

SOUND DIRECTIONS

Best Practices for Audio Preservation

By Mike Casey, Indiana University and
Bruce Gordon, Harvard University

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Contents

Contents

Acknowledgements	vii
Indiana University Sound Directions Staff	vii
Harvard University Sound Directions Staff	vii
1 The Sound Directions Project.....	1
1.1 Project History and Goals.....	1
1.2 Introduction to Institutions.....	3
1.2.1 Indiana University	3
1.2.2 Harvard University	4
1.3 Standards and Best Practices.....	5
1.3.1 Introduction.....	5
1.4 Overview of this Publication	8
2 Personnel and Equipment for Preservation Transfer.....	9
2.1 Preservation Overview.....	9
2.2 Recommended Technical Practices	14
2.2.1 Personnel for Preservation Transfer Work	14
2.2.1.1 Best Practices	14
2.2.1.2 Rationale	14
2.2.1.3 Staff at Indiana and Harvard	14
2.2.2 Studio Signal Chain for Archival Preservation	16
2.2.2.1 Best Practices	16
2.2.2.2 Rationale	16
2.2.2.3 Preservation Studio at the IU Archives of Traditional Music.....	17
2.2.2.4 Preservation Studios at the Harvard College Library’s Audio Preservation Services (HCL-APS).....	22
2.3 Analog Playback.....	24
2.2.3.1 Best Practices	24
2.2.3.2 Analog Playback at Harvard and Indiana	24
2.2.3.3 Playback Problems	29
2.4 Conversion	31
2.2.4.1 Best Practices	31
2.2.4.2 Analog-to-Digital and Digital-to-Analog Conversion at Harvard and Indiana	31
3 Digital Files	33
3.1 Preservation Overview.....	33
3.2 Recommended Technical Practices	38
3.2.1 Target Preservation Format	38
3.2.1.1 Best Practices	38
3.2.1.2 Rationale	38
3.2.1.3 Background	39
3.2.1.4 Use of the BWF <bext> Chunk at Indiana.....	39
3.2.1.5 Use of the BWF <bext> Chunk at Harvard.....	42
3.2.1.6 Institutional Comparison of <bext> Chunk Field Use.....	44
3.2.2 Digital File Types and Uses for Preservation.....	45
3.2.2.1 Best Practices	45
3.2.2.2 Rationale.....	45
3.2.2.3 File Types and Uses at Indiana	45
3.2.2.4 File Types and Uses at Harvard	50
3.2.3 Local Filenames.....	52
3.2.3.1 Best Practices	52

3.2.3.2 Rationale	52
3.2.3.3 Filenames at the IU Archives of Traditional Music	52
3.2.3.4 Filenames at Harvard.....	57
3.2.4 File Data Integrity	58
3.2.4.1 Best Practices	58
3.2.4.2 Rationale.....	58
3.2.4.3 Background	58
3.2.4.4 Redundancy Checks at Harvard and Indiana	59
4 Metadata.....	60
4.1 Preservation Overview	60
4.2 Recommended Technical Practices.....	61
4.2.1 Best Practices	61
4.2.2 Rationale.....	61
4.2.3 Background.....	62
4.2.3.1 Descriptive Metadata.....	62
4.2.3.2 Technical Metadata	62
4.2.3.3 Digital Provenance Metadata (digiprov).....	63
4.2.3.4 Structural Metadata	63
4.2.4 Metadata at Harvard.....	69
4.2.4.1 Preservation Metadata Documents and Their Creation	69
4.2.5 Metadata at Indiana.....	82
4.2.5.1 Audio Technical Metadata Collector (ATMC).....	82
4.2.5.2 Technical Metadata.....	82
4.2.5.3 Digital Provenance Metadata.....	83
4.2.5.4 Structural Metadata	86
5 Storage.....	91
5.1 Preservation Overview.....	91
5.2 Recommended Technical Practices	92
5.2.1 Local, Interim Storage.....	92
5.2.1.1 Best Practices	92
5.2.1.2 Rationale.....	92
5.2.1.3 Local, Interim Storage at the Indiana University Archives of Traditional Music ...	93
5.2.1.4 Local Storage Supplementing Preservation Storage at Indiana University ATM ...	94
5.2.1.5 Local, Interim Storage at Harvard.....	96
5.2.2 Long-Term Preservation Storage.....	97
5.2.2.1 Best Practices	97
5.2.2.2 Rationale.....	97
5.2.2.3 Long-Term Preservation Storage at Harvard.....	97
5.2.2.4 Long-Term Preservation Storage at Indiana	99
6 Preservation Packages and Interchange.....	102
6.1 Preservation Overview.....	102
6.2 Recommended Technical Practices.....	103
6.2.1 Preservation Packages.....	103
6.2.1.1 Best Practices	103
6.2.1.2 Rationale	103
6.2.1.3 Preservation Packages at Harvard	104
6.2.1.4 Preservation Package Technical Practices at Indiana.....	106
6.2.2 Interchange	108
6.2.2.1 Background.....	108
6.2.2.2 Interchange at Indiana	110
6.2.2.3 Interchange at Harvard	110

7 Audio Preservation Systems and Workflows	113
7.1 Preservation Overview.....	113
7.2 Recommended Technical Practices	115
7.2.1 Selection for Preservation	115
7.2.1.1 Best Practices	115
7.2.1.2 Rationale.....	115
7.2.1.3 Selection at Indiana	115
7.2.1.4 Selection at Harvard	116
7.2.2 Quality Control and Quality Assurance	117
7.2.2.1 Best Practices	117
7.2.2.2 Rationale.....	117
7.2.2.3 Background.....	117
7.2.2.4 Quality Control at Indiana	118
7.2.2.5 Quality Control at Harvard	121
7.2.3 Audio Preservation Systems and Workflows	122
7.2.3.1 Best Practices	122
7.2.3.2 Rationale.....	122
7.2.3.3 Audio Preservation System at Indiana	122
7.2.3.4 Workflow at Indiana	128
7.2.3.5 Workflow at Harvard	140
8 Summary of Best Practices	156
Personnel and Equipment for Preservation Transfer	156
Digital Files	156
Metadata	157
Storage	158
Preservation Packages and Interchange	158
Audio Preservation Systems and Workflows.....	158

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A Note to Readers:

Each chapter in this document (other than the first) is divided into two major parts: a preservation overview that summarizes key concepts for collection managers and curators, followed by a section intended for audio engineers, digital librarians, and other technical staff that presents recommended technical practices while summarizing our findings and experience. Collection managers will find many parts of the technical sections useful but, in some cases, may need to engage the audio engineering or digital library communities for assistance in understanding technical topics. Similarly, technical staff may benefit from the broad perspective of the preservation overviews but may want to consult with collection management about the implications of the general principles in these sections for their daily work.

For the purposes of this publication, “Harvard” refers to the Harvard College Library Audio Preservation Services and the Archive of World Music in the Loeb Music Library at Harvard University. “Indiana” and “IU” refer to the Archives of Traditional Music and/or the Digital Library Program at Indiana University. Because both universities are large and complex, this publication cannot and does not represent all audio digitization activities throughout these institutions.

1 The Sound Directions Project

1.1 Project History and Goals

Sound archives have reached a critical point in their history marked by the simultaneous rapid deterioration of unique original materials, the development of expensive and powerful new digital technologies, and the consequent decline of analog formats and media. It is clear to most sound archivists that our old analog-based preservation methods are no longer viable and that new strategies must be developed in the digital domain. Motivated by these concerns, in February 2005 the Indiana University Archives of Traditional Music (ATM) and the Archive of World Music (AWM) at Harvard University began Phase 1 of *Sound Directions: Digital Preservation and Access for Global Audio Heritage*—a joint technical archiving project with funding from the National Endowment for the Humanities Preservation and Access Research and Development program. The goals of Phase 1 of *Sound Directions* were to: a) create best practices and test emerging standards for digital preservation; b) establish, at each university, programs for digital audio preservation that enable us to continue this work into the future and which produce interoperable results; and c) preserve critically endangered, highly valuable, unique field recordings of extraordinary interest.¹

Although the results of our research and development apply to preservation work with all types of audio recordings, the *Sound Directions* partner institutions focused their preservation activity on field recordings—carriers of unique, irreplaceable and historically significant cultural heritage. As caretakers of these collections we must solve the problem of preserving audio resources accurately, reliably, and for the very long term; at the same time we must make our resources readily accessible to those who most need them. These issues have been the subject of work, discussion and study at a number of national agencies and institutional archives, including the Council on Library and Information Resources, the American Folklife Center, the Library of Congress Digital Audio-Visual Prototyping Project,² the AWM and the ATM. Most of us are now approaching audio digitization in similar, deliberately cooperative ways. Yet, there are few published standards or best practices for audio preservation. Committees of the Audio Engineering Society (AES) and the International Association of Sound and Audiovisual Archives (IASA) have written detailed standards and best practices for some, but not all, parts of the audio digitization process. Particularly in the digital part of the preservation chain, best practices are either high-level or non-existent and are not intended to reflect the detail, richness, and experience of real world projects. *Sound Directions* was created in part because of our conviction that the development of best practices and standards in many areas of the preservation chain was the essential next step to insure the preservation

¹ *Sound Directions* has from its inception been conceived as a multi-phased project addressing both preservation of and access to field recordings in the digital domain. In June 2007 the ATM and AWM embarked on an 18-month “Preservation Phase” of the project, again with funding from NEH, through which we will realize the results of our Phase 1 research and development by putting our new digital preservation systems to work preserving and making accessible a substantial number of highly endangered field recordings. A future phase of *Sound Directions* will focus on the development of online access systems for archival field collections.

² The LC website for this project is at: <http://www.loc.gov/rr/mopic/avprot/avprhome.html>. Both *Sound Directions* institutions tracked this project which ended in 2004. In some ways we think that our project has pushed forward some of the issues that the Prototyping Project raised or began addressing. See Carl Fleischhauer, “The Library of Congress Digital Audio Preservation Prototyping Project” (paper presented at the symposium *Sound Savings: Preserving Audio Collections*, Austin, TX, July 24-26, 2003). Also available online: http://www.arl.org/preserv/sound_savings_proceedings/Digital_audio.shtml. Harvard University contributed to the development of the Prototyping Project’s technical and digital provenance metadata schemas.

of fragile and deteriorating audio recordings representing irreplaceable cultural heritage.³

In CLIR's *Folk Heritage Collections in Crisis*, sound preservation consultant Elizabeth Cohen writes, "the development of successful preservation strategies will require the cooperation of computer scientists, data storage experts, data distribution experts, fieldworkers, librarians, and folklorists."⁴ *Sound Directions* was conceived and organized based on the fundamental principle that such collaboration is essential to the task of preserving audio collections in today's world. Collaboration occurred in this project for two primary reasons: first, in the digital domain the expertise and facilities required for audio preservation are distributed across multiple agents and agencies; second, sharing information with others in the global community of sound archivists improves our work and helps us achieve the standardization that is essential to any effective preservation system.

Collaboration occurred within each institution, between the institutions, and between *Sound Directions* and the broader community of sound archivists and specialists around the world. At both Harvard and Indiana, this project involved multiple administrative units and staff including archive administrators/curators, audio engineers, librarians of various sorts (including digital library specialists), computer programmers, digital data managers/storage specialists, subject specialists, and others. Thus, a great deal of inter-professional collaboration was required at each respective university. At a higher yet still fundamental level, Indiana and Harvard staff collaborated as well, acting upon our belief that it should be possible for different institutions to work within their differing workflows and physical settings and still attain preservation through the production of interoperable results. As archives within very different sorts of universities—one public and one private—and with quite different histories, staffing, equipment and workflows, we collaborated throughout the process, at times approaching aspects of our work differently, but always operating with shared goals and toward sharable end results. Communication with the *Sound Directions* Advisory Board enabled us to engage yet a broader community of relevant specialists, including national leaders in the fields of archival audio preservation, digital libraries and information management. Members of the Advisory Board reviewed a draft of this publication. Collaboration with Advisory Board members was supplemented by consultation of additional experts, including archivists and audio engineers in the U.S., Canada, and Europe, whose willingness to share information and advice brought a still broader collaborative network to bear on *Sound Directions* work.

Also motivating our collaborative approach was our desire to render the information generated through our work generalizable to other institutions who want to use the project's innovations but cannot redesign their audio studios nor completely alter their staffing situations in order to do so. Working together and in step with our broader community of collaborators, Indiana and Harvard have developed methods and best practices that are largely system-independent, that can be adopted by other institutions without overhauling their existing operations.

The *Sound Directions* project produced four key results: this publication of our findings and

³ The recordings chosen as test cases for *Sound Directions* were drawn from the rich, outstanding, and unique ethnographic field collections of the Archives of Traditional Music at Indiana University and the Archive of World Music at Harvard University. Field collections were selected based on the following criteria: a) research and cultural value; b) preservation needs; and c) recording format (in order to test the transfer of a range of formats for this research and development project). At Harvard, selected collections included historic field recordings from Egypt, Iraq, Iran, Afghanistan, Pakistan and India—unique documents of cultural history from regions of tremendous interest to Americans today. At Indiana, selected collections included critically important cultural materials such as music of Iraqi Jews in Israel, music from pre-Taliban Afghanistan, music related to the world's longest-running civil war in Sudan, and African-American protest songs from the 1920s through the 1940s.

⁴ Elizabeth Cohen, "Preservation of Audio," in *Folk Heritage Collections in Crisis* (Washington, DC: Council on Library and Information Resources, 2001), 26. Also available online: <http://www.clir.org/pubs/reports/pub96/pub96.pdf>.

best practices, the development of much needed software tools for audio preservation, the creation or further development of audio preservation systems at each institution, and the preservation of a number of critically endangered and highly valuable recordings. All of the above are detailed in this publication, which we believe provides solid grounding for institutions pursuing audio preservation either in-house or in collaboration with an outside vendor. For institutions actively engaged in preservation transfer work themselves, the project created a number of software tools that may be placed into service. The development of these tools reflects both the starting points in this project and the different interests of the two institutions. Harvard University's experience with the vast quantities of metadata required for preservation led them to design and develop the Harvard *Sound Directions* Toolkit. The toolkit is a suite of forty open-source, scriptable, command line interface, audio preservation software tools that streamline workflow, reduce labor costs, and reduce the potential for human error in the creation of preservation metadata and in the encompassing preservation package. Harvard also produced Audio Object Manager for audio object metadata creation and Audio Processing XML Editor (APXE) for collection of digital provenance metadata. To aid selection for preservation, Indiana University developed the Field Audio Collection Evaluation Tool (FACET), which is a point-based, open-source software tool for ranking field collections for the level of deterioration they exhibit and the amount of risk they carry. Indiana also developed the Audio Technical Metadata Collector (ATMC) software for collecting and storing technical and digital provenance metadata for audio preservation. Harvard and Indiana are making their software tools freely available to the preservation community beginning in the fall of 2007, with the exception of ATMC, Audio Object Manager, and APXE, all of which will be released later after further development. A download link for these tools will be posted on the *Sound Directions* website. Many of the tools are referenced throughout this document, and a complete listing of the Harvard *Sound Directions* Toolkit can be found in Appendix 5 where each tool is described, and its use and options are listed. A user's guide for the current version of ATMC, with details on each metadata element, can be found in Appendix 1. All of these tools are key ingredients in the audio preservation systems at each institution, contributing to the enduring preservation of the recordings that are processed by these systems. If we have done our work well, these recordings will speak for our efforts far into the future.

1.2 Introduction to Institutions

1.2.1 Indiana University

The Archives of Traditional Music (ATM)⁵ fosters the educational and cultural role of Indiana University through the preservation and dissemination of the world's music and oral traditions. One of the largest and oldest university-based ethnographic sound archives in the United States, the ATM's holdings cover a wide range of cultural and geographical areas, and include commercial and field recordings of vocal and instrumental music, folktales, interviews, and oral history, as well as videotapes, photographs, and manuscripts. The ATM seeks to fulfill its mission through appropriate acquisitions and by cataloging and preserving its collections for use by educators, researchers, and interested members of the public, including the people from whom the material was collected. The ATM's collections and library contribute to the research and teaching activities of Indiana University, especially the Departments of Folklore and Ethnomusicology, Anthropology, Linguistics; the School of Music; and the interdisciplinary area studies programs that are associated with them. It also serves as a research, teaching, and training center for the IU Ethnomusicology Program. Founded in 1948, the ATM has been a recognized leader in the sound archiving community,

⁵ Indiana University, Archives of Traditional Music. <http://www.indiana.edu/~libarchm/>.

developing in step with technological and theoretical advances in ethnographic research and recorded sound.

At IU, the ATM's primary partner in this project, the IU Digital Library Program (DLP)⁶ is dedicated to the selection, production, and maintenance of a wide range of high quality networked resources for scholars and students at Indiana University and elsewhere, and supports digital library infrastructure for the university. The DLP is a collaborative effort of the Indiana University Libraries, the Office of the Vice President for Information Technology, and IU's research faculty with leadership from the School of Library and Information Science and the School of Informatics. The DLP's current facilities include the Digital Media and Image Center (containing equipment for image, audio, and video capture), the Electronic Text Development Center (supporting creation of scholarly electronic texts), and an extensive server infrastructure for support of digital projects, with life-cycle replacement funding for hardware and software. DLP staff provides expertise in planning, creating, and maintaining digital projects.

1.2.2 Harvard University

The Archive of World Music (AWM) and its technological partner, Harvard College Library Audio Preservation Services (HCL-APS), are both units of the Loeb Music Library⁷ which, in turn, is a component of the Harvard College Library that serves the Faculty of Arts and Sciences at Harvard. The Archive of World Music was established in 1976 and, with the appointment in 1992 of Kay Kaufman Shelemay as Harvard's first senior professor of ethnomusicology, the Archive moved to the Music Library to become one of its special collections. It is devoted to the acquisition of archival field recordings of musics worldwide as well as to commercial sound recordings, videos, and DVDs of ethnomusicological interest. The AWM developed the HCL-APS, a state-of-the-art facility which was an early leader, and continues to provide leadership, in the application of digital technologies to archival audio practice.

Over the past five years HCL-APS has moved toward joining its counterpart, the Harvard College Library Digital Imaging Group (HCL-DIG) in providing top quality service and advice for digitizing media. Both work closely with the Harvard University Library Office for Information Systems on matters of building robust infrastructure and sustainable tools for creating and preserving digital objects via the Digital Repository Service.

The Harvard University Office for Information Systems (OIS)⁸ coordinates all of the Library's online catalogs (HOLLIS, its MARC catalog, OASIS for finding aids, VIA for visual images, and so forth) as well as the highly regarded Library Digital Initiative (LDI), the Digital Repository Service, and innumerable tools that sustain and support online resources. Led by Dale Flecker and Tracey Robinson, OIS is home to nationally recognized experts who advised *Sound Directions*. The Library Digital Initiative in some aspects parallels IU's Digital Library Program. Its mandate is to create the technical infrastructure to support the acquisition, organization, delivery, and archiving of digital library materials, provide experts to advise the community on key issues in the digital environment and enrich the Harvard University Library collections with a significant set of digital resources.

⁶ Indiana University, Digital Library Program. <http://www.dlib.indiana.edu/>.

⁷ Harvard University, Loeb Music Library. <http://hcl.harvard.edu/loebmusic/>.

⁸ Harvard University Library, Office for Information Services. <http://hul.harvard.edu/ois/>.

1.3 Standards and Best Practices

1.3.1 Introduction

It is critical that audio preservation systems use technologies, formats, procedures, and techniques that conform to internationally-developed standards and best practices. These are typically developed by technical experts and, if competently implemented, ensure that the output of a preservation system is high-quality. Standards and best practices also provide a philosophical and ethical foundation for preservation work by outlining expectations and goals for the output of a preservation system along with acceptable means to achieve them. Standards-based technologies will presumably be usable longer, fostering sustainability, and are more likely to generate products that are interoperable. Finally, “non-standard formats, resolutions and versions may not include preservation pathways that will enable long term access and future format migration.”⁹ In this sense we place ourselves all in the same boat by adhering to standards, increasing the likelihood that strategies for migration and access will be developed when it is time to move to new technologies.

Formal standards in preservation-related areas are assessed and ratified by bodies such as the International Organization for Standardization (ISO), the National Information Standards Organization (NISO), the World Wide Web Consortium (W3C) and others. Standards crucial to audio preservation are also developed by organizations such as the Audio Engineering Society (AES), the European Broadcasting Union (EBU), the Library of Congress, the Digital Library Federation (DLF) and others that may not be official national or international standards organizations in the strictest sense, but are charged by various constituencies with providing leadership in this area. The publication of best or recommended practices provides guidance in areas where standards do not yet exist or may never be created. Best practices may also provide strategies, procedures or work plans necessary to successfully implement a standard that has been formally adopted. The *Sound Directions* project has implemented and tested the standards and best practices described below.

⁹ International Association of Sound and Audiovisual Archives, Technical Committee, *IASA-TC 04 Guidelines on the Production and Preservation of Digital Audio Objects: Standards, Recommended Practices, and Strategies* (Aarhus, Denmark: International Association of Sound and Audiovisual Archives, Technical Committee, 2004), 6.

1.3.2 IASA-TC 03: The Safeguarding of the Audio Heritage: Ethics, Principles and Preservation Strategy, Version 3, December 2005

TC 04: Guidelines on the Production and Preservation of Digital Audio Objects¹⁰

IASA-TC 03 provides an overview of key audio preservation topics including selection, preservation transfer, digital archiving basic principles, preservation metadata, format priorities for transfer, and others.

IASA-TC 04 is an important high-level recommended practices document for the preservation of audio in the digital domain. This publication includes detailed recommendations for signal extraction from analog sources, equipment in the digital preservation chain, sample rate and bit depth, characteristics of Preservation Master Files, target preservation file format, guidelines for storage, and others.

In effect, best practices developed during Phase 1 of *Sound Directions* put into action both TC 03 and TC 04 principles, using them to produce detailed practices and procedures as reported in this document.

1.3.3 Capturing Analog Sound for Digital Preservation: Report of a Roundtable Discussion of Best Practices for Transferring Analog Discs and Tapes. NRPB, CLIR, LC¹¹

This report summarizes discussions and recommendations from a meeting of audio preservation engineers that was organized by the Council on Library and Information Resources and the Library of Congress under the auspices of the National Recording Preservation Board. The heart of this document is its detailed discussion of issues relating to the analog playback of both discs and tapes. David Ackerman of Harvard University and three members of the *Sound Directions* Advisory Board—Chris Lacinak, George Massenbourg, and Peter Alyea—were invited to participate in this meeting.

1.3.4 Broadcast Wave Format (BWF or BWAV)¹²

The Broadcast Wave Format, based on the Microsoft WAVE audio file format, was introduced by the EBU in 1996 to allow files to be exchanged between the increasing number of digital audio workstations used in radio and television production. Broadcast Wave is a special type of WAVE file that may contain basic metadata (residing with the file itself) about its audio content, and carries a sample-accurate time stamp that can be used to place related files in the proper sequence. BWF is not a destination for the extensive metadata that must be collected

¹⁰ IASA-TC 04 is available through the website of the International Association of Sound and Audiovisual Archives at <http://www.iasa-web.org/> or in the US through Nauck's Vintage Records: <http://78rpm.com/>. International Association of Sound and Audiovisual Archives, Technical Committee, *IASA-TC 03 The Safeguarding of the Audio Heritage: Ethics, Principles and Preservation Strategy*, ver. 3 ([Budapest]: International Association of Sound and Audiovisual Archives, Technical Committee, December 2005). Also available online: http://www.iasa-web.org/IASA_TC03/IASA_TC03.pdf.

¹¹ Council on Library and Information Resources and Library of Congress, *Capturing Analog Sound for Digital Preservation: Report of a Roundtable Discussion of Best Practices for Transferring Analog Discs and Tapes*, CLIR publication no. 137 (Washington, DC: Council on Library and Information Resources and Library of Congress, 2006). Also available online: <http://www.clir.org/PUBS/reports/pub137/pub137.pdf>.

¹² European Broadcasting Union, "BWF – A Format for Audio Data Files in Broadcasting," ver. 1, Tech 3285 (Geneva: Switzerland: European Broadcasting Union, July 2001), http://www.ebu.ch/CMSimages/en/tec_doc_t3285_tcm6-10544.pdf.

during digital preservation projects. The Broadcast Wave Format itself has become a de facto standard in the audio world. In addition to its widespread use in Europe and Australia, it is specifically recommended by IASA, AES, and the National Academy of Recording Arts and Sciences as the target format for audio preservation.¹³

1.3.5 AES31-3-1999¹⁴

Published by the Audio Engineering Society in 1999, AES31 is an international standard designed to enable simple interchange of audio files and projects between workstations. Part 3 includes a format for the communication of edit decision lists, called Audio Decision Lists (ADLs) in the standard, using ASCII text that is human-readable but also may be parsed by software.

AES31-3 is used in archival work to model the relationship between the source recording and resulting digital files. It provides a standard way to link the various files that are created, sometimes through multiple stops and starts during transfer of a deteriorating source, thereby reconstructing the source recording. Without it, future researchers are left with one engineer's interpretation of the edit points. This standard may also be used for the collection of marker information, or cue points, based on the start and stop times of performances in a digital file. As of this writing, this is not officially supported by the standard, but the data may reside in an ADL in a proprietary section depending on a manufacturer's implementation. AES31-3 is under revision to include this marker information as an official part of the standard, with public release expected soon along with eventual adoption by software manufacturers.

1.3.6 AES SC-03-06 Working Group on Digital Library and Archive Systems, Task Group SC-03-06-A Metadata Harmonization

This emerging standard, developed in consultation with the Library of Congress by the AES in a working group chaired by Harvard's David Ackerman, guides the collection of technical metadata for audio objects, including the source recording and file derivatives, as well as the digitizing process. Mike Casey from Indiana University and *Sound Directions* board member Chris Lacinak are active participants in this working group. The standard was implemented during Phase 1 of *Sound Directions* for the first time in a real world project. Both Indiana University and Harvard University have developed software for the collection of technical metadata using this standard.

1.3.7 Open Archival Information System (OAIS)¹⁵

The Open Archival Information System (OAIS) Reference Model, ISO standard 14721:2003, is a conceptual framework for an archival system dedicated to preserving and maintaining access to digital information over the long term. It describes the environment in which an archive resides, the functional components of the archive itself, and the information infrastructure

¹³ See IASA, Technical Committee, *IASA-TC 04*, 7. See also The Recording Academy, Producers & Engineers Wing and Audio Engineering Society, Technical Committee on Studio Practices and Production, "Recommendation for Delivery of Recorded Music Projects," AES Technical Council Document AESTD 1002.1.03-10; 030930 rev 33 (New York: Audio Engineering Society, 2003), http://www.aes.org/technical/documents/AESTD1002.1.03-10_1.pdf.

¹⁴ Available through: Audio Engineering Society, Standards Committee, "Standards in Print," <http://www.aes.org/publications/standards/>.

¹⁵ Consultative Committee for Space Data Systems, *Reference Model for an Open Archival Information System (OAIS)*, CCSDS 650.0-B-1 Blue Book January 2002 (Washington, DC: CCSDS Secretariat, 2002). Also available online: <http://public.ccsds.org/publications/archive/650x0b1.pdf>.

supporting the archive's processes. Due in part to endorsement by OCLC and RLG, the OAIS Reference Model is used by many libraries, archives, and other cultural heritage institutions as a means of defining their own digital preservation infrastructure. Indiana and Harvard are using OAIS concepts in implementing their digital library object repository systems.

1.3.8 Metadata Encoding and Transmission Standard (METS)¹⁶

The Metadata Encoding and Transmission Standard (METS), developed as an initiative of the Digital Library Federation and maintained by the Library of Congress, specifies an XML document format for packaging metadata necessary for both management of digital library objects within a repository and exchange of such objects between repositories, or between repositories and their users. A METS document is capable of wrapping together all of the descriptive, administrative, and structural metadata for a digital object in many versions, plus references to the object's data files, or optionally, inclusion of the data files themselves. METS is frequently used as the wrapper format for OAIS Submission Information Packages (SIPs), Archival Information Packages (AIPs), or Dissemination Information Packages (DIPs).

1.4 Overview of this Publication

Our purpose in writing this publication is to present the results of research and development carried out by the *Sound Directions* project with funding from the National Endowment for the Humanities in the U.S. Our work has naturally led to some conclusions that are detailed and highly technical along with others that are more general. Both are presented here, in separate sections of each chapter as discussed in the note to readers above.

The work undertaken by the *Sound Directions* project focused largely on what happens after analog-to-digital conversion. We report on our experience with pre-conversion parts of the preservation chain, and even offer a few recommended technical practices, but have not attempted to be exhaustive in these areas. The heart of our work begins with the creation of digital files and continues to long-term preservation storage. This fills a sizeable gap in the audio preservation field as there are no best practices documents that address this part of the preservation pathway in detail. Our aim was to use our real world project to add specificity to the best practices that do exist, as well as to develop best practices in areas where they have not yet been established. These are presented by topic at the beginning of the recommended technical practices section in each chapter and as a group in Chapter 8. *Sound Directions* best practices are based on general principles either widely recognized by the audio preservation community or, in a few cases, newly proposed by our project. While tools, formats, and practices will change over time as our field evolves, these basic principles should remain constant. In some areas that are either out of our scope (analog playback, management of preservation repositories, for example) or are necessarily specific to individual institutions (workflow) we have not developed detailed best practices but report on our own operations, which can be used as a starting point for institutions developing audio preservation systems.

We invite you to continue this conversation on audio preservation issues. Questions, comments, and suggestions may be emailed to the *Sound Directions* project at soundir@indiana.edu.

¹⁶ Library of Congress, "METS: Metadata Encoding & Transmission Standard" (12 July 2007), <http://www.loc.gov/standards/mets/>.

2 Personnel and Equipment for Preservation Transfer

2.1 Preservation Overview

If the primary goal of preservation transfer work is the creation of a surrogate that is an accurate, authentic, and very high quality representation of the original, then both the equipment in the preservation system and the personnel operating it are of key importance. Best practices documents provide guidance for a preservation studio's signal chain and personnel, analog playback, and analog-to-digital conversion, as discussed below. In some cases this guidance is specific, while in others it is necessary to apply the knowledge of an audio engineer to derive particular practices from general statements.

Personnel for Preservation Transfer

Familiarity with obsolete media, its historically accepted qualities and characteristics, its production techniques, playback equipment calibration and equipment maintenance is essential for solid preservation transfer. Such familiarity is in decline. The sophisticated technical equipment used in preservation studios must be operated by appropriately trained personnel. IASA-TC 03 and TC 04, in addition to stating that equipment must be optimally adjusted and maintained, suggest that playback “requires knowledge of the historic audio technologies and a technical awareness of the advances in replay technology.”¹⁷

Fragile audio carriers are damaged by the stress of repeated and inexpert playback attempts and lack of timely intervention in the face of playback problems. The CLIR/LC report, “Capturing Analog Sound,” addresses this directly, suggesting that “there are many areas in which a trained ear and years of experience are by far the most important tools....in some archives, fragile audio recordings are being handled, played, and transferred for digital preservation by staff who have limited experience working with audio recordings or little knowledge about the sonic characteristics and weaknesses of various audio formats.” This report strongly recommends, “audio preservation transfers be done by trained and experienced audio engineers.”¹⁸

Professional audio experience, musical knowledge, and the ability to verify or confute their human perceptions with precise measurement, make audio engineers and technicians, rather than automated systems or untrained students, the best candidates for recognizing playback problems and intervening during archival transfers. In addition, engineers and technicians are equipped with the necessary critical listening skills to ensure that not only playback, but also the performance of the studio signal chain itself, is optimal.

Ideally, an audio preservation workflow would also involve the services of a specialized programmer. Software that automates the mechanistic aspects of the work (such as metadata entry) cuts costs, saves time and reduces the opportunity for human error.

Preservation Studios

Best practices documents contain few specific recommendations for the signal chain in a preservation studio. IASA-TC 04 stipulates: “The combination of reproduction equipment, signal cables, mixers and other audio processing equipment should have specifications

¹⁷ IASA, Technical Committee, *IASA-TC 03*, 6; IASA, Technical Committee, *IASA-TC 04*, 3.

¹⁸ CLIR, “Capturing Analog Sound,” 4 and 15.

that equal or exceed that of digital audio at the specified sampling rate and bit depth. The quality of the replay equipment, audio path, target format and standards must exceed that of the original carrier."¹⁹ The CLIR/LC report discusses the need for accurate monitoring systems to evaluate quality as well as test equipment to evaluate potential problems.²⁰ In addition, the characteristics of the room used for preservation transfer work must be carefully considered.

From recommendations such as these, basic audio engineering principles, and experience, we deduce the following:

- The room in which we monitor transfers can be thought of as an unavoidable lens through which the audio content is experienced. Preservation transfer work is best undertaken in a studio designed as a critical listening space. A critical listening space should have an ambient noise level well below that of the quietest sound we wish to audition when listening at a safe, comfortable, non-fatiguing playback level.²¹ The room should not distort the frequency spectrum of interest, the accuracy of the sonic images, the sense of space, or the timing of the audio content
- If a critical listening space is not possible, then the studio must at least be free from ambient noise, it must be removed from other work areas and traffic, and its acoustic weaknesses should be well understood. Knowing the acoustic weaknesses of the room informs one of the aspects of the sound that can be reliably analyzed by ear and those aspects that cannot. This is vital for the engineer who must be able to make accurate judgments during transfer and when selecting and aligning equipment
- All signal chain components must be professional-quality
- The most direct and clean signal path must be used from source to destination. Signal chain components that are not used for preservation transfer work should be removed to prevent additional noise from entering the system
- The studio should include test/calibration equipment to test and monitor the transfer chain itself for noise as well as to test individual components for performance. During transfer, the test/calibration equipment shall not be inserted between the playback machine and the recorder
- The studio should include a monitoring chain that enables the engineer to monitor the signal directly from both the playback machine and after the analog-to-digital converter to verify the quality of the converted signal

In the digital age, preservation studio signal chain components feed the audio signal into a computer where the audio content is recorded and further processed in the digital domain. The computer-based audio workstation, called a digital audio workstation or DAW, historically required dedicated hardware to efficiently process the audio signal. Integrated, turnkey systems with proprietary hardware and software specifically designed for digital audio processes were commonly used. In recent years, as the processing power on the average desktop computer has increased, these systems have declined considerably in popularity. A standard desktop computer can now handle multiple channels of audio, at least in terms of processing power and memory, although dedicated systems may have advantages for applications that require significant signal processing. It is no longer necessary to invest in

¹⁹ IASA, Technical Committee, *IASA-TC 04*, 8.

²⁰ CLIR, "Capturing Analog Sound," 25.

