is a true and correct copy of the bibliographic and MARC records obtained from the online catalog of the University of Michigan Library. I personally located, identified, and obtained these records. They are the type of records experts in my field have widely accepted and would reasonably rely upon when determining whether a publication was publicly accessible.

35. Pages 1 and 2 are the bibliographic record that shows the same data elements for *Brandstein* as the Library of Congress bibliographic record, and the holdings area shows the book is "On shelf" and its call number is "TK 5986 .M531 2001." Pages 3 and 4 are the MARC record. The first six digits of Field 008 show

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the University of Michigan Library record was created on "011210" (*i.e.*, December 10, 2001) and Field 005 (a field documenting the last time of record modification) shows the MARC record was last revised by the library on "20011210" (*i.e.*, December 10, 2001). Subfield "a" of Field 040 shows "DLC" (*i.e.*, Library of Congress) created the original MARC record, and subfield "d" shows "EYM" modified the record later. According to the *Directory of OCLC Members* (https://www.oclc.org/en/contacts/libraries.html), "EYM" is the OCLC library symbol for the University of Michigan Library. Fields 008, 005, and 040 inform my opinion that the University of Michigan Library used the record created by the Library of Congress for copy cataloging and completed their record on December 10, 2001, making *Brandstein* discoverable in their online catalog from that date on, and the library copy, after processing, would have been publicly accessible by December 17, 2001.

36. Since the MARC record of the University of Michigan Library (**Appendix 1005D**) is a copy cataloging record based on the Library of Congress record for *Brandstein* (**Appendix 1005C**), the two records show the same data elements for the Library of Congress Control Number (Field 010), ISBN (Field 020), classification numbers (Fields 050 and 082), title and editor statement (Field 245), publisher information and copyright date (Field 260), physical description (Field 300), reference and index note (Field 504), subject heading (Field 650), and editors

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(Fields 700). The University of Michigan Library MARC record is different from the Library of Congress record in that Fields 008 and 005 show the local cataloging dates, and Field 490 shows the library does not make the series title searchable. In short, **Appendix 1005D** shows the University of Michigan Library used the Library of Congress record (**Appendix 1005C**) for copy cataloging and completed their record on December 10, 2001, making *Brandstein* discoverable in their online catalog from that date on, and the library copy, after processing, would have become publicly accessible no later than December 17, 2001.

D. Usage Record

37. Actual usage of a publication is reflected by other publications that make reference to it. I used Google Scholar to identify citations to *Brandstein* and compiled a list of early citations. **Appendix 1005E** presents eight citations to *Brandstein* from 2001 and 2002, and the earliest citation appeared in October 2001, supporting my opinion that *Microphone Arrays: Signal Processing Techniques and Applications* edited by M. Brandstein and D. Ward and published by Springer in 2001 would have been publicly accessible no later than December 17, 2001.

E. Summary of My Opinion on *Brandstein*

38. My comparison of the counsel copy of *Brandstein* (**Ex. 1005**) with the used copy I obtained (portions shown in **Appendix 1005A**) and the Library of Congress copy (**Appendix 1005B**) informs my opinion that **Ex. 1005** is authentic.

The Library of Congress copy (**Appendix 1005B**) shows a date stamp of "JUL – 31 2001 CIP" and the library's MARC record (**Appendix 1005C**) shows the cataloging work was completed and the book was sent to processing on September 4, 2001. The University of Michigan Library MARC record (**Appendix 1005D**) shows the library used the Library of Congress record for copy cataloging and completed their record on December 10, 2001. Based on the documents and records I have reviewed, it is my opinion that *Microphone Arrays: Signal Processing Techniques and Applications* edited by M. Brandstein and D. Ward and published by Springer in 2001 would have been publicly accessible no later than December 17, 2001. Citation history shows the earliest citation appeared in October 2001, supporting my opinion of the public availability of this book no later than December 2001.

VIII. Authenticity and Public Availability of *McCowan* (Exhibit 1009)

A. Authentication

39. **Ex. 1009** is a true and correct copy of **"**Near-field adaptive beamformer for robust speech recognition," ("*McCowan*"), by McCowan, Moore, and Sridharan, *Digital Signal Processing*, vol. 12, no. 1 (2002, January), pp. 87-106, that I received from counsel. I understand from counsel that this document is being submitted as Meta Platforms, Inc. Exhibit 1009. Pages 1 and 2 are the covers of a bound volume that show a Library of Congress barcode and identify this volume as "Digital Signal Processing 12 JAN 2002 1" with a call number is "TK 5102.5 .D4463 Set 1." Page

3 is the issue cover that shows "Digital Signal Processing A Review Journal" as the journal title, "1051-2004" as the journal's ISSN (International Standard Serial Number) and Academic Press, an Elsevier Science Imprint, as the publisher. Page 3 identifies this issue as "Volume 12, Number 1, January 2002." Page 4 shows a note indicating the cover photo is about "lower path directivity pattern at 5000 Hz" from "the article by McCowan, Moore, and Sridharan in this issue." Page 5 is the copyright page that shows a "2002" copyright date for this issue and Elsevier Science (USA) as the copyright holder. It also shows this journal is published quarterly by Elsevier Science. This page carries a date stamp of "Library of Congress FEB 19 2002 Copyright Office." Pages 6 to 25 are the full text of *McCowan*, and page 26 is the table of contents of the issue. Page 6 (internal page 87) shows the title, authors, abstract, and author key words ("microphone array; beamforming; near-field; adaptive; superdirectivity; speech recognition"), and indicates this article has a "2002" copyright date and Elsevier Science (USA) is the copyright holder. The top of internal page 87 shows the doi (digital object identifier) of *McCowan* is "10.1006/dspr.2001.0414" and it is available online at http://www.idealibrary.com. **Ex. 1009** shows *McCowan* has a total of 20 pages, including 27 equations, 12 figures, four tables, and 14 references.

40. To authenticate **Ex. 1009**, I searched Google Scholar for *McCowan* and a record link led me to the publisher's webpage for this article at

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https://www.sciencedirect.com/science/article/abs/pii/S1051200401904142. I then used the "Purchase PDF" link to obtain a copy from the publisher Elsevier. This publisher copy is presented as **Appendix 1009A** in this declaration.

41. **Appendix 1009A** is a true and correct copy of *McCowan* that I obtained from Elsevier. I have closely compared **Ex. 1009** with this publisher copy and found them to be the same.

42. *McCowan* (**Ex. 1009**) is in a condition that creates no suspicion about its authenticity. Specifically, my review has found that this copy is not missing any intermediate pages, the text on each page appears to flow seamlessly from one page to the next, and there are no visible alterations to the document. In addition, my analysis found this copy to be the same as the publisher copy. Furthermore, **Ex. 1009** is in the custody of Library of Congress, a national library where, if authentic, it would likely be found. I therefore see no reason to question the authenticity of this copy of *McCowan* (**Ex. 1009**).

B. Library of Congress Records

43. **Appendix 1009B** is a true and correct copy of the bibliographic and MARC records for *Digital Signal Processing* that contains *McCowan*. I personally located, identified and obtained these records from the online catalog of the Library of Congress. They are the type of record experts in my field have widely accepted and would reasonably rely upon when determining whether and when a publication

was publicly accessible.

44. Pages 1 to 3 of **Appendix 1009B** are the bibliographic record that shows details of *Digital Signal Processing*, including the title, publisher, LCCN (Library of Congress Control Number), publication history, current frequency, and ISSN. The "Item Availability" area shows the call number for this journal is "TK5102.5 .D4463 Set 1," the library holds "v.12:no.1" that has an issue date of January 2002, and users can request access to this journal at the Jefferson or Adams Building Reading Rooms.

45. Pages 4 to 6 are the MARC record. The first six digits of Field 008 show the record was created on "900620" (*i.e.*, June 20, 1990), and the "c19919999" code following the record creation date indicates this publication is a continuing resource that began publication in 1991 and has no publication end date. Subfield "a" of Field 040 shows "NSD" (the National Serial Data Program of the Library of Congress) created this MARC record, and subfield "d" shows "DLC" (*i.e.*, the Library of Congress) revised the NSD record later.

46. Field 022 of the MARC record shows the ISSN of this journal. Field 245 shows the journal title, Field 260 shows Academic Press of Duluth, Minnesota has published this journal since 1991, Field 362 shows the journal began with volume 1, number 1 in January 1991, Fields 310 show the publication frequency was "Quarterly" until 2003, and the current publication frequency is "Bimonthly."

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47. Subjects of this journal are represented in Field 050 by a Library of Congress Classification (LCC) number of "TK5102.5" that represents "Telecommunication—General special" and in Field 082 by a Dewey Decimal Classification (DDC) number of "621.382/2" that represents "Signal processing." Subject is also represented in Fields 650 by two Library of Congress subject headings: Signal processing |x Digital techniques ("|x" indicates a topical subdivision), and Signal processing |x Digital techniques |v Periodicals ("|v" indicates a form subdivision). These subject representations support users to conduct subject searches to discover this journal.

48. This MARC record (**Appendix 1009B**) shows that *Digital Signal Processing* is a long-running journal, and the MARC record has made this journal discoverable in the online catalog of the Library of Congress. The date stamp on the Library of Congress copy of *McCowan* shows the library received the issue containing *McCowan* on February 19, 2002, so the MARC record would have been discoverable in the library's online catalog no later than that date. Interested users would have been able to discover this journal by its ISSN, LCCN, title, classification numbers and subject headings.

49. As discussed above, a customary library practice is to use the serial check-in process to document the receipt of a newly received journal issue, and the new issue is processed promptly to make it publicly accessible, usually within a

week after serial check in is completed. The date stamp of **Ex. 1009** shows the issue containing *McCowan* was received by Library of Congress on February 19, 2002, meaning the issue (and *McCowan* contained therein) would have become publicly accessible no later than February 26, 2002, at Library of Congress.

C. Iowa State University Copy

50. To further demonstrate the public availability of *McCowan* I obtained a copy of this article from the library of the Iowa State University. **Appendix 1009C** is a true and correct copy of *McCowan*. Page 1 is the issue cover that shows the call number for this journal is "TK5102.5 .D4463" and page 3 is the copyright page that shows a date stamp of "Library FEB 18 2002 Iowa State University of Science & Technology." Page 4 is the table of contents for volume 12, number 1 that has a "January 2002" issue date and it shows *McCowan* begins on page 87. I have closely compared this copy with the Library of Congress copy (**Ex. 1009**) and the publisher copy (**Appendix 1009A**), and found the three copies to have the same content for *McCowan*. I also found the two library copies (**Ex. 1009** and **Appendix 1009C**) to have the same front matter.

D. Iowa State University Library Records

51. **Appendix 1009D** is a true and correct copy of the bibliographic and MARC records for *Digital Signal Processing* that contains *McCowan*. I personally located, identified, and obtained these records from the online catalog of the Iowa

State University Library. They are the type of record experts in my field would reasonably rely upon when forming their opinions. I used screen captures to produce this appendix.

52. Pages 1 to 4 of **Appendix 1009D** are the bibliographic record that shows this journal is available at "Parks Library Storage Building (TK5102.5 .D4463)" and also available online in five Elsevier databases. Page 3 shows the library holds "v.4 (1994) – v. 15 (2005)" and v.12, 2002 is loanable. Page 4 shows the journal title, subject, genre, publisher, notes, language, format, frequency, and identifiers.

53. Page 5 is the MARC record and the first six digits of Field 008 show the record was created on "900620" (*i.e.*, June 20, 1990), and the "c19919999" code following the record creation date shows this publication is a continuous resource that began publication in 1991 and has no publication end date. Subfield "a" of Field 040 shows "NSD" (the National Serial Data Program of the Library of Congress) created the original record, subfield "d" shows "DLC" (*i.e.*, Library of Congress) revised the record, and another subfield "d" shows "IWA" also revised the NSD record. According to the *Directory of OCLC Members* (https://www.oclc.org/en/contacts/libraries.html), "IWA" is the OCLC library symbol for the library of the Iowa State University. Fields 008 and 040 inform my opinion that Library of Congress and the Iowa State University Library used the

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record created by NSD for copy cataloging.

54. Since the Library of Congress serial MARC record for *Digital Signal Processing* (**Appendix 1009B**) and the Iowa State University Library serial MARC record (**Appendix 1009D**) are based on the same record, these two records have many fields in common, including the Library of Congress Control Number (Field 010), ISSN (Field 022), Library of Congress Classification number (Field 050), title (Field 245), publisher (Field 260), date of publication (Field 362), source of title note (Field 500), and subject heading (Field 650). The Iowa State University Library record is different from the Library of Congress record in that it does not use the Dewey Decimal Classification number, and it reflects more recent cataloging standards because it does not use abbreviations in Field 300 and includes Fields 336, 337 and 338 to indicate the content, media, and carrier information of this publication.

55. This MARC record (**Appendix 1009D**) shows that *Digital Signal Processing* is a long-running journal, and the MARC record has made this journal discoverable in the online catalog of the Iowa State University Library. The date stamp on the library's copy of *McCowan* shows the library received the issue containing *McCowan* on February 18, 2002, so the MARC record would have been discoverable in the library's online catalog by at least that date. Interested users would have been able to discover this journal by its ISSN, LCCN, title, classification

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number and subject heading.

56. As discussed above, a customary library practice is to use the serial check-in process to document the receipt of a newly received journal issue, and the new issue is processed promptly to make it publicly accessible, usually within a week after serial check in is completed. The date stamp of **Appendix 1009C** shows the issue containing *McCowan* was received by the library on February 18, 2002, meaning the issue (and *McCowan* contained therein) would have become publicly accessible no later than February 25, 2002, at Iowa State University Library.

E. Usage Records

57. Actual usage of a publication is reflected by other publications that make reference to it. I used Google Scholar to review citations to *McCowan* and compile a list of early citations. **Appendix 1009E** shows seven citations from 2004 to April 2007, and the earliest citation appeared in October 2004.

F. Summary of My Opinion on *McCowan*

58. The Library of Congress copy of *McCowan* (**Ex. 1009**) is authentic because it is the same as the publisher copy (**Appendix 1009A**). **Ex. 1009** does not contain any markings or changes that cause suspicion of its authenticity. The "FEB 19, 2002" date stamp on **Ex. 1009** indicates the library received this issue containing *McCowan* on February 19, 2002. The library records (**Appendix 1009B**) show their MARC record was a copy cataloging record based on the record created by the

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National Serial Data Program of the Library of Congress, and the MARC record has made this journal discoverable at the library's online catalog by at least February 2002. The copy of *McCowan* from the Iowa State University Library (**Appendix 1009C**) carries a date stamp of "FEB 18 2002" indicating the library received the issue containing *McCowan* on February 18, 2002. The library record (**Appendix 1009D**) shows it was also based on the record created by the National Serial Data Program of the Library of Congress, and the MARC record has made this journal discoverable at the library's online catalog by at least February 2002. The date stamps on **Ex. 1009** and **Appendix 1009C** inform my opinion that *McCowan* would have been publicly accessible no later than February 26 and February 25 of 2002 at the Library of Congress and the Iowa State University Library, respectively. Citation records show that the earliest citation to *McCowan* appeared in October 2004.

IX. Declaration

59. I hereby declare that all statements made herein on my own knowledge are true and that all statements made on information and belief are believed to be true. I declare under penalty of perjury under the laws of the United States of America that the foregoing is true and correct.

Date: 12-19-2023 Executed: Ingrid Hsieh-Yee

Ingrid Hsieh-Yee, Ph.D.

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APPENDIX A

Ingrid Hsieh-Yee

Professor Emerita
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Education

Ph.D. 1990. Library and Information Studies, University of Wisconsin-Madison
Minors: Sociology and Psychology

M.A. 1984. Library and Information Studies, University of Wisconsin-Madison.

M.A. 1983. Comparative Literature, University of Wisconsin-Madison.

B.A. 1980. Foreign Languages and Literature, National Taiwan University.

Work Experience

Professor, School/Dept. of Library and Information Science, Catholic University of America, 1990-2022 (Assistant Professor, 1990-1996; Associate Professor, 1997-2004; Professor 2004-2022; Professor Emerita 2022)

Co-Chair, Dept. of Library and Information Science, Catholic University of America, June 2015-August 2016.

Acting Dean, School of Library and Information Science, Catholic University of America, January 2010-June 2012.

Cataloger, Dept. of Legislative Reference Library, Annapolis, Maryland, 1989-1990.

Lecturer, School of Library and Information Studies, University of Wisconsin-Madison, 1988.

Teaching Assistant, School of Library and Information Studies, University of Wisconsin-Madison, 1986-1988.

Cataloger, Health Sciences Library, University of Wisconsin-Madison, 1984-1986.

Areas of Teaching and Research Interests

Information Organization and Access; Metadata; Cataloging & Classification; Information Architecture; Information Retrieval; Digital Collections; Social Media; Scholarly

Communication; Information Behavior; Health Informatics; Human Computer Interaction; Usability Studies

Grants & Honors

Cultural Heritage Information Management Project. IMLS grant. Amount: $498,741. Period: Aug. 2012 to July 2015. Co-PI with Dr. Youngok Choi.

D.C. Health Information Technology (HIT4): Building Capacity & Providing Access in Our Nation's Capital. Dept. of Labor H2B Training Grant. Grant amount: $4,175,500. Grant period: Nov. 2011 to Dec. 2015. Partner with the Metropolitan School of Professional Studies of the Catholic University of America, Children's National Medical Center, D.C. Department of Employment Services, Holy Cross Hospital, Howard University, Center for Urban Progress, Providence Hospital, and Sibley Memorial Hospital.

Capital Health Careers Project. Department of Labor Healthcare Sector and Other High Growth and Emerging Industries Grant. Grant amount: $4,953,999. Grant period: March 2010 – February 2013. Awarded to a group of healthcare organizations and educational institutions in Washington, D.C. Providence Health Foundation of Providence Hospital (Lead institution). Part of the grant supported the development of a Master's degree program in Information Technology with a concentration in Health Information Technology offered by the School of Library and Information Science.

The Washington D.C. School Librarians Project. IMLS grant. Grant amount: $412,660. Grant period: Aug. 2007 – June 2011. The School partnered with the District of Columbia Public Schools (DCPS) and the District of Columbia Library Association to educate and mentor school media specialists for the DCPS system. PI, Jan. 2010 to June 2011.

SIG Member of the Year, American Society for Information Science and Technology (2009).

Most Outstanding Paper of *OCLC Systems & Services* (2001).

ALISE Research Grant (2001).

Most Outstanding Paper of *OCLC Systems & Services* (2000).

Research Grant from ERIC (1999-2000).

Best Research Paper Award; Association for Library and Information Science Education (1998).

Research Grants, Catholic University of America. 1991, 1992, 1993, 1996, 1998, 1999, 2004, 2005, 2006, 2007, 2013-14.

Cooperative Faculty Research Grant, Consortium of Universities in the Washington Metropolitan Area (1993-1994).

Cooperative Research Grant, Council on Library Resources (1993-1994).

Journal of the American Society for Information Science Best Paper Award (1993).

ASIS/ISI Information Science Doctoral Dissertation Scholarship (1989).

HEA Title IIB Fellowship (Dept. of Education) (1989)

Chinese-American Librarians Association Scholarship (1987).

Beta Phi Mu (1985).

Vilas Fellowship, University of Wisconsin-Madison. 1984

Publications

Hsieh-Yee, I. (2021). Can We Trust Social Media? *Internet Reference Services Quarterly*, 1-15. Published online: 28 July, 2021. 10.1080/10875301.2021.1947433.

Bailey, T., & Hsieh-Yee, I. (2020). Combating the Sharing of False Information: History, Framework & Literacy Strategies, *Internet Reference Services Quarterly*, 24 (1-2), 9-30.

Choi, Y., and Hsieh-Yee, I. (2010). Finding Images in an OPAC: Analysis of User Queries, Subject Headings, and Description Notes. *Canadian Journal of Information and Library Science,* 34(3): 271 – 295.

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NISO Framework for Guidelines for Building Good Digital Collections. 3rd ed. Baltimore, MD: National Information Standards Organization, 2007. Also available online: http://www.niso.org/framework/framework3.pdf (NISO Working Group members: Priscilla Caplan (chair), Grace Agnew, Murtha Baca, Tony Gill, Carl Fleischhauer, Ingrid Hsieh-Yee, Jill Koelling, and Christie Stephenson.)

Choi, Y., Hsieh-Yee, I., and Kules, B. (2007). Retrieval Effectiveness of TOC and LCSH. *Proceedings of the Joint Conference on Digital Libraries*, pp. 233-234.

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a Digital Teaching Commons to Enhance Metadata Teaching, Learning and Research. *Proceedings of the National Conference of the Association of College and Research Libraries, Baltimore, MD*, pp. 26-31.

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Tang, R., Hsieh-Yee, I., and Zhang, S. (2007). User Perceptions of MetaLib Combined Search: An Investigation of How Users Make Sense of Federated Searching." *Internet Reference Services Quarterly,* 12(12): 211-236.

Hsieh-Yee, I., Tang, R., and Zhang, S. (2006). User Perceptions of a Federated Search System. *Proceedings of the Joint Conference on Digital Libraries*, *June 11-15, 2006, Chapel Hill*, p. 338.

Hsieh-Yee, I. (2006). *Organizing Audiovisual and Electronic Resources for Access: A Cataloging Guide*. 2nd ed. Westport, Conn.: Libraries Unlimited.

NISO A Framework of Guidance for Building Good Digital Collections. 2nd ed. Bethesda, MD: National Information Standards Organization, 2004. Framework Advisory Group: Grace Agnew, Liz Bishoff, Priscilla Caplan (Chair), Rebecca Gunther and Ingrid Hsieh-Yee.

Hsieh-Yee, I. (2004). Cataloging and Metadata Education in North American LIS Programs. *Library Resources & Technical Services,* 48(1): 59-68.

Hsieh-Yee, I. (2004). Cataloging and Metadata Education. In Gary E. Gorman (Ed.), *International Yearbook of Library and Information Management 2003: Metadata Applications and Management*, (pp.204-234). London: Facet Publishing.

Yee, P. L., Hsieh-Yee, I., Pierce, G.R., Grome, R., and Schantz, L. (2004). Self-Evaluative Intrusive Thoughts Impede Successful Searching on the Internet. *Computers in Human Behavior,* 20(1): 85-101.

Hsieh-Yee, I. (2003). Cataloging and Metadata Education: A Proposal for Preparing Cataloging Professionals of the 21st Century. A report submitted to the ALCTS-Education Task Force in response to Action Item 5.1 of the *Bibliographic control of Web Resources: A Library of Congress Action Plan*. Approved by the Association for Library Collections and Technical Services. Web version available since April 2003 at http://lcweb.loc.gov/catdir/bibcontrol/CatalogingandMetadataEducation.pdf.

Hsieh-Yee, I. (2002). Cataloging and Metadata Education: Asserting a Central Role in Information Organization. *Cataloging & Classification Quarterly* 34(½): 203-222.

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Hsieh-Yee, I. (2001). ERIC User Services: Changes and Evaluation for the Future. *Government Information Quarterly,* 18: 31-42.

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Logan, E., and Hsieh-Yee, I. (2001). Library and Information Science Education in the Nineties. *Annual Review of Information Science and Technology,* 35: 425-477.

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Hsieh-Yee, I. (1993). Effects of Search Experience and Subject Knowledge on Online Search Behavior: Measuring the Search Tactics of Novice and Experienced Searchers. *Journal of the American Society for Information Science,* 44: 161-174. (Received the 1993 Best JASIS Paper Award.)

Works in Progress

Hsieh-Yee, I. Actionable lessons for debunking false information: A comparative analysis of literacy lessons in communication and library and information science

Hsieh-Yee, I. Representing diverse voices and communities in cataloging and indexing: The evolution of controlled vocabularies

Hsieh-Yee, I. Information Organization and Retrieval in Practice (a monograph scheduled for completion and publication in 2023)

Presentations

Bailey, T. & Hsieh-Yee, I. (February 2020). The Phenomena of Sharing Misinformation and the Need for Information Literacy. Presented at the 2020 Bridging the Spectrum Symposium, Washington, D.C., February 14, 2020.

Hsieh-Yee, I. and Fragan-Fly, J. (May 2018) Trends, Design & Strategies for Digital Scholarship Services. Presented at the 2018 Maryland/Delaware Library Association Conference, Cambridge, MD.

Hsieh-Yee, I. (February, 2018) Research Data Management: What It Takes to Succeed. Presented at the 10th Bridging the Spectrum Symposium, Washington, D.C.

Hsieh-Yee, I. (February, 2017) *Research Data Management: New Competencies and Opportunities for Information Professionals*. Presented at the 9th Bridging the Spectrum Symposium, Washington, D.C.

Hsieh-Yee, I. and Lawton, P. (February, 2017) *Enhancing Catholic Portal Searches with User Terms and LCSH*. Presented at the 9th Bridging the Spectrum Symposium, Washington, D.C.

Hsieh-Yee, I. (2016, October) *Visualizing Data for Information*. Presented at the 2016 Virginia Library Association Conference, Hot Springs, VA.

Hsieh-Yee, I. (2016, August) *Religious Materials Toolbox for Archivists: Solutions to Problems Facing the Profession*. Presented at Archives * Records 2016, Atlanta, GA.

Hsieh-Yee, I. and Lawton, P. (2016, March) *Enhancing Retrieval of Catholic Materials with LCSH Knowledge Structure*. Presented at the 2016 Catholic Library Association Conference, San Diego, CA.

Fagan-Fry, J. and Hsieh-Yee, I. (2016, February) *Approaches to Digital Scholarship at Top Universities around the World: Scholarly Publishing in the Digital Age.* Presented at the 8th Bridging the Spectrum Symposium, Washington, D.C.

Hsieh-Yee and Fagan-Fry, J. (2016, January) *Innovative Services for Digital Scholarship at Top 100 Research Libraries of the World*. Poster presented at the 2016 Annual Conference of the Association for Library and Information Science Education, Boston, Mass.

Hsieh-Yee, I. and Lawton, P. (2015, June). *Crowdsourcing terms for CRRA portal themes.* Poster presented at the third CRRA symposium and annual meeting, Bringing the created toward the Creator: Liturgical art and design since Vatican II. Catholic Theological Union, Chicago, Illinois.

Hsieh-Yee, I. and Lawton, P. (2015, February). *Crowdsourcing terms for thematic exploration in the Catholic Portal.* Poster presented at the 7th Annual Bridging the Spectrum Symposium, Washington, D.C.

Hsieh-Yee, I., James, R., and Fagan-Fry, J. (2015, February). *Support for digital scholarship at top university libraries of the world*. Poster presented at the 7th Annual Bridging the Spectrum Symposium, Washington, D.C.

Hsieh-Yee, I., Zhang, S., Lin, K., and Cherry, S. (2015, February). *Thus said the end users: Summon experience and support for research workflows*. Poster presented at the 7th Annual Bridging the Spectrum Symposium, Washington, D.C.

Yontz, E., Hsieh-Yee, I., & Houston, S. (2015, February). *Healthy Heroes Summer Reading Club: Developing healthy youth at public libraries.* 11th Annual Jean Mills Health Symposium, Greenville, North Carolina.

Yontz, E., Hsieh-Yee, I., and Houston, S. (2015, January). *Healthy youth and libraries: A pilot study.* Association for Library & Information Science Education (ALISE) Annual Conference, Chicago, Illinois.

Hsieh-Yee, I. (2014, May). *Linking CRRA resources to portal themes via authority files*. Presented at the Catholic Research Resources Alliance 2014 Membership Meeting, Marquette, WI.

Hsieh-Yee, I. (2014, April). *Enhancing subject access to CRRA resources*. Presented at the 2014 Catholic Library Association Conference, Pittsburgh, PA.

Hsieh-Yee, I. (2014, January). *Health Information Technology Program: Educational entrepreneurship in action*. Presented at the 2014 annual Conference of the Association for Library and Information Science Education, Philadelphia, PA

Hsieh-Yee, I., Zhang, S., Lin, K., and Cherry, S. (2014, January). *Discovering information through Summon: An analysis of user search strategies and search success*. Paper presented at the 6th Bridging the Spectrum Symposium, Washington, D.C.

Hsieh-Yee, I. (2012, December). *National Digital Stewardship Alliance and SLIS at CUA: An Educational Partnership*. Paper presented at Best Practices Exchange: Acquiring, Preserving, and Providing Access to Government Information in the Digital Era, Annapolis, MD

Choi, Y. and Hsieh-Yee, I. (2010, January). *Finding Images in an OPAC: Analysis of User Queries, Subject Headings, and Description Notes*. Paper presented at 2nd Annual Bridging the Spectrum Symposium, Catholic University of America, Washington, D.C.

Hsieh-Yee, I. and Coogan, J. (2010, January). *Google Scholar vs. Academic Search Premier: What Libraries and Searchers Need to Know*. Paper presented at 2nd Annual Bridging the Spectrum Symposium, Catholic University of America, Washington, D.C.

Hsieh-Yee, I. (2009, November). *Information Science Education: An LIS School's Perspective*. Paper presented at Annual Meeting of the American Society for Information Science and Technology, Vancouver, British Columbia, Canada.