²¹ An example of one room characteristic that must be addressed is noise level. Richard Warren's storage document published in the ARSC Journal recommends a Noise Criteria-level of 20-25 dB for critical listening areas. More generally, he also calls for consideration of the "proper acoustical conditions to prevent the room from distorting the sounds to be studied." Richard Warren, Jr., "Storage of Sound Recordings," *ARSC Journal* 24, no. 2 (1993), 137. Readers are also directed to the publications of the Audio Engineering Society for detailed information on the characteristics of audio studios.

expensive proprietary, dedicated systems for audio preservation transfer work. If this work is relatively simple, emphasizing signal capture without much downstream manipulation, using a carefully designed desktop audio methodology with what is sometimes called a host-based or native audio processing system is a valid approach. Both native and dedicated hardware/software systems are viable options for audio preservation.²²

Analog Playback

Although all points along the preservation chain are important, audio preservation engineers generally agree that playback of analog source recordings is a particularly key juncture at which, if not performed optimally, the quality of the end product will be lessened. According to IASA-TC 04 “any transfer should attempt to extract the optimal signal from the original [as] the original carrier may deteriorate, and future replay may not achieve the same quality, or may in fact become impossible, and secondly, signal extraction is such a time consuming effort that financial considerations call for optimization at the first attempt.”²³ No amount of effort or expense in the remainder of the signal chain can recover information that was not retrieved from the analog original at the moment of playback. TC 04, as well as the CLIR/LC report “Capturing Analog Sound,” provides detailed best practices for the playback of analog recordings.

Both the abilities of staff and the equipment used greatly impact the success of the analog playback stage. The engineer must understand how field recordings carried on obsolete, deteriorating historic formats may be optimally reproduced despite degradation, taking into account specific characteristics of both the individual recording and the format itself. The engineer must also align, calibrate, and verify the performance of the playback machine, which itself must be able to reproduce the recording at the highest fidelity possible.

The analog playback stage must utilize the highest quality copy of the content that is available. For recordings made in the field this is usually, although not always, the original recording. In some cases the original may have deteriorated to the point that a first copy is the highest quality carrier of the content. Locating and identifying the best copy in existence, even if it resides in another archive, will enable the judicious use of preservation resources, prevent duplication of effort, and result in carrying the highest quality version forward into the future. In order to enable future re-consultation for the purpose of assessing past work, analyzing secondary information such as notes on a container, or other reasons, all original recordings should be retained.²⁴

Conversion

If analog playback is one exceptionally key juncture in a preservation system, then analog-to-digital (A/D) conversion is the other. Choices made in both of these areas can dramatically and permanently affect the fidelity of the audio signal that is carried in the digital domain into the future. Ken Pohlmann, in a paper published by CLIR states that “errors introduced by the A/D converter will accompany the audio signal throughout digital

22 This discussion is largely from Francis Rumsey, *Desktop Audio Technology: Digital Audio and MIDI Principles* (Oxford; England, Burlington, MA: Focal Press, 2004), 2 and 156.

23 IASA, Technical Committee, *IASA-TC 04*, 11.

24 See IASA, Technical Committee, *IASA-TC 03*, 7; IASA, Technical Committee, *IASA-TC 04*, 11; and International Association of Sound and Audiovisual Archives, Editorial Group, *Task Force to Establish Selection Criteria of Analogue and Digital Audio Contents for Transfer to Data Formats for Preservation Purposes* (Hungary, October 2003), 5. Also available online: <http://www.iasa-web.org/taskforce/taskforce.pdf>.

processing and storage and, ultimately, back into its analog state.”²⁵ The technical quality of the audio signal can never be made better once converted. IASA-TC 04 suggests that the converter “is the most critical component in the digital preservation pathway.”²⁶

Best practices documents all recommend a professional-quality stand-alone (external) A/D converter rather than one incorporated into the computer’s sound card with its low cost circuitry subject to the electrical noise inside the computer.²⁷ IASA-TC 04 includes minimum specifications for such a converter, although it is often difficult to match specifications to the information provided by any given manufacturer. Pohlmann suggests both measurement and listening tests before purchase but also counsels seeking expert advice, perhaps recognizing that most archives are not capable of conducting scientifically valid tests in either of these areas. It may be necessary to engage the audio engineering community as there does appear to exist an informal, short list of converters that engineers believe are of high-enough quality for preservation transfer work. These tend to range in price from around \$1,000 to \$10,000 and more.

Both the IASA and Pohlmann documents assert that audio transparency—neither adding to nor subtracting from the audio signal present on the analog original—is the most important characteristic for a converter used for preservation transfer. Most converters are not transparent, only the best approach transparency, and the differences are apparently audible to some audio engineers, although they may be subtle.²⁸ However, some feel there may be diminishing returns in analyzing perceivable improvement in quality versus increase in price, especially with professional-quality devices. The performance of A/D converters, many of which use the same brand of converter chip, often relies on other factors such as how well the analog input stage is implemented and the design of the circuitry supporting the chip.²⁹

The characteristics of digital conversion are established at the A/D converter with the choice of sampling rate and word length or bit depth. The audio CD was established with a sampling rate of 44.1 kHz at a bit depth of 16. This combination is now almost universally considered inadequate for audio preservation of analog recordings. There is currently wide agreement on bit depth for preservation transfer of analog sources with 24 bits recommended. A well-designed converter operating at 24 bits will provide a noise floor at the limits of audibility and capture enough dynamic range to make level setting less critical. It will also provide a longer word length to allow for some types of downstream processing stages (of derivative files) that may decrease useful word length.³⁰

There is less agreement on sampling rate and this topic remains somewhat controversial. IASA-TC 04, the CLIR/LC document, the Pohlmann article on converters, along with other

25 Ken C. Pohlmann, “Measurement and Evaluation of Analog-to-Digital Converters Used in the Long Term Preservation of Audio Recordings” (roundtable discussion, “Issues in Digital Audio Preservation Planning and Management,” Washington, DC, March 10-11, 2006). Also available online: <http://www.clir.org/activities/details/AD-Converters-Pohlmann.pdf>.

26 IASA, Technical Committee, *IASA-TC 04*, 6.

27 An external converter is recommended by IASA, Technical Committee, *IASA-TC 04*, 6 and Pohlmann, “Measurement and Evaluation of Analog-to-Digital Converters,” 8 and 12. It is also recommended in Rumsey, “Desktop Audio Technology,” 13.

28 See Pohlmann, “Measurement and Evaluation of Analog-to-Digital Converters,” 2 and IASA, Technical Committee, *IASA-TC 04*, 6.

29 See Pohlmann, “Measurement and Evaluation of Analog-to-Digital Converters,” 6 for example.

30 See Pohlmann, “Measurement and Evaluation of Analog-to-Digital Converters,” 3-4 for reasons to digitize at 24 bit.

sources all recommend higher sampling rates than 44.1 kHz for the following reasons:³¹

- It is important to accurately capture noise, such as clicks and pops on a disc, and other inaudible, high frequency information so that improved signal processing algorithms in the future that are able to take advantage of higher frequency information will have enough data to work as effectively as possible. Some of this noise resides in frequency ranges higher than can be captured at 44.1 kHz
- Higher sampling frequencies enable manufacturers to build better anti-aliasing low-pass filters that operate more efficiently, thereby improving performance within the range of human hearing³²
- Many musical instruments are capable of producing information in higher frequency ranges—including inaudible higher frequency harmonic content that also impacts our perception of sounds
- Higher sampling frequencies provide improved temporal response, or the timing of the arrival of sounds, that in turn improves spatial imaging (the locations of sounds from within a stereo or surround sound-field)
- The limit of human hearing acuity is not yet known, therefore the point of transparency of a recording system cannot be known³³

IASA-TC 04 recommends encoding to linear pulse-code modulation (PCM) with a minimum sample rate of 48 kHz, and for many purposes suggests transferring at 24 bit with a 96 kHz sampling rate. In fact, 24/96 has become the standard choice for audio preservation reformatting. For the reasons listed above, and for the format's wide support and sustainability, both *Sound Directions* institutions have selected 24 bit, 96 kHz linear PCM encoding. For the evaluation of other potential encoding schemes such as 1-bit sigma-delta, the cautious preservationist is well served by the work of Caroline R. Arms and Carl Fleischhauer at the Library of Congress on the sustainability of digital formats.³⁴ This document explores a number of sustainability factors for any digital format including disclosure, adoption, transparency, self-documentation, and impact of patents.

Creating these “high resolution” digital audio files is analogous to practices employed in the still image preservation world, where the term “rich” is sometimes applied to high quality preservation masters. Although there are obvious differences between the media that are the targets of preservation, there are some overlapping issues. According to one imaging tutorial, creating a rich digital master

- provides enough information to eliminate the user's need to use the original;
- satisfies all research, legal, and fiscal requirements related to use;
- supports creation of higher quality derivatives especially if they are processed;
- accommodates future applications as user expectations are likely to be more demanding over time.³⁵

31 IASA, Technical Committee, *IASA-TC 04*, 6; Pohlmann, “Measurement and Evaluation of Analog-to-Digital Converters,” 2-4; and CLIR, “Capturing Analog Sound,” 11.

32 A 44.1 kHz sampling rate requires the converter to have a steep anti-aliasing filter that smears high frequency energy over a wide frequency band within the range of human hearing.

33 This and the previous two points are from Pohlmann, “Measurement and Evaluation of Analog-to-Digital Converters,” 3.

34 Library of Congress, National Digital Information Infrastructure and Preservation Program, “Sustainability of Digital Formats Planning for Library of Congress Collections” (21 May 2007), <http://www.digitalpreservation.gov/formats/>.

35 Paraphrased from Cornell University Library, Research Department, “Moving Theory into Practice: Digital Imaging Tutorial,” Cornell University Library, <http://www.library.cornell.edu/preservation/tutorial/conversion/conversion-03.html>. This document also suggests, for still images, matching the conversion process to the informational content of the original and to scan at that level and no more or less. Note that informational content is more easily defined for still images than for audio recordings, which are time-based media and generally more complex.

2.2 Recommended Technical Practices

2.2.1 Personnel for Preservation Transfer Work

2.2.1.1 Best Practices

***Best Practice 1:** Use audio engineers and technicians with solid technical skills and well-developed critical listening abilities at points in the preservation transfer workflow where their skill is required.*

2.2.1.2 Rationale

Even though this best practice echoes the IASA-TC 04 and CLIR/LC documents discussed above, we feel that it bears restating based on the experience of both institutions hiring and training audio engineers for preservation work.

2.2.1.3 Staff at Indiana and Harvard

Both *Sound Directions* institutions have found that there is no substitute for experience with audio formats and equipment. Nor is there a substitute for the critical listening skills that are informed by such experience. Therefore, both institutions use professional audio engineers with knowledge of historical formats for preservation transfer work. Engineers with excellent technical and critical listening skills who lack experience with some historical formats receive additional training from either in-house sources or outside consultants. Both institutions have utilized programmers as part of the audio preservation staff, but only with the help of external grant funding. Unfortunately, with rapidly changing technology, one-time programming has a short shelf life. Programming staff would ideally be available on an as-needed basis to support the creation of tools as audio transfer workflows evolve.

Harvard has committed to using only audio engineers or technicians for archival transfers. The distinction between an audio engineer and a technician is based upon experience. An audio engineer has a broad range of audio production experience in general, and has experience with historical formats for preservation in particular—along with knowledge of the entire preservation workflow. A technician has some general audio production knowledge, and is skilled in specific tasks of the preservation workflow. Roy Pritts, Past Chairman of the Audio Engineering Society Education Committee, considers a technician to be “...one who takes the prescription or design of an engineer and implements it.”³⁶ In practice, the lines between engineer and technician become less distinct as technicians broaden their experience and education. At that point, one of the challenges might be developing the necessary unit structure in order to support these increasingly valuable individuals in their current role, or expanding their responsibilities, and possibly expanding the unit. Another challenge as we learn and grow is to retain hard-earned knowledge so that it is not lost as personnel eventually change. At the HCL-APS we have instituted a staff knowledge-base wiki in order to document our learning and our procedural changes, and to aid in training new staff in our procedures. We also plan training sessions with subject matter experts when adopting new technologies and methods. At this time, Harvard has committed to assigning a single audio engineer or technician to monitor an entire single transfer only. We do not

³⁶ Roy Pritts, “The Educated Audio Engineer: A Life-Long Learner,” Audio Engineering Society, Education Committee (2007), <http://www.aes.org/education/pritts.html>.

yet see sufficient benefit to dividing one individual's attention among multiple, simultaneous monitoring tasks, but we will continually re-evaluate the benefits and risks.

Our two preservation studios are operated through a first shift by audio engineers and second shift by audio technicians. What would be the third shift time slot is reserved for automated data backup. Rounding-out the staff of HCL-APS is our programmer for the *Sound Directions* project, who together with our lead engineer has developed a suite of scriptable software tools that has streamlined our preservation workflow, thereby allowing the engineers and technicians to focus their talents more effectively on the critical archival transfers.

The Indiana University Archives of Traditional Music has specific experience informing its decision to use audio engineers for preservation transfers. Over many years both talented graduate students and audio engineers have undertaken transfer work and, in fact, we still actively use graduate students for access-only digitizing to fill orders for CDs of our holdings. We have learned through this experience that trained audio engineers with highly-developed critical listening skills and deep technical knowledge are necessary for preservation-quality transfer work for most analog sources. Throughout the *Sound Directions* project we have observed the innumerable decisions that must be made—aligning tape machines and verifying their performance, repairing a deteriorating tape or disc, setting levels, analyzing completed files, monitoring converter performance and studio noise floor, selecting styli, determining track configuration, adjusting azimuth, to name a few— which require technical skill, judgment, and effective critical listening and directly impact the quality of the final product. The ATM does, however, see a role for students with technical aptitude. We have plans to use students from the IU Department of Recording Arts for preservation transfer of non-problematic sources to increase throughput into our preservation system. These students, who have formal audio training, may be considered apprentices and will be trained and supervised by the *Sound Directions* engineer. The ATM produces written procedural documents that guide the work of both audio engineers and audio student workers and serve to document and maintain institutional knowledge.

Both institutions also realize that not every audio engineer is able to understand the principles of archival preservation transfer work which is not only different from, but sometimes at odds with, the commercial sector. It is critical to use engineers who are interested in learning an archival adaptation of their craft and who can commit to fulfilling the preservation mission. These engineers do not always have formal academic training and many have learned their profession through the time-honored apprenticeship tradition. In order to maintain the flexibility to hire the most qualified candidate it is important when advertising for engineering positions to specify technical skills and knowledge that are desirable, including a good sense for music and performance, but not necessarily academic qualifications. Finding personnel who are equally capable in both the playback of obsolete formats and operating modern digital audio workstations is a significant challenge, and it is reasonable to expect that additional training will be necessary to fill gaps in an engineer's knowledge.

2.2.2 Studio Signal Chain for Archival Preservation

2.2.2.1 Best Practices

Best Practice 2: Perform preservation transfers in an appropriately designed, critical listening environment. If such a space is not available, choose a room that is quiet and is removed from other work areas and traffic, and be acutely aware of its sonic deficiencies.

Best Practice 3: Route the signal from the playback machine to the analog-to-digital converter using the cleanest, most direct signal path possible.

Best Practice 4: Design the monitoring chain to allow instant comparison of the signal from the playback machine to the signal that has passed through the analog-to-digital converter.

Best Practice 5: Preservation studios must include test/calibration equipment to test and monitor the transfer chain itself for noise as well as to test individual components for performance. During transfer, the test/calibration equipment shall not be inserted between the playback machine and the recorder.

2.2.2.2 Rationale

The best practices cited above evolved from both our work and the documents discussed in section 2.1. The project's initial objective in this section was to provide a case study of two audio preservation studios. However, our experience was particularly compelling in several areas and these critical best practices emerged. Note that we have not attempted to be comprehensive—the reader must engage the audio engineering community for further specifics on audio preservation studios that we have not covered.

Here is the reasoning behind the above best practices:

- Working in a critical listening environment enables the engineer to accurately evaluate the audio when monitoring the preservation transfer
- Keeping the signal path in the transfer chain as direct and clean as possible helps in meeting the basic preservation objective of obtaining the most accurate, highest-quality transfer of the original possible
- The ability to monitor the signal from both the playback machine and post-A/D converter enables verification of the A/D conversion and allows easier diagnosis of potential problems heard during transfer
- Test/calibration equipment is essential for verifying such things as the studio noise floor and the performance of the signal chain including playback devices

Below are some basic principles that we have found useful in designing our audio preservation studios:

- Design the preservation studio as a critical listening environment and know its limitations
- All signal chain components must be tested so that they are known to be of professional-quality, that they are reliable, and that they do not alter the level or quality of the audio signal at unity
- The most direct and clean signal path from source to destination must be used at all times. There may be no unused devices in the signal path. If there are multiple destination formats

for the transfer, then the signal must be routed in parallel without any daisy-chaining of devices

- Signals shall be split or distributed using only calibrated, high-quality distribution amplifiers, routers, or properly designed and wired balanced cables and patchbays that demonstrably do not degrade the signal
- Use the highest quality signal format present on the source equipment and throughout the chain. For instance, use a balanced signal source rather than an unbalanced signal source

2.2.2.3 Preservation Studio at the IU Archives of Traditional Music

The audio preservation studio at Indiana is a purpose-built facility designed to do one thing—preservation transfer work—simply and well. It was configured to incorporate just enough flexibility to handle this one function. Projects that require flexible routing, usually access-related, are handled in other ATM studios. The preservation studio was intentionally developed within the bounds of a budget typical of a large preservation grant. Audio is handled natively on a standard Dell PC using an excellent external analog-to-digital converter and WaveLab audio recording/editing software. Our goal was to produce preservation-worthy digital objects equivalent in quality to those produced on all but the highest-end systems, and we believe we have been successful, although engineering skill is notably critical to this success. Below is a generic diagram of the signal chain in the ATM Preservation Studio.

Transfer Chain

Monitoring Chain

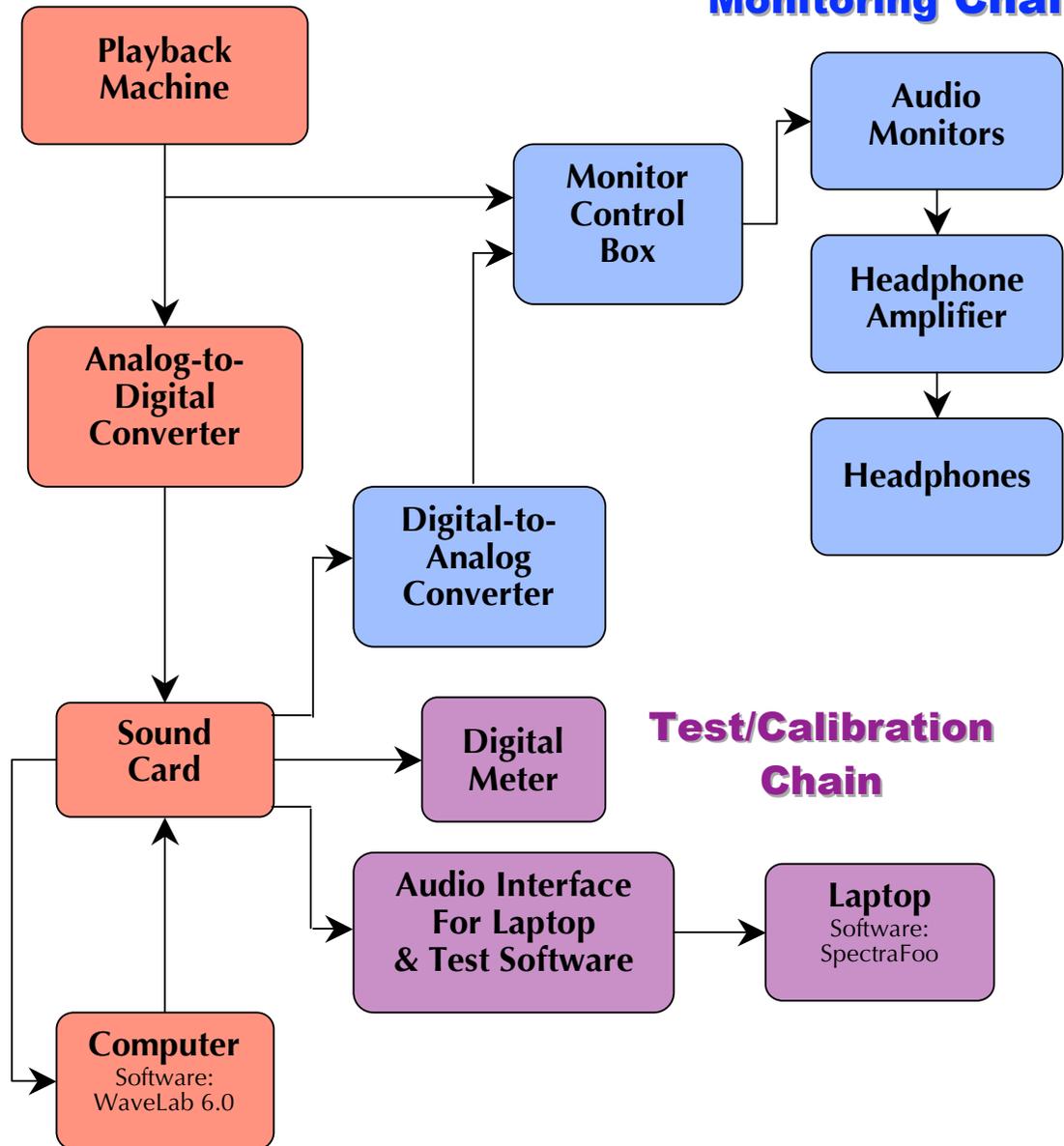


Figure 1: Indiana University ATM preservation studio signal chain

2.2.2.3.1 Indiana University Transfer Chain

Characteristics of the transfer chain:

- The playback machine is hardwired directly to the analog-to-digital converter for tape transfers using one cable that is physically moved as the playback machine changes. Moving this cable is only necessary when moving to a new collection or format
- The signal is split after the playback machine with one side routed to the converter and the other to the monitoring chain. The split is done by a cable built by hand by the engineer taking into account impedance matching issues in our relatively simple signal chain
- We do not object to including a patchbay in the chain but haven't as yet felt it necessary
- If we need additional gain for a tape transfer we can insert a line-level amp into the chain.
- For disc transfers the signal from the turntable is sent to a preamp which is then routed directly to the converter

The audio recording/editing software used at the ATM for preservation work is WaveLab 6.0. We chose WaveLab for its support of both the Broadcast Wave Format and the AES31-3 standard. We also prefer many of its features, such as the audio analysis tools which we use for quality control.

Below are two tables with the specifics of the transfer chain—first, for tape and second, for disc transfers.

Device Type	Device	Channel/Connector	Comments
Playback machines	Studer A810 or Tascam 122 MKII	Analog/XLR	
Analog-to-digital converter	Benchmark ADC1	Digital AES/EBU output/XLR	
Sound card	Lynx AES16	Digital AES/EBU input/DB 25 connector	PCI card
Computer	Dell Optiplex GX620		Pentium 4 processor, 3.8 GHz, 2.0 GB RAM
Slow speed playback machine and line amp	Revox B77 and Gaines Balanced Line Interface	Analog RCA outputs from Revox	For tapes recorded at 1.875 and 0.938 ips. The Gaines device is inserted between the Revox and converter

Table 1: Devices in the transfer chain for tape transfers

Device Type	Device	Channel/ Connector	Comments
Playback machine	Technics SP-15 turntable	Analog RCA outputs from the cartridge	Includes SME 3012 tonearm, Stanton 500 cartridge, various styli
Preamp 1	KAB Souvenir EQS MK12 using flat setting	Analog RCA in/ Analog balanced TRS out	
Preamp 2	Owl 1 using a playback eq curve	Analog unbalanced RCA in and out	Used only when necessary for playback curve. Both preamps are used together to generate flat and equalized files at the same time
Same converter, card, and computer as above			

Table 2: Devices in the transfer chain for disc transfers

2.2.2.3.2 Indiana University Monitoring Chain

Table 3, below, details the monitoring chain in the ATM Preservation Studio.

Device Type	Device	Channel/ Connector	Comments
Digital-to-analog converter	Benchmark DAC1	Digital AES/EBU input/Analog XLR output	
Monitor controller	StudioComm Model 55/ Model 56	Balanced TRS input	Accepts inputs from playback machine and D/A converter. Routes to speakers and provides switches to choose listening source
Audio monitors	Genelec 8040A		
Headphone amp	StudioComm Model 35		
Headphones	Ultrasone Proline 650		

Table 3: Devices in the monitoring chain

2.2.2.3.3 Indiana University Test/Calibration Chain

The test/calibration chain in the ATM preservation studio is a parallel sidechain to the transfer chain, and features a laptop running Metric Halo's test/calibration software, SpectraFoo.³⁷ Because SpectraFoo is only available for the Mac, we use Metric Halo's Mobile I/O as an audio interface for the Mac laptop. SpectraFoo is used for the following:

- Regularly checking the overall noise floor in the studio using a spectrum analyzer
- Reading the results of azimuth adjustments for tape machines during alignment using an x-y display
- Calibrating levels throughout the studio using a level meter
- Checking for consistency in the overall frequency response at the beginning and ends of tapes during transfer using a spectrum analyzer
- Assessing the frequency response of tape machines using a precise level meter

Below is a table that provides details of our test/calibration chain.

Device Type	Device	Channel/Connector	Comments
Audio interface	Mobile I/O 2882	Digital AES/EBU input/ Firewire output	Serves as a high quality interface between the sound card and the laptop
Computer	Apple Powerbook G4 laptop		SpectraFoo is available only for the Mac
Test software	Metric Halo's SpectraFoo 4.0		

Table 4: Devices in the test/calibration chain

We quickly learned through practical experience that test equipment and software are essential for preservation work. While assessing the preservation studio shortly after installation, SpectraFoo revealed a noise floor that appeared high at certain frequencies. Later in the day while taking notes on this problem, the engineer noticed that this spike had decreased dramatically without anything else changing. We finally traced the problem to a dimmer controlling a bank of lights in a nearby room. Dimmers are notorious for creating noise problems in audio systems. Due to the sensitivity of the SpectraFoo tools, we were able to uncover this problem immediately and correct it. It otherwise might have remained undiscovered for some time, putting inappropriate noise into our digital files. SpectraFoo also revealed other less dramatic, though still unwanted, problems with our studio noise floor that we attribute to electromagnetic interference (EMI).³⁸

³⁷ Metric Halo. http://www.mhlab.com/metric_halo/.

³⁸ We tested nearly every location within the ATM for null points (lower interference) by loading the laptop with SpectraFoo and the Mobile I/O on top of a Studer open reel tape machine and rolling it throughout the facility. Some amount of EMI is inevitable in most buildings but care should be taken to minimize it by finding the location with the lowest level. During the process of assessing both of these problems we also engaged university electricians to check all outlets for proper wiring and grounding. As part of this work we were able to match outlets to circuits and to place all preservation studio equipment on a dedicated circuit.

2.2.2.4 Preservation Studios at the Harvard College Library's Audio Preservation Services (HCL-APS)

At this time, HCL-APS is a two-room facility with accompanying machine room, office space, and shelving and supply space. All of the computers and DAWs are accessed from either of two keyboard and monitor positions in either room using a keyboard and video monitor (KVM) switch. Each room is equipped with Genelec S-30DVR M magnetically-shielded studio monitors. Normally, each room has two open reel machines including an Otari MX-5050B MK III for 1 7/8 ips transfers and either an Ampex ATR-102 or Studer 807. Each room is otherwise similarly equipped except that Studio 'A' is designated as the archival disc transfer room, since it has a Technics SP-15 turntable for coarse-groove discs and a Leica MZ75 microscope which swings over the turntable for examining and measuring grooves, an Owl 1 phono pre-amp and a Stanton 310B phono pre-amp. For LP and 45 rpm vinyl discs in studio 'A' there is a Technics SL1200 turntable. Both studio 'A' and studio 'B' were constructed to provide a similarly accurate monitoring environment.

2.2.2.4.1 HCL-APS Preservation Signal Chain

Harvard's preservation studios use an analog signal path that flows from the playback device to a patchbay. The patch points in the bay are either "normalled"³⁹ to a mixing console for monitoring purposes only, or normalled to the inputs of the next logical device such as from the turntable to the phono pre-amplifier or from the phono pre-amplifier to a level-matching device like our Gaines Audio Balanced Line Interface. The patchbay allows us the convenience and flexibility to easily monitor or test the signal at multiple stages in the signal chain.

During the transfer of analog media, the analog signal is connected through the patchbay to the inputs of the Prism AD-2 Analog-to-Digital Converter for digitization. The AD-2 is set for internal synchronization. The AD-2's AES/EBU balanced digital audio outputs are connected directly to the AES/EBU inputs of our Merging Technologies Pyramix Digital Audio Workstation for recording onto our Storage Area Network appliance, or SAN. The Pyramix DAW's AES/EBU output is connected to a Prism DA-2 Digital-to-Analog Converter. The DA-2's analog outputs are normalled in the patchbay to, and monitored through, the mixing console. One of the DA-2's multiple digital outputs are connected to a Metric Halo, Mobile I/O Firewire Interface which is the digital audio interface to the Macintosh G4 or G5 computer running SpectraFoo audio analyzing software. In this configuration we are analyzing the results of the entire digitization signal chain. During transfer we watch SpectraFoo's analysis of the signal for anomalies. This procedure greatly benefited the quality of our transfers. We can see as well as hear potential problems.⁴⁰ Alternatively, we can analyze the digital audio output of the AD-2 with SpectraFoo by connecting one of the AD-2's outputs to the Mobile I/O. In addition, we use an Audio Precision Cascade System Two, Dual Domain Audio Analyzer at specific nodes in the signal chain when necessary. Since we have one Audio Precision system at our two-room facility, this very powerful tool is normally used to address specific problems rather than for general analysis of the transfer. It is worth noting that during transfer the analysis is done from a separate output of the AD-2 so that the signal does not pass through any analyzing equipment or analyzing software on its way to being recorded.

³⁹ "Normalled" means that the devices in question are normally connected unless a patch cord is inserted into the jack; which would break the normal signal flow.

⁴⁰ Similarly to IU's experience, using SpectraFoo, HCL-APS engineers quickly discovered that our Studer A807 open reel tape player was susceptible to an EMI field in a particular orientation, with the quarter-track head stack installed. The problem was solved by modifying the head stack to accept a hum shield and reorienting the Studer until the offending interference was minimized.

For the transfer of digitally-recorded, tape-based media such as DAT, the AES/EBU digital audio output of the DAT player is connected directly to one of the AES/EBU inputs of the Prism AD-2 which is set in D-D mode and receives synchronization from the incoming digital audio signal. The remainder of the signal chain after the AD-2 is as previously mentioned.

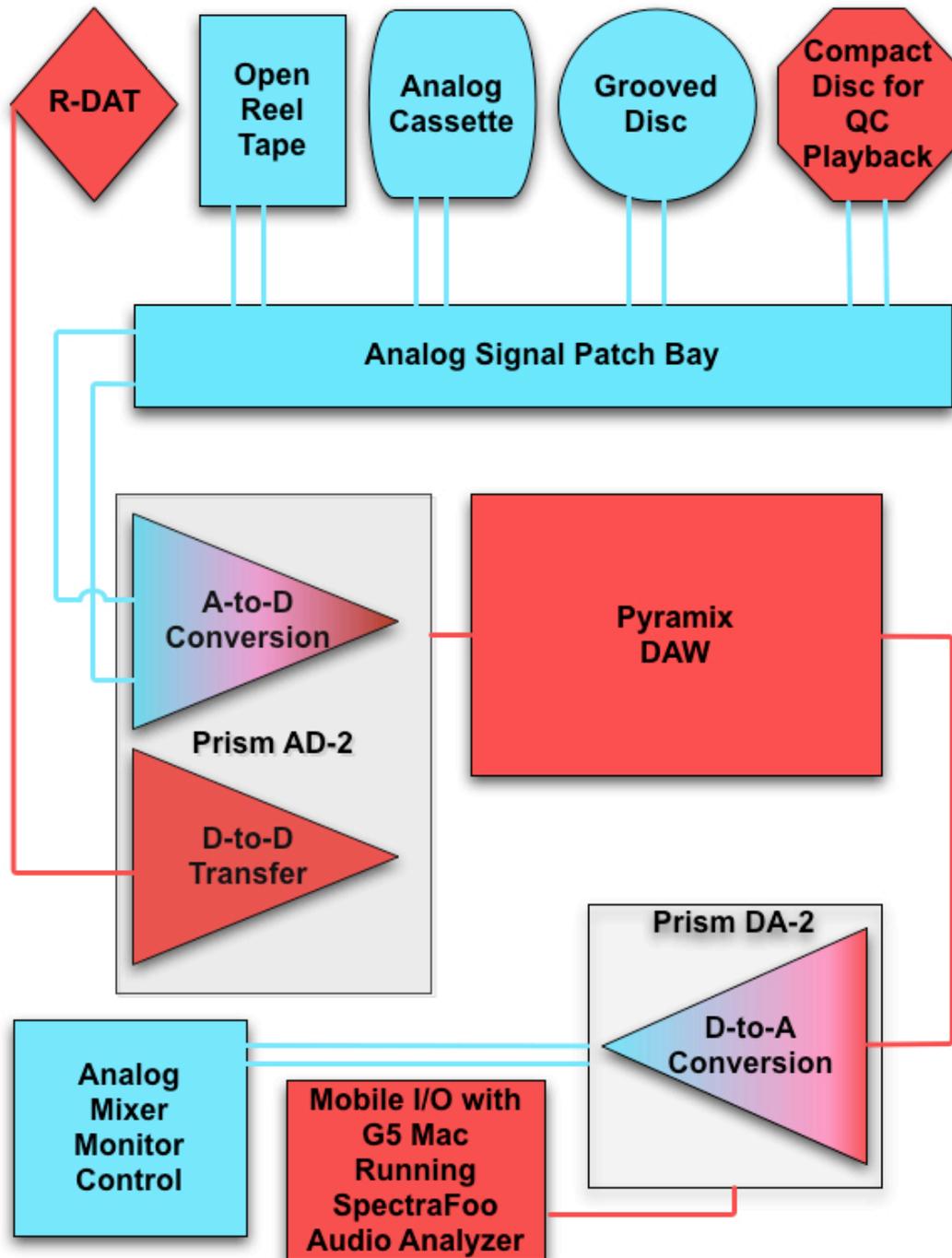


Figure 2: Harvard College Library Audio Preservation Services preservation studio signal chain

2.2.3 Analog Playback

2.2.3.1 Best Practices

The *Sound Directions* project did not attempt to define best practices for analog playback since this topic is covered in detail in both IASA-TC 04 and the CLIR/LC report entitled “Capturing Analog Sound for Digital Preservation.”⁴¹ Rather, we committed to reporting on the equipment, procedures, and techniques that we used in order to create a case study of how two sound archives are able to meet both audio engineering and audio preservation standards and follow best practices. Our report will focus on a few specific areas not covered in the above works in which we gained knowledge that was new to us.

2.2.3.2 Analog Playback at Harvard and Indiana

In general, the *Sound Directions* audio engineers follow all of the recommended practices for analog playback in both IASA-TC 04 and the CLIR report, as well as basic audio engineering practices, including

- aligning tape machines using Magnetic Reference Laboratory (MRL) calibration tapes;
- inspecting the source tape for preservation problems;
- adjusting azimuth (using program material as there are rarely test tones available);
- determining track configurations by listening and by using a magnetic viewer;⁴²
- determining tape thickness using a thickness gauge with enough precision to measure either mils or micrometers;⁴³
- in the absence of test tones, setting levels by previewing tapes (without preservation problems) to determine the average level recorded on the tape;
- assessing the playback eq curve required, if any, as recommended in IASA-TC 04;
- storing the tape in played position, tails out if it is recorded in one direction or heads out if recorded in two directions in order to evenly pack the tape winds.