Hsieh-Yee, I., Menard, E., Ya-Ning Chen, A., Shu-Jiun Chen, S., Kalfatovic, M. R., Wisser. K. M. (2009, November). *Information Organization in Libraries, Archives and Museums: Converging Practices and Collaboration Opportunities*. Presented at Annual Meeting of the American Society for Information Science and Technology, Vancouver, British Columbia, Canada. (Organizer and moderator of this panel.)

Hsieh-Yee, I. and Coogan, J. (2009, July). *Catching up to Google Scholar: The Retrieval Power of Academic Search Premier and Google Scholar*. Poster presented at American Library Association Conference, Chicago, Illinois.

Hsieh-Yee, I., with the CUA Scholarly Communications Project Team. (2009, January). *Digital Scholarship@CUA: Developing an Institutional Repository for CUA*. Poster presented at 1st Annual Bridging the Spectrum Symposium, Catholic University of America, Washington, D.C.

Wise, M., Cylke, K., and Hsieh-Yee, I. **(**2009, January). *Digital Talking Books: Meeting the Needs of the Blind and the Handicapped*. Paper presented at the Bridging the Spectrum Symposium, Catholic University of America, Washington, D.C.

Hsieh-Yee, I. (2009, January). *User Expectations of MERIC*. Presented at the Information Organization Competencies for the 21st Century Discussion Session of the 2009 Conference of the Association for Library and Information Science Education, Denver, Colorado.

Choi, Y., and Hsieh-Yee, I. (2008,November). *Subject Access for Images in an OPAC.* Annual Meeting of the American Society for Information Science and Technology, Columbus, Ohio. (Also co-organized a panel on Retrieving and Using Visual Resources: Challenges and Opportunities for Research and Education.)

Hsieh-Yee, I. (2008, June). *Educating Cataloging Professionals in a Changing Information Environment*. National Taiwan University, Taipei, Taiwan.

Vellucci, S. L., Moen, W.E., Hsieh-Yee, I., Marson, B., and Wisser, K. (2008, January) *Building a Metadata Education and Research Community through MERIC (Metadata Education and Research Information Commons): Demo and Stakeholder Input*. A panel presented at the 2008 Conference of the Association for Library and Information Science Education, Philadelphia, Pennsylvania.

Hsieh-Yee, I., Choi,Y. and Kules, B. (2007, October). *Searching for Books and Images in OPAC: Effects of LCSH, TOC and Subject Domains.* A poster presented at the American Society for Information Science and Technology Annual Meeting, Milwaukee, Wisconsin.

Hsieh-Yee, I. and Coogan, J. (2007, August) *Google Scholar vs. Academic Search Premier: A Comparative Analysis.* Presented to the Faculty and Staff of the University of the District

of Columbia.

Hsieh-Yee, I. and Coogan, J. (2007, June). *Google Scholar vs. Academic Search Premier: A Comparative Analysis*. Presented to the Washington Research Library Consortium Community, Catholic University of America, Washington, D.C.

Hsieh-Yee, I., Choi, Y., and Kules, B.. (2007, June). *What Users Need for Subject Access: Table of Contents or Subject Headings?* A poster presented at the 2007 American Library Association Annual Conference, Washington, D.C., June 2007.

Choi, Y., Hsieh-Yee, I., and Kules, B. (2007, June). *Retrieval Effectiveness of TOC and LCSH.* A paper presented at the Joint Conference on Digital Libraries 2007, Vancouver, Canada.

Vellucci, S. L., Hsieh-Yee, I., and Moen, W.E. (2007, May). *If We Build It, Will They Come? Building a Community of Practice for Metadata Stakeholders.* A poster presented at the Rutgers University Research Day, Bridgeton, New Jersey.

Hsieh-Yee, I. (2007, May). *Federated Searching: User Experience & Perceptions.* International Conference on Information Organization & Retrieval, National Taiwan University, Taipei, Taiwan.

Hsieh-Yee, I. (2007, May). *Search Performance of Google Scholar and Academic Search Premier.* International Conference on Information Organization & Retrieval, National Taiwan University, Taipei, Taiwan.

Hsieh-Yee, I. (2007, May) *MERIC: Building a Digital Commons for Metadata Education & Research.* International Conference on Information Organization & Retrieval, National Taiwan University, Taipei, Taiwan.

Hsieh-Yee, I., and Coogan, J. (2007, March/April). *A Comparative Analysis of Google Scholar and Academic Search Premier.* Poster presented at the Association of College & Research Libraries 13th National Conference, Baltimore, Maryland.

Vellucci, S. L. and Hsieh-Yee, I. (2007, March/April) *They Didn't Teach Me That in Library School! Building a Digital Teaching Commons to Enhance Metadata Teaching, Learning and Research*. On-site presentation and Webcast by Elluminate. A contributed paper presented at the Association of College & Research Libraries 13th National Conference, Baltimore, Maryland. The acceptance rate for contributed paper was 20%. This paper was one of 10 conference papers chosen for live webcast during the conference.

Moen, W., Hsieh-Yee, I. and Vellucci, S.L. (2007, January) *A DSpace Foundation for a Teaching & Research Commons: The Metadata Education and Research Information Commons.* A poster session presented at the Open Repositories Conference 2007, San Antonio, Texas.

Tang, R., Hsieh-Yee, I., and Zhang, S. (2006, November) *User Perception of MetaLib Combined Search*. Paper presented at the Annual Meeting of the American Society for Information Science and Technology, Austin, Texas, Nov. 2006.

Hsieh-Yee, I. (2006, November). *Federated Searching: User Perceptions, System Design, and Library Instructions.* Paper presented at the Annual Meeting of the American Society for Information Science and Technology, Austin, Texas. (Panel organizer, moderator, presenter).

Hsieh-Yee, I. (2006, November). *Building a Digital Teaching Commons to Enhance Teaching and Learning: The MERIC Experience and Challenges.* Paper presented at the Annual Meeting of the American Society for Information Science and Technology, Austin, Texas. (Panel organizer, moderator, presenter)

Hsieh-Yee, I. (2006, September). *Search Performance of Google Scholar and Academic Search Premier.* Paper presented at the ERIC Publishers Meeting, Washington, D.C.

Hsieh-Yee, I., Zhang, S., and Rong Tang, R. (2006, June). *User Perceptions of a Federated Search System*. Poster presented at Joint Conference on Digital Libraries, Chapel Hill, North Carolina.

Hsieh-Yee, I. and Zhang, S. (2006, June). *Preparing Users for Federated Search: Implications of a MetaLib User Perceptions Study.* Paper presented at the 2006 Ex Libris User Groups of North America Conference, Knoxville, Tennessee.

Hsieh-Yee, I. (2006, January). *MERIC Organizations and Navigation*. Paper presented at the 2006 ALISE Annual Conference, San Antonio, Texas.

Hsieh-Yee, I. (2006, January). *Metadata and Cataloging Education: Recommended Competencies.* Paper presented at the 2006 ALISE Annual Conference, San Antonio, Texas.

Hsieh-Yee, I. (2005, November). *Digital Library Evaluation: Progress & Next Steps.* Presentation at the Annual Meeting of the American Society for Information Science & Technology, Charlotte, North Carolina.

Hsieh-Yee, I. (2005, August). *Providing Access to Digital Content: Issues for DL Managers.* Presentation at MDK12 Digital Library Steering Committee Meeting, Columbia, Maryland.

Hsieh-Yee, I. (2005, April). *Enhancing Teaching and Learning: The Role of School Library Media Specialists.* Presentation at Meeting of the Baltimore County Public School System School Media Specialists, Baltimore, Maryland.

Hsieh-Yee, I. (2005, January). *Subject Access and Users: Insights & Inspirations from Marcia J. Bates.* Paper presented at the Historical Perspectives SIG, 2005 Conference of the Association for Library and Information Science Education, Boston, Massachusetts.

Hsieh-Yee, I. (2005, January). *Electronic Resource Management: Practice, Employer Expectations, & CE Interests.* Paper presented at Technical Services Education SIG, 2005 Conference of the Association for Library and Information Science Education, Boston, Massachusetts.

Hsieh-Yee, I. (2004, October). *Library Professionals for the Digital Age: Competencies & Preparation.* Paper presented at Bibliographic Access Management Team meeting, Library of Congress, Washington, D.C.

Hsieh-Yee, I. (2004, January). *Cataloging and metadata expertise for the digital era.* Presented at Preparing 21st Century Cataloging and Metadata Professionals: A Workshop for Educators and Trainers, San Diego and sponsored by ALCTS, ALISE, LC, and OCLC.

Hsieh-Yee, I. (2004, January). *Educating catalogers for the digital era.* Paper presented at the Technical Services SIG, 2004 Conference of the Association for Library and Information Science Education, San Diego.

Hsieh-Yee, I. (2003, July). *Cataloging Education for the 21st Century.* A presentation at the Library of Congress, Washington, D.C.

Hsieh-Yee, I. (2002, January) *Metadata Education and Research Priorities: A Delphi Study of Metadata Experts.* Presentation at the 2002 Conference of the Association for Library and Information Science Education, New Orleans.

Hsieh-Yee, I. (2001, November). *A Delphi Study of Metadata: Preliminary Findings.* Poster session at the 2001 Annual Meeting of the American Society for Information Science & Technology, Washington, D.C.

Hsieh-Yee, I. (2001, June). *Resources on Asian American Children: Analysis of Retrieval by Search Engines and WorldCat.* Presentation at the National Conference on Asian Pacific American Librarians, San Francisco.

Hsieh-Yee, I. (2001, January). *Delphi Study on Metadata: Project Design.* Presentation at Research Awards Session, Association for Library & Information Science Education, Washington, D.C.

Hsieh-Yee, I. (2000, May). *Web Search Behavior Research: Progress and Implications.* Presentation at the Symposium on Evaluating Library and Information Science Research, University of Wisconsin-Madison, Madison, Wisconsin.

Hsieh-Yee, I. (2000, March). *ERIC User Services: Evaluation in a Decentralized Environment.* Presentation at the National ERIC Joint Directors/Technical Meeting, Arlington, Virginia.

Hsieh-Yee, I. (2000, January). *Enhancing Learning with Web Technology.* Presentation at Faculty Conversations, Catholic University of America, Washington, D.C.

Hsieh-Yee, I. (2000, January). *From Surrogates to Objects: CUA's Approaches to Organizing Electronic Resources.* Paper presentation at the Annual Conference of the Association for Library and Information Science Education, San Antonio, Texas.

Yee, P., and Hsieh-Yee, I. (1997, November). *Individual Differences in Search Behavior on the WWW.* A poster session presented at the 38th Annual Meeting of the Psychonomic Society, Philadelphia, Pennsylvania.

Hsieh-Yee, I. (1997, April). *Research + Marketing + Preparation = Job!* Presented at the "Workshop on Resume and Interview Techniques," Special Libraries Association, Student Chapter, Catholic University of America, Washington, D.C.

Hsieh-Yee, I. (1997, February). *Creating CyberCatalogers: Education and Training.* Presentation at ALA's Midwinter Meeting, Washington, D.C.

Hsieh-Yee, I. (1997, February). *Search Tactics of Web Users in Searching for Texts, Graphics, Known Items and Subjects: A Search Simulation Study.* Presented at the Conference of the Association for Library and Information Science Education, Washington, D.C.

Hsieh-Yee, Ingrid. "Beginning Your Special Library/Information Center Career." Presented at SLA's "Career Day," Jan. 11, 1997, Catholic University of America.

Hsieh-Yee, I. (1996, September). *The Roles of Library and Information Scientists in Managing Electronic Information.* Presentation at Hamilton College, Clinton, New York.

Hsieh-Yee, I. (1996, May). *The Future of Cataloging as a Profession.* Presented at "The Cataloging Forum, Library of Congress, Washington, D.C.

Hsieh-Yee, I. (1994, October). *The Impact of the Internet on OPACs.* Presented at the Third Workshop on User Interfaces for OPACs, Library of Congress, Washington, D.C.

Reports

Hsieh-Yee, I., with Knowledge Management Competencies and Performance Action Group of the Federal Knowledge Management Initiative. "From Knowledge Management Competencies to Improved Organizational Performance." April 9, 2009.

Hsieh-Yee, I., with Knowledge Practices Action Group of the Federal Knowledge Management Initiative. "KM Practice in Government Agencies: Findings and Recommendations." April 9, 2009.

Hsieh-Yee, I. "Delphi Study on Metadata." 2001. Three quarterly reports submitted to the Association for Library and Information Science Education.

Hsieh-Yee, I. "College Students' Information Channels: Patterns of Use and Possible Factors in Channel Selection." 1995. Submitted to the Catholic University of America.

Hsieh-Yee, I. "The Information-Seeking Patterns of Scholars and Their Use of an Online Information System." 1994. Submitted to the Council on Library Resources.

Book Reviews

Review of *The Measurement and Evaluation of Library Services*, by Sharon L. Baker and F. Wilfrid Lancaster. *Information Processing and Management* 30 (1994): 450-52.

Review of *Subject Access to Films and Videos, by Sheila S. Intner and William E. Studwell; and Cataloging Unpublished Nonprint Materials*, by Verna Urbanski with Bao Chu Chang and Bernard L. Karon. *Information Processing and Management* 30 (1994): 449-50.

Review of *Automated Information Retrieval in Libraries: A Management Handbook*, by Vicki Anders. *Journal of Library and Information Science* 19 (1993): 98-100.

Review of *Full Text Databases*, by Carol Tenopir and Jung Soon Ro. *Information Processing and Management* 28 (1992): 667-68.

Review of *Descriptive Cataloging for the AACR2R And USMARC: A How-to-Do It Workbook*, by Larry Millsap and Terry Ellen Ferl. *Information Processing and Management* 28 (1992): 809-11.

Review of *MARC Manual: Understanding and Using MARC Records*, by Deborah J. Byrne. *Information Processing and Management* 28 (1992): 537-38.

Service

Professional Associations and Societies

- Library of Congress. RDA Training Program for the Profession. Co-authored with Tim Carlton. 2013-2014.
- 2014 Digital Preservation Outreach & Education Survey. Contributed to the design of the survey, 2014.
- National Digital Stewardship Alliance. Outreach Committee. 2011-2014.
- National Digital Stewardship Residency Program. Advisory Group, 2012-2013.
- FEDLINK Health Information Technology Advisory Council, 2011-2015.

- 2012 Joint Conference on Digital Libraries. Program Planning Committee, Pre-Conference Proposals Review Committee, 2012
- Catholic Research Resources Alliance. Five-Year Strategic Plan Task Force, 2011-2012
- Institute of Museum and Library Services. Grant reviewer. 2004, 2005, 2010.

- Association for Library and Information Science Education.
  * ALISE Bodan Wynar Research Paper Award Committee, 2015, 2016, 2017
  * ALISE Eugene Garfield Dissertation Award Competition, Jury, 2013, 2014
  * ALISE Research Grant Competition Committee. Chair, 2012
  * Pratt-Severn Faculty Innovation Award. Chair, 2009, 2010
  * ALISE Doctoral Poster Jury, 2012
  * "Information Organization Competencies for the 21st Century" Discussion session leader. 2009 Conference of the Association for Library and Information Science Education.
  * Assisted Technical Services SIG Convener in organizing a program, ""Building a Metadata Education and Research Community through MERIC (Metadata Education and Research Information Commons): Demo and Stakeholder Input" for the 2008 ALISE conference.
  * Association for Library Collections and Technical Services/Association for Library and Information Science Education (ALCTS/ALISE) Metadata Education and Research Information Center (MERIC) Advisory Board, Co-Chair (with Sherry Vellucci), 2005-2007. Chair, 2008-2009 (leading the effort to build MERIC, a repository and collaborative space for metadata educators, practitioners, and researchers)
  * Technical Services SIG, Convener, 2004-2005. Organized a program on "Electronic Resources Management: Current Practices, Employer Expectations, and Teaching Strategies" for the 2005 conference in Boston, Massachusetts.
  * Technical Services SIG, Convener, 2003-2004. Organized a program on "Organizing Information with Metadata: Desired Competencies and Teaching Innovations" for the 2004 conference.
  * Technical Services SIG, Convener, 1999-2000. Organized a program on "Teaching the Organization of Electronic Resources" for the 2000 conference.
  * Curriculum SIG, Co-convener (with Sibyl Moses), 1996-97. Organized a program on "Government Information Policy" for the 1997 conference.

- American Society for Information Science & Technology.
  * Reviewer, Conference program panel submissions and poster submissions, 2005, 2006, 2007, 2009, 2011, 2012, 2013, 2014, 2015, 2016, 2017
  * Nomination Committee, 2009-2011
  * Information Science Education Special Interest Group. American Society for Information Science and Technology. Chair-Elect, 2007-2008. Chair 2008-2009.
  * Committee on Information Science Education. 1999-2006.
  * Committee on Information Science Education. Organizing Committee for an orientation program for students at ASIS annual meetings, 1999-2001
  * Committee on Information Science Education. Sub-committee on Student Welfare (focusing on issues related to master's education), 1998-2001

* SIG ED. Organizing Committee for the "Seminar on Research and Career Development" for junior researchers. 1995-96 (chair), 1997-2001
* ISI Doctoral Dissertation Proposal Scholarship Jury, 1997; 2001, 2002
* Pratt-Severn Best Student Research Paper Award Jury. Chair. 1997
* 1998 Midyear Meeting (referee of contributed papers), 1997
* Organizer and moderator of the ASIS Doctoral Forum and the Doctoral Research Seminar 1994-1995
* SIG Human Computer Interaction. Chair-Elect, Chair, 1993-1995
* Doctoral Forum Award Jury, 1995
* Best Student Paper Award Jury, 1995

- American Library Association.
  * Committee on Accreditation, External Review Panelist, 2009- (site visiting team 2013-2014; site visiting team 2016-2017)
  * Association for Library Collections and Technical Services Task Force on Competencies and Education for a Career in Cataloging, member, 2008-2009
  * Facilitator for "What They Don't Teach in Library School: Competencies, Education and Employer Expectations for a Career in Cataloging," an Association for Library Collections and Technical Services Preconference, June 22, 2007 in Washington, D.C. Also a local liaison for bringing this program to the Catholic University of America.
  * Facilitator for a discussion on "Effect of Electronic Resources on Technical Services" at ALA's Midwinter Meeting held in Feb. 1997 in Washington, D.C.
  * International Relations Committee, Subcommittee Task Force for IFLA and China, 1994-1997

- Virginia Association of School Librarians. Scholarships and Awards Committee. 2010-2012

- Federal Knowledge Management Initiative, Knowledge Management Practices Action Group. Member. 2009 (leading the effort to build a knowledge management repository)
- Federal Knowledge Management Initiative, Knowledge Management Competencies & Learning Action Group. Member. 2009 (developing an action plan for helping government knowledge workers and government agencies to develop knowledge management competencies)
- National Center for Education Statistics. Technical Review Panel. 2008.
- External evaluator for a case of promotion to full professorship. University of Tennessee. 2008.
- National Information Standards Organization (NISO). Advisory Board, Revision of "IMLS Framework of Guidance for Building Good Digital Collections," 2004, 2007.
- Library of Congress, Bibliographic Control of Web Resources: A Library of Congress Action Plan. Principal Investigator of Action Item 5.1, focusing on cataloging and metadata education for students and new librarians, 2002-2003. (worked with the Association for Library Collections and Technical Services, Education Task Force)

- Chinese American Librarians Association
  * Chinese American Librarians Association Outstanding Library Leadership Award in Memory of Dr. Margaret Chang Fung, Award Committee, 2016-2017
  * Achievement Award Jury, 2000-2001
  * CALA Goal 2000 Task Force, 1997
  * Scholarship Committee, 1995, 1996-1997 (chair)
  * Board of Directors, 1994-1997
  * Publication Committee, 1993-1995
  * International Relations Committee, 1993-1996

- Sailor[SM] Assessment Advisory Group (An impact study of Sailor, Maryland's Public Information Network), 1995

- Editorial boards
  Journal of Library & Information Science. Editorial Board, 2012-
  Chinese American Librarians Association, *Occasional Papers Series*. Editorial Board, 2009-2016.
  Library Quarterly. Editorial Board, 2003-2008
  Bulletin of the Medical Library Association, 1994-97
  Newsletter editor for the Chinese American Librarians Association, 1989-92

- Referee for the following journals
  Information Processing and Management
  Journal of Digital Information
  Journal of Education for Library and Information Science
  Journal of Information Science
  Journal of Library & Information Science
  Journal of Library Metadata
  Journal of the American Society for Information Science & Technology
  Library and Information Science Research
  Library Quarterly

- Expert reviewer, "Digital Library" course, Evaluation module, University of North Carolina, Chapel Hill, 2007-2008.
- Expert reviewer, "Information Organization" course, University of Michigan, Ann Arbor. 2007.

Catholic University of America

- Academic Senate Committee on Appointments and Promotions, 2005-2008
- Academic Senate Committee on Committees and Rules, 2009-2012, 2019-2022
- Academic Senate representative (for School of Arts & Sciences), 2017-2020

- Academic Senate representative (for School of Library & Information Science) 2003-2012
- Academic Senate Library Committee, Interim Chair (2007), Member, 2008-2012
- Academic Senate Committee on Computing, 1995-2003

- President's Administrative Council, 2010-2012
- Deans' Council, 2010-2012
- Academic Leadership Group, 2010-2012
- Graduate Board, 2010-2012

- School of Arts & Sciences, Committee on Appointments and Promotions, 2015-2019
- School of Arts & Sciences, Academic Council, 2015-2016.
- School of Arts & Sciences, Ordinary Professor Group, 2013-
- Doctoral Dissertation Defense Committee, Chair, Dept. of Psychology, 2016, 2017, 2018, 2019, 2020, 2021, 2022
- Doctoral Dissertation Defense Committee, Chair, Dept. of Education, 2014, 2015, 2017, 2018, 2019
- Doctoral Dissertation Defense Committee, Chair, National Catholic School of Social Services, 2019, 2020, 2021
- Doctoral Dissertation Defense Committee, Chair, School of Nursing, 2006, 2008
- CUA Scholarly Communication Project Team, Member (2007), Chair, 2008-2009
- Dean Search Committee, 1992-1994, 1998-1999, 2002-2003, 2006-2007
- Fulbright Review Panel, 2006
- CUA Service Learning Advisory Board, 2001-2002
- CUA Faculty Conversations on Enhancing Teaching and Learning through Technology, Planning Group, 1999-2001
- CUA Initiative on Technology and Teaching, 1998-2001

Dept. of Library and Information Science

- Admissions Committee, 2007-2009, Chair 2010-2012, Member 2013-2015, Member 2018-2022
- Appointments and Promotions Committee, 1991-2022
- Student advisement, 1990-2022
- 4+1 Bachelor to Master program, Program Advisor, 2014-2022
- LIS Minor, Advisor, 2017-2022
- Curriculum Committee, 2020-21 (Member), 2021-2022 (Chair)
- Scholarship & Awards Committee, 2016-2018, 2020-21
- Comprehensive examination editor, 2020-21
- Symposium, Colloquium, Lecture Series Committee, May 2018-May 2020 (Chair)
- Symposium and Colloquium Committee, fall 2016-May 2018
- Bridging the Spectrum Symposium session moderator, 2018, 2019, 2020, 2021

- Organization of Information Core Course Outcomes Assessment, 2019, 2020
- The Information Professions in Society Core Course Outcomes Assessment, 2019, 2020
- Community Services Librarianship, Course of Study Review and Revision, 2020
- Organization of Information, Course of Study Review and Revision, 2019
- Organization of Information, Competency and Course Offerings Review, 2018-2019
- Accreditation presentation, Chair, June 2015-August 2016
- Interim Co-Chair, June 2015-August 2016.
- Blended/OWL Learning Committee, spring 2016-2018
- Technology Committee, fall 2016-2017
- Comprehensive examination editor, 2016-2017, reader (every year since 1990)
- LIS Advisory Board, 2015-2016 (chair); fall 2016- May 2018 (member)
- Committee on Planning and Assessment, 2015-2016 (chair)
- Senior Faculty Committee, 2014-2016.
- Accreditation Steering Committee, 2014-2016 (Chair, 2015-2016)
- Accreditation Students Standard Committee, co-chair, 2014-2016
- Accreditation Mission, Goals, and Objectives Standard Committee, co-chair, 2014-2016
- Accreditation Curriculum Standard, member 2014-2-16
- Accreditation Administration and Finance Standard, member 2014-2016
- Cultural Heritage Information Management Project (IMLS-funded), Co-PI, 2012-2015
- Cultural Heritage Information Management Forum (scheduled for June 2015), Co-Organizer, 2013-2015
- Health Information Technology Interim Review Committee, 2015 (chair)
- Health Sciences Librarianship Advisory Group, 2015- (chair)
- Comprehensive examination editors, 2013-2014, 2016-2017
- National Digital Stewardship Alliance liaison, 2011-2014
- Advisory Board, Chair 2010-2012
- Academic Honesty Committee, Chair, 2008-2012
- Blended Learning Committee, 2010-2012
- Colloquium Committee, 2010-2012
- Comprehensive Examination Administration, 2010-2012
- Cultural Heritage Information Management Advisory Committee, 2010-2012 (chair), 2013-
- Curriculum Committee, 1991-2003, 2007-2009, Chair 2010-2012, member 2013-
- Curriculum Subcommittee on Comprehensive Examination, Chair 2009-2012
- Health Information Technology Advisory Board, Chair 2010-2012. Member 2013-
- Health Sciences Advisory Committee, 2009, Chair 2010-2012. Member 2013-
- HIT Expert Forum, Chair 2012. Member 2013-2015
- Health Information Technology Student Group Advisor, 2011-2012
- State Council for Higher Education of Virginia, SLIS Representative, 2010-2012
- Symposium Planning Committee, 2010-2012
- Website Management Team, Chair, 2010-2012
- Urban School Librarianship Project (IMLS-Funded), PI, 2007-2011 (chair, 2010-11)

- Failing Grades Committee, 1995-1997 (chair), 2000-2001 (chair), 2004-2005 (chair), 2007 (chair)-2011
- Faculty Search Committee, 1994-1998, 2002-2004, 2006 (chair), Fall 2007-2009, Chair fall 2009-2012
- Recruitment Committee, Chair 2010-2012
- Strategic Planning Committee, Chair 2010-2012
- Technology Committee, 2010-2012
- Accreditation Advisory Committee, 2007-2009
- Accreditation Coordinating Committee, 2007-2009
- Accreditation Steering Committee, 2007-2009
- SLIS Advisory Group, 2007-2009
- Accreditation Curriculum Standard Committee, Co-chair, 2007-2009
- Accreditation Faculty Standard Committee, Co-chair, 2007-2009
- LSC 551 Information Organization Review Team, Co-chair, 2008-2009, 2015-2016.
- Curriculum Subcommittee on Portfolios, 2009
- LSC 555 Information Systems in Libraries and Information Centers Review Team, contributor, 2008-2009
- Redesign of LSC 730 Use and Users of Libraries and Information. 2009-
- Development of a metadata institute that was taught as LSC 715 Organization of Internet Resources in 2008. The institute is being revised and will be offered in 2010 under a new course title.
- Development of lesson plans, assignments, and evaluation rubrics for LSC 606, Cataloging and Classification, for the School's NCATE accreditation. 2008
- Howard and Mathilde Rovelstad Scholarship Committee, Chair, 2004-2007
- Assistant Dean Search Committee, Chair, Fall 2007
- Liaison to the Association for Library Collections and Technical Services to bring its preconference program, Cataloging Education and Employer Expectations, to CUA during the 2007 American Library Association Annual Meeting in Washington, D.C.
- Organizer of the colloquium presentation and reception for Tamar Sadeh of Ex Libris on PRIMO June 2007
- Practicum review and design (work with potential supervisors, such as the American Indian Museum internship description revision) 2006-
- Comprehensive examinations (edits, proctoring, and grading), 1990-
- SLIS Web site redesign: Comments and suggestions. Fall 2007
- Conducted surveys of current students and alumni in preparation for the 2005 re-accreditation, 2004-2005
- Technology Committee, 1992-1999 (chair, 1996-1998), 2002-2003 (member)
- Colloquia Committee 1997-1999, 2002-2003.
- Advisor of the CUA Student Chapter of the American Society for Information Science and Technology, 2002-2003
- Visiting Professor Search Committee, 1999, 2000, 2001
- Leader, Participation in the CORC experiment, 1999-2000
- Advisor of the Special Libraries Association Student Chapter, 1993-1999; the group was recognized for outstanding leadership by SLA in 1999.