The above is a partial list focusing on tape transfers; further details may be found in the publications cited above.

The Preservation Studios at Indiana and Harvard contain similar and, in many cases, the same playback equipment, as detailed below.

2.2.3.2.1 Tape

The ATM at IU uses the following quarter-inch open reel tape machines for preservation transfer work:

- Studer A810 full-track playback head, 3.75 ips to 30 ips
- Studer A810 half- and quarter-track 2-channel playback heads, 3.75 ips to 30 ips
- Revox B77 MKII half- and quarter-track playback heads, 0.938 (15/16) ips and 1.875 (1 7/8) ips

41 CLIR, “Capturing Analog Sound.”

42 Arnold B-1022 Magnetic Viewer. Plastiform Division of Arnold. 1000 E. Eisenhower Ave, Norfolk, NE 68702-1567. 402-371-6100 ext. 176.

43 An example of a thickness gauge is the Mitutoyo dial caliper No. 7326S, which is calibrated in mils. This device, along with others, may be purchased from Precision Graphic Instruments, Inc. 1-800-280-6562.

<http://pgiinc.com>. HCL-APS uses an inch-micrometer whose thimble is divided into .001-inch (1 mil) graduations and has a vernier scale with graduations in tenths of a mil.

The HCL-APS uses the following quarter-inch open reel tape machines:

- Ampex ATR-102 half- and quarter-track, 4-speed with vari-speed controller
- Studer A807 3-speed with full-track, half-track/two-track, and quarter-track head stacks
- Otari 5050-B MK III half- and quarter-track used only in conjunction with an Eventide Harmonizer as controller for playback of 1.875 ips tapes

Very few quarter-inch open reel tape machines are manufactured today and archives typically must use older units for preservation transfer work. The condition of older playback machines must be carefully assessed and they are likely to need repair work. Both of the ATM's Studers were refurbished by ATR Services to bring them up to their original performance specification.⁴⁴ In addition, the playback heads on professional-quality open reel machines are usually optimized for the professional tape playback speeds of 15 and 30 ips. The holdings of sound archives are often recorded at slower speeds, particularly 3.75 and 7.5 ips. For example, we estimate that roughly 55-60% of the Archives of Traditional Music's open reel holdings are recorded at 7.5 ips, 25-30% at 3.75 ips, around 5% at 1.875 ips, a handful at both 0.938 ips and 15 ips, and none at 30 ips. It became obvious that this was a problem when aligning the machines at 3.75 ips. Above 2 kHz each machine was down anywhere from 1 to 5 dB from 0VU, with -4 dB typical, depending on the specific machine, channel, and frequency. This is not acceptable performance for preservation work, so we arranged through JRF Magnetic Sciences to have custom playback heads with smaller head gaps manufactured by Flux Magnetics.⁴⁵ Typical head gap length is 120 microinches. The gap on the custom heads is 80 microinches and the lamination thickness is thinner, making them more responsive to high frequencies at slower speeds. By modifying the machine's playback equalizers we shifted the transition frequency (above which the high frequency reproduction adjustment is made when aligning the machines), moving it down to 3.15 kHz. With these two changes our tape machines are now flat to about 10 kHz and up less than 1 dB at higher frequencies when operating at the slower speeds.

Both the ATM and HCL-APS use Tascam 122 MKIII cassette tape machines for preservation work. While we like the idea of using the currently popular Nakamichi Dragons for this work because they adjust for azimuth automatically, neither institution yet owns one. We have found it quite easy to adjust azimuth on the Tascam machines while the tape is playing, using the screw located at the lower left of the head assembly.

The original performance specifications, and in some cases procedures for calibration, for most equipment can be found in the operations or repair manuals. These documents are indispensable to preservation studios.

2.2.3.2.2 Field discs

Both the ATM and the HCL-APS use a Technics SP-15 Direct Drive Turntable with an SME Tonearm for playback of field discs. Both also have Technics SL-1200 Direct Drive Turntables for playback of 45 rpm and 33.33 rpm LP discs. The ATM purchased a number of differently shaped and sized custom styli from Expert Stylus in the U.K. to optimize the interface with both the size and individual wear pattern of any given disc.⁴⁶ Similarly, HCL-APS also has a wide selection of styli from which to choose. For general cleaning of discs that can tolerate cleaning, both institutions use a Keith Monks record-cleaning machine with the appropriate

44 ATR Services. <http://www.atrservice.com/>.

45 JRF Magnetic Sciences. <http://www.jrfmagnetics.com/>.

46 Expert Stylus Company. Omega House. 50 Harriotts Lane. Ashtead. Surrey KT21 2QB. United Kingdom and Northern Ireland. info@expertstylus.co.uk.

cleaning solution.

This is a fairly typical disc setup for archives. With the current popularity of vinyl LP's amongst some audiophiles there is a large selection of high-end disc-related equipment available. While some of this gear may well be higher-quality, most of it is prohibitively expensive for archives and does not enable playback of discs that were recorded at 78 rpm or are larger than twelve inches.

At Indiana, discs requiring a playback equalization curve (many field discs do not) are transferred both with and without the curve at the same time in one pass, and both files are preserved to maintain maximum flexibility into the future. The signal through the flat setting of the KAB preamp generates the Preservation Master File. The signal is also routed through an Owl 1 preamp using a playback curve and this file becomes a Preservation Master-Intermediate as discussed in Chapter 3, below. There is some disagreement among preservation engineers on whether a disc that needs a playback equalization curve should be transferred flat with the curve added later (either in the digital domain or by looping back through an analog device, both of which currently have technical disadvantages), or whether the engineer should choose the curve and apply it in the analog domain during the transfer. This latter course has the disadvantage of relying on both the engineer's choice and the curves available in the preamp. The CLIR/LC report recommends transferring flat if the specific playback curve is unknown, which is nearly always the case with ATM field discs. Our procedure, which we find easy to implement, accomplishes both at the same time. We define the unequalized version as the primary preservation object and the equalized version as an optimized stand-in for the Preservation Master File.

To further our skills in playing field discs, and to develop a specific playback and transfer methodology, IU consulted with disc expert Eric Jacobs of The Audio Archive.⁴⁷ From Jacobs we learned procedures for using a microscope as a diagnostic tool, both for examining problems on discs and to aid in the selection of a stylus.⁴⁸ We also acquired procedures for turntable setup. In addition, Jacobs provided us with techniques for cleaning lacquer discs, including a cleaning solution that he developed based on researching existing mixtures used in sound archives.

Here are the basic procedures used to transfer discs at Indiana:

A. Assessment

1. Identify disc coating and base materials.
2. Evaluate the condition of the disc. Cotton gloves are worn when handling lacquer discs in particular as acidic fingerprints can cause imprints in the nitrocellulose, leading to increased noise and local chemical reactions.
3. Decide whether cleaning is appropriate to the condition of the disc and the materials it is made of.
4. Decide whether to transfer first, then clean, or vice-versa. If the disc is showing

⁴⁷ The Audio Archive. <http://www.theaudioarchive.com/>.

⁴⁸ The microscope is a used, rebuilt Nikon SMZ-2B stereo microscope with WF 15x eyepieces focus block, a boom stand, 10mm/100 div scale reticle, new 2X SMZ Aux lens, new 21v 150w fiber optic illuminator, new annular ring light guide, calibration certification for scope, metric stage micrometer 1mm/100Div for checking reticle. It was purchased from Bi Optic. www.biopticinc.com.

signs of deterioration it will be transferred first, then cleaned (if possible) and re-transferred.

B. Cleaning

1. Discs are cleaned and rinsed by hand using brushes obtained from the Disc Doctor, then rinsed and vacuumed dry using a Keith Monks record cleaning machine. Discs may be rinsed twice on the record cleaning machine, especially lacquers with significant palmitic acid on the disc surface, to eliminate any remaining contaminants as well as cleaning solution residue.
2. Each disc size and surface material has its own set of brushes for cleaning.
3. Lacquer (nitrocellulose laminate) discs are cleaned using a solution that is 0.25 parts Tergitol 15-S-3, 0.25 parts Tergitol 15-S-9, 98.5 parts deionized water, and 1 part clear ammonia. The ammonia is an optional palmitic acid solvent used on lacquers only. The advantage of the ammonia is that it minimizes the amount of mechanical scrubbing required to remove palmitic acid as well as minimizes the amount of exposure of the laminate to water, as water can cause the laminate to swell and delaminate. Discs with a compromised lacquer layer (i.e. cracks or signs of delamination) should not be cleaned with an aqueous solution as this will accelerate delamination.
4. Lacquer discs receive a final rinse with a solution that is 99.75 parts deionized water and 0.25 parts Disc Doctor solution. The Disc Doctor solution is used to lower the surface tension, allowing water to push down into the grooves for more thorough rinsing.
5. Aluminum discs are sometimes cleaned with Isopropyl alcohol applied with lint-free Kimwipes. This is not very effective in our experience, but we have not yet found anything more effective.

C. Turntable Setup

The following alignments are done for each disc or for each collection of discs depending on the variations within a collection.

1. Height adjustment or vertical tracking angle (VTA): The tonearm assembly (tonearm, cartridge, and stylus) is placed near the center of the disc in playback position. The height of the tonearm at the pivot is then adjusted so that the tonearm is parallel to the disc surface from the cartridge to the outside edge of the disc. This adjustment results in the stylus being perpendicular to the disc surface for optimal frequency response and minimal distortion.
2. Longitudinal balance and vertical tracking force (VTF): It is important to begin with a neutral longitudinal balance before setting tracking force. This is achieved by moving the rider weight to the null position and adjusting the main balance weight along the tone-arm extension until the arm floats just above bare contact with the disc. Once balance is achieved we set the tracking force to around 2.5 to 3.5 grams. This is just a starting position and may change once we listen to the playback. We have found that too little tracking force can affect the sound quality and too much tracking force can cause damage to softer surface materials. Also, too much VTF can compress the dynamics of the audio signal.

3. Anti-skate: The anti-skate is set equal to the amount of vertical tracking force used. 3.5 grams of tracking force will result in a setting of 3.5 for anti-skate. If there are skips or other tracking problems with the disc the anti-skate may be set lower (to help move the stylus forward) or higher (to help hold the stylus back) than the tracking force. The angle of the anti-skate thread is set to a 90° angle at the outer groove of the disc on the Technics SP-15 turntable. Note that the skating force is greater at the outer grooves than the inner groove due to the fact that the groove velocity is higher at the outer grooves, so there is more friction force (which generates the later anti-skating force). Setting the anti-skate thread at a 90° angle at the outer groove maximizes the anti-skate force where it is most needed.
4. Horizontal alignment (Zenith): The next step is measuring the horizontal tracking angle (HTA) and selecting an alignment curve that minimizes distortion. The specific horizontal tracking angle controls the distortion profile created from tracking angle errors that result from the various alignment curves that place the null points—or points of no distortion—in various places on the disc. We use the Baerwald alignment curve because it distributes the tracking distortion equally over the entire disc (whereas the Lofgren alignment minimizes tracking distortion in the middle groove region at the expense of higher distortion at the inner and outer grooves). Horizontal alignment is calculated by measuring the distance from the spindle to the innermost groove and the spindle and the outermost groove. A tool developed by Eric Jacobs enables us to determine from these measurements the proper distance from the spindle to the tonearm pivot for the best Baerwald Alignment, yielding the least amount of total distortion.⁴⁹
5. Cartridge azimuth: We adjust the cartridge azimuth by placing a small mirror on the turntable and resting the stylus on it. When viewed in this way any departure from vertical is accented and easily visible. If any discrepancy is perceived we adjust until proper azimuth is achieved.

D. Preamp Setup

1. All discs are recorded stereo into Preservation Master Files, making use of a stereo cartridge to capture the signal from both groove walls. For mono discs, the signal is converted to mono in a Production Master File, providing some intrinsic noise reduction as the recorded signal is doubled while noise that exists in only one channel is not.
2. For noise reduction to be maximized when converting to mono, both groove walls must be recorded at exactly the same level. Levels are matched by placing the KAB preamp in mono, engaging the vertical switch, and adjusting the balance for the least amount of audible program material. This technique makes use of the fact that in vertical mode the left and right channels are placed 180° out of phase. This channel balancing is done for each disc, after which the preamp is switched back to stereo and lateral for transfer.

⁴⁹ For further information see embedded PDF file from: Michael Fremer, *21st Century Vinyl: Michael Fremer's Practical Guide to Turntable Set-Up* (DVD. Musicangle.com, 2006). PDF file is also available online at: <http://www.musicdirect.com/products/featured/img/FREMER/VFREMERSETUPDVD/21stCenturyVinyl.pdf>.

E. Groove Examination

1. Disc grooves are inspected using a microscope with a built-in reticle to help determine proper stylus size and to identify potential tracking problems such as collapsed groove walls and cross-cut grooves. In addition, the use of the microscope can identify sources of groove noise due to groove wear as well as sources of playback distortion due to a poor pressing or worn stamper, thus setting expectations for the quality of the source material and ultimately the transfer.
2. The distance between the top of the groove walls and the distance between the bottom of the groove walls is measured in micrometers (also known as microns or 1/1000 mm). The resulting numbers are entered into a stylus calculator designed by Eric Jacobs, which provides a stylus size that will fit perfectly into the grooves. Although we always listen to different styli, we find that this calculation quickly places us close to or at the best stylus size, minimizing the number of times we must play sections of the disc. It is, however, important to note that an over- or undersized stylus sometimes yields better results for discs that have suffered groove wall damage.

F. Tracking Problems

Engineers who transfer discs all have their own bag of tricks for dealing with skips or other tracking problems. A few that we have found useful include

- adding more, or less, anti-skate to counter-balance the direction that the stylus jumps from a skip;
- using more, or less, tracking force to keep the stylus in the grooves. One convenient way to do this that we learned from Jacobs is to temporarily place small M5 washers that weigh about 0.25 gram each on the headshell. This enables the engineer to hear the effect of extra force quickly and to keep working rather than stopping and adjusting the VTF using the counterweight at the rear of the arm. Once the best tracking force is determined, the washers are counted and removed, and the appropriate adjustment is made to the VTF counterweight. The washers are not used to actually apply the extra force if it appears they might resonate or rattle during the transfer;
- guiding the head shell with a hand or fingers may help to keep the stylus in the grooves;
- using a horsehair brush (a soft two-inch paint brush) to apply gentle pressure to the headshell. Sometimes the tonearm can be better controlled by applying pressure to the rear of the tonearm (at the counterweight) rather than at the headshell. Frequently, this method is the only way to get through a heavily damaged disc;
- employing a viscous damping trough (if the tonearm has this option). The trough can limit dynamics subtly on the one hand, but can also make some transfers of damaged discs more consistent and easier.

The microscope allows us to see problems on the disc close-up and, with experience, make an educated guess as to what technique(s) is likely to work best.

2.2.3.3 Playback Problems

The only difficult, persistent preservation problem encountered so far at Indiana in the playback of open reel tapes is severe squealing on Scotch 175-brand tapes. Degradation of magnetic tape is complex and not yet well understood, sometimes requiring multiple procedures to

render a tape playable. This squealing was *not* accompanied by significant shedding and did not appear to be classic Sticky Shed Syndrome for which there is a temporary remedy—baking—available. For many years, preservation engineers believed that squealing without shedding was basically caused by a loss of lubricant from the tape. However, recent work by Richard Hess in consultation with a group of scientists, audio engineers, and tape specialists, has demonstrated that what has been termed loss of lubricant in open reel tapes is likely caused by a number of factors not yet completely understood. His work also suggests that the mechanism by which baking (also called incubation) renders a Sticky Shed Syndrome tape playable has also been misunderstood. Hess has proposed a new category using the term Soft Binder Syndrome (SBS) for all polyester-backed tapes that exhibit sticking, squealing, and abnormal shedding.⁵⁰ Hess further theorizes that the degradation process results in the lowering of the tape's glass transition temperature to below room temperature, and that playback in cold conditions (below this reduced temperature) might stop squealing. We tested Hess' theory by acclimating one Scotch 175 tape from the Henry Glassie collection along with a playback machine to the 50° F, 30% RH climate at the IU Auxiliary Library Facility (ALF). First attempts to transfer this tape failed. We then used a freezer at ALF, and acclimated the tape to 38° F, 24% RH. This was ultimately successful, but only after one failed attempt and an additional 24 hours of storage in these cold conditions.

Indiana has encountered a number of cassettes that would not play and, in all but two cases, re-housing the tape into a new outer shell solved the problem. One tape for which re-housing did not work was twisted and reversed in a number of places so that the back of the tape mistakenly made contact with the playback head. In addition, the tape pack had loosened so much that it could no longer wind to the end because it would not fit. We were able to work through this tape slowly, opening the shell and straightening it out as necessary, transferring all but the last five minutes of the second side, which were unplayable. This process took a full day. Fortunately, the last five minutes were available on a backup open reel tape copy that was recorded before the cassette original had deteriorated to this extent. The second cassette, from the same collection, was suffering from a similar problem although somewhat worse. We could have invested an estimated two days in capturing as much content from it as possible. Instead, the ATM Director and Archivist, after examining the nature of the content on the tape, recommended that we transfer the backup open reel in its place.

At Harvard, we encountered many playback problems during the process of digitizing a collection of R-DAT field recordings dating from 1992 to 2004. The most challenging problem was misalignment of the field recorder. To remedy this we selected the worst examples of mis-tracking tapes, and sent the field recorder with them to be "de-aligned" to match those tapes in hopes of a successful transfer. This resulted in a slight increase of successful transfers. However, some tracking errors could not be eliminated. In addition, we discovered mis-tracking due to speed changes during start and stop of the field recorder at the time of recording. We were unable to compensate for these tracking errors. Further tracking errors occurred during playback due to poor head-to-tape contact—perhaps from curling, damage or poor storage. We found that acclimating the tape in the R-DAT player for at least twenty to thirty minutes, increased our success rate. This led us to believe that R-DAT head-to-tape contact is improved by the higher temperatures of the interior of the playback machine. Other errors encountered were due to tape damage and loss of oxide. We found no correlation between tape errors and age or brand of tape.

50 Richard L. Hess, "Tape Degradation Factors and Predicting Tape Life" (paper presented at the 121st AES Convention, San Francisco, Calif., October 5-8, 2006).

2.2.4 Conversion

2.2.4.1 Best Practices

Because analog-to-digital conversion is covered in the works cited above, this topic is another in which we agreed to report on our choices and practices, rather than research best practices. For specific recommendations, see IASA-TC 04 and the article by Ken Pohlmann, published by CLIR.

2.2.4.2 Analog-to-Digital and Digital-to-Analog Conversion at Harvard and Indiana

At Harvard, the choice of a Prism AD-2 analog-to-digital converter and a DA-2 digital-to-analog converter was based upon the accuracy of the conversion and the usefulness of the feature set. The converter pair's apparent accuracy when comparing the pre-conversion and post-conversion signal was a significant factor in our decision, but chief among useful features that influenced the purchase were the multiple digital inputs and outputs and, in the case of the AD-2, the ability to adjust input sensitivity from -28 dBFS to 0 dBFS. Multiple digital outputs provide a simultaneous source for recording and real time signal analysis. An adjustable input sensitivity can compensate for low-level recordings without adding undue amounts of noise that might occur if a previous gain stage were pushed significantly past its optimal range. Additionally, the AD-2's input balance can be adjusted in steps of one one-hundredth dB. The DA-2 is switchable among its multiple digital inputs and allows monitoring of multiple signals without the engineer having to change connections. The Prism AD-2 and DA-2 reduce setup time and make our workflow more efficient than would a more limited design converter pair.

Indiana currently uses two-channel Benchmark ADC1 and DAC1 converters for preservation transfer work and is very satisfied with their functionality and performance. We began the project with converters from another company that did not consistently function properly as described in the section on quality control in Chapter 7, but switched to the Benchmarks. The ADC1 features two stages of gain, the first stage with three selectable levels that are optimized for signal to noise performance, and the second stage that uses either a 41-detent potentiometer or a continuously variable, 10-turn gain range calibration trimmer for finer input gain adjustments. It also features multiple digital outputs and, although we don't currently use it, the ability to simultaneously output multiple sample rates and bit depths.

Engineers at both institutions have found that our converters typically do not seem to perform best near 0 dBFS but have a sweet spot that is significantly lower where they simply sound better. With 24 bit recording there is plenty of dynamic range available so that levels can be set lower without penalty and there is no need to drive the converter at high levels. Therefore we do not try to set levels so that peaks are just below 0dBFS.

Sample rate and word length is another area in which the *Sound Directions* project did not propose to recommend best practices but to report on our choices. Both the ATM and HCL-APS now digitize analog sources as PCM audio at 96 kHz sample frequency and 24 bit word length. This seems the best compromise among coding format, file size and audio fidelity. We say *compromise* because we realize that choosing a coding, a sample frequency and a word length is a limiting choice in an ever-changing landscape of audio formats and tools. At this time, the wide support for PCM audio at this resolution should ensure that our digital objects have a fairly long life before migration to the next form becomes necessary.

For digital-to-digital transfer and digital file conversion, Harvard neither re-samples nor up-samples tape-based and file-based digital audio objects to a higher resolution for preservation. We retain the original audio object's resolution. We transfer tape-based digital audio objects such as DAT, by routing the DAT's AES/EBU digital audio signal through our Prism AD-2 in Digital-to-Digital mode using the incoming digital audio as the synchronization master for the AD-2, and then to our Pyramix DAW that gets its synchronization from the incoming signal. The Pyramix project's sample rate and word depth match those of the incoming audio. During the transfer we watch for error indications on the AD-2 as well as monitoring the SpectraFoo analysis. We import file-based digital audio objects directly into Pyramix in their original form if possible. If the original file is not supported in Pyramix, we convert the file to Broadcast Wave at the original sample rate and word length, and import that BWF into Pyramix.

3 Digital Files

This chapter explores critical issues related to the creation of preservation digital files, the recommended target file format for preservation, the selection of local filenames, and the verification of file data integrity.

3.1 Preservation Overview

Digital File Preservation Surrogates

All analog and physical digital audio recording formats deteriorate over time, degrading much more rapidly than paper-based archival documents. Audio recordings rely upon reproduction technology that adds wear, fosters deterioration, and eventually becomes obsolete.⁵¹ For these reasons, audio preservation has always required the creation of duplicate copies, often called surrogates, which act as stand-ins for the original recording.⁵²

It is now widely accepted that, for both technical and economic reasons, the preservation of audio must rely upon transfer to, and storage in, the digital domain.⁵³ As stated in IASA-TC 04, “the integration of audio into data systems, the development of appropriate standards, and the wide acceptance of digital audio delivery mechanisms have replaced all other media to such an extent that there is little choice for sound preservation except digital storage approaches.”⁵⁴ There is general agreement among sound archivists that data file formats are preferable to physical carriers containing digital audio streams, such as DAT and CD, for ensuring data security, monitoring data integrity, and managing preservation assets.⁵⁵ Our target preservation format is now the digital file, which must be managed over the long term in such a way as to ensure data integrity and trigger action when its specific format becomes obsolete. These files may reside on a number of different types of carriers—data tape, hard drives, optical discs—each with their own strengths and weaknesses. Audio preservation, then, requires a long-term responsibility to the digital file.⁵⁶ If digital files are properly created and well documented, they not only represent the best chance for preservation of the target content, but carry a number of technical advantages over any analog surrogates that might be produced, including the possibility of using automated processes in the creation of copies that are of the same fidelity as the original files.

51 This reproduction technology includes not only playback machines, but such things as tape heads, disc cartridges and styli, tools used to adjust the machines, service manuals used in repairing the machines, and format-related supplies such as empty reels and splicing tape. It also includes the expertise necessary to optimally use, align, and repair playback devices. All of these will become obsolete in time. For some formats, obsolescence is already a serious issue.

52 See IASA, Technical Committee, *IASA-TC 04*, 4. The term “surrogate” is particularly used in the image preservation world.

53 These reasons include the near obsolescence and increasing cost of ¼” analog tape as well as past preservation problems with this format, the consequent unavailability of tape machines, the difficulty in maintaining the highest possible quality when migrating analog formats, and the flexibility and consequent expanded access options that digital technologies afford, among others.

54 IASA, Technical Committee, *IASA-TC 04*, 2.

55 See IASA, Technical Committee, *IASA-TC 04*, 48 and 65; IASA, Technical Committee, *IASA-TC 03*, 8. See also Kevin Bradley, “Memory of the World Programme: Risks Associated with the Use of Recordable CDs and DVDs as Reliable Storage Media in Archival Collections - Strategies and Alternatives,” CI/INF/2006/1 REV (Paris: UNESCO, October 2006), <http://unesdoc.unesco.org/images/0014/001477/147782E.pdf>.

56 This particular way of stating this idea comes from: Humanities Advanced Technology and Information Institute, University of Glasgow and National Initiative for a Networked Cultural Heritage, “Preservation,” Chapter XIV in *The NINCH Guide to Good Practice in the Digital Representation and Management of Cultural Heritage Materials*, ver. 1.1 (NINCH, 2003), 198-213. Also available online: <http://www.nyu.edu/its/humanities/ninchguide/>.

The digital file surrogate will gradually become the highest quality record of the original as the original becomes more difficult or impossible to play due to media degradation and/or format obsolescence. In time, it will become the only surviving record, bearing the original content for use by future generations. Even if a recording has not yet deteriorated greatly, digitization requires significant resources and it is unlikely that the means to repeat this work will be available. These certainties carry many significant implications for audio preservation. Therefore, utmost care must be taken in the creation of surrogates, and digital file roles within the institution must be clearly defined and understood to ensure appropriate creation and handling of different types of files.

Target File Format

There is general agreement in the audio preservation field that the Wave file format (.wav) or, as more specifically recommended by IASA, AES, and the National Academy of Recording Arts and Sciences, the Broadcast Wave Format (abbreviated BWF or BWAV), is the best target preservation format.⁵⁷ Many audio file formats have been developed and used over the years, some of which are now obsolete. The wider the use of a format within a professional environment, the greater the chance that it will be broadly accepted and supported, including the development of tools for migration to future file formats. Broadcast Wave is widely accepted in the archival and professional audio engineering communities with literally hundreds of thousands of hours preserved in BWF files. The Broadcast Wave Format is non-proprietary, restricted in definition, contains assigned locations for specific metadata, and has a sample-accurate time stamp. This is ideal for archival preservation.

The European Broadcasting Union (EBU) introduced the Broadcast Wave specification⁵⁸ in 1996 to facilitate the exchange of audio files between the increasing number of digital audio workstations on different computer platforms in radio and television production. The format is not new and is based on the widely adopted Microsoft Wave file format. BWF files are essentially .wav files but more narrowly defined with additional restrictions on the types of audio that they may contain. A BWF file also contains several extra data chunks⁵⁹—the most commonly used of which is called the <bext> or Broadcast Audio Extension chunk—into which basic metadata may be entered. The audio contained in a BWF file can be reproduced by any software that can read a .wav file, although software applications that do not support the format will not be able to access the metadata in the <bext> chunk. A Broadcast Wave file also carries an embedded sample-accurate time stamp that references the source timeline, fixing its position in time. The time stamp facilitates the sequencing of related files on any computer workstation that supports the format.

The Broadcast Wave Format was developed by the EBU as an “open” format—not tied to any specific manufacturer’s hardware or software—allowing software developers and standards organizations to support the format as an industry standard without the need to address proprietary issues. BWF is neither application nor platform specific. The underlying file format—.wav—is proprietary, owned by Microsoft. A BWF file maintains the usual .wav extension, although some manufacturers and many users mistakenly believe that the extension should be .bwf. To counter this misunderstanding, the term BWAV is increasingly used to refer to the format even though BWF is the original abbreviation.

57 See IASA, Technical Committee, *IASA-TC 04*, 7. See also The Recording Academy, Producers & Engineers Wing and Audio Engineering Society, “Recommendation for Delivery of Recorded Music Projects.” Note that later in this section *Sound Directions* specifically recommends BWF as the target format for preservation.

58 The BWF specification is found in: EBU, “BWF – A Format for Audio Data Files in Broadcasting.”

59 A chunk—a basic building block of a digital file—is a self contained collection of data that contains a header followed by the data itself. A file will contain a number of chunks.

The Broadcast Wave Format allows for the entry of metadata into the header of the .wav file. Because of the relatively small amount of information that may be included in the header and the difficulty involved with editing each file when metadata is updated, the *Sound Directions* Advisory Board suggested that, for archival institutions, a file header is not an appropriate location for a complete, authoritative version of metadata relating to a file. Nor is it a substitute for a full, external metadata creation and storage system. From within our context—research archives with large audio holdings—we view the BWF <bext> chunk as an appropriate location for “catastrophic metadata,”⁶⁰ that is, the information necessary to interpret the contents of the file in the absence of the metadata system. In effect this is a middle ground, where a limited amount of metadata that is unlikely to change and is easily obtained is included in the file header.

While <bext> chunk metadata is insufficient it is important, and should be supplied for eventualities such as these:

- Files are exchanged with another institution and accompanying metadata is either lost or not available
- Files are disseminated to researchers who then, without permission, further distribute files without accompanying metadata. In this case, header metadata would at least suggest ownership and point to the origin of the files⁶¹
- Files are disseminated to researchers who lose the accompanying metadata
- The metadata system either is temporarily unavailable or fails, or is not working properly during a planned dissemination that includes a database lookup on the fly for the most recent version of metadata
- Files are made available to an access system that is owned by an outside organization and archive-supplied metadata is either lost or not used
- A file is corrupt, the filename is not obtainable, and the file must be examined using a hex editor to gain clues as to its contents
- The organization that produced the files does not yet have a sustainable metadata system but is relying on maintaining accompanying metadata in an unsustainable format such as a document produced by proprietary text-editing software
- The name of the file is changed unintentionally

In addition, embedding metadata in the file itself might obviate the need for spoken announcements or slates that identify content.

In a future that includes federated digital libraries holding content from a number of different organizations as well as widespread privately-owned access systems for audio content of all types, these scenarios do not seem implausible to us. There is ongoing research in the digital library world into methods for machine-to-machine transfer and understanding of complex digital objects, including both multiple versions of primary objects and metadata, that aims to reduce the risk of content becoming divorced from its appropriate context.⁶²

Characteristics of Preservation Files

General guidance on the characteristics of preservation files in the BWF format and the

⁶⁰ This term was coined by John Spencer of BMS/Chace.

⁶¹ Of course, files disseminated to researchers may not be .wav files, in which case the point is moot. We are also not considering here intellectual property issues and contracts with researchers that govern when, or if, files may be available for download to researchers.

⁶² See, for example: Open Archives Initiative, Object Reuse and Exchange.
<http://www.openarchives.org/ore/>.

procedures used to create them can be found in both of the IASA best practices documents as well as in the ARSC/AAA report that guided audio preservation in the 1980's and 90's when analog tape was the target preservation format. This document cleverly points out that the philosophical purpose of re-recording for the sound archivist is "saving history, and not rewriting it." Historical accuracy is the goal and, therefore, a "1-to-1" preservation transfer of the original was advocated.⁶³ Unmodified transfer to the new format without subjective alteration—forgoing the temptations of denoising or editing—remains a widely shared principle of audio preservation processes in the digital world.

In addition to a "Preservation Master," the digital preservation workflow generates numerous types of files that serve different functions and have different characteristics, including files for online streaming for example or files from which physical products such as CDs can be made. These are described and explained in detail in the Recommended Technical Practices section below. It is necessary to define the characteristics of each type of file produced in the preservation workflow to enable future understanding of the role that a file plays, its value to the institution, and its relationship to other files. Metadata that identifies a file's function or role must be preserved along with the file's content.

The simple act of copying digital files does not incur the loss of fidelity inherent in analog copying. If done within closed systems using error correction and including verification using a checksum,⁶⁴ one digital copy should be identical to the next, with no loss of fidelity from one "generation" to the next as would have been the case copying recordings from one magnetic tape to another. (The integrity of data in digital files must be routinely checked, however. Data integrity is the subject of section 3.2.4.)

Filenames

Creating structured, consistent, and well-formed local filenames supports local interoperability, parse-ability, and efficient use for the preservation workflow only as long as the files are controlled locally, that is, within the audio studio and/or curatorial unit. Files produced in the preservation workflow require logically and consistently created unique identifiers that match the content of the files with the metadata belonging to it. These unique identifiers may be as simple as a number string that identifies a single file. For a digital repository with a completely automated preservation workflow and robust external systems for handling metadata, a simple number string may be all that is ever needed.

On the other hand, filenames can reflect catalog and preservation workflow information in a human-readable form in order to facilitate a workflow that is either entirely or partly manual. A meaningful local filename may provide a significant convenience, supporting a particular workflow by quickly providing basic identification of files without the need to refer to a metadata system. At this point in the development of audio preservation systems, many institutions still rely on local filenames to carry content and to function as primary identifiers.

What is critical to understand, though, is that filenames are not a reliable means of storing information. They are for identification purposes only, and although they may contain catalog and workflow information, they should not be relied upon as a database for cataloging or other metadata. (An external metadata system should fulfill this function.)

⁶³ Association for Recorded Sound Collections, Associated Audio Archives Committee, *Audio Preservation: A Planning Study* (Silver Spring, MD: Association for Recorded Sound Collections, 1988), 21 and 107.

⁶⁴ Computer-based storage media are formatted so that they are essentially error free. A system using such media will warn the user if giving errors. This is quite different from digital audio formats such as DAT or CD that require many levels of error protection and will, if necessary, interpolate data to correct errors.

Filenames and file paths are especially significant to Audio Decision Lists (ADLs). The ADL uses the file path and filename to locate related media. The workflow must account for the filename's role in this function. Typically, this means that, if upon deposit the repository replaces a filename with its own unique identifier, then upon extraction from the repository the original filenames must be re-instated. (Alternatively, upon ingest, the ADL could be rewritten to reflect the unique identifier so that the filenames in the deposited ADL will be correct upon extraction.)

File Data Integrity

Once a preservation file is created, assurance that it has not been modified accidentally or purposefully during transmission from one location to another or during long-term storage is necessary. Without this assurance it is not possible to claim that the preservation surrogate is an authentic representation of the original. For preservation of the content information, the OAIS reference model requires fixity information that documents the data integrity checks which ensure that content has not been altered in an undocumented manner.⁶⁵ Preserved digital content must be checked at regular intervals for data integrity.⁶⁶

Verification of data integrity is typically done using a form of redundancy check applied to a file or group of files called a hash algorithm such as Message-Digest algorithm 5 (MD5) that substitutes or transposes the data into a relatively small number called a digest. This number serves as a digital fingerprint or signature of the target data. An algorithm such as MD5 is substantially more accurate than error detection algorithms used in storage devices and network protocols. This is a one-way process: it is impossible to reconstruct the target data from the digest and it is statistically highly unlikely that two sets of data can have the same digest. The term 'checksum' is widely used generically for this process even though, strictly speaking, it refers to just one simple form of redundancy check. In the library and archive community, MD5 is widely used and there are many software tools available for its implementation. Another algorithm used in this community is SHA-1.⁶⁷

After a file is created, a Message Digest (often referred to as a checksum) is generated for the file using software that produces the checksum value based upon an algorithm applied to the data in the file. The integrity of the file may then be verified at any point by comparing the originally-generated checksum value with a regeneration of the checksum from the file in question. If the two values match, it is highly likely that the data in the file remains unaltered. The algorithm is not an audio comparison utility such as may be found in some audio software suites. Nothing is inserted into the file when a checksum is generated—the value is stored in a separate checksum file by the generating software. This value may also be stored with the technical metadata related to the target file in an institution's database or other metadata system.