- COA planning Committee, Task Force on Electronic Presentation of SLIS Reports (team leader) 1997-1998
- COA Planning Committee, Subcommittee on Technology 1996-1998
- NLM practicum coordinator, 1997-1998
- Computer Literacy Workshops: Assisted with the development and evaluation of the workshops, 1996-1998
- Leader, Participation in the InterCat project, 1995-1997

APPENDIX B

MARC 21 Reference Materials

Part VII: A Summary of Commonly Used MARC 21 Fields
Part VIII: A List of Other Fields Often Seen in MARC Records
Part IX: The Leader
Part X: Field 008 for Books

Part VII:

A Summary of Commonly Used MARC 21 Fields

This is a summary of the MARC 21 tags used most frequently by libraries in entering their own bibliographic records. For full listings of all MARC 21 tags, indicators, and subfield codes, see *MARC 21 Format for Bibliographic Data*.

In the explanations on these pages:

Tags -- The tags (3-digit numbers) are followed by the names of the fields they represent. In this summary, and in the *MARC 21 Format for Bibliographic Data*, if a tag can appear more than once in one bibliographic record, it is labeled repeatable (R). If it can only be used once, it is labeled non-repeatable (NR). For example, a catalog record can have several subjects, so the tags for subject added entries (6XX) are labeled repeatable (R).

Indicators -- The use of indicators is explained in fields where they are used. Indicators are one-digit numbers. Beginning with the 010 field, in every field -- following the tag -- are two character positions, one for Indicator 1 and one for Indicator 2. The indicators are not actually defined in all fields, however. And it is possible that a 2nd indicator will be used, while the 1st indicator remains undefined (or vice versa). When an indicator is undefined, the character position will be represented by the character # (for blank space).

Subfield codes -- All the data in each field (beginning with the 010 field) is divided into subfields, each of which is preceded by a delimiter-subfield code combination. The most common subfield codes used with each tag are shown. Each subfield code is preceded by the character $, signifying a delimiter. The name of the subfield follows the code.

In general, every field MUST have a subfield 'a' (**$a**). One exception that is often seen is in Field 020 (ISBN), when the ISBN information (subfield **$a**) is unavailable but the price (subfield **$c**) is known. Some subfields are repeatable. In this summary, repeatability is noted for only the more common repeatable subfields.

Examples: Examples follow the explanation for each field. For clarity, one space has been placed between the tag and the first indicator, one space has been placed between the second indicator and the first delimiter- subfield code, and one space has been inserted between the delimiter-subfield code and the subfield data.

010 Library of Congress Control Number -- (LCCN)
(NR, or Not Repeatable
Indicators undefined.

Subfield used most often:

$a -- Library of Congress control number

```
Example:    010 ##  $a ###86000988#
```

020 International Standard Book Number -- (ISBN)
(R, or Repeatable)

Indicators undefined.

Subfields used most often:

$a -- International Standard Book Number
$c -- Terms of availability (often a price)
$z -- Cancelled/invalid ISBN (R)

```
Example:    020 ##  $a 0877547637
```

040 Cataloging source -- (NR)

Indicators undefined.

Subfields used most often:

$a -- Original cataloging agency
$c -- Transcribing agency
$d -- Modifying agency (R)

```
Example:    040 ## $a DLC
                   $c DLC
                   $d gwhs
```

100 Main entry -- Personal name -- (primary author)
(NR; there can be only one main entry)

Indicator 1: Type of personal name entry element

0 -- Forename
1 -- Surname (this is the most common form)
3 -- Family name

Indicator 2 undefined.

Indicator 2 became obsolete in 1990. Older records may display 0 or 1

Subfields used most often:

$a -- Personal name
$b -- Numeration
$c -- Titles and other words associated with a name (R)
$q -- Fuller form of name
$d -- Dates associated with a name (generally, year of birth)

```
Example:    100 1# $a Gregory, Ruth W.
```

```
                    $q (Ruth Wilhelme),
                    $d 1910-
```

130 Main entry -- Uniform title -- (NR)

Indicator 1: Nonfiling characters

0-9 -- Number of nonfiling characters present (for initial articles, including spaces)

Indicator 2 undefined.

Indicator 2 became obsolete in 1990. (See 100 above.)

Subfields used most often:

$a -- Uniform title
$p -- Name of part/section of a work (R)
$l -- Language of a work
$s -- Version
$f -- Date of a work

```
Example:    130 0# $a Bible.
                   $p O.T.
                   $p Psalms.
```

240 Uniform title (NR)

Indicator 1: Uniform title printed or displayed

0 -- Not printed or displayed
1 -- Printed or displayed (most common)

Indicator 2: Nonfiling characters

0-9 -- Number of nonfiling characters present (for initial articles, including spaces)

Subfields used most often:

$a -- Uniform title
$l -- Language of a work
$f -- Date of a work

```
Example:    240 10 $a Ile mystérieuse.
                   $l English.
                   $f 1978
```

245 Title Statement (NR)

Indicator 1: Title added entry

(Should the title be indexed as a title added entry?)

0 -- No title added entry

(indicates a title main entry; i.e. no author is given)

1 -- Title added entry

(the proper indicator when an author given in 1XX; the most common situation)

Indicator 2: Nonfiling characters

0-9 -- Number of nonfiling characters present, including spaces; usually set at zero, except when the title begins with an article; e.g., for *The robe*, the second indicator would be set to 4. The letters *T*, *h*, *e*, and the space following them are then ignored in alphabetizing titles. The record will be automatically filed under "*r*" -- for *Robe*.

Subfields used most often:

$a -- Title proper
$h -- Medium (often used for non-book media)
$p -- Name of part/section of a work (R)
$b -- Reminder of title (subtitles, etc.)
$c -- Remainder of title page transcription/Statement of responsibility

```
Example:    245 14 $a The DNA story :
                   $b a documentary history of gene
                      cloning /
                   $c James D. Watson, John Tooze.
```

246 Varying form of title (R)

Indicator 1: Note/title added entry controller
1 -- Note, title added entry
3 -- No note, title added entry

Indicator 2: Type of title
-- No information provided
0 -- Portion of title
1 -- Parallel title
4 -- Cover title
8 -- Spine title

Subfield used most often:
$a -- Title proper

```
Example:    246 3# $a Four corners power review
```

250 Edition statement (NR)

Indicators undefined.

Subfield used most often:
$a -- Edition statement

```
Example:    250 ## $a 6th ed.
```

260 Publication, distribution, etc. (Imprint) (R)

Indicator 1: Sequence of publishing statements
-- No information provided

Indicator 2: Undefined

Subfields used most often:

$a -- Place of publication, distribution, etc. (R)
$b -- Name of publisher, distributor, etc. (R)
$c -- Date of publication, distribution, etc. (R)

```
Example:    260 ## $a New York :
                   $b Chelsea House,
                   $c 1986.
```

300 Physical description (R)

Indicators undefined.
Subfields used most often:

$a -- Extent (number of pages) (R)
$b -- Other physical details (usually illustration information)
$c -- Dimensions (cm.) (R)
$e -- Accompanying material (for example, "teacher's guide" or "manual")

```
Example:    300 ## $a 139 p. :
                   $b ill. ;
                   $c 24 cm.
```

440 Series statement / Added entry--Title

This field was made obsolete in 2008 to simplify the series statement. See 490 and 830.

490 Series statement (No added entry is traced from field) (R)

Indicator 1: Specifies whether series is traced (whether an 8XX tag is also present)

0 -- Series not traced
1 -- Series traced (8XX is in record)

Indicator 2 undefined.
Subfield used most often:

$a -- Series statement (R)
$v -- Volume number (R)

```
Example:    490 1# $a Colonial American craftsmen
```

500 General note (R)

Indicators undefined.
Subfield used most often:

$a -- General note (Used when no specialized note field has been defined for the information. Examples: Notes regarding the index; the source of the title; variations in title; descriptions of the nature, form, or scope of the item.)

```
Example:    500 ##  $a Includes index.
```

504 Bibliography, etc. note (R)

Indicators undefined.

Subfield used most often:

$a -- Bibliography, etc. note

```
Example:    504 ## $a Includes bibliographical
                     references.
```

505 Formatted contents note (R)

Indicator 1: Type of contents note

0 -- Complete contents

1 -- Incomplete contents (used with multivolume set when some volumes are not yet published)

2 -- Partial contents

Indicator 2: Level of content designation

-- Basic

Subfield used most often:

$a -- Formatted contents note

```
Example:    505 0# $a Pride and prejudice -- Emma
                     -- Northanger Abbey.
```

520 Summary, etc. note (R)

Indicator 1: Display constant controller

-- Summary

1 -- Review

2 -- Scope and content

3 -- Abstract

Indicator 2 undefined

Subfields used most often

$a -- Summary, abstract, or annotation

$b -- Expansion of summary note

```
Example:    520 ## $a This basic guide to parliamentary
                     procedure tells how to conduct
                     and participate in a meeting
                     properly.
```

600 Subject added entry -- Personal name (R)

Indicator 1: Type of personal name entry element

0 -- Forename

1 -- Surname (this is the most common form)
3 -- Family name

Indicator 2: Subject heading system/thesaurus (identifies the specific list or file which was used)

0 -- Library of Congress Subject Headings
1 -- LC subject headings for children's literature
2 -- Medical Subject Headings
3 -- National Agricultural Library subject authority file
4 -- Source not specified
5 -- Canadian Subject Headings
6 -- Répertoire de vedettes-matière
7 -- Source specified in subfield $2

(Note regarding Sears subject headings: The MARC 21 format does not provide an assigned indicator for Sears subject headings. Therefore, an indicator of 7 is used, and the MARC defined code "sears" is placed in subfield $2.)

Subfields used most often:

$a -- Personal name (surname and forename)
$b -- Numeration
$c -- Titles and other words associated with a name (R)
$q -- Fuller form of name
$d -- Dates associated with a name (generally, year of birth)
$t -- Title of a work
$v -- Form subdivision (R)
$x -- General subdivision (R)
$y -- Chronological subdivision (R)
$z -- Geographic subdivision (R)
$2 -- Source of heading or term (used with 2nd indicator of 7)

```
Example:    600 10 $a Shakespeare, William,
                   $d 1564-1616
                   $x Comedies
                   $x Stage history.
```

```
Example:    600 10 $a Shakespeare, William,
                   $d 1564-1616
                   $x Knowledge
                   $z Rome
                   $v Congresses.
```

Notice that subfields $v, $x, and $z in the 600 field are repeatable. Subfields $v, $x, $y, and $z do not have to be in alphabetical order. They will be in the order prescribed by the instructions given by the subject heading system.

610 Subject added entry -- Corporate name (R)

*Indicator 1:*Type of corporate name entry element

0 -- Inverted name (not used with AACR2)
1 -- Jurisdiction name
2 -- Name in direct order

Indicator 2: Subject heading system/thesaurus.
See indicator 2 under 600

Subfields used most often:
$a -- Corporate name or jurisdiction name as entry element
$b -- Subordinate unit (R)
$v -- Form subdivision (R)
$x -- General subdivision (R)
$y -- Chronological subdivision (R)
$z -- Geographic subdivision (R)
$2 -- Source of heading or term (used with 2nd indicator of 7)

```
Example:    610 10 $a United States.
                   $b Army Air Forces
                   $v Biography.
```

650 Subject added entry -- Topical term (Most subject headings fit here.) (R)

Indicator 1: Level of subject
-- No information provided

Indicator 2: Subject heading system/thesaurus
(identifies the specific list or file which was used)
0 -- Library of Congress Subject Headings
1 -- LC subject headings for children's literature
2 -- Medical Subject Headings
3 -- National Agricultural Library subject authority file
4 -- Source not specified
5 -- Canadian Subject Headings
6 -- Répertoire de vedettes-matière
7 -- Source specified in subfield $2

Note regarding Sears subject headings: The MARC 21 format does not provide an assigned indicator for Sears subject headings. Therefore, an indicator of 7 is used, and the MARC defined code "sears" is placed in subfield $2.)

Subfields used most often:
$a -- Topical term
$v -- Form subdivision (R)
$x -- General subdivision (R)
$y -- Chronological subdivision (R)
$z -- Geographic subdivision (R)
$2 -- Source of heading or term used with 2nd indicator of 7)

```
Example:    650 #0 $a Theater
                   $z United States
                   $v Biography
                   $v Dictionaries.
```

Notice that subfields $v, $x, and $z in the 650 field are repeatable. Subfields $v, $x, $y, and $z do not have to be in alphabetical order. They will be in the order prescribed by the instructions given by the subject heading system.

651 Subject added entry -- Geographic name (R)

Indicator 1: undefined.

Indicator 2: Subject heading system/thesaurus.

See indicator 2 under 600

Subfields used most often:

$a -- Geographic name
$v -- Form subdivision (R)
$x -- General subdivision (R)
$y -- Chronological subdivision (R)
$z -- Geographic subdivision (R)
$2 -- Source of heading or term (used with 2nd indicator of 7)

```
Example:    651 #0 $a United States
                   $x History
                   $v Chronology.
```

Notice that subfields $v, $x, and $z in the 651 field are repeatable. Subfields $v, $x, $y, and $z do not have to be in alphabetical order. They will be in the order prescribed by the instructions given by the subject heading system.

700 Added entry -- Personal name (R)

Indicator 1: Type of personal name entry element

0 -- Forename
1 -- Surname (this is the most common form)
3 -- Family name

Indicator 2: Type of added entry

-- No information provided (most common; co-authors, editors, etc.)
2 -- Analytical entry (The values for Indicator 2 changed in 1994 with Format Integration, and older records may display additional values. An analytical entry involves an author/title of an item contained in a work.)

Subfields used most often:

$a -- Personal name
$b -- Numeration
$c -- Titles and other words associated with a name (R)
$q -- Fuller form of name
$d -- Dates associated with a name (generally, year of birth)
$e -- Relator term (such as ill.) (R)
$4 -- Relator code (R)

```
Example:    700 1# $a Baldridge, Letitia.
```

710 Added entry -- Corporate name (R)

Indicator 1: Type of corporate name entry element

0 -- Inverted name (not used with AACR2)

1 -- Jurisdiction name

2 -- Name in direct order

Indicator 2: Type of added entry.

See Indicator 2 under 700

-- No information provided

2 -- Analytical entry

Subfields used most often:

$a -- Corporate name or jurisdiction name as entry element

$b -- Subordinate unit (R)

```
Example:    710 2# $a Sunburst Communications (Firm)
```

740 Added entry -- Uncontrolled related/analytical title (R)

Indicator 1: Nonfiling characters

0-9 -- Number of nonfiling characters present (for initial articles, including spaces)

*Indicator 2:*Type of added entry. See Indicator 2 under 700

-- No information provided

2 -- Analytical entry

(This field was redefined in 1994 with Format Integration. Prior to 1994, the field was also used for variant titles, such as a different wording on a spine title. In records created since Format Integration, those variant tiles appear in a 246 field.)

Subfield used most often:

$a -- Title

```
Example:    740 02 $a Uncle Vanya.
```

800 Series added entry -- Personal name (R)

Indicator 1: Type of personal name entry element

0 -- Forename

1 -- Surname

3 -- Family name

Indicator 2 undefined.

Subfields used most often:

$a -- Personal name

$b -- Numeration

$c -- Titles and other words associated with a name (R)

$q -- Fuller form of name

$d -- Dates associated with a name (generally, year of birth)
$t -- Title of a work (the series)
$v -- Volume number

```
Example:    800 1# $a Fisher, Leonard Everett.
                  $t Colonial American craftsmen.
```

830 Series added entry -- Uniform title (R)

Indicator 1 undefined.
Indicator 2: Nonfiling characters
0-9 -- Number of nonfiling characters present (for initial articles, including spaces)
Subfield used most often:
$a -- Uniform title
$v -- Volume number

```
Example:    830 #0 $a Railroads of America (Macmillan)
```

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Part VIII:

A List of Other Fields Often Seen in MARC Records

001 Control number
003 Control number identifier
005 Date and time of latest transaction
006 Fixed-length data elements -- additional material characteristics
007 Physical description fixed field
008 Fixed length data elements (See Part X)
022 International Standard Serial Number (ISSN)
037 Source of acquisition
041 Language code
043 Geographic area code
050 Library of Congress call number
060 National Library of Medicine call number
082 Dewey Decimal classification number (the one recommended by the Library of Congress; locally-assigned call numbers may appear elsewhere)
110 Main entry -- Corporate name (less frequent under AACR2 rules)
256 Computer file characteristics
263 Projected publication date (indicates a CIP -- Cataloging in Publication -- record)

306 Playing time
508 Creation/production credits note
510 Citation/references note (review sources)
511 Participant or performer note
521 Target audience note (first indicator: 0 = reading grade level, 1 = interest age level, 2 = interest grade level, 3 = special audience characteristics, 4 = motivation interest level)
530 Additional physical form available note
538 System details note
586 Awards note
656 Index term -- Occupation
730 Added entry -- Uniform title
852 Location
856 Electronic location and access
9XX Reserved for local use. (They are used by vendors, systems, or individual libraries to exchange additional data)

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Part IX:

The Leader

There are 24 positions in the Leader, numbered from 00 to 23. For fuller explanation, see the *MARC 21 Format for Bibliographic Data*.

00-04 Record length (calculated by the computer for each record)
05 Record status
- a = increase in encoding level
- c = corrected or revised
- d = deleted
- n = new
- p = increase in encoding from prepublication (previous CIP)

06 Type of record
- a = language material
- c = printed music
- d = manuscript music
- e = cartographic material
- f = manuscript cartographic material
- g = projected medium
- i = nonmusical sound recording
- j = musical sound recording
- k = 2-dimensional nonprojectable graphic
- m = computer file
- o = kit
- p = mixed materials
- r = 3-dimensional artifact or naturally occurring object
- t = manuscript language material

07 Bibliographic level
- a = monographic component part
- b = serial component part

c = collection
d = subunit
i = integrating resource
m = monograph/item
s = serial

08 Type of control
= no specified type
a = archival

09 Character coding scheme
= MARC-8
a = UCS/Unicode

10 Indicator count (always "2")

11 Subfield code count (always "2")

12-16 Base address of data (calculated by the computer for each record)

17 Encoding level
= full level
1 = full level, material not examined
2 = less-than-full level, material not examined
3 = abbreviated level
4 = core level
5 = partial (preliminary) level
7 = minimal level
8 = prepublication level (CIP)
u = unknown
z = not applicable

18 Descriptive cataloging form
= non-ISBD
a = AACR2
i = ISBD
u = unknown

19 Multipart resource record level
= Not specified or not applicable
a = Set
b = Part with independent title
c = Part with dependent title

20 Length of the length-of-field portion (always "4")

21 Length of the starting-character-position portion (always "5")

22 Length of the implementation-defined portion (always "0")

23 Undefined (always "0")

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Part X:

Field 008 for Books

Field 008 is used for Fixed Length Data Elements ("Fixed Field Codes"). There are 40 character positions in field 008, numbered from 00-39. Undefined positions must contain either a blank (#) or a fill character (|). Positions 00-17 and 35-39 are defined the same way for all media.

The information shown here for positions 18-34 applies only to books. For explanation of all the positions below and for positions 18-34 for other media, see the *MARC 21 Format for Bibliographic Data*.

Note that field 008 has no indicators or subfield codes.

00-05 Date entered on file (YYMMDD),
where Y=year, M=month, and D=day

06 Type of date/publication status:
- b = no dates given; B.C. date involved
- e = detailed date
- s = single known date/probable date
- m = multiple dates
- r = reprint/reissue date (Date 1) and original date (Date 2)
- n = dates unknown
- q = questionable date
- t = publication date and copyright date
- | = no attempt to code

07-10 Date 1/beginning date of publication

11-14 Date 2/ending date of publication

Date fields contain the year(s) of publication. The type of date(s) in these elements are specified in fixed field element 06: Type of date/publication status. (For further details, see the field 008 description in the *MARC 21 Format for Bibliographic Data*.)

15-17 Place of publication, production, or execution
For example:
- pk# = Pakistan
- cau = California (US)

(For a full list of codes used in these positions, see the *MARC Code List for Countries*.)

18-21 Illustrations (up to 4 codes):
- # = no illustrations
- a = illustrations
- b = maps
- c = portraits
- d = charts
- e = plans
- f = plates
- g = music
- h = facsimiles
- i = coats of arms
- j = genealogical tables
- k = forms
- l = samples
- m = phonodisc, phonowire, etc.
- o = photographs
- p = illuminations
- | = no attempt to code

22 Target audience:
- # = unknown or not specified
- a = preschool

b = primary
c = pre-adolescent
d = adolescent
e = adult
f = specialized
g = general
j = juvenile
| = no attempt to code

23 Form of item:
= none of the following
a = microfilm
b = microfiche
c = microopaque
d = large print
f = braille
r = regular print reproduction
s = electronic
| = no attempt to code

24-27 Nature of contents (up to 4):
= no specified nature of contents
a = abstracts/summaries
b = bibliographies (is one or contains one)
c = catalogs
d = dictionaries
e = encyclopedias
f = handbooks
g = legal articles
i = indexes
j = patent document
k = discographies
l = legislation
m = theses
n = surveys of literature
o = reviews
p = programmed texts
q = filmographies
r = directories
s = statistics
t = technical reports
u = standards/specifications
v = legal cases and notes
w = law reports and digests
z = treaties
| = no attempt to code

28 Government publication:
= not a government publication
i = international intergovernmental
f = federal/national
a = autonomous or semi-autonomous component
s = state, provincial, territorial, dependent, etc.
m = multistate
c = multilocal

l = local
z = other type of government publication
o = government publication -- level undetermined
u = unknown if item is government publication
| = no attempt to code

29 Conference publication:
0 = not a conference publication
1 = conference publication
| = no attempt to code

30 Festschrift:
0 = not a festschrift
1 = festschrift
| = no attempt to code

31 Index:
0 = no index
1 = index present
| = no attempt to code

32 Undefined (since 1990) (Earlier records may contain the values 0 or 1)
= Undefined
| = no attempt to code

33 Literary form:
0 = not fiction (not further specified)
1 = fiction (not further specified)
c = comic strips
d = dramas
e = essays
f = novels
h = humor, satires, etc.
i = letters
j = short stories
m = mixed forms
p = poetry
s = speeches
u = unknown
| = no attempt to code

34 Biography:
= no biographical material
a = autobiography
b = individual biography
c = collective biography
d = contains biographical information
| = no attempt to code

35-37 Language:
A three-letter code. For example: eng fre ger spa rus ita

(For a full list of codes used in these positions, see the *MARC Code List for Languages*.)

38 Modified record:
= not modified
x = missing characters (because of characters unavailable in MARC character set)
s = shortened
d = "dashed-on" information omitted

r = completely romanized/printed cards in script
o = completely romanized/printed cards romanized
| = no attempt to code

39 Cataloging source:
= national bibliographic agency
c = cooperative cataloging program
d = other sources
u = unknown
| = no attempt to code

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APPENDIX 1003A

DIGITAL SIGNAL PROCESSING

M. Brandstein · D. Ward (Eds.)

Microphone Arrays

Signal Processing Techniques and Applications



Springer

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The study and implementation of microphone arrays originated over 20 years ago. Thanks to the research and experimental developments pursued to the present day, the field has matured to the point that array-based technology now has immediate applicability to a number of current systems and a vast potential for the improvement of existing products and the creation of future devices. This text is organized into four sections which roughly follow the major areas of microphone array research today. Parts I and II are primarily theoretical in nature and emphasize the use of microphone arrays for speech enhancement and source localization, respectively. Part III presents a number of specific applications of array-based technology. Part IV addresses some open questions and explores the future of the field. The result is a text that will be of utility to a large audience, from the student or practicing engineer just approaching the field to the advanced researcher with multi-channel signal processing experience.

ISBN 978-3-540-41953-2
90000
9 783540 419532
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Microphone Arrays

Signal Processing Techniques and Applications



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ISBN 3-540-41953-5 Springer-Verlag Berlin Heidelberg New York

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Printed in Germany

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Typesetting: Camera-ready copy by authors
Cover-Design: de'blik, Berlin
SPIN: 10836055 62/3020 5 4 3 2 1 0 Printed on acid-free paper

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Preface

The study and implementation of microphone arrays originated over 20 years ago. Thanks to the research and experimental developments pursued to the present day, the field has matured to the point that array-based technology now has immediate applicability to a number of current systems and a vast potential for the improvement of existing products and the creation of future devices.

In putting this book together, our goal was to provide, for the first time, a single complete reference on microphone arrays. We invited the top researchers in the field to contribute articles addressing their specific topic(s) of study. The reception we received from our colleagues was quite enthusiastic and very encouraging. There was the general consensus that a work of this kind was well overdue. The results provided in this collection cover the current state of the art in microphone array research, development, and technological application.

This text is organized into four sections which roughly follow the major areas of microphone array research today. Parts I and II are primarily theoretical in nature and emphasize the use of microphone arrays for speech enhancement and source localization, respectively. Part III presents a number of specific applications of array-based technology. Part IV addresses some open questions and explores the future of the field.

Part I concerns the problem of enhancing the speech signal acquired by an array of microphones. For a variety of applications, including human-computer interaction and hands-free telephony, the goal is to allow users to roam unfettered in diverse environments while still providing a high quality speech signal and robustness against background noise, interfering sources, and reverberation effects. The use of microphone arrays gives one the opportunity to exploit the fact that the source of the desired speech signal and the noise sources are physically separated in space. Conventional array processing techniques, typically developed for applications such as radar and sonar, were initially applied to the hands-free speech acquisition problem. However, the environment in which microphone arrays is used is significantly different from that of conventional array applications. Firstly, the desired speech signal has an extremely wide bandwidth relative to its center frequency, meaning that conventional narrowband techniques are not suitable. Secondly, there

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is significant multipath interference caused by room reverberation. Finally, the speech source and noise signals may located close to the array, meaning that the conventional far-field assumption is typically not valid. These differences (amongst others) have meant that new array techniques have had to be formulated for microphone array applications. Chapter 1 describes the design of an array whose spatial response does not change appreciably over a wide bandwidth. Such a design ensures that the spatial filtering performed by the array is uniform across the entire bandwidth of the speech signal. The main problem with many array designs is that a very large physical array is required to obtain reasonable spatial resolution, especially at low frequencies. This problem is addressed in Chapter 2, which reviews so-called superdirective arrays. These arrays are designed to achieve spatial directivity that is significantly higher than a standard delay-and-sum beamformer. Chapter 3 describes the use of a single-channel noise suppression filter on the output of a microphone array. The design of such a post-filter typically requires information about the correlation of the noise between different microphones. The spatial correlation functions for various directional microphones are investigated in Chapter 4, which also describes the use of these functions in adaptive noise cancellation applications. Chapter 5 reviews adaptive techniques for microphone arrays, focusing on algorithms that are robust and perform well in real environments. Chapter 6 presents optimal spatial filtering algorithms based on the generalized singular-value decomposition. These techniques require a large number of computations, so the chapter presents techniques to reduce the computational complexity and thereby permit real-time implementation. Chapter 7 advocates a new approach that combines explicit modeling of the speech signal (a technique which is well-known in single-channel speech enhancement applications) with the spatial filtering afforded by multi-channel array processing.

Part II is devoted to the source localization problem. The ability to locate and track one or more speech sources is an essential requirement of microphone array systems. For speech enhancement applications, an accurate fix on the primary talker, as well as knowledge of any interfering talkers or coherent noise sources, is necessary to effectively steer the array, enhancing a given source while simultaneously attenuating those deemed undesirable. Location data may be used as a guide for discriminating individual speakers in a multi-source scenario. With this information available, it would then be possible to automatically focus upon and follow a given source on an extended basis. Of particular interest lately, is the application of the speaker location estimates for aiming a camera or series of cameras in a video-conferencing system. In this regard, the automated localization information eliminates the need for a human or number of human camera operators. Several existing commercial products apply microphone-array technology in small-room environments to steer a robotic camera and frame active talkers. Chapter 8 summarizes the various approaches which have been explored to accurately locate an individ-

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ual in a practical acoustic environment. The emphasis is on precision in the face of adverse conditions, with an appropriate method presented in detail. Chapter 9 extends the problem to the case of multiple active sources. While again considering realistic environments, the issue is complicated by the presence of several talkers. Chapter 10 further generalizes the source localization scenario to include knowledge derived from non-acoustic sensor modalities. In this case both audio and video signals are effectively combined to track the motion of a talker.