In general practice an MD5 is calculated for a file using the entire contents of the file. In the case of a BWF file this includes both the metadata in the <bext> chunk as well as the audio content. In a preservation workflow it is often desirable to edit only the metadata while maintaining the ability to separately verify the audio content. This is possible by using a "chunk-specific" checksum that is applied to the audio content only, as described in section 3.2.4.4 below.

⁶⁵ CCSDS, *OAIS*, 19 and 65.

⁶⁶ See IASA, Technical Committee, *IASA-TC 03*, 9.

⁶⁷ MD5 has been cracked, and it is now easy to produce "collisions" where two different sets of data have the same message digest. This means that while MD5 is suitable for protecting against accidental file changes such as corruption during storage or errors during transmission, it may not be effective within a security context that is concerned with malicious behavior. Similar weaknesses have recently been demonstrated with another widely used hash function, SHA-1.

3.2 Recommended Technical Practices

3.2.1 Target Preservation Format

3.2.1.1 Best Practices

Best Practice 6: Use the Broadcast Wave Format for the preservation of audio.

Best Practice 7: Use the <bext> chunk of the Broadcast Wave Format for metadata that is needed to interpret the contents of a file in the absence of accompanying metadata. Do not use it as the authoritative source or for metadata that may change over time.

Best Practice 8: Include the local name for the file in the Description field.

Best Practice 9: Use the Time Reference field to provide a time stamp showing the location of the file on a reference or destination timeline.

3.2.1.2 Rationale

As discussed above, the Broadcast Wave Format is ideal for archival preservation because it is non-proprietary, restricted in definition, and has assigned locations for specific metadata and a sample-accurate time stamp. BWF is a restrictive file format that can contain either MPEG (Layer I or II), or PCM audio only. This is a distinct advantage over the general WAVE file format that could contain any of the following codecs: PCM, ADPCM, μ -law, GSM, CELP, SBC, TrueSpeech and MPEG-Layer III. This restriction will be valuable in identifying the audio coding of objects. A further advantage of the broadcast wave file is the Broadcast Audio Extension chunk, also known as the <bext> chunk. The <bext> chunk can contain essential metadata. There is now strong support in the audio industry for BWF, with many professionals using it because the time stamp can be transferred between digital audio workstations. Applications that require audio files to be placed in either a specific order or at a specific location can read the time stamp without the need to refer to a separate Edit Decision List. This is particularly useful within an archival setting for preservation transfers that must be stopped and restarted due to speed changes or preservation problems. This results in multiple files that must be sequenced in the proper order for presentation to an end user. In the past, workstations typically used a proprietary method for storing the time stamp in a sound file, and the time stamp would rarely be useable with other workstation software. The format is supported by industry leaders including Avid (maker of Digidesign's Pro Tools), Merging Technologies (maker of Pyramix), Yamaha (maker of Steinberg's WaveLab) and MAGIX (maker of Samplitude).

Non-changing metadata is entered into the BWF <bext> chunk to identify the contents of a file in the event that it is detached from its metadata, as discussed above. Including the local name for the file ensures that it is available within the file itself if the name is unintentionally modified or if it is intentionally removed during normal ingestion into a preservation repository.

3.2.1.3 Background

All of the metadata entered into the BWF <bext> chunk, plus much more, is also captured

in external technical metadata systems at both Indiana University and Harvard University. As discussed above, we consider the BWF header as an appropriate location for “catastrophic” metadata only at this time.

Note that the time stamp by itself is not enough to present seamless content when overlaps in multiple preservation files from the recording process are edited out. In this case, an AES31-3 ADL is also required.

Also note that some organizations populate the BWF header from an in-house database, requiring little effort by the audio engineer. Others have database applications that make use of this metadata to find specific files. Because we have robust external search and discovery utilities this latter use is of less value to us.

3.2.1.4 Use of the BWF <bext> Chunk at Indiana

In WaveLab, and other software, it is possible to establish defaults so that the metadata in the BWF <bext> chunk copies from file to file, making its collection easy and efficient. At the ATM, nearly all of the BWF metadata remains the same throughout a collection; typically, we must update only the shelf number (incrementing by one digit), the original filename (by cut and paste), and the origination date (at the start of each day) when moving from one recording to the next within a collection.

3.2.1.4.1 Description Field

This field is designed to contain a free description of the sound sequence and may include up to 256 ASCII characters. To aid software applications that only display a short sequence, the EBU recommends that the essence of the description be contained within the first 64 characters. If the string contains fewer than 256 characters, the last one is followed by a null character.

Here are the data elements we use:

[Collection ID] [Source Object ID] [File Use] [IUCAT Title Control Number] [Original Filename]

Where:

- Collection ID = ATM collection accession number
- Source Object ID = the unique number that identifies the source of the file’s content—usually the ATM shelf number
- File Use = the role that the recording plays within the unit (ATM)
- IUCAT Title Control Number = unique identifier for the collection within the Indiana University Integrated Library System
- Original Filename = the filename assigned to the file

A minimum of explanatory text introduces some of the elements. Here is an example:

File content: Collection 75-025-F number EC 5730. File use: Preservation Master. IUCAT Title Control Number: BAA0584BT. Original filename: atm_75025_ec5730_010101_pres_20070130

Explanation:

- The first sentence contains perhaps the two most important elements of this string—the

collection accession number and individual recording shelf number—from which all documentation available at the ATM may be located. This sentence contains fewer than 60 characters including spaces

- The IUCAT Title Control Number is the unique number generated by the IU Integrated Library System for the bibliographic record representing an ATM collection
- The name of the file is included as the last element as a safeguard against the accidental changing of the name, which would de-link it from its metadata. It is also possible that a filename might intentionally change within a digital library setting, with the appropriate reconnection of metadata to the new name and the tracking of the old name. In this case, we think it might be useful to either the digital library or the organization that generated the file to have a redundant copy of the original filename residing within the file itself. Note that on Windows and most modern file systems, filenames are stored separate from the files themselves. Adding the original filename into the <bext> chunk ensures that it will reside with the actual file

3.2.1.4.2 Originator field

This field was created to contain the name of the originator/producer of the audio file and may contain up to 32 characters. Although some DAWs will legitimately enter a reference to themselves in this field, the ATM chooses to use this area to indicate the organization or institution that created the file. The ATM uses this statement: *IU Archives of Traditional Music*, which is, fortunately, exactly 32 characters.

3.2.1.4.3 Originator Reference field

The purpose of this field is to carry a unique identifier. The EBU has designed a format for this field, which is one of many types of unique or persistent identifiers that might be used. Because unique identifiers will be associated with ATM content by the IU Digital Library Program, the ATM does not use this field.

3.2.1.4.4 Origination Date field

This field contains the date that the file was created using the standard ISO 8601 form of yyyy-mm-dd (year-month-day).

3.2.1.4.5 Origination Time field

This field documents the time that the file was created. We do not enter any data into this field, reasoning that within our setting documenting the exact time that a file was created has little value compared to the time it takes to add this information.

3.2.1.4.6 Time Reference

This field contains the time stamp, documenting the start position on the reference or destination timeline that this file should occupy. For the first Face (or side of a recording) the value in this field is 0 as automatically generated by WaveLab. Additional Faces for a recording will have a time stamp value greater than 0, representing their in-order location on the timeline. In situations where multiple files were created for one Face, the time stamp for each subsequent file would also be greater than 0, reflecting their appropriate positions on the timeline in reference to the first file.

3.2.1.4.7 Coding History

The Coding History field is designed to hold data on the digitizing process including signal chain specifics, sample rate and bit depth, and other elements. It is defined as a collection of strings, each presented on a separate line, containing a history of the coding processes applied to the file. Each variable within a string is separated by a comma. A new line is added when the coding history related to the file is changed, and each line should end with a carriage return and line feed which are automatically added by WaveLab. According to the EBU, each line should contain these elements, as appropriate to the coding history being described:⁶⁸

- Coding algorithm. String begins with “A=” For example: *A=ANALOG, PCM, MPEG1L3*, and others
- Sampling frequency. String begins with “F=”
- Bit-rate, for MPEG coding only. String begins with “B=”
- Word length. String begins with “W=”
- Mode—this corresponds to sound field, such as mono, stereo, or dual-mono. String begins with “M=”
- Text, free string—a free ASCII-text string for in-house use. The EBU suggests documenting devices in the signal chain and analog source recording formats in this field. String begins with “T=”

At Indiana, we include three lines of coding history in our BWF files for the digitization of analog recordings. The first documents the analog source recording, the second contains data on digitization chain, while the third records information on the storage of the file.

For example:

```
A=ANALOG,M=mono,T=Studer A810; SN3690; 15 ips; open reel tape,
A=PCM,F=96000,W=24,M=mono,T=Benchmark; ADC1; SN00252; A/D,
A=PCM,F=96000,W=24,M=mono,T=Lynx; AES16; DIO,
```

Line 1 reads: an analog open reel tape with a mono sound field was played back on a Studer A810 tape machine with serial number 3690. Tape speed was 15 ips.

While the EBU document suggests including the tape brand and product number as the last element, we prefer a general designation of the format for several reasons: it is more useful to know the format than the specific brand and it avoids the need to interpret the brand information and playback machine data to identify the format. When a range of formats— analog cassettes, discs, DATs and others—are routinely digitized this interpreting might become unnecessarily difficult. In addition, the format remains constant through an entire collection (the brand and product number may or may not), providing one less element that requires data entry for each source recording.

Line 2 reads: the tape was digitized in mono mode using a Benchmark ADC1 A/D converter with serial number 00252 at 96 kHz sample rate with a bit depth of 24 bits.

⁶⁸ European Broadcasting Union, Production Technology Management Committee, “Format for the <CodingHistory> Field in Broadcast Wave Format files, BWF,” EBU Technical Recommendation R98-1999 (Geneva, Switzerland: European Broadcasting Union, 1999), http://www.ebu.ch/CMSimages/en/tec_text_r98-1999_tcm6-4709.pdf.

Line 3 reads: the tape was stored as a 96/24 mono file using a Lynx AES16 digital input/output interface.

If we apply additional coding processes to produce a derivative file we add a fourth line in the header of the derivative file. For example:

```
A=PCM,F=44,100,W=16,M=mono,T=Steinberg; WaveLab 6; Resampler, Waves L2; Dither; DAW,
```

This line reads: A 16 bit, 44.1 kHz file was created using the WaveLab 6 Resampler and Waves L2 Dither in the Digital Audio Workstation.

3.2.1.5 Use of the BWF <bext> Chunk at Harvard

In the <bext> chunk we populate the Description, Originator, OriginatorReference, OriginationDateTime, TimeReference, and the Version fields. Entered in the Description field is the owner supplied name of the audio file such as AWM_RL_0001_AM_01_01, which means Archive of World Music Reel number 0001, Archival Master, side 1, file 1. The Originator field is the name of the Pyramix DAW that created the file, either PyramixOne or PyramixTwo. The OriginatorReference field contains the USID, or Unique Source Identifier. It is a string that can be broken down into the Country Code (CH for Switzerland), Organization Code (MTI for Merging Technologies, Inc.), Serial Number (PYRAMIX16934), Origination Time (153213), and a Random Number (745780579). At this time our tools support the use of the UMID (Unique Material Identifier) and the Coding History (description of coding processes applied to the audio data), but we are not using either of them in our current process.

<bext> chunk fields		
Field	Use	Example Data
Description	owner supplied name (filename from originating unit)	AWM_RL_0001_AM_01_01
Originator	originating DAW	PyramixOne
OriginatorReference	USID	CHMTIPYRAMIX16934153213745780579
OriginationDate	file creation date (yyyy-mm-dd)	2007-01-10 (Although OriginationDate and OriginationTime are stored separately in the <bext> chunk, our tools display date & time as one field labeled OriginationDateTime.)
OriginationTime	file creation time (hh-mm-ss)	15-32-13 (Although OriginationDate and OriginationTime are stored separately in the <bext> chunk, our tools display date & time as one field labeled OriginationDateTime.)
TimeReference	1st sample count since midnight	2374
Version	version of BWF	Our DAW produces version 0 Our tools produce version 1 (current version) The version difference is in the SMPTE UMID that we do not use, and for which we write all zeros
UMID	Unique Material Identifier	Supported by our tools but not currently used
CodingHistory	ASCII strings describing coding process applied to the audio data	Supported by our tools but not currently used

Table 5: Harvard <bext> chunk fields

3.2.1.6 Institutional Comparison of <bext> Chunk Field Use

contrasting uses of <bext> chunk fields		
Field	HCL-APS Use	IU ATM Use
Description	owner supplied name	Collection ID, Source Object ID, File Use, IUCAT Title Control Number, Original Filename
Originator	originating DAW	originating institution
OriginatorReference	USID	not used
OriginationDate	file creation date (Although OriginationDate and OriginationTime are stored separately in the <bext> chunk, our tools display date & time as one field labeled OriginationDateTime.)	file creation date
OriginationTime	file creation time (Although OriginationDate and OriginationTime are stored separately in the <bext> chunk, our tools display date & time as one field labeled OriginationDateTime.)	Not used
TimeReference	1st sample count since midnight	1st sample count since midnight
Version	version of BWF	version of BWF
UMID	supported in tools but not currently used	not used
CodingHistory	supported in tools but not currently used	used to describe the digitization of analog recordings

Table 6: Harvard and Indiana <bext> chunk fields

3.2.2 Digital File Types and Uses for Preservation

3.2.2.1 Best Practices

Best Practice 10: *Clearly define the purpose(s) of every digital file surrogate created for preservation or access. This metadata must be preserved so that files are managed in a way that is appropriate to their defined role and value in the archive.*

Best Practice 11: *Develop specific, written guidelines on the characteristics of primary preservation surrogates, and the procedures used to produce them, within the context of the institution's workflow.*

Best Practice 12: *Develop written guidelines on handling both technical and content edits.*

3.2.2.2 Rationale

Preservation (or Archival) Master Files contain the most authentic and highest-quality representation of the original or source recording to be carried into the future. For this reason, their production and use must be specifically defined and tightly controlled. Likewise, the creation of other types of preservation or access files must be defined and controlled to assure that the target content is accurately presented to end users.

3.2.2.3 File Types and Uses at Indiana

3.2.2.3.1 Background

Our aim in this area was to add specificity and detail to the international standards discussed in the preservation overview above by defining such notions as “unmodified,” “without subjective alteration,” and “historical accuracy” within the context of daily preservation transfer work, and all of the anomalies and problems that it presents. Towards this goal, Indiana University developed the definition of Preservation Master Files presented below.

3.2.2.3.2 Preservation Master Files

The ATM calls its first, and primary, digital file produced from the transfer process a Preservation Master File. We define this file type as containing complete, unaltered data from the source audio object exactly as reproduced by the playback machine. It functions as a carrier of the raw material from the transfer. This file (or its ADL) establishes the reference timeline if there is no Preservation Master–Intermediate file (see below). A Preservation Master File is always produced during preservation transfer projects.

A Preservation Master File

- contains a complete, unaltered stream from the playback machine. This stream begins when the field recording machine is placed in record and ends when it is taken out of record;
- is produced without data reduction (either lossless or lossy compression);
- is produced without any equalization other than that inherent in the playback machine (NAB or IEC1 equalization on an open reel tape machine, for example. Disc equalization is a special case considered below);

- is not signal processed, contains no gain or level changes (including normalizing) or dithering in the digital domain;
- contains no added spoken announcements or slates. This type of information is handled by metadata in the BWF <bext> chunk and in our metadata system;
- is recorded at a high resolution and high sampling rate—24 bit/96 kHz;
- is a Broadcast Wave file with metadata contained in the BWF <bext> chunk;
- is created by an audio preservation engineer in a studio designed for preservation transfer work.

Here is how we handle some specific issues and problems that emerge during transfer:

- If there is unrecorded silence in the middle of the source recording, with content beginning again afterwards, this is retained in the Preservation Master File to accurately represent the source recording
- Unrecorded tape after the end of content, when the field recording machine has clearly been taken out of record, is not retained. Unrecorded tape at the beginning of content, before the field recording machine was placed in record, is not retained
- Signal level from the playback machine and into the computer is never adjusted *during* the transfer in order to preserve the source recording's dynamic range in the digital file. The exception is when discrete, unrelated performances are recorded onto the source, separated by the turning on and then off of the recording machine. In this case, recording levels may be adjusted to maximize the signal to noise ratio of each discrete performance
- Discs requiring a playback equalization curve are transferred both with and without the curve at the same time in one pass, and both files are preserved to maintain maximum flexibility into the future. The file produced from the unequalized transfer becomes the Preservation Master, while the equalized version is defined as a Preservation Master–Intermediate (see below)

The Preservation Master File is defined as the primary object produced by the preservation transfer project.

3.2.2.3.3 Preservation Master–Intermediate Files

The ATM at IU defines a Preservation Master–Intermediate File as a faithful representation of the source object, optimized with a type of post-processing that does not lead to the substantive loss of any content. Although it is not the raw material from the transfer, it is a valid stand-in for the Preservation Master. It represents some stringently-defined decisions and provides a file type that may be used in ways that the Preservation Master cannot because of the strict way in which it is defined. This file type is created only when necessary and represents one important step towards a more user-friendly presentation of the content.

A Preservation Master–Intermediate File

- may carry “technical” edits that solve technical problems from the transfer. For example, skips and other tracking problems on a disc may be edited out only if it does not lead to loss of content or loss of ability to interpret content;
- does not contain any edits other than technical edits described above;
- may include the application of a disc playback equalization curve;
- is never signal processed (denoised, for example) other than the disc eq curve;
- contains the content from one Preservation Master File, that is, a Preservation Master–Intermediate is created for each individual Preservation Master File;

- is the same bit depth and sample rate as the Preservation Master File;
- always establishes the reference timeline to which further derivatives, including Production Masters, are created. This file type is often used in situations where the timeline from the Preservation Master cannot easily be carried forward as described below;
- is a Broadcast Wave Format file with metadata contained in the <bext> chunk.

For example, a Preservation Master–Intermediate File is created when a disc has a locked groove in the middle that prevents the stylus from moving forward. The listener hears the same word(s) repeated any number of times until the transfer engineer is able to move the stylus forward by hand. Similarly, a number of stylus drops may be necessary at the beginning of a disc to get it started. These technical problems are addressed by editing in the Preservation Master–Intermediate File. This alters the timeline from the Preservation Master; therefore, the reference timeline is established by the Intermediate. Similarly, re-pitching part of a file to make content understandable will result in an altered timeline and is accomplished using a Preservation Master–Intermediate File.

Edits are not made if content will be lost, or if the edit changes the aural context in such a way as to make the content unclear. Sometimes a skip alerts the listener to a problem on the disc and makes it clear that the loss of a word was due to disc problems, not engineer error. For example, a skip forward resulting in a loss of content may be impossible to fix due to groove problems. Removing such a skip results in a jarring jump from one part of the content to another with nothing to indicate why it happened. In cases like this, the skip is *not* edited out.

Bands on the disc or the entire disc side may be topped and tailed as necessary to edit out repeated needle drops at the start of the disc or repeated content at the end due to a locked groove. Some disc noise is left in the file during the topping and tailing process for use by signal processing algorithms. This noise is left at the tail as a first choice, if possible, but may also be left at the head.

Authenticity of the final product depends somewhat on the skill and judgment of the audio engineer, although the “raw” Preservation Master File is always kept in addition to the Preservation Master–Intermediate.

Preservation Master–Intermediate Files are also created when a disc requires a playback equalization curve. In this case, a single transfer produces both an unequalized Preservation Master and an equalized Preservation Master–Intermediate.

3.2.2.3.3.1 Rationale for the Preservation Master–Intermediate File Type

The word intermediate can be defined as situated between two points, stages, or things. In our case, the Preservation Master–Intermediate is a proxy for the Preservation Master File, representing a step towards the creation of a Production Master File (described below) that is then used to produce all further derivatives, especially deliverables.

We have chosen *not* to perform these technical edits in either Preservation Master or Production Master Files for a number of reasons:

- Preservation Masters are strictly defined as the unaltered material from the transfer and we do not wish to loosen or expand this definition
- Production Masters may be signal processed to provide a further optimized version

for end users. Signal processing tools grow in their capabilities each year, and we fully expect to re-create some Production Masters for this and other reasons over time. If these types of technical edits were made in Production Masters they would be lost and need to be re-done

- It is also possible to carry technical edits in an ADL so that the Preservation Master is left intact, but this requires software that can efficiently separate the file around each edit point, and then render in place multiple files while adjusting the time stamp. WaveLab does not do this easily or efficiently. Additional software with this capability is expensive; hence our use of an alternative that affords some flexibility in this area.⁶⁹ Note that in response to this problem, Harvard developed a tool called “adlconsolidate” that will be available as part of the *Sound Directions* Toolkit.

A workflow using a Preservation Master–Intermediate File type still adheres to the key principle of providing a common reference (or destination) timeline for all manifestations of the recorded work. Timelines are discussed in Chapter 4, section 4.2.3.4.1.2. The Preservation Master–Intermediate is a valid stand-in for the Preservation Master, and the reference timeline extends from it forward through all downstream derivatives. We do not anticipate reasons to return to the Preservation Master itself except in the rarest of cases, but it is retained as verification of an accurate preservation transfer process.

3.2.2.3.4 Production Master Files

Production Master is a term historically used in the audio, video, and film industries to refer to an object that provides a source for the production of further copies. At Indiana University, a Production Master File is used to generate all further derivatives, particularly deliverables that will be used by researchers. A Production Master is a representation of the source audio object that is optimized as determined by the ATM Archivist and the ATM Associate Director for Recording Services. This optimization may include (rarely) editing of the content and/or signal processing. When an optimized representation is not necessary—when the Preservation Master Files are considered sufficient for the creation of deliverables—a Production Master that is a clone of the Preservation Master is produced.

A Production Master File

- contains the content of one “Face” (for example, one side of a disc or one track on a tape) of a source recording. If multiple Preservation Master Files were created during transfer of one Face of an analog recording, these files will be edited together into one Production Master;
- may have either the same or different bit depth, sample rate, and number of channels⁷⁰ as the Preservation Master File depending on end use and production workflow;
- may be edited as directed by the ATM Archivist and/or ATM Director;
- may be signal processed using procedures established by the Associate Director for Recording Services;
- is a BWF file with metadata in the <bext> chunk;
- may exist in different versions. For example, a 24/96 and a 16/44.1 Production Master may be produced at different times to satisfy production and/or workflow demands.

If transfer of a source audio object results in multiple Preservation Master Files for one Face, the Production Master File will carry the complete, combined content of this Face for

⁶⁹ Steinberg’s Nuendo will do all of this more efficiently. As of this writing, the price to academic institutions is \$1,000.

⁷⁰ Mono field discs, for example, are often transferred stereo for technical reasons but presented to end users in mono.

seamless presentation to the end user. The edits necessary to combine the files are carried by an AES31-3 ADL, allowing the Preservation Master Files to remain intact.

Edits in a Production Master File are based on content issues with decisions made at the curatorial level. Content deemed unquestionably irrelevant for research use or designated as restricted by the collector are examples of what may be removed from a Production Master. Underlying Preservation Master Files that contain complete, unaltered content are always retained.⁷¹

Production Masters may be signal processed for increased clarity and understandability, or for easier listening, following guidelines that outline a signal processing aesthetic for research use. Underlying Preservation Master Files that contain unaltered sound are always retained.

The ATM generates Production Masters for *all* preserved content for the following reasons:

1. To limit handling of the Preservation Master Files to rare cases where a new Production Master must be created. At Indiana University, deliverables are produced by the Digital Library Program, not the ATM. This means that someone outside of the content-generating unit works with the files. While we fully trust the DLP it makes sense structurally, over time, to have this safeguard in place for both ATM and DLP staff

2. So that no analysis is needed when accessing files to produce deliverables—the procedure is simply to find the Production Master. If we chose to create this file type for only some purposes—to retain the work of signal processing, for example—it would be necessary to first search for this file then, if not found, look for a Preservation Master-Intermediate. If this file was not found then the search would continue for a Preservation Master for use in producing deliverables

3. To provide flexibility in accessing content. Production Masters can be stored somewhere other than deep preservation storage in a location that provides quicker and easier access. This is important to the ATM, which receives a steady stream of orders that are currently filled by burning CDs.

All Production Master Files are currently created at high resolutions and sample rates—24/96—to facilitate delivery of high resolution files to end users, which we think is likely in the future.

The main disadvantage of generating Production Masters for all content is an automatic doubling of storage space. This is not an issue at Indiana University, at least at this time. With

⁷¹ Note that content editing may be handled successfully in several ways and at several levels. One strategy is to keep restricted content in a Production Master and let a downstream access system exclude the material using metadata. For example, a SMIL file could be used with the Production Master (or a deliverable file) to point to only the content that may be played. With this approach, restricted content does not need to be edited out of a source file since SMIL will simply not point to the restricted areas. Another approach is to replace restricted material in a Production Master with silence. This has the advantage of physically removing the material while also signaling its removal, all the while maintaining the reference timeline so that the Production Master is still in-sync with the Preservation Master. The need for editing based on content restrictions was not encountered at IU during the project, although we are sure that it will be necessary at some point given the nature of our collections. Therefore, we have not actually researched either of these procedures. We do have a number of issues in this area that we will consider, including: the general principle that restricted content should stay as close to ATM control as possible, the possibility that someone could hack the SMIL format and gain access to restricted material, the feasibility of supporting the SMIL format at IU into the future, and the possibility that access for certain content may be possible only for the community that created it. This is a decision that will ultimately be made by ATM and DLP staff together, and it is possible that we will implement multiple procedures.

storage costs steadily decreasing, and audio files relatively small compared to other audio-visual formats such as video or data from scientific experiments, we don't think that this will be an issue for many institutions in the foreseeable future.

3.2.2.3.5 Deliverables

Online access to ATM content will be provided by the IU Digital Library Program through an access system developed or modified to handle field collections. This system does not yet exist, although preliminary work to define its functionality is underway as part of an internally-funded grant project. In addition, the focus of Phase 1 of *Sound Directions* at Indiana University is preservation. For these reasons we are not currently generating derivatives for use in either a campus- or web-based access system. We anticipate, however, the automated production of deliverables from Production Masters at the appropriate time.

3.2.2.4 File Types and Uses at Harvard

When we speak of file types, we are referring to the audio file in either BWF or RealAudio, keeping in mind that these files are merely the raw audio assets (source files) for their respective interpreting documents such as an AES31-3 ADL in the case of BWF or a SMIL file in the case of a RealAudio asset. We use five preservation file types at the Archive of World Music: pre-archival, Archival Master, pre-production, Production Master, and Delivery Master.

3.2.2.4.1 pre-archival

The pre-archival file is the result of AES31-3 export from the DAW digitization project of the original audio object, in which the files from the digitization are created and their time stamp is adjusted in the appropriate relation to midnight,⁷² or the zero point or zero hour, in the project timeline, by editing and rendering in place. The pre-archival file naming convention is identical to that of the Archival Master File—as the pre-archival is a temporary file used for preparing the Archival Master so that no further operations on the Archival are necessary once it is created. The pre-archival is not deposited in the Digital Repository.

3.2.2.4.2 Archival Master

The Archival Master is the BWF file or set of files that, together with the AES31-3 ADL, comprise the unadulterated digital surrogate of the source audio object for preservation. The Archival Master is the source-file part of the result of our software script “makearchival” operation on the pre-archival AES31 export. Once created, the Archival Master is untouched by manual operations. We use the abbreviation “AM” for Archival Master. For example: AWM_RL_0001_AM_01_01

3.2.2.4.3 pre-production

The pre-production is the first intermediary file in the creation of deliverables. It is the file whose purpose is de-noising and file consolidation prior to making the Production Master. The pre-production naming convention remains the same as the Archival Master again because it is temporary and not deposited.

⁷² The term “midnight” refers to the fact that many nonlinear editing systems have an unlimited timeline that could span more than one day including negative time before the zero point, or normal project start point—midnight being the zero point where positive time begins.

3.2.2.4.4 Production Master

The Production Master is the second and potentially permanent intermediary file for the purpose of creating a deliverable. In most cases the Production Master is deposited and kept indefinitely, although as a Production Master it can be remade to suit different delivery purposes. Since in our current workflow we need only make a single delivery file per Face, and that delivery file can be repurposed by creating additional SMIL navigation documents, the Production Master can remain in the repository until such time that it becomes obsolete. The Production Master is the stage at which sample rate conversion happens. It is the result of our software script “makeproduction” operation on the pre-production project. Again, this file is not altered by any manual operations once created. We use the abbreviation “PM” for Production Master. For example: AWM_RL_0001_PM_01_01

3.2.2.4.5 Delivery Master

The Delivery Master is the source file we use for patron access to the audio content. It is the result of encoding the sample rate converted Production Master into RealAudio through the action of the “makedeliverable” script. Its interpreting and navigational document is the SMIL file accessed via an assigned URN in the online catalog. We use the abbreviation “DM” for Delivery Master. For example: AWM_RL_0001_DM_01_01

3.2.3 Local Filenames

3.2.3.1 Best Practices

These best practices refer to locally-generated filenames that are human-readable and carry meaning:

Best Practice 13: Use ASCII letters (a-z), ASCII digits (0-9), underscores and hyphens, and be aware of the implications of using any other characters in filenames.

Best Practice 14: Do not use diacritic marks or any non-printing characters.

Best Practice 15: Reserve the period (full stop) for the file extension at the end of the filename.

Best Practice 16: Do not use values in file elements that might change over time.

Best Practice 17: The first element should identify the unit that created the file.

Best Practice 18: Make filename elements more detailed and/or specific as they are read from left to right.

Best Practice 19: Identifiers used in the filename should correspond to those used with physical objects and in existing catalogs.

Best Practice 20: Include a sequence indicator in the filename.

Best Practice 21: For derivative files, use the same name as the master file with the addition of an element that indicates the derivative's type.

3.2.3.2 Rationale

The details of local filenames are necessarily institution-specific, and while we believe that the models presented below may be helpful to many organizations, they will not be applicable to everyone. For this reason we have issued a list of general practices that are recommended for maximum interoperability and ease of use for filenames that carry meaning. This list incorporates both best practices reported by other organizations and those from our own experience. Both *Sound Directions* institutions use file-naming conventions that clearly identify preservation files, their relation to the catalog, and their respective roles in the workflow, but we differ in the extent of detail that we include in filenames. These conventions are offered in this document only as examples of what we have found useful in our workflows, and not as prescriptions for successful preservation. Note that both institutions have external metadata systems that store content information included in filenames.

3.2.3.3 Filenames at the IU Archives of Traditional Music

3.2.3.3.1 Background

The Archives of Traditional Music generates filenames that carry content and are human-readable. While metadata is available for all files, we find that the convenience of meaningful filenames makes it simpler and quicker to manage folders and directories of files as the content and use of any given file can be determined at a glance. This is especially true when filling orders for ATM collections when only particular types of files—signal processed, for example—are needed. Indiana University does not yet have a fully functioning preservation repository and we do not know how ingestion into our future repository will affect our use of filenames.

The ATM worked with the IU Digital Library Program to update and refine its naming scheme while the DLP was also creating preservation repository requirements for filenames. This process yielded a number of recommendations.⁷³ Here are basic guidelines that we used that encourage maximum interoperability and efficient use over time and that appear to have general agreement in the field:⁷⁴

- Keep filenames as short as possible
- Include only alphanumeric characters plus the special characters underscore, hyphen, and period (before the file extension only)
- Use lowercase letters
- Do not use spaces
- Do not use any values that might change over time
- Do not duplicate other names used in the same organization
- If carrying meaning in names, and if there is an existing catalog, use numbers/letters that match what is already cataloged

We also made use of these more specific recommendations:

- The first character of the filename should be an ASCII letter as many programming and metadata languages place this restriction on their identifiers. Filenames should be usable as identifiers in these languages (e.g., section ID's in a METS document)
- The use of camelCase (first word starts with lower case, second word begins with upper case) for adjacent words to increase readability is acceptable, if necessary
- While periods are permissible in filenames, it is *highly recommended* that they be avoided in the base of the name. A period is used to separate the base name from the extension that specifies the type of file. Some programs assume that there is only a single period in a filename, and will behave strangely if multiple periods are present
- Filename elements should indicate more specific detail as they are read from left to right as alphabetical listings of files in a directory or folder are more easily understood with this organization
- Elements of the filename should be separated by either hyphens or underscores for

⁷³ See Indiana University, Digital Library Program, "Filename Requirements for Digital Objects, Indiana University Digital Library Program," Indiana University (2 November 2006), <http://wiki.dlib.indiana.edu/confluence/display/INF/Filename+Requirements+for+Digital+Objects> for many of these.

⁷⁴ See, for example: Online Computer Library Center, Research Libraries Group, "Recommendations for Digitizing for RLG Cultural Materials" (2006), http://www.rlg.org/en/page.php?Page_ID=220; UCSD Libraries, Digital Library Program, "A Naming Protocol for Digital Content Files" (10 February 2006), <http://tpot.ucsd.edu/Cataloging/MASU/naming%20protocol.pdf>; and Indiana University, DLP, "Filename Requirements."

readability and automated processing. It may be easier for later parsing to avoid mixing the two

- The first element in the filename should identify the unit that produced the file to easily identify it in a digital library setting that includes files from many organizations
- When possible, the digital object's "primary" identifier (the identifier appearing in the filenames) should correspond to an identifier in use for the original (physical) object. This makes it easy to match files with physical objects without the need to refer to metadata
- If the transfer of a physical object resulted in multiple files, the filename must include a sequence number
- Dates used in filenames should follow the ISO 8601 standard (yyyy-mm-dd). Among other benefits, this allows file directory listings to be sorted into correct chronological order
- A derivative file must have the same name as the master file, except the "base" filename should have an indication of the derivative's type appended. This will make it easy to identify master-derivative relationships. Derivative files may or may not be a different file type, with a different extension, than the master file

3.2.3.3.2 General ATM Filename Form

The general form for Indiana University ATM filenames is:

unit_collectionID_sourceObjectID_physicalSequenceIndicator_fileUse_signalProcessingFlag_date.fileExtension

Where:

- *unit* = atm
- *collectionID* = shortened ATM accession number
- *sourceObjectID* = ATM shelf number
- *physicalSequenceIndicator* = identification of what part of the source recording is represented by the file
- *fileUse* = the role that the file plays within the unit.
- *signalProcessingFlag* = identification that the file was signal processed
- *date* = year, month, day using the ISO 8601 standard
- *fileExtension* = extension that identifies the type of file, automatically placed at the end by software

3.2.3.3.3 ATM Filename Elements

Unit: *atm_*

All Archives of Traditional Music filenames begin with "atm" followed by an underscore. This identifies the derivation of the file within a larger digital library setting and increases the chances of a globally unique name.

collectionID: *atm_86507_*

Collections are identified at the ATM through accession numbers that serve as their primary identifier. Filenames contain a shortened version of the accession number, dropping the hyphens and last letter. For example, accession number 86-507-F (the 507th collection accessioned in 1986—the "F" designates a field collection) becomes 86507 in a filename.

sourceObjectID: *atm_86507_ot4214_*

Each physical recording receives a unique identifier called a shelf number by which it is stored on a shelf in the ATM vault. Filenames contain a version of the shelf number with a lower case prefix and no spaces. Some of our shelf numbers contain the symbol for inches (") which is replaced with a lowercase letter n in the filename.

physicalSequenceIndicator: *atm_86507_ot4214_020101_*

The digitization of one analog recording often results in multiple digital files. The `physicalSequenceIndicator` documents the relationship of the digital file to the analog source recording, notating what portion of the source recording is included in the digital file. This enables multiple files for one source to be sequenced, based on their relationships to the physical, analog recording.