Part III of this text details some specific applications of microphone array technology available today. Microphone arrays have been deployed for a variety of practical applications thus far and their utility and presence in our daily lives is increasing rapidly. At one extreme are large aperture arrays with tens to hundreds of elements designed for large rooms, distant talkers, and adverse acoustic conditions. Examples include the two-dimensional, harmonic array installed in the main auditorium of Bell Laboratories, Murray Hill and the 512-element Huge Microphone Array (HMA) developed at Brown University. While these systems provide tremendous functionality in the environments for which they are intended, small arrays consisting of just a handful (usually 2 to 8) of microphones and encompassing only a few centimeters of space have become far more common and affordable. These systems are intended for sound capture in close-talking, low to moderate noise conditions (such as an individual dictating at a workstation or using a hands-free telephone in an automobile) and have exhibited a degree of effectiveness, especially when compared to their single microphone counterparts. The technology has developed to the point that microphone arrays are now available in off-the-shelf consumer electronic devices available for under $150. Because of their growing popularity and feasibility we have chosen to focus primarily on the issues associated with small-aperture devices. Chapter 11 addresses the incorporation of multiple microphones into hearing aid devices. The ability of beamforming methods to reduce background noise and interference has been shown to dramatically improve the speech understanding of the hearing impaired and to increase their overall satisfaction with the device. Chapter 12 focuses on the case of a simple two-element array combined with postfiltering to achieve noise and echo reduction. The performance of this configuration is analyzed under realistic acoustic conditions and its utility is demonstrated for desktop conferencing and intercom applications. Chapter 13 is concerned with the problem of acoustic feedback inherent in full-duplex communications involving loudspeakers and microphones. Existing single-channel echo cancellation methods are integrated within a beamforming context to achieve enhanced echo suppression. These results are applied to single- and multi-channel conferencing scenarios. Chapter 14 explores the use of microphone arrays for sound capture in automobiles. The issues of noise, interference, and echo cancellation specifically within the car environment are addressed and a particularly effective approach is detailed. Chapter 15 discusses the applica-

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tion of microphone arrays to improve the performance of speech recognition systems in adverse conditions. Strategies for effectively coupling the acoustic signal enhancements afforded through beamforming with existing speech recognition techniques are presented. A specific adaptation of a recognizer to function with an array is presented. Finally, Chapter 16 presents an overview of the problem of separating blind mixtures of acoustic signals recorded at a microphone array. This represents a very new application for microphone arrays, and is a technique that is fundamentally different to the spatial filtering approaches detailed in earlier chapters.

In the final section of the book, Part IV presents expert summaries of current open problems in the field, as well as personal views of what the future of microphone array processing might hold. These summaries, presented in Chapters 17 and 18, describe both academically-oriented research problems, as well as industry-focused areas where microphone array research may be headed.

The individual chapters that we selected for the book were designed to be tutorial in nature with a specific emphasis on recent important results. We hope the result is a text that will be of utility to a large audience, from the student or practicing engineer just approaching the field to the advanced researcher with multi-channel signal processing experience.

Cambridge MA, USA — *Michael Brandstein*
London, UK — *Darren Ward*
January 2001

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Part I

Speech Enhancement

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1 Constant Directivity Beamforming

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Abstract. Beamforming, or spatial filtering, is one of the simplest methods for discriminating between different signals based on the physical location of the sources. Because speech is a very wideband signal, covering some four octaves, traditional narrowband beamforming techniques are inappropriate for hands-free speech acquisition. One class of broadband beamformers, called constant directivity beamformers, aim to produce a constant spatial response over a broad frequency range. In this chapter we review such beamformers, and discuss implementation issues related to their use in microphone arrays.

1.1 Introduction

Beamforming is one of the simplest and most robust means of *spatial filtering*, i.e., discriminating between signals based on the physical locations of the signal sources [1]. In a typical microphone array environment, the desired speech signal originates from a talker's mouth, and is corrupted by interfering signals such as other talkers and room reverberation. Spatial filtering can be useful in such an environment, since the interfering sources generally originate from points in space separate from the desired talker's mouth. By exploiting the spatial dimension of the problem, microphone arrays attempt to obtain a high-quality speech signal without requiring the talker to speak directly into a close-talking microphone.

In most beamforming applications two assumptions simplify the analysis: (i) the signals incident on the array are narrowband (the *narrowband assumption*); and (ii) the signal sources are located far enough away from the array that the wavefronts impinging on the array can be modeled as plane waves (the *farfield assumption*). For many microphone array applications, the farfield assumption is valid. However, the narrowband assumption is never valid, and it is this aspect of the beamforming problem that we focus on in this chapter (see [2] for techniques that also lift the nearfield assumption).

To understand the inherent problem in using a narrowband array for broadband signals, consider a linear array with a fixed number of elements separated by a fixed inter-element distance. The important dimension in measuring array performance is its size in terms of operating wavelength. Thus for high frequency signals (having a small wavelength) a fixed array will appear large and the main beam will be narrow. However, for low frequencies

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2 Superdirective Microphone Arrays

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Abstract. This chapter gives an overview of so-called superdirective beamformers, which can be derived by applying the minimum variance distortionless response (MVDR) principle to theoretically well-defined noise fields, as for example the diffuse noise field. We show that all relevant performance measures for beamformer designs are functions of the coherence matrix of the noise field. Additionally, we present unconstrained and constrained MVDR-solutions using modified coherence functions. Solutions for different choices of the optimization criterion are given including a new solution to optimize the front-to-back ratio. Finally, we present a comparison of superdirective beamformers to gradient microphones and an alternative generalized sidelobe canceler (GSC) implementation of the superdirective beamformer.

2.1 Introduction

What is "super" about a superdirective microphone array? Compared to the standard delay-and-sum beamformer a superdirective array achieves a higher directivity. Therefore, "super"-directivity indicates that summing is not the optimal choice for combining sensor signals, if optimal directivity is desired. The term directivity describes the ability of a beamformer to suppress noise coming from all directions without affecting a desired signal from one principal direction.

A short historical overview in [6] shows that superdirectivity (or supergain) in connection with array processing was first mentioned in the first half of the last century. The solutions provided at that time were of academic interest only, since a lot of practical problems occurred which restricted the use of the theoretical work. The main reasons for failure were the self-noise and the gain and phase errors of the microphones. In order to overcome these problems a first constrained solution was published by Gilbert and Morgan in 1955 [15]. Early applications with slight modifications were seismic and sonar techniques [5]. It was not until the 90's that supergain was connected to microphone applications. Research in hearing aids highlighted the advantages of fixed beamformers over adaptive solutions [17]. Modern designs of superdirective beamformers include nearfield assumptions and the possibility to adapt the constraining to the actual problem.

This chapter is organized as follows: Section 2.2 introduces the measures to judge the different designs. In section 2.3 the optimal design will be derived

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3 Post-Filtering Techniques

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Abstract. In the context of microphone arrays, the term post-filtering denotes the post-processing of the array output by a single-channel noise suppression filter. A theoretical analysis shows that Wiener post-filtering of the output of an optimum distortionless beamformer provides a minimum mean squared error solution. We examine published methods for post-filter estimation and develop a new algorithm. A simulation system is presented to compare the performance of the discussed algorithms.

3.1 Introduction

What can be gained by additional post-filtering if the Minimum Variance Distortionless Response (MVDR) beamformer already provides the optimum solution for a given sound field?

Assuming that signal and noise are mutually uncorrelated the MVDR beamformer minimizes the noise power (or variance) subject to the constraint of a distortionless look direction response. The solution can be shown to be optimum in the Maximum Likelihood (ML) sense and produces the best possible Signal to Noise Ratio (SNR) for a narrowband input [1]. However, it does not maximize the SNR for a broadband input such as speech. Furthermore, the MVDR beamformer does not provide a broadband Minimum Mean Squared Error (MMSE) solution. The best possible linear filter in the MMSE sense is the multi-channel Wiener filter. As shown below the broadband multi-channel MMSE solution can be factorized into a MVDR beamformer followed by a single-channel Wiener post-filter. The multi-channel Wiener filter generally produces a higher output SNR than the MVDR filter. Therefore, additional post-filtering can significantly improve the SNR, which motivates this chapter.

The squared error minimized by the single-channel Wiener filter is the sum of residual noise and signal distortion components at the output of the filter. As a result, linear distortion of the desired signal cannot be avoided entirely if Wiener filtering is used. Additional Wiener filtering is advantageous in practice, however, because signal distortions can be masked by residual noise and a compromise between signal distortion and noise suppression can be found. Using MVDR beamforming alone often does not provide sufficient noise reduction due to its limited ability to reduce diffuse noise and reverberation.

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4 Spatial Coherence Functions for Differential Microphones in Isotropic Noise Fields

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Abstract. The spatial correlation function between directional microphones is useful in the design and analysis of the performance of these microphones in actual acoustic noise fields. These correlation functions are well known for omnidirectional receivers, but not well known for directional receivers. This chapter investigates the spatial correlation functions for Nth-order differential microphones in both spherically and cylindrically isotropic noise fields. The results are used to calculate the amount of achievable cancellation from an adaptive noise cancellation application using combinations of differential microphones to remove unwanted noise from a desired signal. The results are useful in determining signal-to-noise ratio gains from arbitrarily positioned differential microphone elements in microphone array applications.

4.1 Introduction

The spatial correlation function is important in the design of optimal beamformers that maximize the signal-to-noise ratio (SNR), source direction finding algorithms, the calculation of actual SNR gain from arrays, and other array signal processing areas. The space-time correlation functions are well known for omnidirectional receivers in two specific environments: spherically and cylindrically isotropic noise fields. One area of large concern that has been a topic of ongoing work has been the design and performance of directional differential microphone systems. One application of these systems is in adaptive noise cancellation schemes. In order to predict the expected performance gains of these adaptive cancellation systems, the spatial correlation functions between directional microphones are required. Results are presented here for the specific cases of general orientation for first-order differential microphones in both spherically and cylindrically isotropic fields. Specific results are given for the general Nth-order cases for differential arrays that have collinear axes.

4.2 Adaptive Noise Cancellation

The use of adaptive noise cancellation in communication devices has been under investigation for more than two decades [1], [2]. The early studies predicted SNR gains on the order of 10 dB and higher. However, it was

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5 Robust Adaptive Beamforming

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Abstract. This chapter presents robust adaptive beamforming techniques designed specifically for microphone array applications. The basics of adaptive beamformers are first reviewed with the Griffiths-Jim beamformer (GJBF). Its robustness problems caused by steering vector errors are then discussed with some conventionally proposed robust beamformers. As better solutions to the conventional robust beamformers, GJBFs with an adaptive blocking matrix are presented in the form of a microphone array. Simulation results and real-time evaluation data show that a new robust adaptive microphone array achieves improved robustness against steering vector errors. Good sound quality of the output signal is also confirmed by a subjective evaluation.

5.1 Introduction

Beamforming is a technique which extracts the desired signal contaminated by interference based on directivity, i.e. spatial signal selectivity [1]–[5]. This extraction is performed by processing the signals obtained by multiple sensors such as microphones, antennas, and sonar transducers located at different positions in the space. The principle of beamforming has been known for a long time. Because of the vast amount of necessary signal processing, most research and development effort has been focused on geological investigations and sonar, which can afford a higher cost. With the advent of LSI technology, the required amount of signal processing has become relatively small. As a result, a variety of research projects where acoustic beamforming is applied to consumer-oriented applications, have been carried out [6].

Applications of beamforming include microphone arrays for speech enhancement. The goal of speech enhancement is to remove undesirable signals such as noise and reverberation. Among research areas in the field of speech enhancement are teleconferencing [7]–[8], hands-free telephones [9]–[11], hearing aids [12]-[21], speech recognition [22]–[23], intelligibility improvement [24]–[25], and acoustic measurement [26].

Beamforming can be considered as multidimensional signal processing in space and time. Ideal conditions assumed in most theoretical discussions are not always maintained. The target DOA (direction of arrival), which is assumed to be stable, does change with the movement of the speaker. The sensor gains, which are assumed uniform, exhibit significant distribution. As a result, the performance obtained by beamforming may not be as good as

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6 GSVD-Based Optimal Filtering for Multi-Microphone Speech Enhancement

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Abstract. In this chapter a class of multi-microphone signal enhancement techniques is described, which are based on a Generalized Singular Value Decomposition (GSVD) and which amount to a specific optimal filtering problem when the so-called desired response signal cannot be observed. When applying this GSVD-based optimal filtering technique to noise reduction in multi-microphone speech recordings, simulations show that this technique has a better noise reduction performance than standard beamforming techniques for all reverberation times and that it is more robust to deviations from the nominal situation, e.g. encountered in uncalibrated microphone arrays. If the GSVD-based procedure is used to create speech as well as noise references, as in a Generalized Sidelobe Canceler (GSC) structure, and an adaptive noise canceler (ANC) is added, the noise reduction performance can still be improved. Because computational complexity may appear as a bottleneck for this algorithm, recursive GSVD-updating and downsampling techniques are needed to make this signal enhancement technique amenable to real-time implementation.

6.1 Introduction

In many speech communication applications, like audio-conferencing and hands-free mobile telephony, the recorded speech signals are corrupted by acoustic background noise and far-end echo signals (see Fig. 6.1). This is mainly due to the fact that the speaker is located at a distance from the recording microphones, allowing the microphones to record the noise and echo sources too. Background noise and far-end echo signals cause a signal degradation which can lead to total unintelligibility of the speech and which decreases the performance of speech coding and speech recognition systems. Therefore efficient acoustic noise and echo reduction algorithms are called for.

Several single-microphone techniques for noise reduction in speech have been proposed that are based on a (generalized) singular value decomposition. While early techniques only dealt with white noise [1][2], more advanced techniques also incorporated colored noise [3][4]. However, since these techniques are single-microphone speech enhancement techniques, they only perform a (signal-adaptive) frequency filtering on the noisy speech signal.

When performing multi-microphone speech enhancement, both the frequency and the spatial characteristics of the speech and the noise sources can be exploited. Recently a multi-microphone noise reduction technique for

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7 Explicit Speech Modeling for Microphone Array Applications

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Abstract. In this chapter we address the limitations of current approaches to using microphone arrays for speech acquisition and advocate the development of multi-channel techniques which employ an explicit model of the speech signal. The goal is to combine the advantages of spatial filtering achieved through beamforming with knowledge of the desired time-series attributes. We offer two examples of algorithms which incorporate this principle. The first is a frequency domain approach illustrating the utility of model-based microphone array processing for improved speech spectral analysis. The second is a time domain procedure which highlights the benefits of speech modeling/spatial filtering fusion for the purpose of improving speech quality.

7.1 Introduction

A major thrust of microphone array research has focused on improving the spatial filtering capability of the system in its operating environment. The previous chapters have directly addressed a number of the methods which have proven beneficial in this regard. The goal of this chapter is to present an alternative and complementary strategy which emphasizes the incorporation of explicit speech modeling into the microphone array processing. As will be shown, by combining specific knowledge of the desired time-series attributes with the advantages of multi-channel spatial filtering, this approach can significantly improve the quality of the speech signal and its derived quantities obtained in challenging acoustic environments.

Much of this work is motivated from current research in the field of single-channel speech systems where, due to necessity perhaps, improvements have been obtained by focusing on the underlying content of the signal of interest in addition to its environmental degradations. There is a rich history of work addressing the use of single channel methods for speech enhancement. Summaries of these techniques may be found in texts on the subject, such as [1–3]. While capable of improving perceived quality in restrictive environments (additive noise, no multipath, high to moderate signal-to-noise ratio (SNR), single source), these approaches do not perform well in the face of reverberant distortions, competing sources, and severe noise conditions. In recent years, sophisticated speech models have been applied to the enhancement problem. In addition to utilizing the periodic features of the speech, as in the case of comb filtering, these systems exploit the signal's mixture of harmonic and

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Part II

Source Localization

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8 Robust Localization in Reverberant Rooms

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Abstract. Talker localization with microphone arrays has received significant attention lately as a means for the automated tracking of individuals in an enclosure and as a necessary component of any general purpose speech capture system. Several algorithmic approaches are available for speech source localization with multi-channel data. This chapter summarizes the current field and comments on the general merits and shortcomings of each genre. A new localization method is then presented in detail. By utilizing key features of existing methods, this new algorithm is shown to be significantly more robust to acoustical conditions, particularly reverberation effects, than the traditional localization techniques in use today.

8.1 Introduction

The primary goal of a speech localization system is accuracy. In general, estimate precision is dependent upon a number of factors. Major issues include (1) the quantity and quality of microphones employed, (2) microphone placement relative to each other and the speech sources to be analyzed, (3) the ambient noise and reverberation levels, and (4) the number of active sources and their spectral content. The performance of localization techniques generally improves with the number of microphones in the array, particularly when adverse acoustic effects are present. This has spawned the research and construction of large array systems (e.g. 512 elements) [1]. However, when acoustic conditions are favorable and the microphones are positioned judiciously, source localization can be performed adequately using a modest number (e.g. 4 elements) of microphones. Performance is clearly affected by the array geometry. The optimal design of the array based on localization criteria is typically dependent on the room layout, speaking scenarios, and the acoustic conditions [2]. In practice, many of these design considerations are very dependent on the specific application conditions, the hardware available, and non-scientific cost criteria. In an effort to make its applicability as general as possible, this chapter will focus primarily on speech localization effectiveness as a function of the acoustic degradations present, namely background noise and reverberations, rather than attempt to address more specific environmental scenarios.

In addition to high accuracy, these location estimates must be updated frequently in order to be useful in practical tracking and beamforming appli-

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9 Multi-Source Localization Strategies

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Abstract. Localization of multiple acoustic sources is an important task in many practical applications. However, in most cases adopted models are not fully adequate to describe real scenarios. In particular, in the presence of reverberation, the signal model should explicitly take into account both signals radiated by multiple sources and reflections. In alternative to Generalized Cross-Correlation methods, array processing concepts can be effectively applied to multi-source localization in reverberant environments. In this chapter, main features and limitations of wideband array processing approaches are briefly reviewed. Furthermore, a new integrated localization and classification system, based on a robust frequency-domain Time Delay Of Arrival (TDOA) estimation followed by a spatial clustering of raw location estimates, is presented. The proposed method efficiently incorporates TDOA and array processing concepts in an unified approach. Results on simulated data are supplied.

9.1 Introduction

Many practical applications rely on microphone arrays to localize multiple acoustic sources. Direct examples include automated systems for surveillance, videoconferencing and hands-free speech acquisition [1]. Localization data is also used indirectly for a variety of other tasks: dereverberation of speech, fault prediction and analysis in machinery, cuing and tracking of television cameras, speaker verification, etc.

Ideally, localization methods should exploit knowledge of the particular acoustical transfer functions associated with the sources and sensors. However, over reliance on these functions in the form of excessive calibration to a specific environment may result in limited system flexibility and poor adaptation capabilities to different and varying acoustical conditions. This suggests that localization algorithms should be based on a compact and robust parameterization of the transfer functions.

If reverberation is present, a general signal model should explicitly take into account both the signals radiated by multiple sources and their respective multi-path reflections. The effects of reflective surfaces in closed environments can be studied in a rigorous manner by solving the wave equations with proper boundary and initial conditions. This task is generally too expensive and can be very sensitive to numerical errors and changes in assumptions about the environment. A physically acceptable approximation may be derived from

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10 Joint Audio-Video Signal Processing for Object Localization and Tracking

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Abstract. Applications such as videoconferencing, automatic scene analysis, or security surveillance involving acoustic sources can benefit from object localization within a complex scene. Many single-sensor techniques already exist for this purpose. They are, e.g., based on microphone arrays, video cameras, or range sensors. Since all of these sensors have their specific strengths and weaknesses, it is often advantageous to combine information from various sensor modalities to arrive at more robust position estimates.

This chapter presents a joint audio-video signal processing methodology for object localizing and tracking. The approach is based on a decentralized Kalman filter structure modified such that different sensor measurement models can be incorporated. Such a situation is typical for combined audio-video sensing, since different coordinate systems are usually used for the camera system and the microphone array.

At first, the decentralized estimation algorithm is presented. Then a speaker localization example is discussed. Finally, some estimation results are shown.

10.1 Introduction

Microphone arrays are becoming increasingly important for applications related to the recording of speech or music for subsequent processing, transmission or storage. In most of these cases, the goal is to extract information by listening. In object localization and tracking, however, things are different. There, the information conveyed by acoustic waves is not related to the pattern of the signal they carry but to the position of the sound emitting source.

Of course, sound waves are not the only physical quantity that can be used for source localization. Other sensors may, for example, measure the range of the source or detect its retinal image. Since all sensors have their specific strengths and weaknesses, there is no single modality for object localization that always outperforms all others. Therefore, it is desirable to integrate the information of various sensor modalities to exploit different physical phenomena.

A practical example for such a multi-sensor system may comprise a microphone array and a video camera, for example. Either sensor can be used to estimate a (single-sensor) source position. Taken together, they can be viewed as a joint audio-video object localization system. Unfortunately, such

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Part III

Applications

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11 Microphone-Array Hearing Aids

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Abstract. Microphone-array hearing aids provide a promising solution to the problems encountered by hearing-impaired persons when listening to speech in the presence of background noise. This chapter first discusses implementation issues and performance metrics specific to the hearing-aid application. A review of previous work on microphone-array hearing aids includes systems with directional microphones, fixed beamformers, adaptive beamformers, physiologically-motivated processing, and binaural outputs. Recent simulation results of one promising adaptive beamforming system are presented. The performance of microphone-array hearing aids depends heavily on the acoustic environments in which they are used. Additional information about the level of reverberation, number of interferers, and relative levels of interferers encountered by hearing-aid users in everyday situations is required to quantify the benefit of microphone-array hearing aids and to select the optimal processing strategy.

11.1 Introduction

Hearing impairments affect 10–16% of the total adult population and at least one-third of the elderly population [1–5]. Recent surveys in the United States indicate that roughly 20% of the hearing-impaired population (5–6 million people) use hearing aids, but only 60% of users are satisfied with their devices [6].

Users' overall satisfaction with their hearing aids is highly dependent on the situations in which they are used. While 90% of hearing aid users indicated that they were satisfied with their hearing aid for one-on-one listening, 62% were satisfied in small group settings, 45% in workplaces and restaurants and only 25% in large groups [7]. In an analysis of why hearing aid owners do not use their hearing aids, the second most prevalent reason, given by 25% of respondents, was that the hearing aids "did not work" in background noise [8].

These complaints are not surprising given the greater susceptibility of hearing-impaired listeners to interference from background noise. Normal-hearing listeners typically exhibit speech reception thresholds[1] (SRT's) of

[1] The speech reception threshold is the signal-to-interference ratio at which speech is 50% intelligible.

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12 Small Microphone Arrays with Postfilters for Noise and Acoustic Echo Reduction

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Abstract. This chapter presents arrays with two microphones and a postfilter for noise reduction and acoustic echo cancellation. The postfilter algorithm exploits the spatial coherence of the microphone signals. In contrast to single-microphone enhancement algorithms, it does not need an explicit noise power spectral density estimate. An analysis of the mean square error reveals that the coherence properties of the microphone signals are of paramount importance for the performance of the postfilter. Coherence measurements of signals in various acoustic environments are presented. The influence of the directivity and orientation of the microphones on the measured coherence is discussed and rules for the design of the acoustic interface are given. Finally, applications of this approach are presented. The two-microphone algorithm is employed to reduce the non-stationary noise in the voice intercom of a computed tomography scanner. It is also combined with echo cancellers to be used in a robust desktop conferencing device.

12.1 Introduction

Hands-free operation of voice communication terminals presents a challenging signal processing task. The relatively large distance between speaker and microphones, the feedback of acoustic echoes, and the "anywhere and anytime" paradigm of mobile communications can all contribute to considerably disturbed speech signals. To achieve a reasonable communication quality the hands-free terminal must therefore reduce disturbing environmental acoustic noise as well as acoustic echoes in received speech signals.

Because of their ease of implementation and use, single-microphone speech enhancement systems are favored in many applications. However, multi-microphone systems have considerable advantages over single-microphone systems when the noise is non-stationary or the speech is reverberated. Multi-microphone systems take the spatial correlation of sound fields into account. The spatial correlation can be exploited to dereverberate the desired speech signal and to reduce noise and acoustic echoes. The simplest system which takes advantage of some of these benefits is the two-microphone array.

In contrast to larger arrays, the two-microphone approach relies less on the beamforming gain of the array but more on the noise and echo suppression of a postfilter. The postfilter combines and processes the two microphone signals in order to compute an estimate of the clean speech signal. The array can be easily implemented using widely available stereo A/D converters.

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13 Acoustic Echo Cancellation for Beamforming Microphone Arrays

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Abstract. Acoustic feedback from loudspeakers to microphones constitutes a major challenge for digital signal processing in interfaces for natural, full-duplex human–machine speech interaction. Two techniques, each one successful on its own, are combined here to jointly achieve maximum echo cancellation in real environments: For one, acoustic echo cancellation (AEC), which has matured for single-microphone signal acquisition, and, secondly, beamforming microphone arrays, which aim at dereverberation of desired local signals and suppression of local interferers, including acoustic echoes. Structural analysis shows that straightforward combinations of the two techniques either multiply the considerable computational cost of AEC by the number of array microphones or sacrifice algorithmic performance if the beamforming is time-varying. Striving for increased computational efficiency without performance loss, the integration of AEC into time-varying beamforming is examined for two broad classes of beamforming structures. Finally, the combination of AEC and beamforming is discussed for multi-channel recording and multi-channel reproduction schemes.

13.1 Introduction

For natural human–machine interaction, acoustic interfaces are desirable that support seamless full-duplex communication without requiring the user to wear or hold special devices. For that, the general scenario of Figure 13.1 foresees several loudspeakers for multi-channel sound reproduction and a microphone array for acquisition of desired signals in the local acoustic environment. Acoustic signal processing is employed to support services such as speech transmission, speech recognition, or sound field synthesis offered by communication networks or autonomous interactive systems. Such hands-free acoustic interfaces may be tailored for incorporation into a wide variety of communication terminals, including teleconferencing equipment, mobile phones and computers, car information systems, and home entertainment equipment.

For signal acquisition, microphone arrays allow spatial filtering of arriving signals and, thus, desired signals can be enhanced and interferers can be suppressed. With full-duplex communication, echoes of the loudspeaker signals will join local interferers to corrupt the desired source signals. Beamforming, however, does not exploit the available loudspeaker signals as reference information for suppressing the acoustic echoes. This is accomplished by acoustic

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14 Optimal and Adaptive Microphone Arrays for Speech Input in Automobiles

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Abstract. In this chapter, speech enhancement and echo cancellation for hands-free mobile telephony are discussed. A number of microphone array methods have been tested and results from car measurements are given. Traditional methods such as linearly constrained beamforming and generalized sidelobe cancelers are discussed as well as array gain optimization methods. An *in situ* calibrated method which gives an overall improved performance is also presented. Algorithms such as Least-Squares (LS) and Normalized-Least-Mean-Squares (NLMS) are used to find optimal weights. Improved performance using an LS-method is shown, but at the cost of increased numerical complexity limiting its implementation in real-time applications. By introducing subband processing this issue can be avoided. The results show a noise suppression of approximately 18 dB and hands-free loudspeaker suppression of the same order.

14.1 Introduction: Hands-Free Telephony in Cars

The increased use of mobile telephones in cars has created a greater demand for hands-free, in-car installations. The advantages of hands-free telephones are safety and convenience. In many countries and regions hand-held telephony in cars is prohibited by legislation. The car manufacturers also prohibit such use since it will interact with other electronic devices in the car such as air bags, navigation equipment, etc. This means that a mobile telephone should be properly installed and an external antenna should be used. However, by installing the microphone far away from the user, a number of disadvantages, such as poor sound quality and acoustic feedback from the far-end side, are introduced. This means that some form of filtering is required in order to obtain sound quality comparable to that of hand-held telephony. This filtering operation must suppress the loudspeaker, as well as background noise and room reverberation, without causing severe speech distortion. A number of potential methods will be presented to address this problem.

For automobile applications, there has also been the desire to replace many hand-controlled functions with voice controls. The signal degradations in this context have many similarities to those encountered in distant-talker speech recognition applications. A study of recognition in car environments was presented in [1,2]. However this topic is beyond the primary goal here and is the specific subject of Chapter 15.

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15 Speech Recognition with Microphone Arrays

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Abstract. Microphone arrays can be advantageously employed in Automatic Speech Recognition (ASR) systems to allow distant-talking interaction. Their beamforming capabilities are used to enhance the speech message, while attenuating the undesired contribution of environmental noise and reverberation. In the first part of this chapter the state of the art of ASR systems is briefly reviewed, with a particular concern about robustness in distant-talker applications. The objective is the reduction of the mismatch between the real noisy data and the acoustic models used by the recognizer. Beamforming, speech enhancement, feature compensation, and model adaptation are the techniques adopted to this end. The second part of the chapter is dedicated to the description of a microphone-array based speech recognition system developed at ITC-IRST. It includes a linear array beamformer, an acoustic front-end for speech activity detection and feature extraction, a recognition engine based on Hidden Markov Models and the modules for training and adaptation of the acoustic models. Finally the performance of this system on a typical recognition task is reported.

15.1 Introduction

During the last decade research on ASR technology has made significant advances. As a result, high performance systems are now available for situations where there is a good match between testing and training conditions [1,2]. However, these same systems tend to suffer from a limited robustness to variability in their operating environment [3–8].

One of the most attractive potential features of ASR technology is the flexibility afforded through hands-free interaction. Not being encumbered by a hand-held or head-mounted microphone may be of considerable utility to the user. Of particular concern is the distant-talker case where the user is beyond the normal acquisition range of the system microphone[1] (e.g. at more than one meter in the case of an omnidirectional microphone). For ASR applications of moderate/high language complexity this represents a very ambitious task.

The development of distant-talker ASR will allow for the expansion of voice activated technology into a number of areas where it has until now

[1] The distinctions among different types of microphones will not be treated here. However, the choices available and their relative characteristics should be specifically addressed in the design of the application.

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