For Preservation Master and Preservation Master–Intermediate files, the `physicalSequenceIndicator` *always* consists of a series of three, two-digit numbers with a leading zero (if necessary). The first number indicates which Face of the source analog recording is represented in the file. A Face is a group of one or more streams (audio channels or tracks) that are meant to be played synchronously, such as one direction of a tape or one side of a disc.

The second number in the `physicalSequenceIndicator` documents which Region of the source analog recording is represented in the file. Faces may be divided into Regions if necessary, each characterized by a change in a basic characteristic of the format within the Face. If a tape is recorded at 15 ips but switches to 7.5 ips for its remainder, then the Face would contain two Regions, one for each tape speed. A Face must have at least one Region, by definition.

Faces may also be divided into Parts, if necessary, and the third number indicates which Part of the Face or Region is represented in the file. A Part is created when a transfer must be stopped for reasons other than a change in format, resuming in a second digital file that is labeled Part 2. For example, a physical problem with a tape may necessitate stopping the transfer to fix the problem, resuming again in a second digital file.

Note that Regions are created (conceptually) on analog sources while Parts refer to digital files. Parts may be created at either the Face or Region level as needed.

The `physicalSequenceIndicator` always consists of just one number for Production Master Files, which contain the content from one complete Face, by definition. Therefore, designations for Region and Part are omitted from these filenames.

Examples:

- *010101* = contains content from Face 1, Region 1, Part 1 of the source recording. There may or may not be a Face 2, Region 2, or Part 2
- *020101* = Face 2, Region 1, Part 1. There must also be a file containing Face 1
- *010201* = Face 1, Region 2, Part 1. There must also be a file containing Face 1, Region 1
- *02* = Face 2. By definition, this means that all of the content from Face 2 is included, regardless of the number of Regions and Parts

fileUse: *atm_86507_ot4214_020101_pres_*

This element identifies the assigned use or role of the file within the ATM.

Examples:

- *pres*: This string is used for a Preservation Master File
- *presInt*: This string is used for a Preservation Master–Intermediate File
- *prod*: This string is used for a Production Master File. A Production Master File always includes two numbers within this element that indicate bit depth and sample rate. For example, *prod2496* = a 24 bit, 96K Production Master

signalProcessingFlag: *atm_86507_ot4214_02_prod2496_sproc*

Although our metadata system will supply this information, it is very useful to see at a glance which files have been signal processed as we often choose these to fill orders. The word *sproc* is added between the *fileUse* and *date* elements if a file is signal processed (in a Production Master), however, this element is absent if the file is flat or unaltered.⁷⁵

date: *atm_86507_ot4214_020101_pres_20070925*

The date that the file was created is included in the *date* element using the ISO 8601 standard without hyphens to avoid mixing hyphens with underscores in our filenames.

fileExtension: *atm_86507_ot4214_020101_pres_20070925.wav*

The file extension is the (usually) three letter code automatically placed at the end of a filename by software. This extension is NOT added into the filename by ATM workers. All of our preservation files are currently in the Broadcast Wave Format, which uses the *.wav* extension.

3.2.3.3.4 ATM Filename Examples

Preservation Master Files

Audio file: *atm_67934_ot290_010101_pres_20060915.wav*

This is a Preservation Master File of source OT 290 from collection 67-934-F. This file contains the content from Face 1 (direction or track 1 of the tape), Region 1, Part 1. The file was created Sept 15, 2006.

Related ADL file: *atm_67934_ot290_010101_pres_20060915.adl*

Preservation Master–Intermediate Files

atm_54323_10n348_020101_presInt_20061121.wav

This is a Preservation Master–Intermediate File for 10" disc number 348 from collection 54-323-F. This file contains the content from Face 2 (side B of the disc), Region 1, Part 1. The file

⁷⁵ Note that we do not consider dithering as signal processing within this context as it is a basic engineering practice—always done when appropriate—and does not need to be indicated in a filename. Note also that a playback equalization curve applied to a file is also not considered signal processing in this sense. Both of these examples, which technically qualify as signal processing, are tracked through metadata.

was created November 21, 2006.

Production Master Files

atm_85553_cass458_02_prod2496_20060423.wav

This is a 24 bit, 96 kHz Production Master File from cassette 458 from collection 85-553-F, containing all of the content for Face 2 (side B of cassette). The file was created April 23, 2006.

Signal processed files

atm_54673_10n3467_02_prod2496_sproc_20070224.wav

3.2.3.4 Filenames at Harvard

We view filenames as temporary conveniences that should be used for their benefit as human readable, curatorially-derived identifiers. It is helpful, and we feel it is essential in the workflow, to have a standard naming convention, and to stick to that convention in order to avoid confusion.

Our naming convention for preservation objects is derived from the catalog or shelf number and denotes the following:

- creating unit name abbreviation such as AWM for Archive of World Music
- original object type such as RL for reel
- shelf number such as 0001
- volume number such as V n when the shelf number pertains to more than one item, where V=volume, and n =the item number
- additional specifier such as S n when the volume contains more than one item, where S=the media type such as T for Tape or D for Disc, etc., and n =the item number
- preservation file type or role such as
 - AM for Archival Master,
 - PM for Production Master,
 - DM for Delivery Master
- incrementing Face number such as 01 for side 1
- incrementing file number such as 01 for the first file in the timeline

In addition, the local filename may include a pseudo-random unique identifier created by the originating DAW.

Example: *AWM_DAT_139_AM_01_01_{BB4BB2F2-A64C-4AAA-8B2C-D34506FDF7E7}*

Note: The above example is from a single item R-DAT, and so has neither volume nor additional specifier in its filename.

As an example of the fleeting nature of preservation filenames, the above-mentioned file's assigned Object ID in the Digital Repository Service (DRS) is 6459098, and if one were to download the file from the repository, it would be labeled 6459098.wav. Since we store the original name in the <bext> chunk description field, examination of that field would reveal the name as *AWM_DAT_139_AM_01_01_{BB4BB2F2-A64C-4AAA-8B2C-D34506FDF7E7}*. In addition, examination of the metadata for the file in the DRS would reveal the same filename (called "owner supplied name" by the DRS). The original filename and file path are

stored as separate metadata in both the DRS and the METS Archival Information Package. We could have chosen any name for the file. The preservation of that file does not depend upon the form of the filename. Only the uniqueness of the object ID is significant where there are robust external systems for handling metadata.

3.2.4 File Data Integrity

3.2.4.1 Best Practices

Best Practice 22: *Generate a checksum for every file that has enduring value. This includes both audio and non-audio files created during the preservation process.*

Best Practice 23: *Generate a checksum as soon as possible after a file is created—usually after basic work with the file is completed.*

Best Practice 24: *Consider the checksum value critical technical metadata. Store the value in the system used for other technical metadata, with backup copies kept in multiple physical locations.*

Best Practice 25: *Verify the checksum before trusting any file that has been moved, copied, or had its header edited.*

3.2.4.2 Rationale

As discussed in the overview, the integrity of every file created for preservation must be verified over time. Generating the checksum soon after a file is created provides a baseline in case there are problems during the preservation workflow, or during storage or transmission. In order to ensure that checksum values remain available in the face of system failures or other disasters, we should avoid susceptibility to a single point of failure. For example, the repository might be structured to store checksums within the deposit package as well as on a separate file system for metadata storage, with backup copies kept in multiple physical locations. The integrity of files in long-term preservation storage must be verified regularly.

3.2.4.3 Background

Both *Sound Directions* institutions experienced problems related to either checksum generation or verification that proved instructive.

The ATM experienced one case of failed checksum verification in an audio file that we believe was caused by correcting BWF metadata and forgetting to generate a new checksum. To confirm that the audio was not compromised we compared the file with a version on another hard drive using the audio file compare process in WaveLab. We also reversed the phase in one file and compared it to the other to check for differences. The process of verifying this audio data would have been far simpler had a chunk-specific checksum (as produced by a Harvard tool described below) been available for the audio data only.

In the initial implementation of Harvard's SAN at the Archive of World Music, the SAN software used a file system translation utility to allow Windows and Macintosh to read from and write to a single file system. During the production of RealAudio deliverables, the translation utility caused file corruption to the SAN volumes that manifested as spurious

audio appearing in the deliverable files when tested. This discovery was made before a checksum had been generated for the files. Had we not tested the files, and then generated a checksum, the resulting corrupted files would have been validated—even though they were corrupt. It is important to generate a checksum as soon as possible after the basic work is completed. However, it is also important to verify the quality of the file that you wish to validate before you run a checksum. This experience confirmed our belief in the importance of quality control, and led us to adopt a split file system on our SAN—thereby eliminating the translation utility at the cost of the increased labor of a split file system.

3.2.4.4 Redundancy Checks at Harvard and Indiana

Both IU and Harvard use the MD5 hash algorithm (a type of “checksum”) for verifying data integrity of every file. This algorithm was chosen because it is widely used in the digital library community and there are many tools available that support it. Although we have some concern about malicious behavior, we do not feel that a completely secure encryption algorithm is as necessary in our environments as it might be in other fields. We also reason that malicious action would likely render the content of the files themselves unusable since they would be the primary target.

The technical metadata collection software used at each institution generates an MD5 digest that is stored with other technical metadata, which is placed in preservation packages and stored in our preservation repositories.

In general practice an MD5 is calculated for a file using the entire contents of the file. In the case of the Broadcast Wave Format, metadata in the <bext> chunk of the file may be edited later either by hand or using a script as a normal part of the preservation workflow. When this happens, the originally generated MD5 no longer matches the edited file—even if the audio data portion of the BWF file remains perfectly intact—and there is no way to verify that the audio data is unchanged. Therefore, using a simple MD5 as a means of verifying data integrity throughout a production process such as this is impossible. An alternative approach developed at Harvard is to generate an MD5 on only the sound data portion of the BWF, and use that message digest as a means to verify that the audio payload of the file is unchanged even after the metadata in the header has been edited. Our “bwavinfo” software tool’s “-md5” function will validate only the audio payload portion of BWF files whose metadata has been purposely edited during the preservation workflow. Using an audio chunk-specific checksum may also aid the verification of data integrity into the future, revealing problems created through programming mistakes or as files are handled within a preservation repository.

Indiana uses a Windows software program called FastSum to verify its MD5 values.⁷⁶ Our technical metadata collector will generate, but not yet verify, these values. FastSum is available as both shareware and freeware and supports several different user interfaces.

⁷⁶ FastSum. <http://www.fastsum.com/>.

4 Metadata

4.1 Preservation Overview

Metadata provides the framework for digital audio preservation and thus forms an essential component of the virtual object. More specifically, various kinds of metadata document the identity of a recording such as its title and call number, the performer's name or the occasion of the performance, the format of the original recording and how the original was played back. Other forms of metadata document the kinds of digital copies produced [i.e. Archival Master, de-noised Production Master, streaming Delivery Master], where the digital files are stored, the relationships between the object's virtual and original manifestations and, importantly, how the digital files were made. All of this information is critical to our ability to find what we have in the first place, and in the long run to automatically migrate it to the digital file formats of the future. The significant amount of time spent collecting accurate metadata is never wasted. Effectively, there is no preservation without metadata. Fortunately, some of the software developed by *Sound Directions* automates the collection of this data, thus saving time and limiting the opportunity for human error.

The categories of metadata discussed below include descriptive, structural and administrative—with administrative further divided into its subcategories: technical, rights management, and digital provenance. There are other names that are used in place of or in addition to these. For example, the term “preservation metadata” is used, notably by the PREMIS project, to refer to any metadata that “a repository uses to support the digital preservation process.” PREMIS' preservation metadata may include data elements from each of the categories listed above.⁷⁷

Curators will perhaps be most familiar with descriptive metadata which is, in effect, cataloging information: the content of the object, the names of performers and producers, the subjects of the recording, and the recording's key component parts. Access and property rights metadata enable us to govern who may listen to a file. For instance, may the general public listen, only those with a particular password, or only the staff of the institution? This is where the specifics of rights information for use of the recording may be spelled out. Technical metadata records characteristics of the original formats of the recordings such as playback speed, track format, or playback equalization. Structural metadata can define relationships among a group of digital audio files. If, for instance, a single performance occupies four reels of tape and both Archival Masters and deliverable audio files have been made of the entire performance, the structural metadata will link the files to each other and to descriptions of the physical items so that it is clear at a glance what one has and what belongs with what.

Digital provenance metadata, often called process history or abbreviated as digipro, describes in detail the entire preservation process from the analog or digital transfer and conversion, to the digital repository deposit. This information is essential if we are to manage files and migrate them automatically in years to come. Message digest information, sometimes referred to as a checksum, is also vital, and is used by programs to check for file errors that would indicate that refreshing or reversion is needed. Potentially, metadata can be used to identify and group like digital objects for the purpose of uniformly treating them in a future mass data

⁷⁷ Preservation Metadata: Implementation Strategies (PREMIS), *Data Dictionary for Preservation Metadata: Final Report of the PREMIS Working Group* (Dublin, OH: OCLC and RLG, 2005), ix. Also available online: <http://www.oclc.org/research/projects/pmwg/premis-final.pdf>.

migration. In order to capitalize on labor-saving processes of the future, information about what we have done to create these files should be recorded now.

A metadata creation workflow in an audio archive must match appropriately trained individuals with the task at hand. Curators and content specialists play critical roles in the construction of metadata. It is generally they who provide cataloging or descriptive metadata, of course, but they may also provide valuable information in determining where the performance boundaries reside within a recording. Similarly, it is the engineers and technologists who have the knowledge necessary to document the technical characteristics of the original recording, the resulting digital files, and the processes used to transform one to the other. Subject-matter experts and technologists must collaborate to ensure that structural metadata faithfully represents the original object. Essentially, the archival tasks of sorting and ordering physical objects move into a virtual arena and object-based workflows must be adjusted to accommodate the new considerations.

Further, the rubrics (titles, call numbers, filenames and so forth) used by engineers and technologists ideally should correspond to those that appear in print catalog records or finding aids. In a perfect world, automated software tools would harvest descriptive metadata from online cataloging or finding aids to insure that all information about a single recording “matches.” This work will always require close communication between curators and technologists to establish clear understandings of preservation values and expectations.

4.2 Recommended Technical Practices

4.2.1 Best Practices

Best Practice 26: *Validate all generated metadata against the schema of a published standard, or against a copy of a locally agreed upon schema.*

Best Practice 27: *Generate valid audio object (technical) metadata for all physical and digital audio objects in the preservation workflow.*

Best Practice 28: *Generate valid digital provenance (process history) metadata that describes each process event in the preservation workflow.*

Best Practice 29: *If a transfer must be stopped and restarted, resume in a new digital file that contains overlaps in content. Use an AES31-3 ADL to document the edits needed to present seamless audio as originally recorded on the source audio object.*

Best Practice 30: *Maintain a common timeline that references the Preservation (Archival) Master and all further file manifestations of the content.*

4.2.2 Rationale

Without metadata, digital audio preservation is not possible. We might have a collection of digital audio files as a result of an archival transfer, but unless those files possess just one of the metadata fields of a Broadcast Wave file, the time stamp, there is no way to accurately determine the files’ relationships in time. Without the structural metadata of an Audio Decision List (ADL), we cannot be certain of the transitions between the files. Without

technical metadata about the files we cannot easily migrate them. Without information about how they were created and by whom, there is no way to judge what we have. We can make the highest quality transfers, with the finest equipment available, but unless we record and maintain the requisite metadata, essentially all we have is a bunch of files with an uncertain past and an even less certain future.

4.2.3 Background

The table below shows the different categories of metadata, and gives examples of each. For the purposes of this document we will refer to each individual subcategory of administrative metadata separately—keeping in mind their greater role.

Descriptive Metadata	Cataloging data encoded in MARC or MODS
Administrative - Rights Management Metadata	Rights management metadata governs access to files. (The <i>Sound Directions</i> project did not address this topic.)
Administrative - Technical Metadata	Tape Speed, Oxide Coating, Groove Width, Sample Rate, Word Length, Coding, Noise Reduction, Condition Comments
Administrative - Digital Provenance Metadata	Process History (digipro) for: archival transfer, sample rate conversion, de-noising or any DSP event, deliverable creation
Structural Metadata	AES31-3 Archival ADL, Audio Object Face, Region & Stream, BWF Time Stamp, PQ Marks, SMIL document, METS document (in its role documenting relationships using its <structMap>)

Table 7: Categories of metadata

4.2.3.1 Descriptive Metadata

Descriptive metadata is dedicated to curatorial information, rather than technical. This data identifies the object and its performances in a collections and patron research-centered manner. While it is clear that descriptive metadata plays an important role in the preservation process, the development of best practices for descriptive metadata was considered outside the scope of this phase of the *Sound Directions* project. Both IU and Harvard have existing descriptive metadata creation workflows for field audio. The best practices developed during this project phase therefore focus on technical, structural, and digital provenance metadata.

4.2.3.2 Technical Metadata

Technical metadata describes the immediate technical attributes of a physical or file-based audio object including specifications that enable access to the content. Both Harvard and Indiana implemented the technical metadata standard emerging from the Audio Engineering Society's AES SC-03-06 Working Group, labeled AES-X098-B in its draft form. This as-yet-unreleased standard provides the vocabulary for describing analog audio formats, physical digital audio formats and file-based digital audio formats. This vocabulary takes the form of an XML schema. It also provides for the collection of a specific type of structural metadata as described below along with minimal descriptive metadata. This schema provides for the collection of technical metadata in a number of broad categories, including, but not limited to

- physical properties such as base material, oxide material, groove orientation, and track configuration;
- object dimensions such as height, width, depth, unwound length and shape;
- signal characteristics such as playback speed, sound field, and noise reduction;
- digital file characteristics such as audio data encoding, sample rate, bit depth, and byte order;
- condition—comments on preservation problems with the object.

Note that only categories that pertain to the specific format of the source object are used.

4.2.3.3 Digital Provenance Metadata (digiprov)

Digital provenance metadata, or the process history, documents the “who, what, where, and how” of all preservation activity from transfer through the creation of deliverables. It records the various components of each process such that any part can be evaluated and/or readily reproduced. Each instance of processing of an audio object that has an input and an output is documented as an event. The *digiprov* document is both a means to record preservation processes for the distant future and a vital tool in the more immediate preservation workflow—as it facilitates teamwork and multiple project management by enabling recall of processes and settings at critical points in the workflow.

Both Harvard and Indiana implemented the metadata standard emerging from the Audio Engineering Society’s AES SC-03-06 Working Group, labeled AES-X098-C in its draft form. Like AES-X098-B, this standard takes the form of an XML schema and is not yet publicly available. The *digiprov* metadata collected using this standard is capable of describing processing events in minute detail. These events include such processes as the archival transfer and AES31-3 ADL export, any digital signal processing for production, the sample rate and word length conversion during the creation of derivatives, including deliverables, just to name a few. Each event is listed with the devices used, all their inputs, internal modules, parameters and settings, and outputs, along with the input media and the output media of the event. In addition, auditing information attributing the process to a specific operator, location, and time is recorded for each event.

4.2.3.4 Structural Metadata

Structural metadata “ties the components of a complex or compound resource together and makes the whole usable.”⁷⁸ It can be used to express the relationships among objects. It can describe the relative positions of the objects’ streams on the carrier, the boundaries of format-based regions of physical objects, and the mapping of the digital objects’ audio streams through the use of pan automation data. Structural metadata can also be used to define areas of interest within an audio object through the use of marker information.

⁷⁸ Robin Wendler, “LDI Update: Metadata in the Library,” *Harvard University Library Notes*, no. 1286 (July/August 1999), 4-5. Also available online: http://hul.harvard.edu/publications/hul_notes_pdfs/HULN_1286.pdf.

4.2.3.4.1 Types of Structural Metadata

During the *Sound Directions* project we generated the following types of structural metadata:

- Documentation of the source Audio Object structure
- AES31-3 Audio Decision List (ADL)
- Broadcast Wave Format time stamp
- Marker information defining areas of interest
- METS <structMap>

Background information for source Audio Object structure and AES31-3 ADL are provided below. We discuss the Broadcast Wave Format time stamp in Chapter 3 and refer to it in our discussion of timelines in this chapter. We refer to the use of marker information throughout this document and specifically in the context of the ADL and our workflows. Use of the METS <structMap> is discussed in the section on preservation packages in Chapter 6.

4.2.3.4.1.1 Source Audio Object Structure

The AES-X098-B standard provides for the documentation of the structure of the audio object that is the source for preservation activity. There are four hierarchical levels of structure that are documented:

Audio Object

Audio object is a general term used to describe digital files and physical audio carriers. This is the root of the hierarchy and contains metadata that pertains to the entire object. Each audio object is described by a single instance document in a strict one-to-one mapping.

Face

A Face is a solitary stream, a sequence of solitary streams, or a group or sequence of groups of synchronous streams. For example, a disc recorded on two sides contains two Faces—each side is a Face. A tape that has content recorded in two directions also has two Faces—each direction, encompassing one track to be played by itself or multiple tracks intended for synchronous playback, is one Face. Every audio object must contain at least one Face.

Region

A Region is the partitioning of a Face based upon format. For example, if the first part of a Face is recorded at 15 ips, and the second part of the Face is recorded at 7.5 ips, then the Face would contain two Regions—one for each tape speed. A Face must have at least one Region.

Stream

A Stream is an individual channel of audio information contained within a Region. Each Region must have one or more Streams. Interleaved audio channels in a digital file are documented as separate individual Streams. For multiple audio files that represent multi-channel audio, each file is a separate audio object.

4.2.3.4.1.2 Timelines

A timeline is simply an abstraction that helps us understand the temporal relationships of audio samples, and a timeline provides a reference for marker information that delineates areas of interest in those audio samples. An audio object, as a collection of audio samples, possesses its own timeline. The object's timeline has a point at which the first sample of the file resides, the start time, and it has a duration, or length of the playback through all of the object's Faces. In the source/destination model of editing, the timeline is the guiding reference. A digital audio recording/editing project has a source timeline where recorded files derive their time values, and a destination timeline where the edited composition resides. Within a preservation workflow it is useful to conceptualize a reference timeline: a common timeline that each manifestation of the recorded work references. That is, if the second song on a recording begins at 3 minutes and 40 seconds, this would continue to be true in every derivative created.

When we digitize an audio object into BWF files, each file contains in its <bext> chunk a field called "TimeReference." The TimeReference field stores the time offset in number-of-samples from the recording device's zero point (midnight) to the first sample of the recording. This is the start time of the file. The BWF file also contains a data chunk. The data chunk stores the number of sample frames in the file. The number of sample frames (samples) defines the duration of the file. In an AES31-3 ADL, and in all of our metadata documents, time values are expressed in terms of a start time and duration. A BWF file's time values are defined by the record start time (TimeReference) in combination with the duration (number of sample frames) as illustrated below.

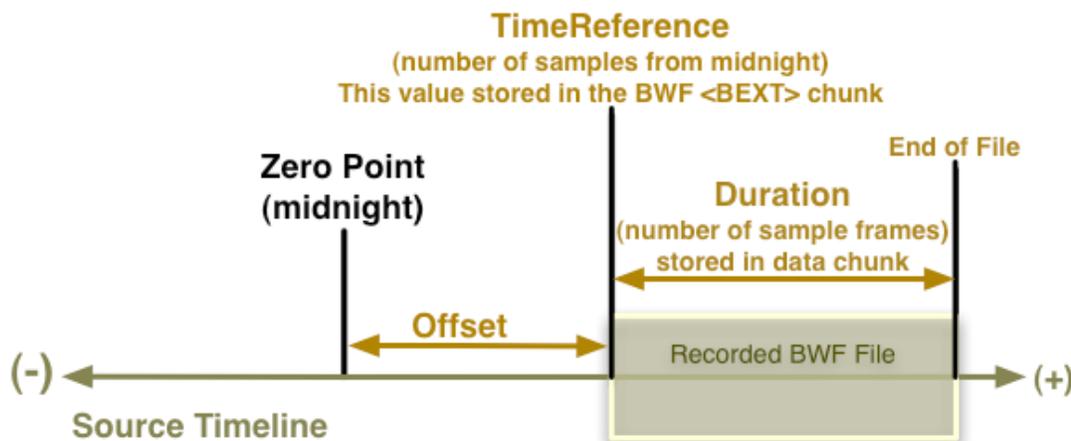


Figure 3: Source timeline

The source timeline and the destination timeline are completely separate representations of time. In the source/destination editing model, as documented in an AES31-3 ADL, the edits are performed from the Source-in point (SrcIn) on the source timeline, to the Destination-in point (DestIn) on the destination timeline. The Destination-out point (DestOut) on the destination timeline indicates the end of the edit clip. The Destination-in point subtracted from the Destination-out point defines the number of samples to be taken from the source BWF file. It may be useful to know that all file-related time values in an AES31-3 ADL are based upon numbers of SMPTE frames, and for purposes of readability are expressed in the ADL in terms of hours, minutes, seconds, frames, and sample remainder. The following diagram is an example of a simple source/destination edit. The edit presents in the destination timeline only the desired portion of the BWF source file, and the edit is nondestructive of the source file.

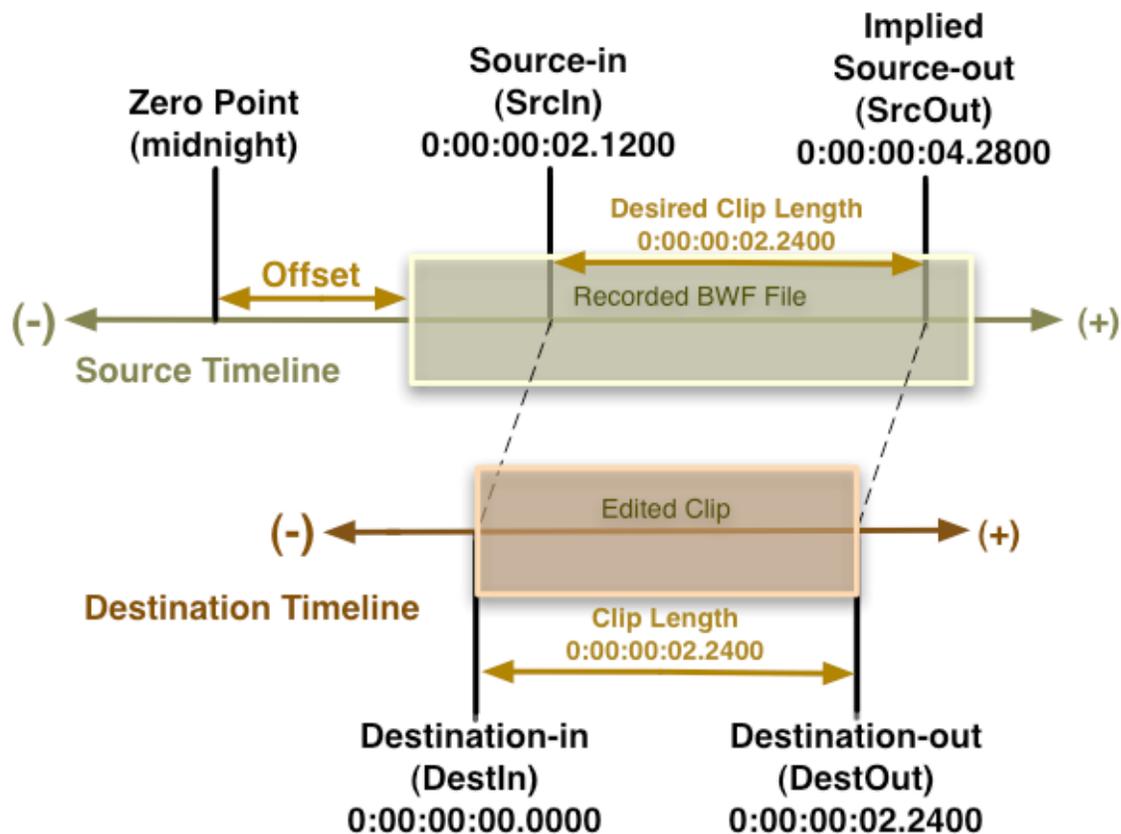


Figure 4: Source and destination timelines

4.2.3.4.1.3 AES31-3 ADL

In our workflows, an ADL documents the relationship between one Face of the source audio object and the digital files created from the preservation transfer of the Face. In the case where stopping the transfer and restarting to continue become necessary, the Face will be represented by more than one digital audio file with overlapping content. These files must be edited together in order to provide the end user with seamless audio as recorded on the original object. To meet the requirement that Preservation Master Files contain unaltered data exactly as reproduced by the playback machine, these edits are expressed in the ADL while the underlying Preservation Master Files remain untouched.

Using a pan list, an ADL can document the pan automation data for the streams of a Face. This data can be used by a DAW's software to position individual streams within the sound field of a project.

An ADL can also document markers in reference to content in a digital file. These markers are used to define areas of interest in the content, and are often used to delineate performances. This functionality is not officially part of the AES31-3 standard as of this writing but has been implemented in proprietary sections of the ADL by at least one manufacturer. The upcoming revision of the standard will include a standardized section for this metadata. Once adopted by manufacturers, this will provide a standards-based method for harvesting this metadata.

When we export a project as an AES31-3 ADL and a media set of BWF files, we not only have a group of digital audio files containing the original object's content, we also have a complement of structural metadata that tells us the exact relationships in time of all the files,

where to transition or cross-fade from one file to the next, as well as pan automation values for the audio streams and a list of the start and stop markers delineating areas of interest.

Below is an example of an AES31-3 ADL for the first side of a stereo audiocassette. In this example, the source files from the cassette transfer are discrete (non-interleaved) left and right channels. Due to the limits of the page, the SOURCE_INDEX URL's wrap on these pages. The major sections of the ADL are:⁷⁹

- <VERSION> Contains a unique ID for this ADL, verifies the version of the standard, and contains information on the software that produced the ADL
- <PROJECT> Contains information relating to the specific project including a title assigned by the user and the date the ADL was created
- <SYSTEM> Contains optional information about setup parameters of the system (workstation) that produced the ADL
- <SEQUENCE> Provides global information about ADL contents and status
- <TRACKLIST> An optional track list where track numbers correspond to the destination channels in the event list
- <SOURCE_INDEX> List of source material used in the project. Each source is identified with an incrementing number beginning with 1. (F) identifies the source as a file. The URL that follows is the location of the file. The string of numbers and letters following this URL is a unique source identifier (USID) from the BWF Originator Reference field. Next is the start position for the file on the destination timeline (BWF time stamp), followed by the duration of the file
- <EVENT> This is the main edit list. (Cut) = a simple cut edit. The first timecode value is the in-point for the source file (SrcIn), the second value is the location for the edited clip on the destination timeline (DestIn), and the third value is the end of the clip on the destination timeline (DestOut)
- <PAN_LIST> This section will carry pan information in the upcoming revised standard
- <MARK_LIST> This section will carry marker metadata in the upcoming revised standard

⁷⁹ The full specification for the AES31-3 standard may be obtained from the Audio Engineering Society. See AES, Standards Committee, "Standards in Print," <http://www.aes.org/publications/standards/>.

```

<ADL>
  <VERSION>
    (ADL_ID)          "06,64,43,52,01,01,01,04,01,02,03,04," } ADL identifiers
    (ADL_UID)         fa3eacf1-3387-4ba1-8ea3-234751c34b85
    (VER_ADL_VERSION) 01.00
    (VER_CREATOR)     "Pyramix" } Info. on software and ADL schema
    (VER_CRTR)        05.00.03.03 } versions
  </VERSION>

  <PROJECT>
    (PROJ_TITLE)      "CD_32157_Archival_Side_1" } Project info. supplied
    (PROJ_ORIGINATOR) "Merging Technologies S.A." } by software
    (PROJ_CREATE_DATE) 2007-01-10T20:33:06
    (PROJ_NOTES)       ""
    (PROJ_CLIENT_DATA) ""
  </PROJECT>

  <SYSTEM>
    (SYS_SRC_OFFSET) 00|00|00.00*0000 } Originating system
    (SYS_BIT_DEPTH)  24                } setup parameters
    (SYS_AUD_CODECC) "BWF"
    (SYS_GAIN)        0.00
  </SYSTEM>

  <SEQUENCE>
    (SEQ_SAMPLE_RATE) S96000 } Global info. on ADL
    (SEQ_FRAME_RATE)  30      } contents and status
    (SEQ_ADL_LEVEL)   1
    (SEQ_CLEAN)        FALSE
    (SEQ_SORT)         0
    (SEQ_MULTICHAN)    FALSE
    (SEQ_DEST_START)  00|00|00.00*0000
  </SEQUENCE>

  <TRACKLIST>
    (Track)  1  "Input L-R" } Track names
    (Track)  2  "Input L-R" } (1 if mono., 2 if stereo.)
  </TRACKLIST>

  <SOURCE_INDEX>
    (Index) 0001 ← File 1
    (F) "URL:file://localhost//Volumes/V20/DRS_DEPOSITS/
    Kirchner/Deposit_01/CD_32157/archival/CD_32157_Archival_Side_1_Media/
    CD_32157_AM_01_01_{F50301AB-CDD5-4401-A2F2-47DA7C58A3B8}.wav"
    CHMTIPYRAMIX16934153137385150454 00|00|00.00*0000
    ← Duration → 00|45|56.09*2382 "CD_32157_AM_01_01" N ← Start position
    (Index) 0002 ← File 2
    (F) "URL:file://localhost//Volumes/V20/DRS_DEPOSITS/
    Kirchner/Deposit_01/CD_32157/archival/CD_32157_Archival_Side_1_Media/
    CD_32157_AM_01_01_{D28AAF03-471B-4A12-8A1E-3F561218057D}.wav"
    CHMTIPYRAMIX16934153213745780579 00|00|00.00*0000
    ← Duration → 00|45|56.09*2382 "CD_32157_AM_01_01" N ← Start position
    </SOURCE_INDEX>
    (BWF time stamp)

  <EVENT_LIST>
    (Entry) 0001
    Source In (Cut) I 0001 1 1
    00|00|00.00*0000 00|00|00.00*0000
    00|45|56.09*2381
    Out Point (Rem) NAME "CD_32157_AM_01_01 (1)"
    Destination (Entry) 0002
    (Cut) I 0002 1 2
    00|00|00.00*0000 00|00|00.00*0000
    00|45|56.09*2381
    (Rem) NAME "CD_32157_AM_01_01 (2)"
  </EVENT_LIST>

  <PAN_LIST>
    (PP) 1 00|00|00.00*0000 -100.0 0.0 } *Pan list -
    (PP) 2 00|00|00.00*0000 100.0 0.0 } controls channel mapping
  </PAN_LIST>

```

Source project material:
Locations, names, identifiers, durations of wav files underlying the ADL

Edit List:
For each file - Starting point in file, beginning and ending position on time-line, displayed filename

```

      (PP) 1 00|00|00.00*0000 -100.0 0.0 } *Pan list -
      (PP) 2 00|00|00.00*0000 100.0 0.0 } controls channel mapping
</PAN_LIST>

<MARK_LIST>
(MK-PQ-START) 0 00|00|02.20*2560 _ "Symphony #102 Movement 1"
(MK-PQ-END) 0 00|09|01.10*0000 _ "CD Track Stop"
(MK-PQ-START) 0 00|09|53.08*0000 _ "Symphony #102 Movement 2"
(MK-PQ-END) 0 00|16|44.18*0000 _ "CD Track Stop"
(MK-PQ-START) 0 00|17|11.20*2560 _ "Symphony #102 Movement 3"
(MK-PQ-END) 0 00|23|02.19*0640 _ "CD Track Stop"
(MK-PQ-START) 0 00|23|12.21*0640 _ "Symphony #102 Movement 4"
(MK-PQ-END) 0 00|28|11.00*0000 _ "CD Track Stop"
(MK-PQ-START) 0 00|28|33.03*1920 _ "Brass and Chamber Orchestra Movement 1"
(MK-PQ-END) 0 00|36|54.08*0000 _ "CD Track Stop"
(MK-PQ-START) 0 00|36|59.02*1280 _ "Brass and Chamber Orchestra Movement 2"
(MK-PQ-END) 0 00|45|54.11*0640 _ "CD Track Stop"
</MARK_LIST>
</ADL>

```

*Marker List:
Position and name for each
marker placed in the file

* These sections follow the upcoming revision of the AES31-3 standard.

4.2.4 Metadata at Harvard

4.2.4.1 Preservation Metadata Documents and Their Creation

The Harvard College Library's Audio Preservation Services has created software tools to aid in the collection of metadata, in the creation of the various metadata documents, and in the automation of those tasks.

4.2.4.1.1 Source Audio Object Metadata

The source audio object, which we call original audio object, is described in an XML metadata document called *Audio Object Metadata*. This first document created in our preservation workflow contains technical, structural and descriptive metadata about the original object. The original audio object metadata document is created using a java application called "AudioObjectManager" that was written here at Harvard. In this application a document template is customized to present the data fields pertinent to the source object, and then, if the object is physical rather than digital, those fields are completed by the operator. If the original audio object is a digital audio file, rather than a physical object, then the application can parse the file and retrieve data for most pertinent fields. One of the fields for the digital audio file will be for a MD5 checksum that the application will perform on the digital object. The remainder of data, such as condition comments and data not discoverable from the original file, is hand entered.

Below we see the selection of the appropriate root properties of the physical audio object in order to configure the document template in the AudioObjectManager software application.

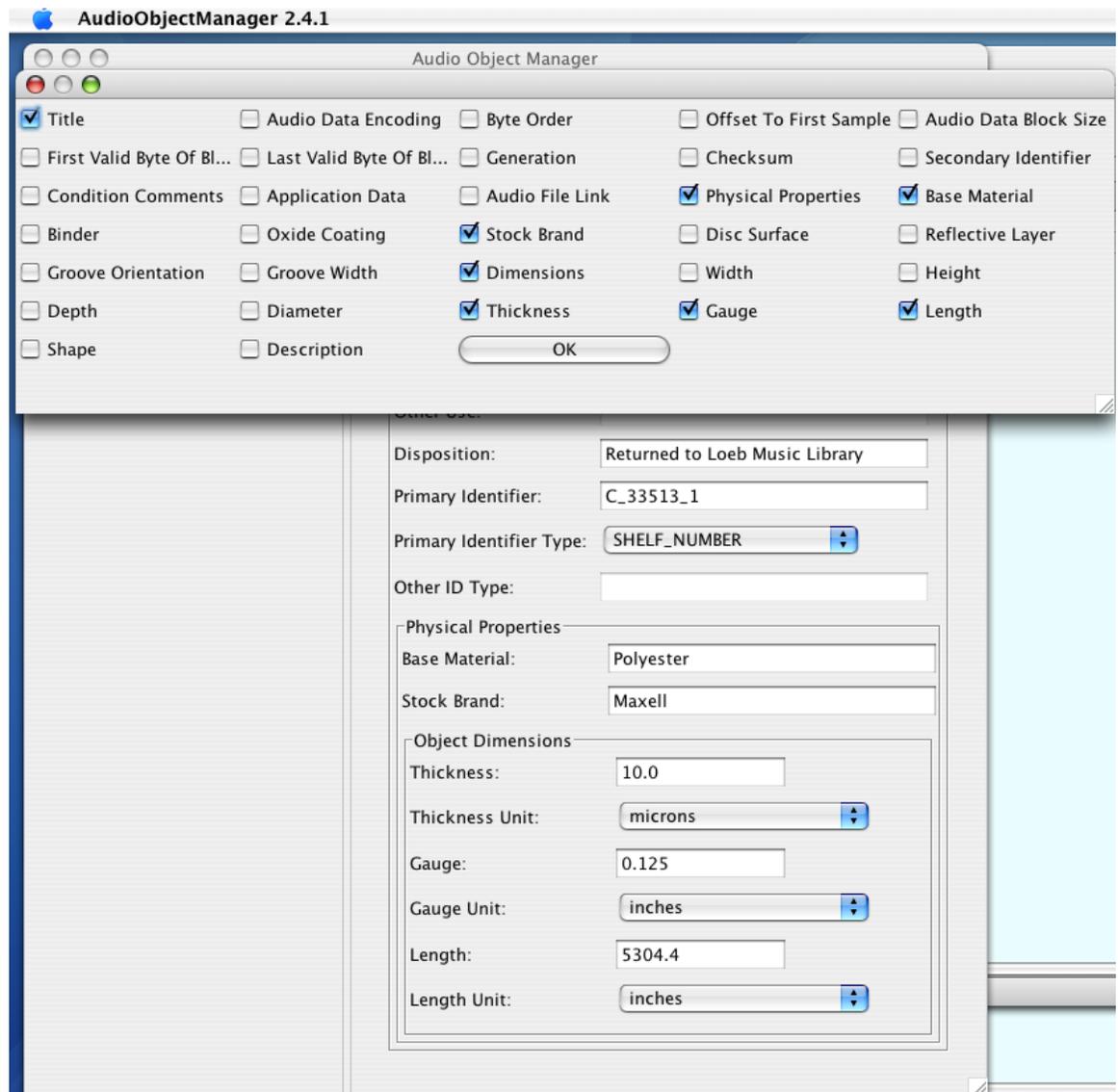


Figure 5: Configuring the template in AudioObjectManager

This image shows the document with the root of the object selected in the document tree on the left and the root properties displayed to the right.

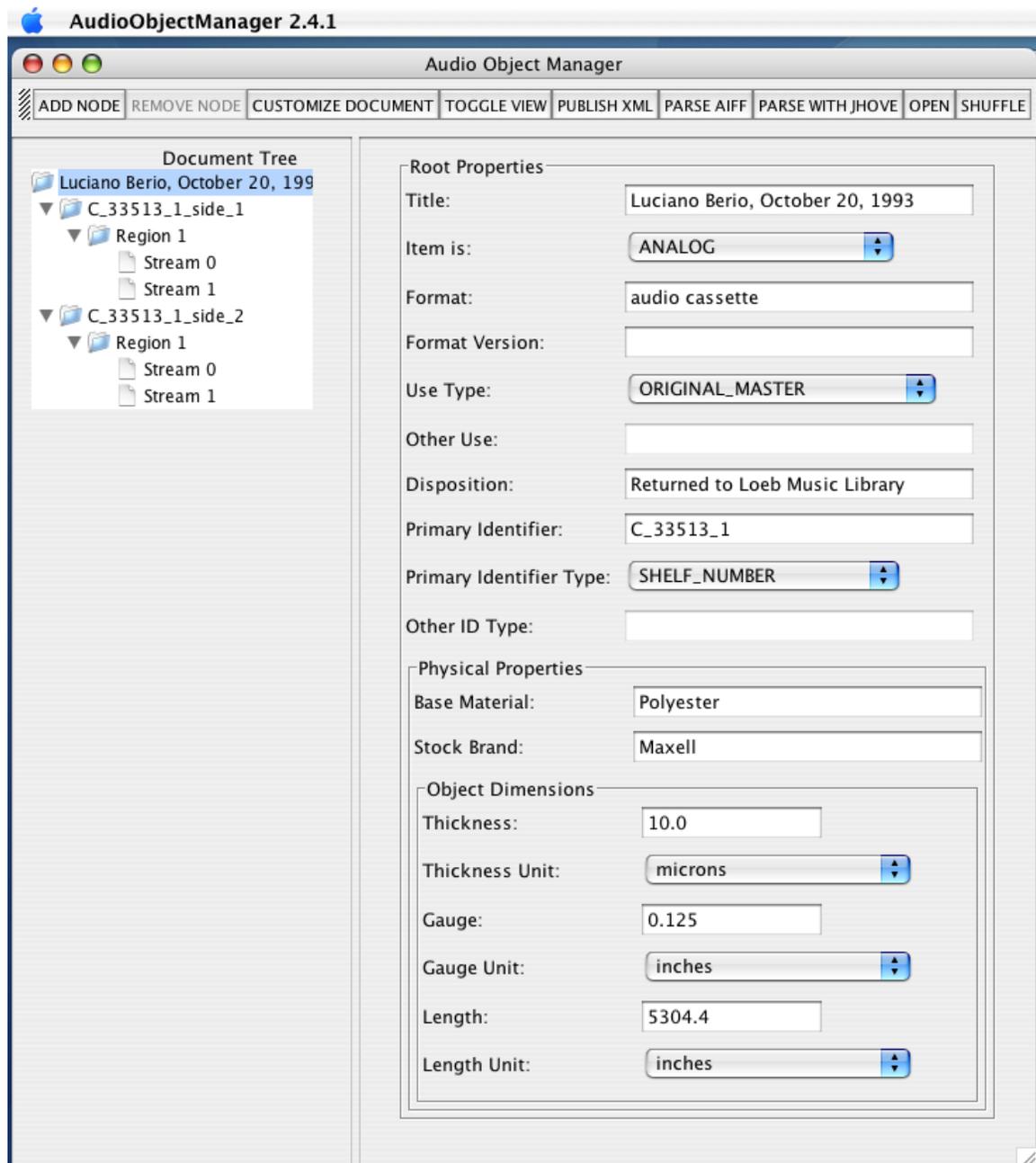


Figure 6: Root properties in AudioObjectManager

Nodes representing Faces, Regions and Streams are added and completed as needed. Below we see the first Face selected and its properties displayed.

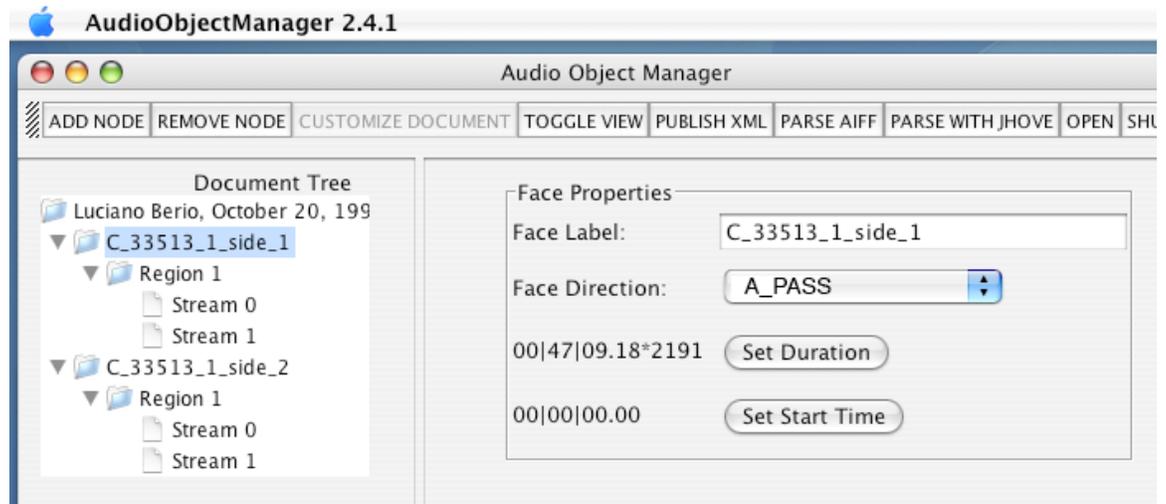


Figure 7: Face properties in AudioObjectManager

Next we see the first Region selected and its properties displayed.

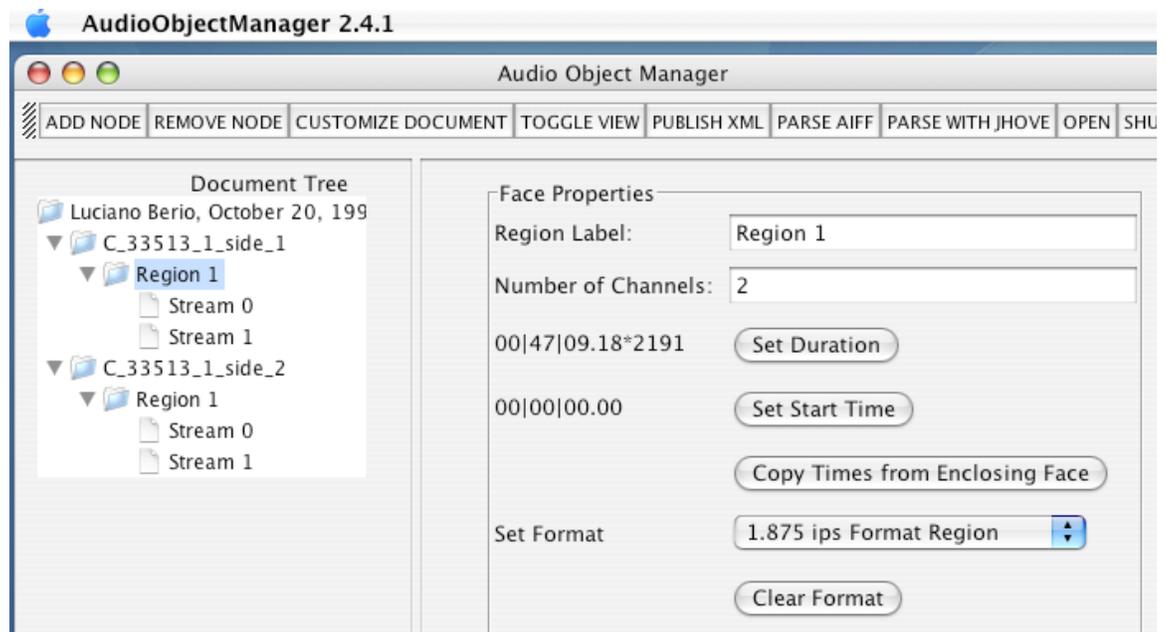


Figure 8: Region properties in AudioObjectManager

The Region's format is created by toggling the view of the document and filling in the appropriate fields.

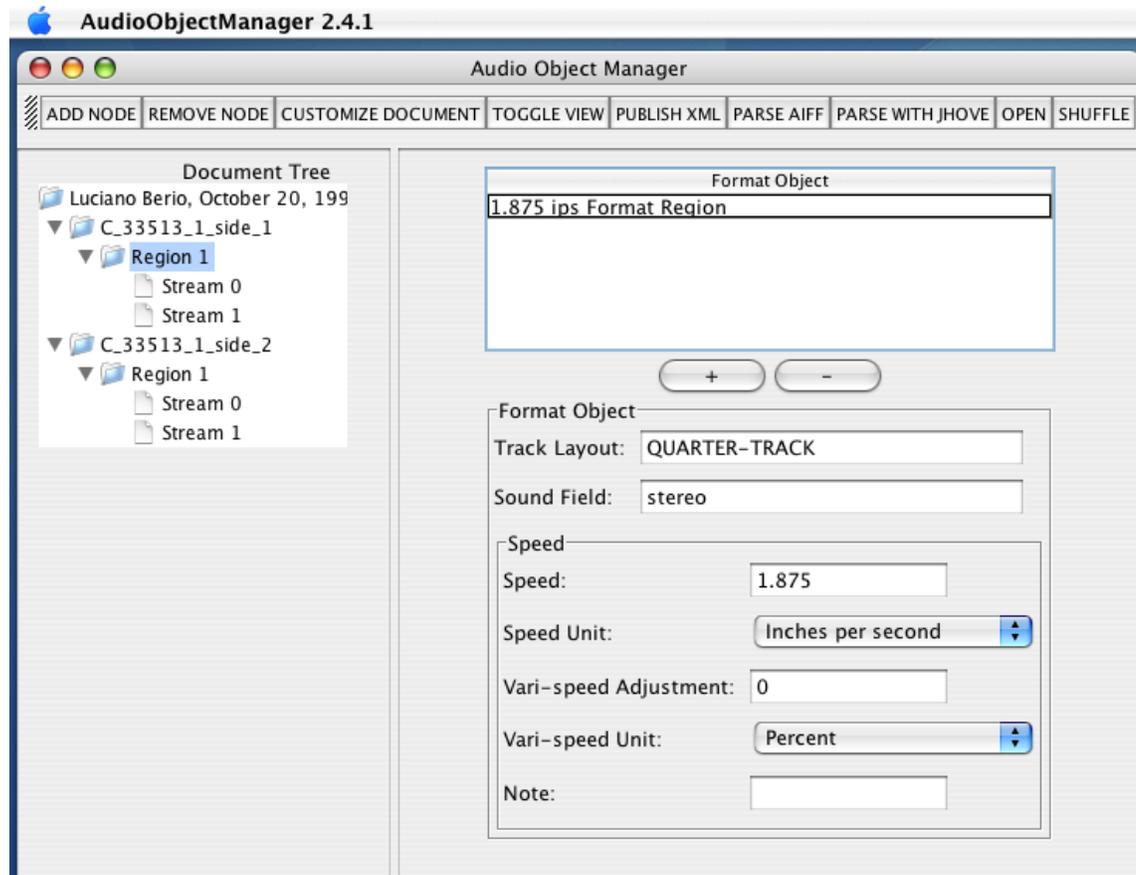


Figure 9: Configuring the format in AudioObjectManager

Here we see the first and second Streams of the first Region selected and displayed.

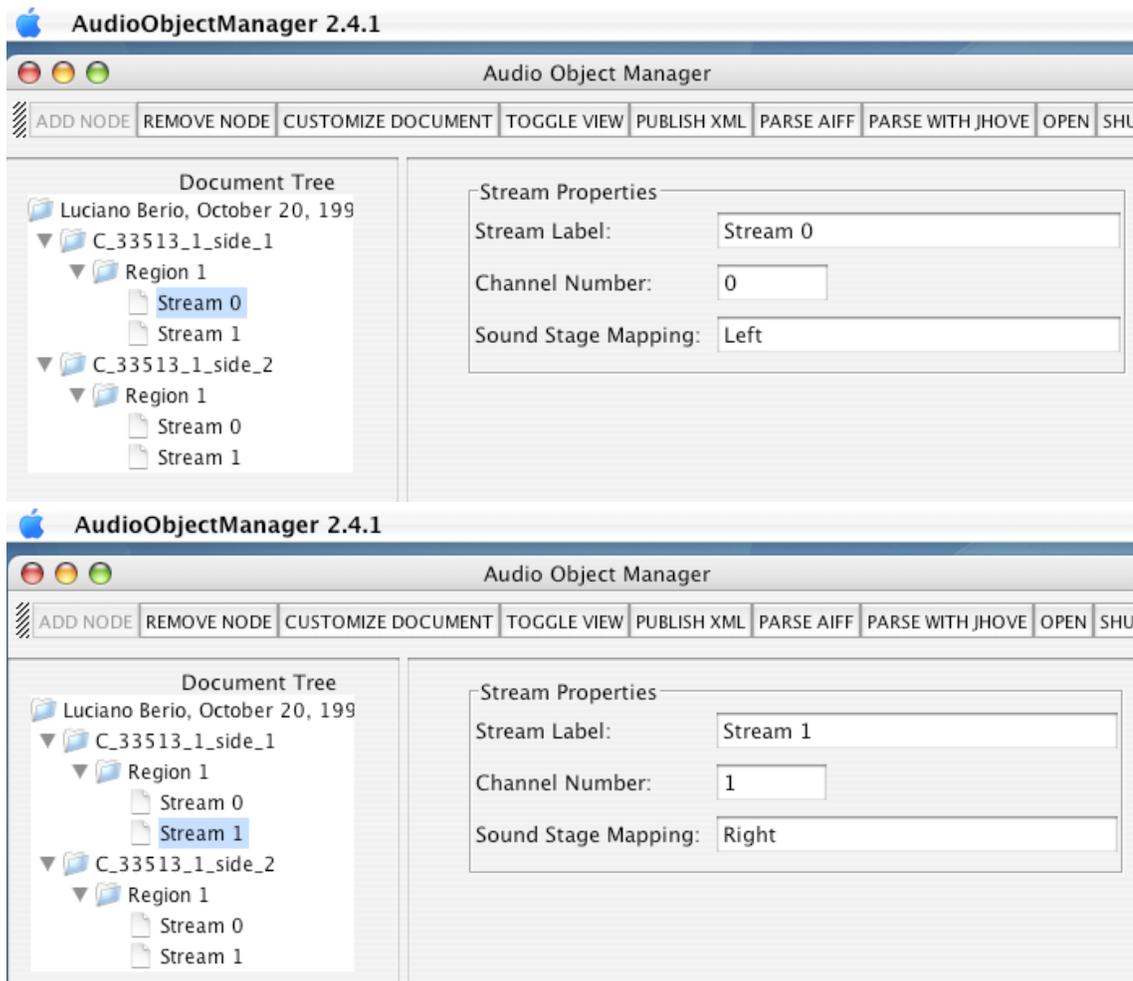


Figure 10: Stream properties in AudioObjectManager

Finally, observe the resulting published document as XML code.

```
<?xml version="1.0" encoding="UTF-8"?>
<audioObject xmlns:xlink="http://www.w3.org/1999/xlink"
  xmlns:tcf="http://www.aes.org/tcf"
  xmlns:xsd="http://www.w3.org/2001/XMLSchema"
  xmlns="http://www.aes.org/audioObject"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="http://www.aes.org/audioObject http://hul.harvard.edu/ois/xml/xsd/drs/
audioObject.xsd"
  ID="_11559222219020.2609892689404265"
  title="Luciano Berio, October 20, 1993" analogDigitalFlag="ANALOG"
  disposition="Returned to Loeb Music Library" schemaVersion="1.03b"
  xsi:type="audioObjectType">
  <format>audio cassette</format>
  <physicalProperties>
    <baseMaterial>Polyester</baseMaterial>
    <stockBrand>Maxell</stockBrand>
    <dimensions>
      <gauge unit="inches">0.125</gauge>
      <length unit="inches">5304.4</length>
      <thickness unit="microns">10.0</thickness>
    
```

```

</dimensions>
</physicalProperties>
<use useType=»ORIGINAL_MASTER«/>
<primaryIdentifier identifierType=»SHELF_NUMBER«>C_33513_1</primaryIdentifier>
<face ID=»_11559242914940.9381934972814039« direction=»A_WIND«
  audioObjectRef=»_11559222219020.2609892689404265« label=»C_33513_1_side_1«>
<timeline>
  <tcf:startTime frameCount="30" timeBase="1000"
    videoField="FIELD_1" countingMode="NTSC_NON_DROP_FRAME">
    <tcf:hours>0</tcf:hours>
    <tcf:minutes>0</tcf:minutes>
    <tcf:seconds>0</tcf:seconds>
    <tcf:frames>0</tcf:frames>
    <tcf:filmFraming framing="NOT_APPLICABLE" xsi:type="tcf:ntscFilmFramingType"/>
  </tcf:startTime>
  <tcf:duration frameCount="30" timeBase="1000"
    videoField="FIELD_1" countingMode="NTSC_NON_DROP_FRAME">
    <tcf:hours>0</tcf:hours>
    <tcf:minutes>47</tcf:minutes>
    <tcf:seconds>9</tcf:seconds>
    <tcf:frames>18</tcf:frames>
    <tcf:samples sampleRate="S96000">
      <tcf:numberOfSamples>2191</tcf:numberOfSamples>
    </tcf:samples>
    <tcf:filmFraming framing="NOT_APPLICABLE" xsi:type="tcf:ntscFilmFramingType"/>
  </tcf:duration>
</timeline>
<region ID="_11559243209520.04244254843681039"
  formatRef="_11559243270550.5106679149708852"
  faceRef="_11559242914940.9381934972814039" label="Region 1">
  <timeRange>
    <tcf:startTime frameCount="30" timeBase="1000"
      videoField="FIELD_1" countingMode="NTSC_NON_DROP_FRAME">
      <tcf:hours>0</tcf:hours>
      <tcf:minutes>0</tcf:minutes>
      <tcf:seconds>0</tcf:seconds>
      <tcf:frames>0</tcf:frames>
      <tcf:filmFraming framing="NOT_APPLICABLE" xsi:type="tcf:ntscFilmFramingType"/>
    </tcf:startTime>
    <tcf:duration frameCount="30" timeBase="1000"
      videoField="FIELD_1" countingMode="NTSC_NON_DROP_FRAME">
      <tcf:hours>0</tcf:hours>
      <tcf:minutes>47</tcf:minutes>
      <tcf:seconds>9</tcf:seconds>
      <tcf:frames>18</tcf:frames>
      <tcf:samples sampleRate="S96000">
        <tcf:numberOfSamples>2191</tcf:numberOfSamples>
      </tcf:samples>
      <tcf:filmFraming framing="NOT_APPLICABLE" xsi:type="tcf:ntscFilmFramingType"/>
    </tcf:duration>
  </timeRange>
  <numChannels>2</numChannels>
  <stream ID=»_11559243815370.024875668740057644«
    label=»Stream 0« faceRegionRef=»_11559243209520.04244254843681039«>
    <channelAssignment channelNum=»0« mapLocation=»Left«/>
  </stream>
  <stream ID=»_11559243896970.36782358265958637«
    label=»Stream 1« faceRegionRef=»_11559243209520.04244254843681039«>

```

```

        <channelAssignment channelNum=»1» mapLocation=»Right»/>
    </stream>
</region>
</face>
<face ID=»_11559245144540.03726796072569449» direction=»B_WIND»
  audioObjectRef=»_11559222219020.2609892689404265» label=»C_33513_1_side_2»>
  <timeline>
    <tcf:startTime frameCount=»30» timeBase=»1000»
      videoField=»FIELD_1» countingMode=»NTSC_NON_DROP_FRAME»>
      <tcf:hours>0</tcf:hours>
      <tcf:minutes>47</tcf:minutes>
      <tcf:seconds>12</tcf:seconds>
      <tcf:frames>18</tcf:frames>
      <tcf:samples sampleRate=»S96000»>
        <tcf:numberOfSamples>2191</tcf:numberOfSamples>
      </tcf:samples>
      <tcf:filmFraming framing=»NOT_APPLICABLE» xsi:type=»tcf:ntscFilmFramingType»/>
    </tcf:startTime>
    <tcf:duration frameCount=»30» timeBase=»1000»
      videoField=»FIELD_1» countingMode=»NTSC_NON_DROP_FRAME»>
      <tcf:hours>0</tcf:hours>
      <tcf:minutes>34</tcf:minutes>
      <tcf:seconds>18</tcf:seconds>
      <tcf:frames>4</tcf:frames>
      <tcf:samples sampleRate=»S96000»>
        <tcf:numberOfSamples>2339</tcf:numberOfSamples>
      </tcf:samples>
      <tcf:filmFraming framing=»NOT_APPLICABLE» xsi:type=»tcf:ntscFilmFramingType»/>
    </tcf:duration>
  </timeline>
</region ID=»_11559245637120.8879793908058797»
  formatRef=»_11559243270550.5106679149708852»
  faceRef=»_11559245144540.03726796072569449» label=»Region 1»>
  <timeRange>
    <tcf:startTime frameCount=»30» timeBase=»1000»
      videoField=»FIELD_1» countingMode=»NTSC_NON_DROP_FRAME»>
      <tcf:hours>0</tcf:hours>
      <tcf:minutes>47</tcf:minutes>
      <tcf:seconds>12</tcf:seconds>
      <tcf:frames>18</tcf:frames>
      <tcf:samples sampleRate=»S96000»>
        <tcf:numberOfSamples>2191</tcf:numberOfSamples>
      </tcf:samples>
      <tcf:filmFraming framing=»NOT_APPLICABLE» xsi:type=»tcf:ntscFilmFramingType»/>
    </tcf:startTime>
    <tcf:duration frameCount=»30» timeBase=»1000»
      videoField=»FIELD_1» countingMode=»NTSC_NON_DROP_FRAME»>
      <tcf:hours>0</tcf:hours>
      <tcf:minutes>34</tcf:minutes>
      <tcf:seconds>18</tcf:seconds>
      <tcf:frames>4</tcf:frames>
      <tcf:samples sampleRate=»S96000»>
        <tcf:numberOfSamples>2339</tcf:numberOfSamples>
      </tcf:samples>
      <tcf:filmFraming framing=»NOT_APPLICABLE» xsi:type=»tcf:ntscFilmFramingType»/>
    </tcf:duration>
  </timeRange>
</numChannels>2</numChannels>

```

```

<stream ID=»_11559245736880.6900223089837544»
  label=»Stream 0» faceRegionRef=»_11559245637120.8879793908058797»>
  <channelAssignment channelNum=»0» mapLocation=»Left»/>
</stream>
<stream ID=»_11559245766480.932796829852367»
  label=»Stream 1» faceRegionRef=»_11559245637120.8879793908058797»>
  <channelAssignment channelNum=»1» mapLocation=»Right»/>
</stream>
</region>
</face>
<formatList>
  <formatRegion ID=»_11559243270550.5106679149708852»
    label=»1.875 ips Format Region» ownerRef=»_11559243209520.04244254843681039_1
    1559245637120.8879793908058797»>
    <speed>
      <speedCoarse unit=»Inches per second»>1.875</speedCoarse>
      <varispeedAdjustment unit=»Percent»>0</varispeedAdjustment>
    </speed>
    <trackLayout>QUARTER-TRACK</trackLayout>
    <soundField>stereo</soundField>
  </formatRegion>
</formatList>
</audioObject>

```

4.2.4.1.2 Audio Object Metadata for Digital Files

Every digital audio file that is created in the preservation workflow gets an MD5 checksum and its own corresponding audio object metadata XML document. The document is created by a software tool such as *jhovemeta*⁸⁰ or *rameta*⁸¹ and is similar to the original audio object XML document, except that it is not hand generated. An example XML document may be found in Appendix 2.

4.2.4.1.3 Digital Provenance Metadata (digipro) (digipro)

The digipro document, or process history for the preservation workflow, is created using a Mac OS custom software application, called the Audio Processing XML Editor (APXE)⁸² that was designed specifically for quick documentation of the work history in the audio preservation process. With APXE, the work history is documented across a series of steps referred to as processing events. Each processing event documents a signal path from reproducer to recorder, or a data processing path from one device to another, as well as input and output media. The audio processing history is stored as XML data utilizing the AES-X098-B Process History Metadata Schema. The digipro document is generated at the start of each preservation project and is updated upon completion of every critical task in the workflow so that we have a complete record of those processes during initial preservation and for future reference. Most of the Harvard *Sound Directions* Toolkit records its actions in digipro documents named for their particular stage of the workflow that are finally merged into one digipro document during the creation of the deposit package. The digipro document is often used to facilitate a team effort on a project or to ensure consistency in processing a

⁸⁰ “*jhovemeta*” is a tool in the Harvard *Sound Directions* Toolkit that generates audio object metadata for a specified BWF file.

⁸¹ “*rameta*” is a tool in the Harvard *Sound Directions* Toolkit that generates audio object metadata for a specified RealAudio file.

⁸² APXE was written for Harvard University by programmer Robert La Ferla using Exolab Castor and Apache Xerces <http://www.castor.org> <http://www.apache.org>

large collection of similar objects whose preservation spans long periods of time. Regardless of its immediate usefulness in the preservation workflow, it is our permanent record of, and clear path back to, what was done in the preservation of an object.

In Figure 11, we see an open APXE digiproov document showing in the top panel a list of six process events in the preservation of an R-DAT tape, beginning with the archival transfer and AES31-3 export, then to the rendering of the multiple archival media files into a single file for production purposes, next to sample rate conversion (SRC) by a scripted sample rate conversion tool, then to a scripted dithering process of the SRC'd files, next to the scripted sample rate conversion of the exported ADL, and finally to the scripted RealAudio encoding of the deliverable files. In the middle panel is the media pool for the event selected, and below that is the opened, virtual, media patchbay—a graphic representation of connections—showing the routing of audio streams and their respective sources and destinations. In this instance, the media labeled “AWM_DAT_161 - Region 1” is the original audio object, and the media labeled “AWM_DAT_161_Archival_Side_1” is the exported archival transfer AES31-3 ADL. To the right side of the media patchbay is the timeline master. The timeline master is the media whose timeline is the reference for the result of a process event. Normally, the original object provides the master timeline.

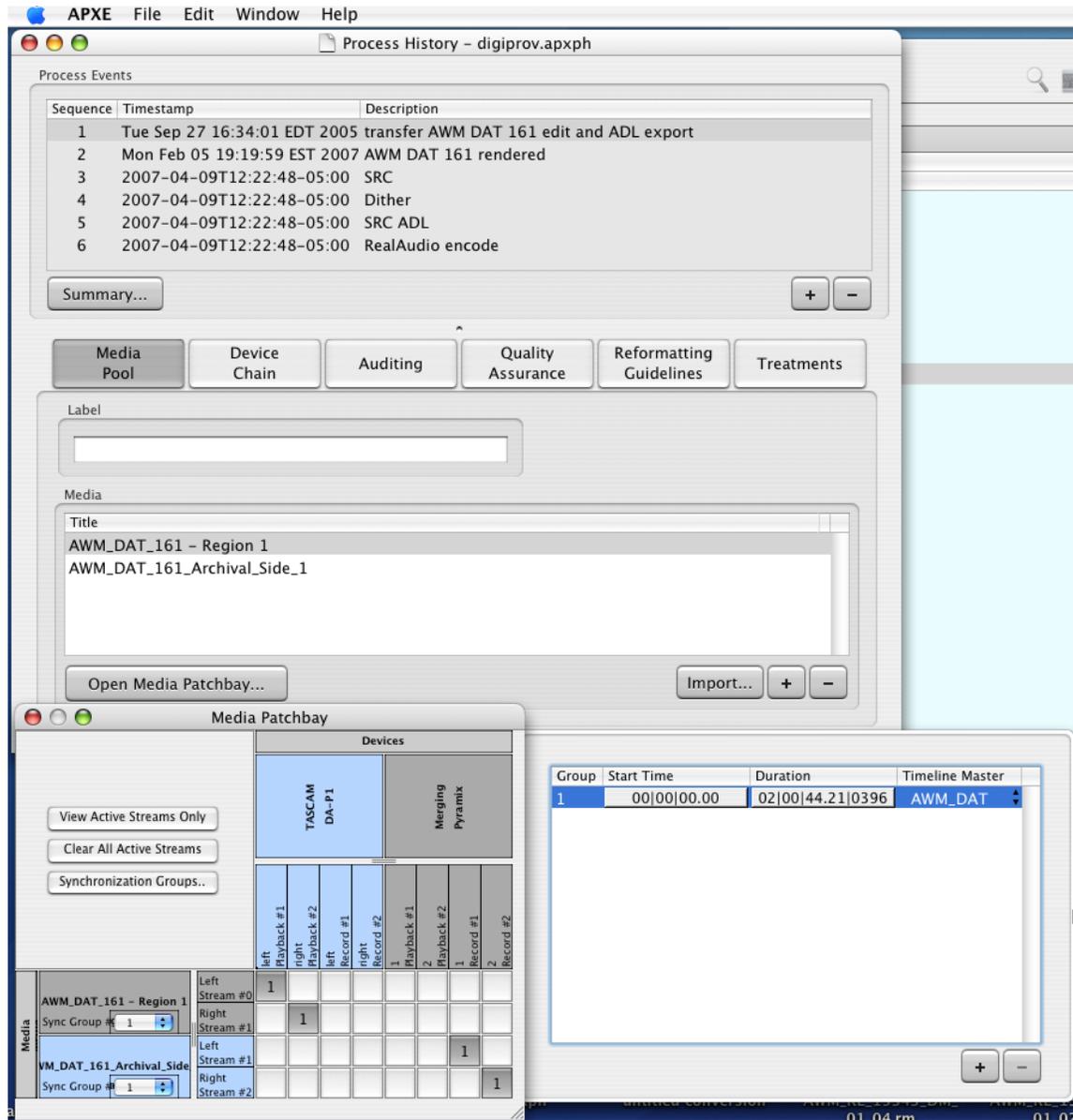


Figure 11: Process event in APXE

The image below shows the device chain for the first process event. The synchronization master for the chain is the TASCAM DAT player, indicated by the square wave icon on the left. The synchronization master is the digital audio device that provides the master digital sample clock reference for downstream digital audio devices in the chain.

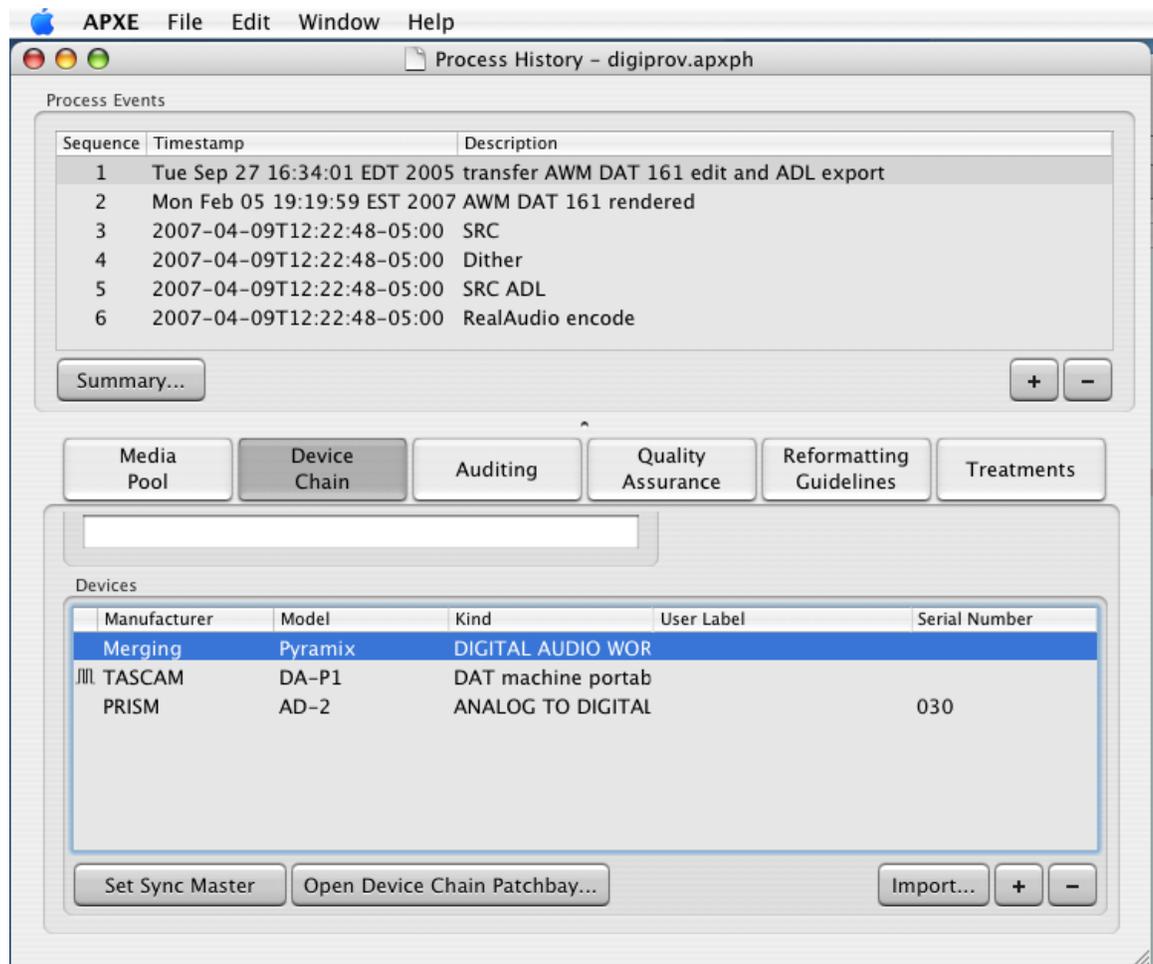


Figure 12: Device chain in APXE

The next image shows the overview of the DAW in the device chain and some of its parameters. Indicated in the other tabs would be the physical input and output channels, any profiles, modules such as de-noising, and media channels pertaining to streams. The virtual patchbay for the device can also be displayed as well as a logical diagram of the connections in a device chain. Each device in the chain gets completely described in a similar manner, and the entire documentation process is repeated for each process event. Templates for particular devices can be saved and used as a time-saving convenience when building a device chain. Templates for an entire device chain can also be created and saved for re-use. The ability to use templates not only saves time but helps reduce human error and promote consistency in the metadata.

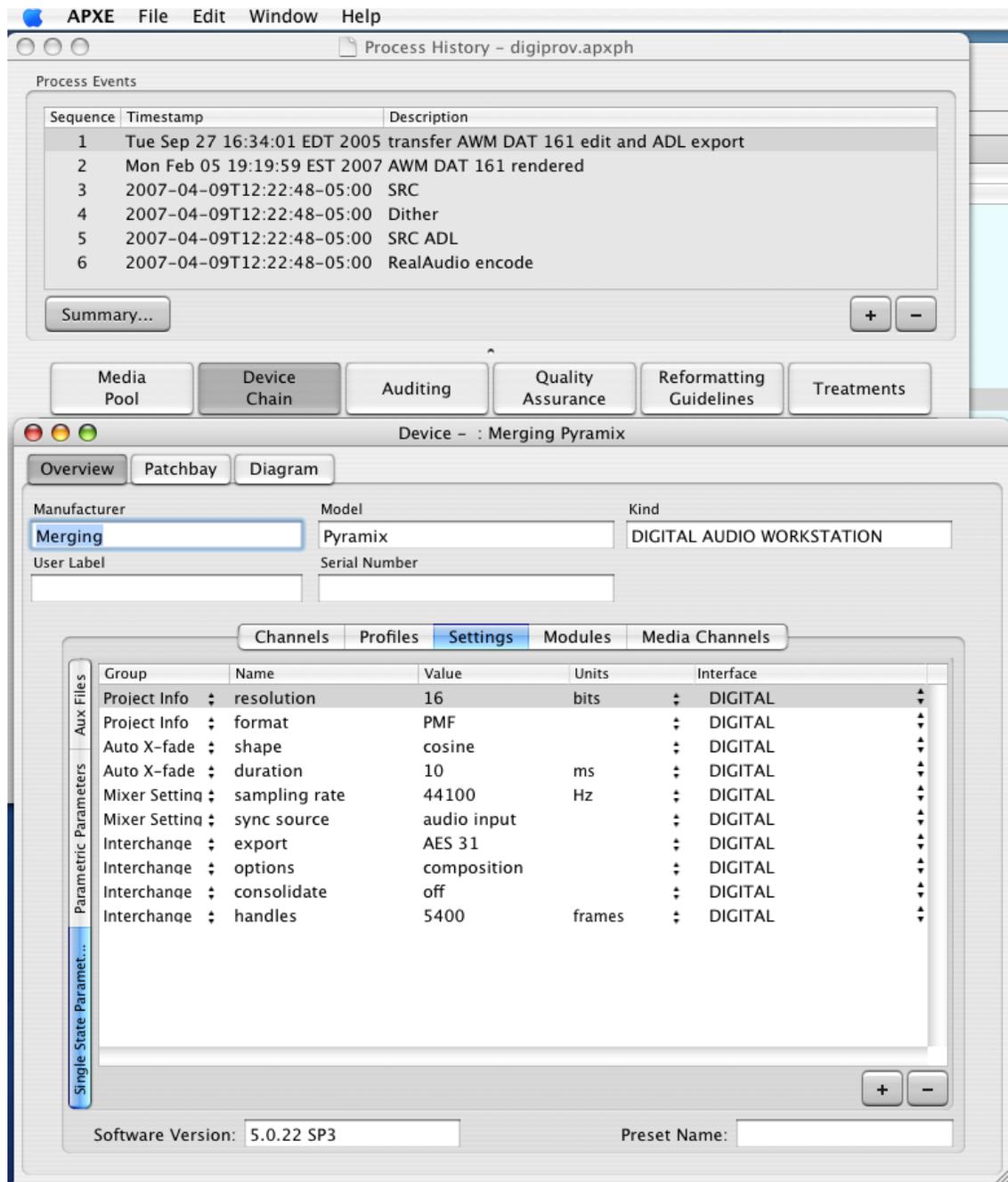


Figure 13: Device settings in APXE

4.2.5 Metadata at Indiana

During this project the ATM at Indiana University served as a test site for the emerging AES standards for the collection of technical and digital provenance metadata. We applied the latest versions of the schemas—they are not yet public—in developing software that will hold and, in some cases, automatically collect this metadata. Our software tool is named ATMC—the Audio Technical Metadata Collector—and it will be developed for release as open source software within a project funded by NEH that begins June, 2007. A complete list of metadata elements used in ATMC with definitions for each is included in Appendix 1.

4.2.5.1 Audio Technical Metadata Collector (ATMC)

ATMC supports the collection and generation of metadata on the source audio object that is the target for preservation transfer, digital files created during transfer, and the digitization process. This tool currently supports features allowing the user to do the following:

- Enter and edit a wide variety of technical and structural metadata for audio objects
- Enter and edit audio object evaluations to support tracking physical and aural degradation over time
- Parse audio files to automatically collect relevant metadata
- Generate MD5 checksum information
- Enter and edit parent/child relationship information for audio objects
- View relationships diagrams that show related objects in a tree-like graph
- Enter and edit processing history (digital provenance) information for events undertaken with audio objects
- Export AES-compliant XML for audio objects and processing history
- Store data in an Oracle database to support preservation management of, and research in, ATM collections over time. There is also a local-only version that writes files to a local folder

ATMC is a Java Swing-based application that uses Tomcat and Axis to communicate to a server via a SOAP protocol. The server uses JDBC to connect to an Oracle database which stores the audio object information in XML form. The application also incorporates a portion of software from JHOVE (JSTOR/Harvard Object Validation Environment) to assist with parsing audio files and extracting metadata.

4.2.5.2 Technical Metadata

Technical metadata is collected for both the source audio object that is the target for preservation activity and all digital files that are the result of such activity.

4.2.5.2.1 Source Audio Object

Characteristics of the source audio object are entered in two tabs of the ATMC interface that are labeled General and Format. The first includes fields for data common to all recordings, including some elements that may be considered descriptive metadata, such as generation, use, and disposition. The second provides a location in which to enter format-specific data and contains only the fields that are applicable to the specific format of the object, such as tape thickness or disc surface material. Additional format attributes such as base material, sound field, and brand/product number are documented by Region in the structure part of the interface as described in the section on structural metadata, below. Format-related preservation problems such as cinching or windowing on open reel tapes are documented in a separate evaluation area that also provides a location to note sonic problems such as distortion or hum.

4.2.5.2.2 Digital Files

ATMC parses digital audio files to gather technical metadata and to create a record of the file in the database. Data collected includes byte order, sample rate, bit depth, and block alignment along with other elements. While parsing the file ATMC generates an MD5 hash, storing the checksum value alongside other technical metadata.

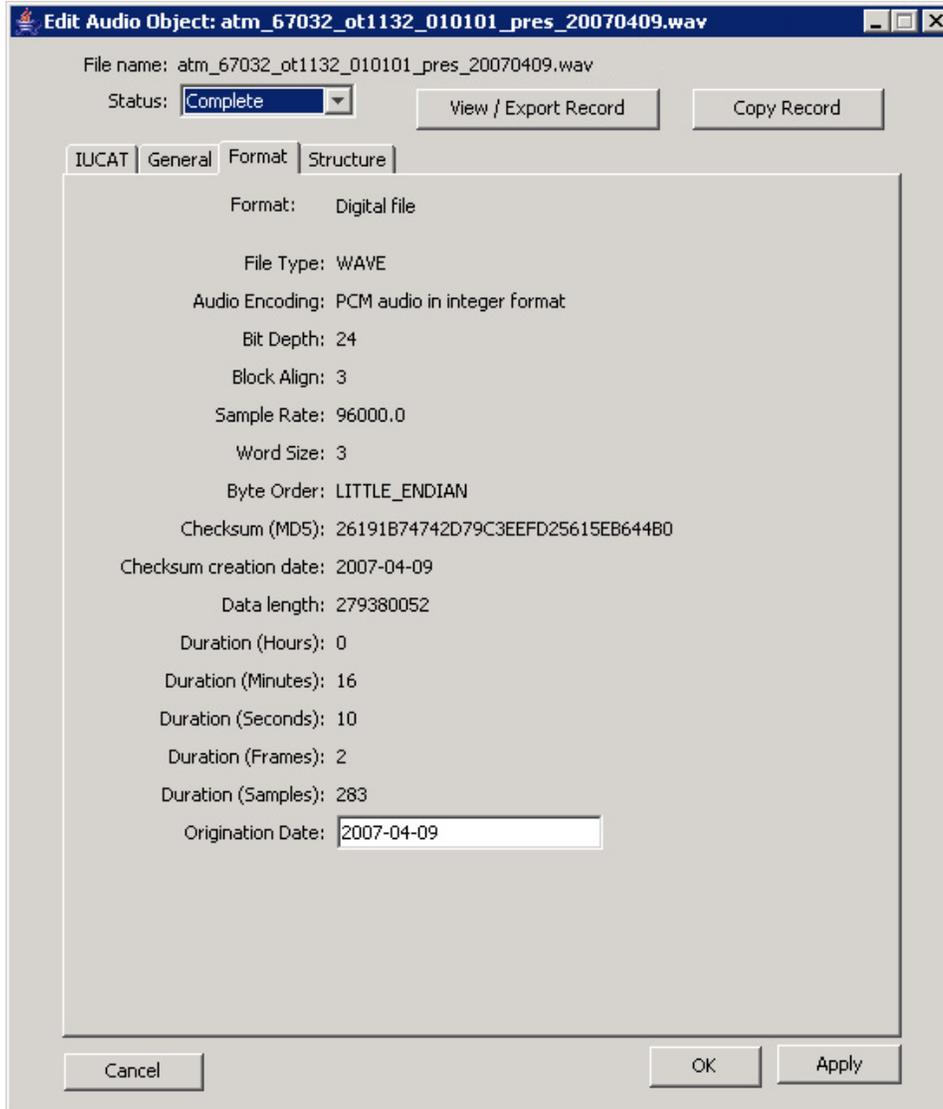
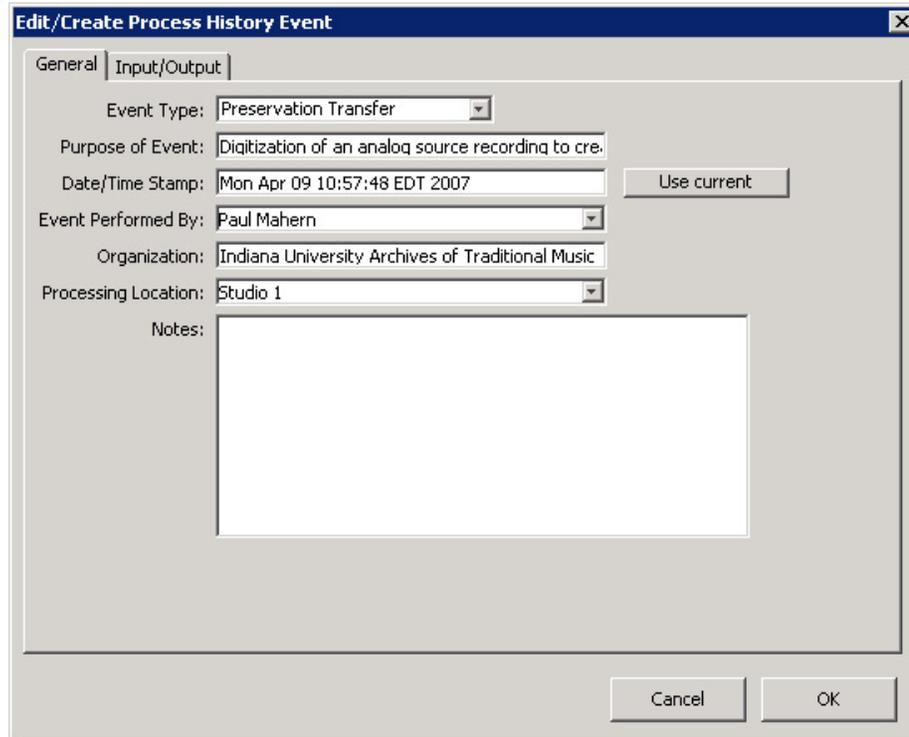


Figure 14: Digital file metadata in ATMC

4.2.5.3 Digital Provenance Metadata

In ATMC, processing history metadata is accessed from the General tab. The first step is to document the event:



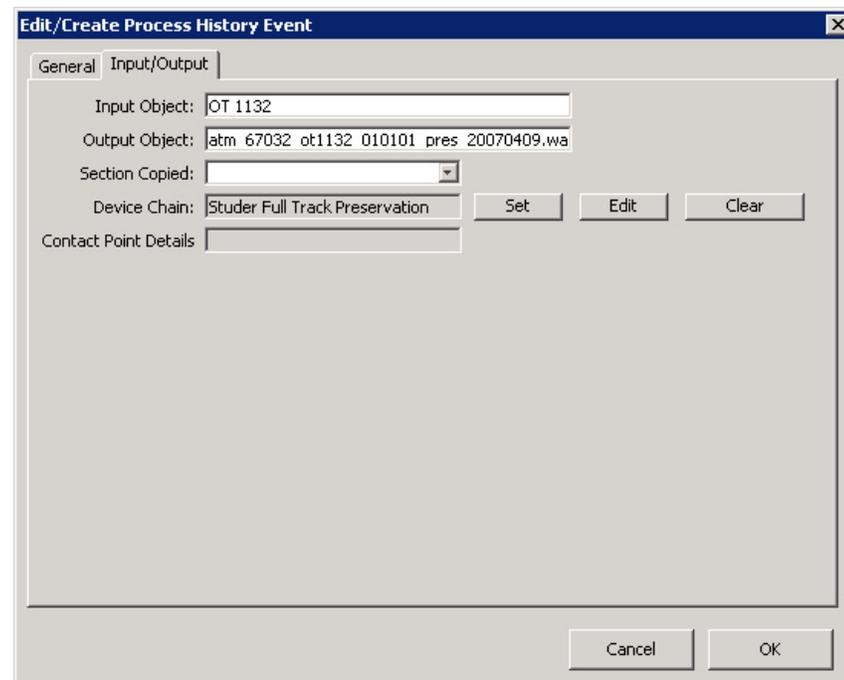
The screenshot shows the 'Edit/Create Process History Event' dialog box with the 'General' tab selected. The fields are as follows:

- Event Type: Preservation Transfer
- Purpose of Event: Digitization of an analog source recording to cre.
- Date/Time Stamp: Mon Apr 09 10:57:48 EDT 2007 (with 'Use current' button)
- Event Performed By: Paul Mahern
- Organization: Indiana University Archives of Traditional Music
- Processing Location: Studio 1
- Notes: (Large empty text area)

Buttons at the bottom: Cancel, OK.

Figure 15: Processing history event in ATMC

The next step is to designate the input and output objects for the event and attach the device or tool chain that was used. The input object for this event is an open reel tape with the shelf number OT 1132. The output object is a Preservation Master File with the local filename displayed in the appropriate field, below.



The screenshot shows the 'Edit/Create Process History Event' dialog box with the 'Input/Output' tab selected. The fields are as follows:

- Input Object: OT 1132
- Output Object: atm 67032 ot1132 010101 pres 20070409.wa
- Section Copied: (Empty dropdown)
- Device Chain: Studer Full Track Preservation (with 'Set', 'Edit', and 'Clear' buttons)
- Contact Point Details: (Empty text field)

Buttons at the bottom: Cancel, OK.

Figure 16: Processing history input and output in ATMC

All tools in the device chain can be opened and values added or edited for any parameter on the device.

Edit/Create Tool

File Location:

Nickname:

Tool Type:

Manufacturer:

Model Number:

Serial Number:

Software Version:

Settings: (4 exist)

Channels: (2 exist)

Modules: (0 exist)

Figure 17: Device documentation in ATMC

Edit Tool Settings

Name	Interface	Group	Value	Units	Transition
Speed	Analog Stepped		15	inches per second	None
EQ	Analog Stepped		NAB		None
Reproduction Level	Analog Free		4db above 250	nWb/m	None
Azimuth	Analog Free		Adjusted		None

Figure 18: Device settings in ATMC

Finally, connections between devices are documented using the virtual patchbay in ATMC:

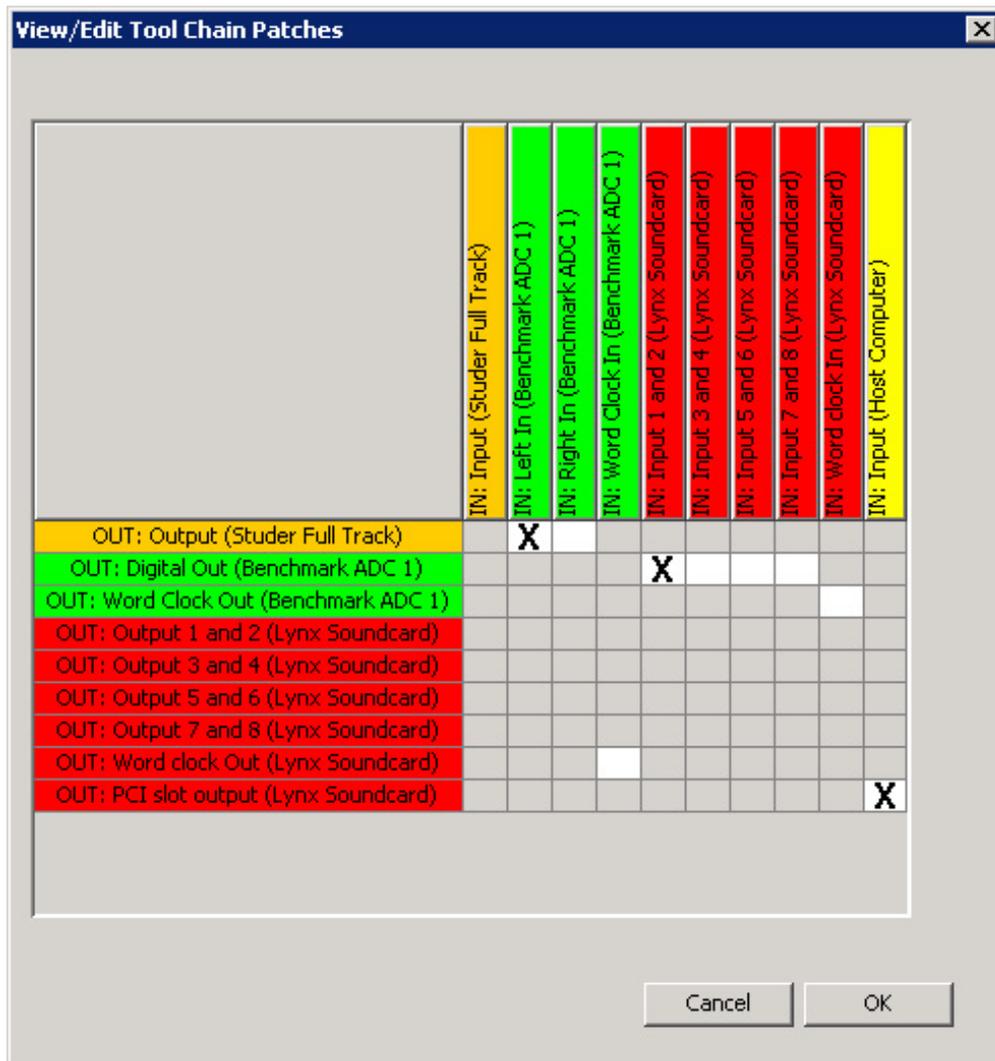


Figure 19: Virtual patchbay in ATMC

Chains of devices/tools are created in advance and simply attached to the processing history instance. Each device in a chain carries default values for each parameter and these are automatically included. The engineer need only update values that are different from the default.

4.2.5.4 Structural Metadata

IU ATM structural metadata for the source audio object and for digital files is presented below.

4.2.5.4.1 Source Audio Object

The structure inherent in the source audio object is documented in ATMC in the part of the interface devoted to it.

Here is an example from ATMC that documents an open reel tape that switched playback speeds on Face 1. Face 2 was recorded at only one speed. Data from Face 1, Region 1 is shown.

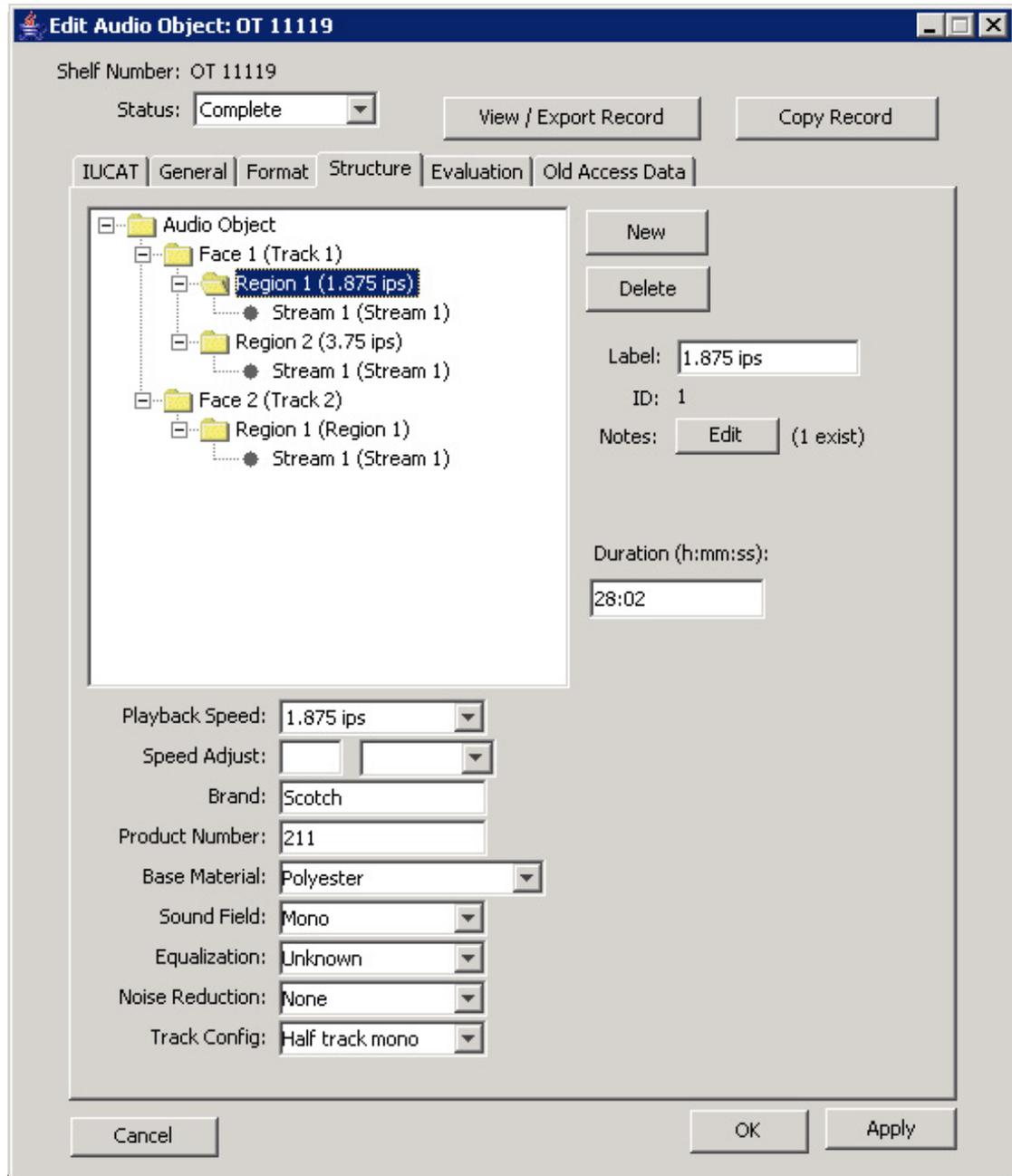


Figure 20: Audio object structure in ATMC

4.2.5.4.2 Source Audio Object and Derivative Relationships

Within ATMC it is also possible to document the relationships that exist between the source audio object and any derivatives, whether analog or digital. Relationships between objects are identified when specifying the input and output objects from a digital provenance event. That is, the event is often a transformation that results in the creation of the output object from the input.

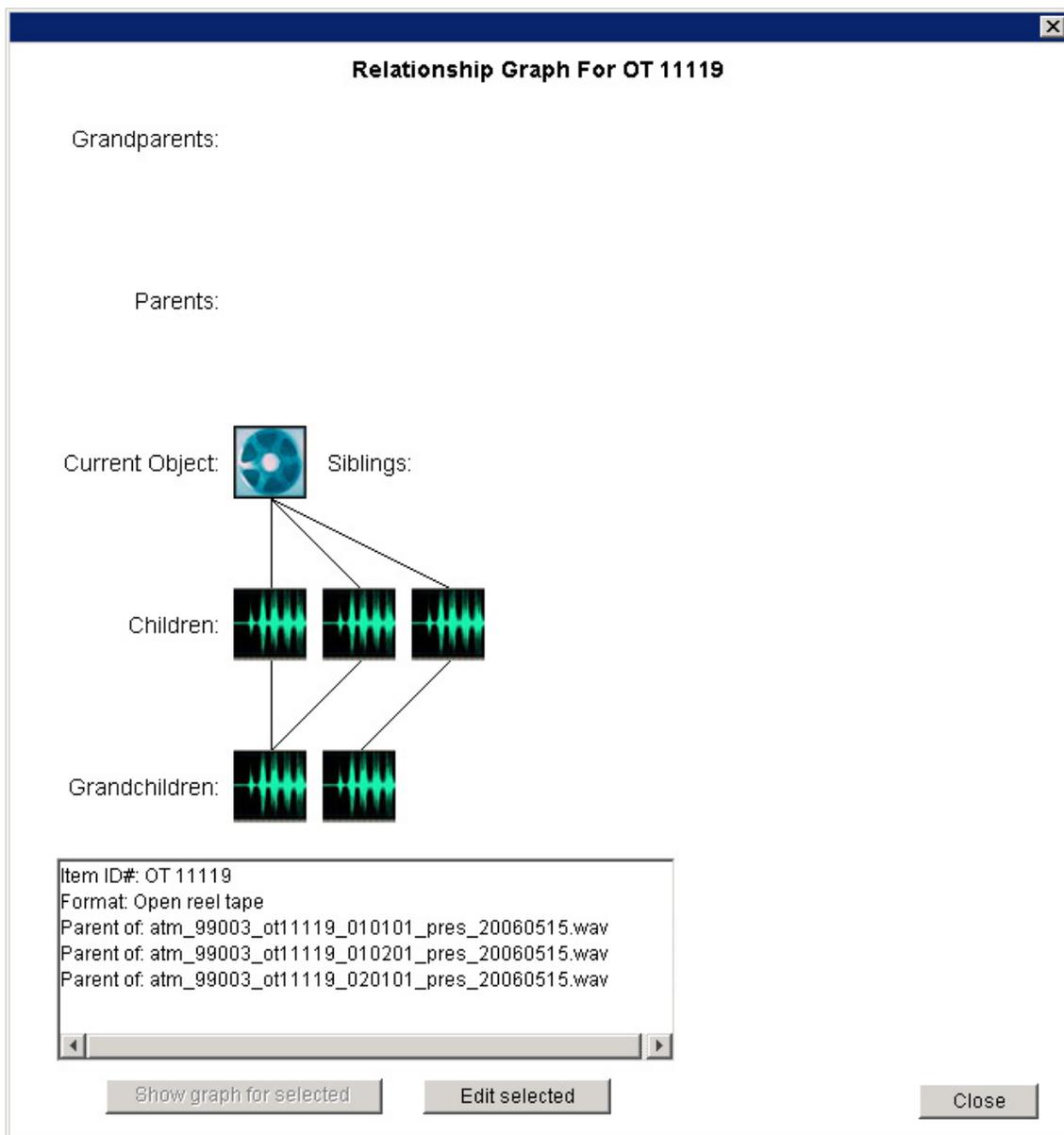


Figure 21: Relationships between audio objects in ATMC

4.2.5.4.3 Digital Files

AES31-3 ADLs (Audio Decision Lists) are used to carry edit information necessary to present a seamless listening experience to an end user without the need to permanently edit the underlying Preservation Master Files. They are also used to automatically gather the time offsets and descriptive metadata attached to markers that are placed in digital files at specific areas of interest. We produce an ADL in the following situations:

- Transfer of a source recording must be stopped and restarted, producing multiple digital files for one Face of the source audio object with overlaps of content
- Markers that bound areas of interest (performances, for example) are placed in a digital file

In the first example, the ADL edits together multiple files using edit information to model the source recording. In the second example, time offset metadata for areas of interest is carried by an ADL that points to the underlying Preservation Master File that does not contain

markers. We also produce another ADL that references the Production Master.

Because the ATM renders edited Production Master Files that carry the reference timeline, we do not rely on ADLs for the presentation of content. In a limited number of cases we would require the ADL in order to produce new Production Masters. The role of the ADL can vary depending on the number of files that are needed during preservation transfer. If only one digital file is required for one Face of a source recording, the ADL functions as the initial carrier of marker metadata only. This is true even if a Preservation Master–Intermediate File is used in which “technical” edits are made—this file type is a stand-in for the Preservation Master and establishes the reference timeline (see section 3.2.2.4.2).

If multiple digital files are necessary for one Face of a source recording, the ADL carries both edit information and marker metadata. Here are two scenarios in which this occurs:

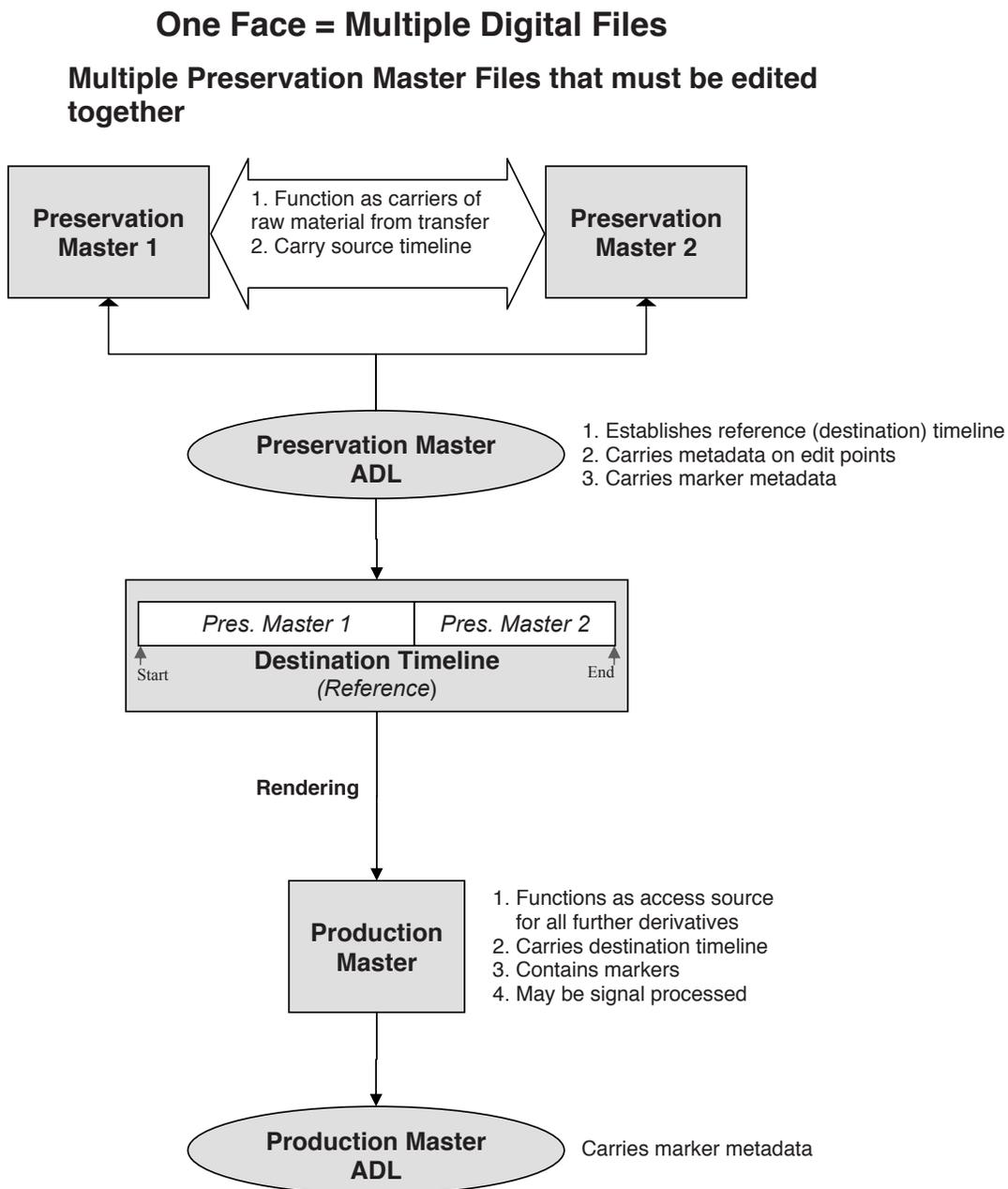


Figure 22: Illustration of workflow for editing multiple files for one Face

One Face = Multiple Digital Files

Multiple Preservation Master Files that must be edited together *and* require “technical” edits

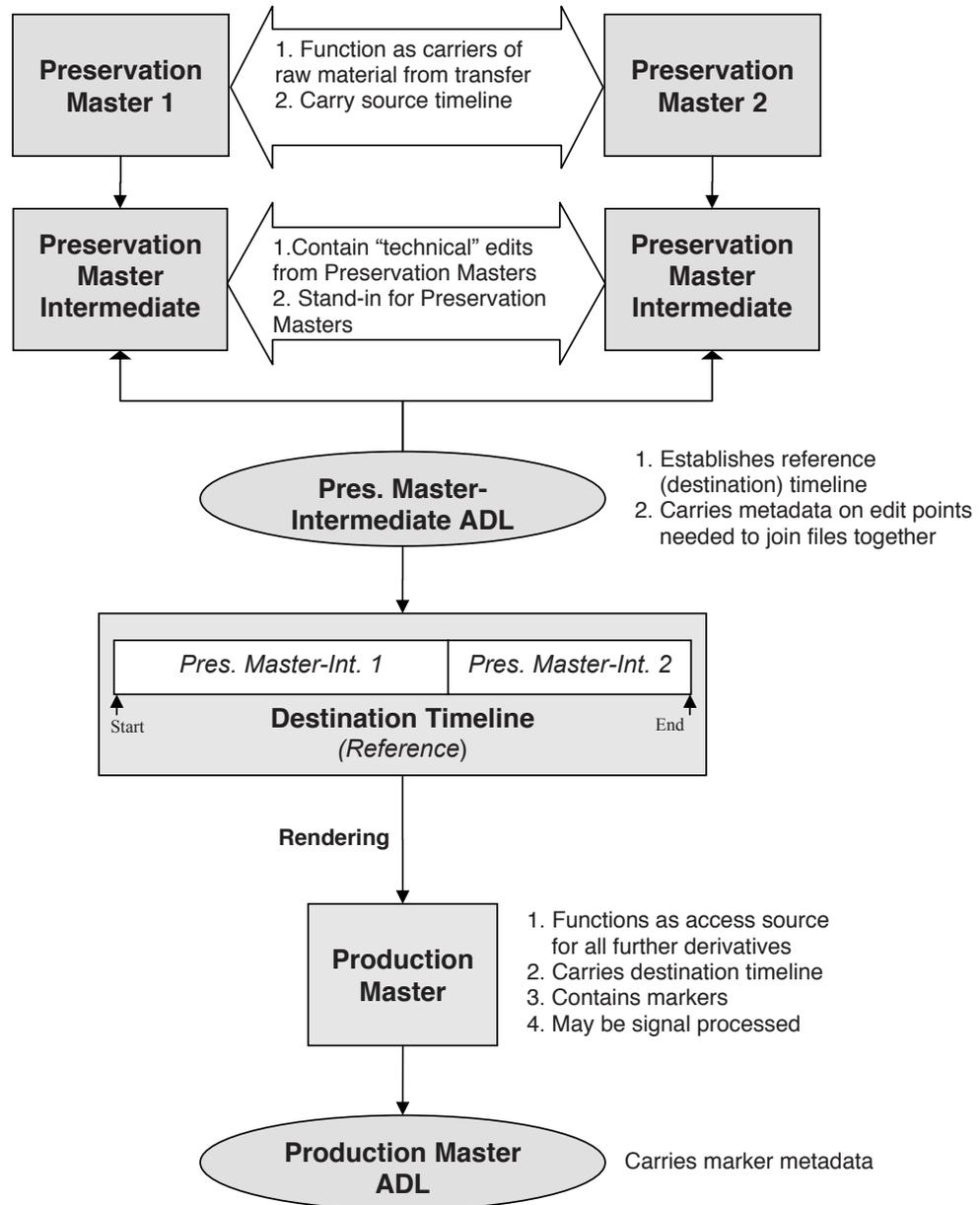


Figure 23: Illustration of workflow for “technical” edits and editing multiple files for one Face

Markers that bound areas of interest are placed in the Production Master File by the Project Assistant, working both from documentation provided by the collector and processing instructions from the ATM Archivist. The time offsets for these markers, along with the labels attached to them, are carried in an ADL and then exported to the METS document. Further descriptive metadata related to these time points is placed in an index carried in a Word document. During the next phase of the *Sound Directions* project we will investigate alternatives to recording this information in Word documents, with the goal of moving to a more machine-readable format.

5 Storage

5.1 Preservation Overview

Long-term Preservation Storage

Long-term storage has been probably the thorniest issue in audio preservation for the past fifty years if not more. Particularly in the U.S., we have argued for decades about what, if any, digital format is preservation-worthy or sufficiently robust to be worth the cost of reformatting. Historically, sound archivists have focused on the importance of the carrier—how long will it last? Can we predict when it will fail? A major paradigm shift was born around 1990 when international archivists began questioning the search for an eternal carrier.⁸³ This change in strategy, now widely accepted, focuses on preserving the content, not the carrier. Once content is digitized, this new strategy relies upon regular migration from one carrier to another in the digital domain. It therefore must take into account the need for a feasible migration path in the near- to medium-term—otherwise, resources are ill-used and retrieval costs are high. Based on a growing corpus of research, IASA-TC 04 recommends the use of digital mass storage systems for the carriers of preserved content, preferably managed storage—that is, digital storage where the characteristics and condition of files are carefully monitored—and experience internationally supports this approach. While “no target format is a permanent solution,”⁸⁴ digital mass storage offers the option of automated routines including migrating outdated formats forward in an automated fashion along with regular data-integrity checks of the contents. This greatly reduces both the opportunities for error and the resources required in the performance of these tasks.

Digital mass storage systems form essential components of preservation and access programs at the National Library of Australia, at many European national radio archives, and at archives such as the Swiss Memoriav.⁸⁵ In the U.S. and Canada, major universities such as Harvard, Indiana, and the University of California; the Library of Congress with its new National Audio-Visual Conservation Center in Culpeper, Virginia and others are building and maintaining such systems as the consensus grows that this storage approach will work for the future. An important ongoing initiative spearheaded by OCLC’s RLG Programs, the National Archives and Records Administration, and the Center for Research Libraries has worked to define the characteristics of a reliable digital repository and develop the means to certify such a repository.⁸⁶ The terms “preservation repository” and “trusted digital repository” refer to these types of storage solutions that provide preservation services beyond bit-level storage.

Both IASA-TC 03 and TC 04 outline core principles of digital storage.⁸⁷ These publications were produced for the audio preservation community but the same principles are found in various digital library publications. The recommendations in our report are based on these works and on our experience implementing preservation repositories and using digital mass storage systems.

83 Dietrich Schüller, “Preserving the Facts for the Future: Principles and Practices for the Transfer of Analog Audio Documents in the Digital Domain,” *AES Journal* 49, no. 7/8 (July/August 2001), 618.

84 IASA, Technical Committee, *IASA-TC 04*, 3.

85 The experiences and decisions of a number of institutions who have been using digital mass storage since 1992 appear in: IASA, Technical Committee, *IASA-TC 04*, 49-51.

86 Center for Research Libraries and Online Computer Library Center, “Trustworthy Repositories Audit & Certification (TRAC): Criteria and Checklist” (3 September 2007), <http://www.crl.edu/content.asp?l1=13&l2=58&l3=162&l4=91>.

87 IASA, Technical Committee, *IASA-TC 03*, 9; IASA, Technical Committee, *IASA-TC 04*, 51-52.

It is important to distinguish between digital preservation repositories and systems known as digital asset management systems (DAMS) or media asset management systems (MAMS). DAMS/MAMS tend to be commercial systems that are tailored for digital object and file management, versioning, and access in a media production environment rather than in an archival environment. For this reason, DAMS/MAMS products are more easily integrated with studio software tools such as audio editors. A DAMS/MAMS product could be used as a component of a sound archive's digital workflow and/or digital preservation infrastructure, but it likely does not provide on its own all of the features and services necessary for digital preservation, such as ongoing file integrity checking.

Local, Interim Storage

Because digital storage carriers may fail at any time without warning, planning must include safe, local, interim storage for all files with enduring preservation value. Files are typically not sent to long-term preservation storage immediately; both technologists and curators require access to files as they complete work with them and as they consult with each other. At least one backup copy of all files must be kept at all times.⁸⁸

5.2 Recommended Technical Practices

5.2.1 Local, Interim Storage

5.2.1.1 Best Practices

Best Practice 31: *Generate backup copies of all preservation files as soon as possible after creation, and preferably at the end of each complete cycle of work during which new data would normally be created.*

Best Practice 32: *Verify that backup copies are all unchanged by re-generating message digest (checksum) values before trusting that the copies are valid.*

Best Practice 33: *Store backup copies on a separate device, in a physically separate location from the original files.*

5.2.1.2 Rationale

Preservation files must be protected through the use of backup copies at all times. Ideally, these copies should be produced at the end of each complete cycle of data-creating work, particularly for files from source recordings that are actively deteriorating. It may be possible to generate copies less often for recordings that can be played again in the case of storage failures, but since the creation of redundant copies is neither labor-intensive nor costly in media, there should be very compelling reasons to even consider not making backups.

⁸⁸ IASA, Technical Committee, *IASA-TC 03*, 9.

5.2.1.3 Local, Interim Storage at the Indiana University Archives of Traditional Music

Indiana uses an entry-level NAS (Network Attached Storage) device for interim storage. This file server contains four hard drives and is RAID-5 protected, although it does not have a hot spare that can immediately take the place of a failed disk. We learned midway through the project that keeping only one instance of files on this device was not sufficient, even though RAID-5 protects against the failure of one hard drive. A series of errors by an IT support person working to recover from the failure of one hard drive resulted in the permanent loss of all data contained on the NAS. Fortunately, we had implemented a policy that backups of all files would also be stored in at least one other location—either a local workstation drive or a temporary location in the IU Massive Data Storage System—and this enabled us to recover the results of many months of work. We also experienced a more limited failure, where a project worker accidentally deleted a folder on the NAS, thinking that the deletion was occurring on a local drive. Again, we were able to restore the data from the redundant backup.

Here are the details of our interim storage solution:⁸⁹ Dell PowerVault 745N NAS purchased in late 2005, Pentium 4, 3.4 GHz, 4 GB RAM, four 200 GB hard drives, two internal 1 GB/s NIC, RAID-5 striped with parity, C: 10 GB System/Boot drive, NTFS. D: 688 GB Data drive, NTFS, Price: \$5,147.

This device is managed by CITO, the College of Arts and Sciences' IT office. Documentation of this device—how it was configured, installed, and tested—is maintained both by CITO and the ATM.

At the ATM, our audio workstations and NAS are all attached to a Gigabit Ethernet switch. Since the NAS' network connection and disks are shared by all of the workstations using the NAS, if more than one person is copying large amounts of data to the NAS at the same time, it is extremely slow. This may become a serious issue in the future when we bring a second preservation studio online and work to increase throughput within our system.

Here are some of the ways in which we use the NAS for interim storage:

- The NAS is accessible only to the Project Engineer, Project Assistant, and ATM Associate Director through a drive mapped to each person's workstation. Other staff do not have access
- All workstations with access are placed on the same switch for fastest transfer speed
- The Project Engineer uploads Preservation Master Files, ADLs, and MD5 files to the NAS at the end of each work day, then verifies the checksum
- The Project Assistant copies files from the NAS to a workstation for consultation (not updating) and QC, then verifies the checksum
- The Project Assistant uploads Production Master Files, ADLs, and MD5 files to the NAS each day, then verifies the checksum
- The NAS holds final versions of all files until ingestion into the preservation repository
- While work with a collection is ongoing, backup copies are maintained on engineer or assistant workstation hard drives or in a temporary location in the IU mass storage system in addition to the NAS
- As storage space on the NAS is filled, content must be moved to other locations. Eventually it will be ingested into the preservation repository and then deleted from the

⁸⁹ The NAS, along with computer workstations for the audio engineer and project assistant, was purchased with one-time funding from the IU University Information Technology Services.

NAS. Currently, content is moved to the IU mass storage system as a stopgap measure because the IU digital repository is not yet ready to ingest audio

5.2.1.4 Local Storage Supplementing Preservation Storage at Indiana University ATM

There is another question related to local, interim storage that we addressed at the IU Archives of Traditional Music during this project: at this point in the development of preservation repositories, should copies be retained for a longer period of time by the unit whose holdings are the target of preservation transfer? Typically, original and backup files are stored locally until content is ingested into a preservation repository, at which point they are deleted. Is it worthwhile to retain redundant copies of preserved content locally for five years, ten years, or forever? A recent article in *D-Lib Magazine* provided some food for thought:

As libraries and other institutions embark on the digital preservation process, judgment must be used to balance risk against the maturity of the process. Documents that are extremely rare or whose loss might cause considerable financial, environmental, or cultural disasters should not be entrusted to a relatively immature process. We would like to say that we will preserve our cultural heritage materials in perpetuity; however the unknown—and, furthermore, unknowable—digital landscape suggests that any such guarantee would be inadvisable at this point.⁹⁰

Digital files created from deteriorating analog sources recorded in the field will quickly become the best—in some cases, only—copy available. Few digital repositories have been tested by failure and there is little data available on dealing with the many different types of threats to digital data residing in a preservation repository. These threats include media faults, media/hardware obsolescence, software/format obsolescence, human error, loss of metadata, malicious attack, natural disaster, and failure of organizations.⁹¹

Preservation files created for content held at the Archives of Traditional Music are destined for long-term storage and access through a digital preservation repository currently under development by the Indiana University Digital Library Program (DLP). This repository will manage preserved assets from many sources at IU. Although on the same campus, the DLP is a separate administrative unit from the ATM. These two units have worked closely on a number of projects, establishing a solid, trusted, and mutually beneficial relationship in the process. Even with the new preservation repository, the ATM has decided to maintain an on-site copy of Preservation Master Files for an interim period because

- the IU digital preservation repository is still under development;
- the ATM does not yet have a service level agreement with the DLP;
- the DLP does not yet have a service level agreement with the IU Massive Data Storage System which will provide underlying long-term storage;
- the preservation repository field itself is not yet mature;
- there is little experience anywhere in planning for, and recovering from, problems;
- failure modes are not yet well understood in the field due to lack of experience.

It must be noted that the IU digital preservation repository, like others around the country, is undergoing careful development using current standards and best practices to ensure the integrity and longevity of the data it is charged with preserving. There is much experience

⁹⁰ Ronald Jantz and Michael J. Giarlo, "Architecture and Technology for Trusted Digital Repositories," *D-Lib Magazine* 11, no. 6 (June 2005), <http://www.dlib.org/dlib/june05/jantz/06jantz.html>.

⁹¹ Mary Baker, et al., "A Fresh Look at the Reliability of Long-term Digital Storage," in *EuroSys Proceedings: Proceedings of the 2006 EuroSys Conference, Leuven, Belgium, April 18-21, 2006* (New York: ACM Press, 2006), 221-34. Also available online: <http://www.lockss.org/locksswiki/files/3/30/Eurosys2006.pdf>.

protecting digital data among corporate IT professionals that can be utilized. The ATM strongly believes that a digital library repository represents the best strategy for preservation into the future. However, we feel that there is a window of time—perhaps 5-10 years—during which it is prudent to maintain under our direct control redundant copies of files. During this time period we expect the digital library community to gain significant experience in developing and managing preservation repositories so that a certified trusted digital repository is not only possible, but has been demonstrated. It must also be noted that managing local storage to supplement preservation requires a certain amount of both commitment and expertise to be successful. If these are not present, the storage effort may not be usable.

The ATM considers this type of locally controlled, interim storage as last-ditch: in the face of catastrophe, if all else fails, this one additional redundant copy of preservation files stored outside of the preservation repository might save content. For this reason we are not attempting to store all files from the preservation process, which would involve significantly more time and expense, but Preservation Masters only, from which all other types of files can be regenerated.

There are three formats often considered for temporary storage for a 5-10 year time period that can handle the large files recorded at 24/96, and that are seen as feasible by some—reasonably priced and technologically manageable—for archives like the ATM: DVD, hard drives stored offline on shelves,⁹² and data tape. All must be managed, with a planned migration necessary if this time window extends beyond 3-5 years, depending on the format. We have chosen data tape using the LTO (Linear Tape–Open) format for this purpose. LTO was introduced in 1998, developed jointly by HP, IBM, and Quantum as an “open format” technology in the sense that users have multiple sources of tape drives and tape media. This is due to intentional decisions by the developers to make the specification available to all potential manufacturers for a reasonably priced IP licensing fee. We have chosen the LTO format for the following reasons:

- Preservation problems with DVD are not yet well understood or tested, expensive testing software is necessary to manage any optical disc program, and more human intervention than we can manage is necessary to migrate from optical disc
- Hard drives may fail sooner than data tape and may be less reliable, hard drives are intended to remain powered up, there are no tests on drives that are infrequently used, and there are reports of lost data from drives sitting on shelves for too long
- LTO is the most “open” of the data tape formats and is supported by at least 30 companies and is a leader in the midrange tape drive segment of the market
- There is a clear roadmap to the future for LTO that now extends to six generations enabling informed decision-making and management. A drive can read tapes from its own generation and the two previous generations and write data to its own and the immediate prior generation in the prior generation format
- LTO supports Write Once, Read Many (WORM)
- Each LTO tape holds 400 GB, uncompressed, for LTO Generation 3

The ATM suggests, if content is destined for a preservation repository, conducting an analysis of the repository’s current stage of development as well as the experience of staff who manage it. If the above study warrants, and if it is technically and economically feasible, maintain a local, interim copy of all Preservation Master Files for a temporary period of time.

⁹² Very few people recommend this. A more robust solution would be a number of hard drives arranged in a RAID array in, for example, a network attached storage device. This is considerably more expensive.

5.2.1.5 Local, Interim Storage at Harvard

At Harvard College Library's Audio Preservation Services, all digitizing, editing, processing, and metadata creation up to the point of transfer to the Digital Repository Service (DRS) is done on twenty interim storage, logical volumes residing on our Storage Area Network (SAN), which is a four-terabyte, EMC CLARiiON storage appliance connected via Fibre Channel to the DAWs. The volumes are accessed using the SAN client software.⁹³ Half of the volumes are formatted as NTFS for direct use by the Pyramix workstations, and the other half are formatted as HFS+ for use by our Macintosh G4 and G5 where we mirror the NTFS files, using symbolic links, for testing deliverables and final creation of the deposit package. The NTFS volumes on the SAN are backed up nightly over the network to SAIT™ tape by Harvard College Library's Information Technology Services. Those backups are retrievable going back one month. The HFS+ volumes on the SAN are backed up each night, locally, using Dantz Retrospect™ to VXA™ tape.

Even though we normally move or copy files from NTFS to HFS+ (if we move them at all), we occasionally need to copy files from HFS+ to NTFS. This is accomplished through our NAS (Network Attached Storage) appliance using Macintosh File Services. The NAS is not backed up because it is very temporary storage, and used only as a bridge between file systems.

Our choice of a local workspace storage system was based upon the desire for cross-platform compatibility to accommodate the various tools used in our workflow. In pursuit of that compatibility, the SAN was originally configured for a single file-system with dual-platform read/write access. In practice, we found that the storage client software's reliance upon third party file-system translation utilities (for cross-platform read and write to a single file-system) made the storage system unreliable. Rather than jeopardize our content or delay the schedule for *Sound Directions*, we abandoned the idea of cross-platform read/write to a single file-system, and adopted a split file-system approach. WindowsXP reads and writes to NTFS only. Mac OSX reads and writes to HFS+, and can read NTFS but cannot write to it. We admit that this approach is less than ideal, but it is at least stable and reliable.

⁹³ A "client" is a piece of software that lets one negotiate access to one's storage. For instance, it manages access permissions for each user such as the ability to read and write to a volume or read-only.

5.2.2 Long-Term Preservation Storage

5.2.2.1 Best Practices

Best Practice 34: Use mirroring techniques for redundancy of online preservation and access storage, and migrate the storage environment as technology changes.

Best Practice 35: Use off-site data tape for near-line storage and tape clones with periodic media refresh.

Best Practice 36: Regenerate message digest (checksum) values periodically, and when accessing files, to verify that all copies are unchanged.

Best Practice 37: Implement systems that generate periodic reports about the condition of stored objects, and allow for ad hoc reporting of those conditions such as preservation risk factors and confidence levels.

Best Practice 38: Monitor digital audio formats, the technical environment in which they are used, and the service requirements of the user community. Look for usability threats or opportunities and implement an appropriate preservation action plan.

5.2.2.2 Rationale

Failures are a factor in all systems. A digital audio object's usability is dependent upon the reliability of the data and the systems that support that data. It is therefore vital that both the data and system integrity be monitored for failures and potential failures, and it is also vital that the systems have sufficient redundancy to sustain failures while maintaining uninterrupted service and integrity of objects. These best practices are best supported by a digital preservation repository system.

Digital audio file formats become obsolete. Software applications required for the use of those formats also become obsolete. User requirements change and may demand the richer feature sets of newer formats. It is vital that a preservation repository recognize both the threat of obsolescence and the opportunities provided by feature-rich file formats, and consult with collection owners to take appropriate action. Such actions might include either a format migration or the commitment to preserve an obsolete format and supporting application.

5.2.2.3 Long-Term Preservation Storage at Harvard

5.2.2.3.1 Background

Harvard's Digital Repository Service was developed as a vital part of the infrastructure for the Library Digital Initiative—a comprehensive program to develop the University's capacity to manage digital information. The DRS is a preservation and access repository available to any Harvard affiliate or administrative unit. The DRS is committed to preserving access to eligible, library-like content—that which supports research, scholarship and pedagogy—content that has inherent, persistent value and is intended to be stored indefinitely.

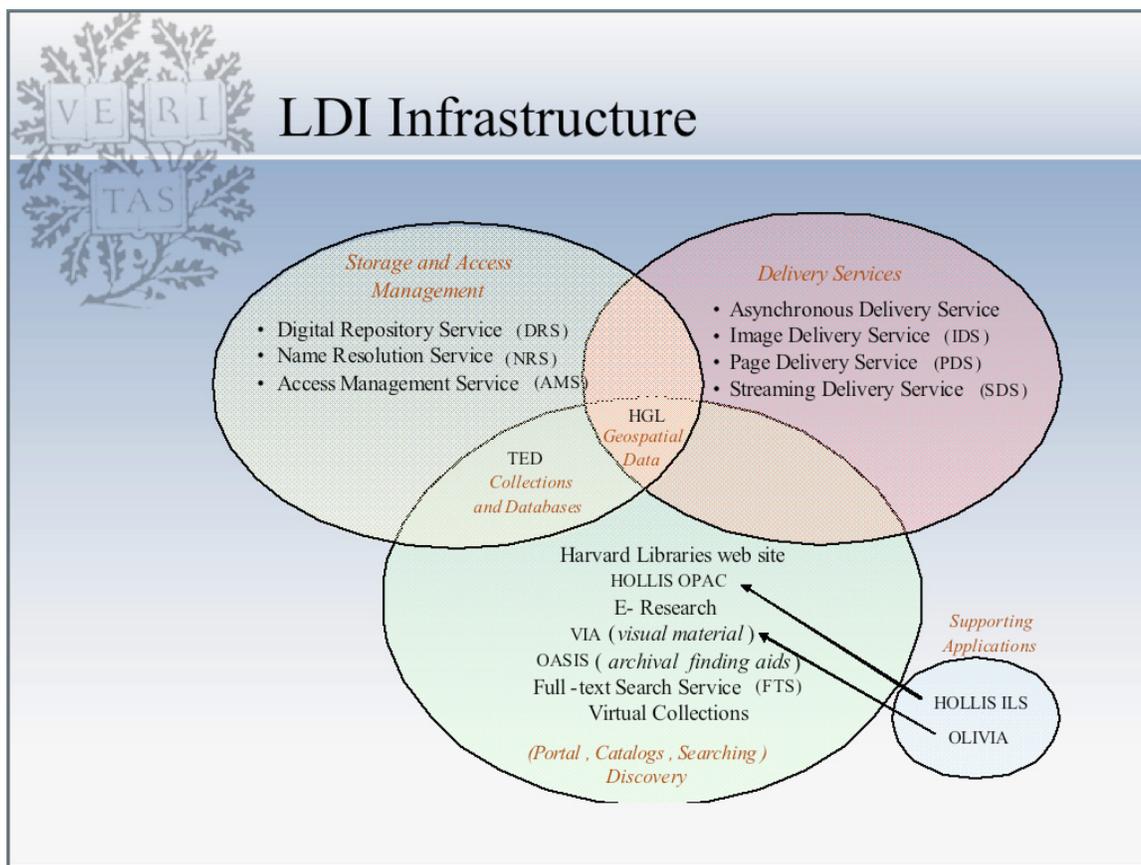


Figure 24: Library Digital Initiative at Harvard

The Digital Repository Service has been in production operation since the fourth quarter of 2000. It is used by 28 administrative units and 5 reformatting laboratories or depositing agents. It holds over 6 million objects, totaling more than 24 terabytes of data in 12 formats.

The appropriate unit of curatorial management is the object—the digital expression of an intellectual work—not the file. The digital content and its metadata comprise the object. In the DRS, the following core metadata is required for each file within the object:

- DRS object ID—a unique numeric identifier assigned upon deposit
- MD5 checksum used for data integrity upon ingest and beyond
- Insertion date
- MIME type and format type
- Owner code (FHCL.MUSI)
- Billing code for project level management (FHCL.MUS_0001)
- Owner-supplied identifier—used to describe the managed object in a curatorially-significant way (AWM_DAT_172_AM_01_01_{52A7EEB3-1ED4-4FA3-8385-C008F6F047F5})
- Access (public, Harvard-only, staff-only)

An audio preservation package in the DRS may contain Broadcast Wave Format files, RealAudio files, and SMIL files to support complex delivery of streaming media. All relationships among files are described using METS and AES31-3 ADLs.

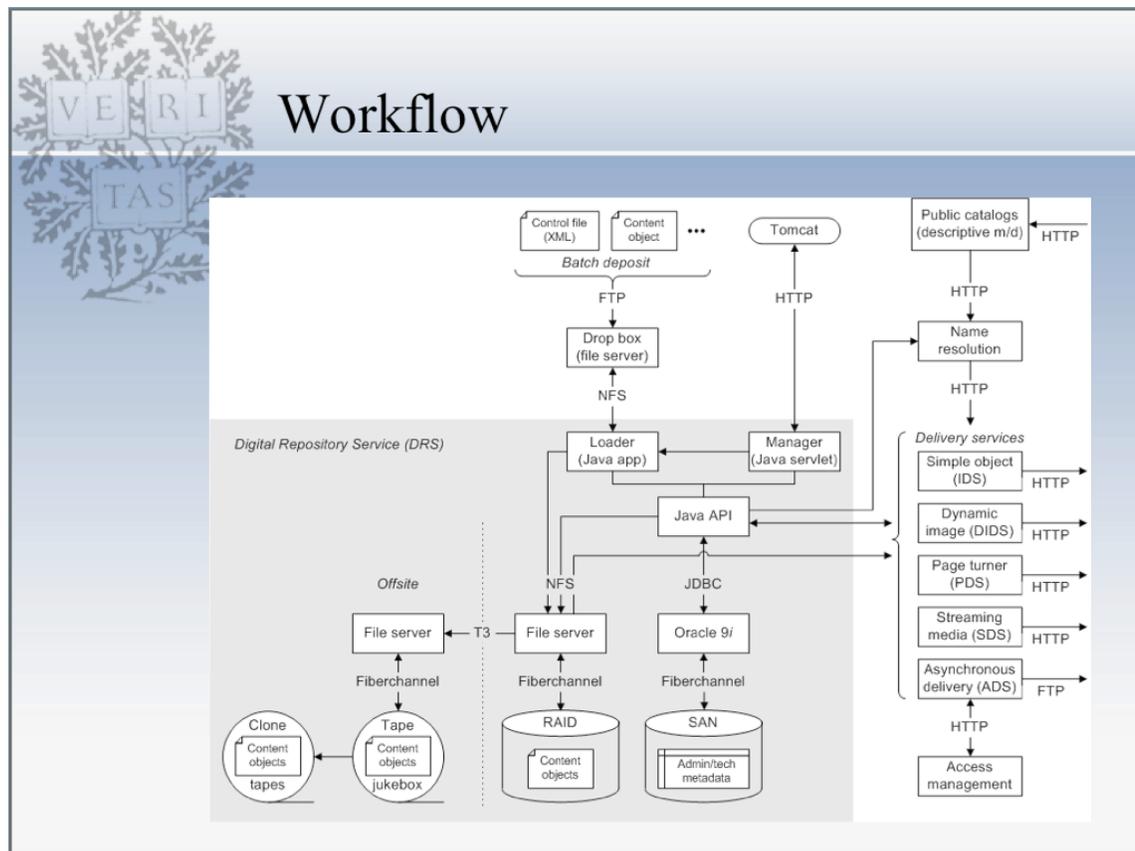


Figure 25: Digital Repository Service diagram

The DRS is tightly integrated with Streaming Delivery Service. The RealAudio server streams the data, and SMIL provides playlist capabilities. Delivery and administration services are Java web applications. The administrative metadata is structured as an Oracle 9i database, and is stored on a Fibre Channel EMC CLARiON SAN. Online content is stored on a Fibre Channel RAID appliance. All servers and services are monitored 24/7 and are housed in a University Information Systems data center. Near-line content is stored off-site in a Fibre Channel tape jukebox with off-line tape clones.

5.2.2.4 Long-Term Preservation Storage at Indiana

5.2.2.4.1 Background

The Indiana University (IU) Digital Library Program is currently in the process of designing and implementing a digital preservation repository to support the storage, preservation, and delivery of digital objects. This repository project, along with the larger Digital Library Program, is jointly funded by the IU Libraries and University Information Technology Services, and is intended to serve the digital access and preservation needs of library and archive collections from across the university. While work to date on the IU digital preservation repository has largely been focused on access needs, our goal is to develop it into a preservation repository capable of being certified as an OAIS-compliant trusted digital repository, through the emerging trustworthy repositories audit and certification process.⁹⁴

⁹⁴ The Center for Research Libraries and Online Computer Library Center, *Trustworthy Repositories Audit & Certification: Criteria and Checklist*, ver. 1.0 (Chicago, IL: CRL; Dublin, OH: OCLC, 2007). Also available online: <http://www.crl.edu/PDF/trac.pdf>.

5.2.2.4.2 Fedora

The Fedora repository system is being used as the basis for the IU digital preservation repository. Fedora (Flexible Extensible Digital Object and Repository Architecture)⁹⁵ is an open source digital object storage system developed jointly by Cornell University Information Science and the University of Virginia Library. The Fedora architecture is implemented as a set of Web services. Core repository functions are separated from utilities that act on the repository (e.g., submission tools, search systems, metadata harvesting providers, etc.), allowing external utilities to be replaced or upgraded without changes to the digital objects. Fedora is extremely flexible, both in terms of the data stored and how it may be accessed. Objects in Fedora can contain an unlimited number of “datastreams,” which may contain digital media files or metadata about those files. Datastreams may be stored in locally managed disk space or may be distributed across the Web. Data in locally-managed space is stored in a straightforward manner, making manipulation by traditional file system tools easy, including backup and restore functions. Each datastream may be associated with one or more Web services to provide “just-in-time” data transformations when users make requests.

Fedora’s flexible nature allows it to be used as the foundation for a wide variety of applications for digital libraries, archives, institutional repositories, and learning object systems. Fedora has a growing user community, with implementations at more than 30 organizations around the world, including the University of Virginia, Tufts University, the Technical University of Denmark, the University of Hull, the US National Science Digital Library, and the Australian ARROW project. The community has become active in developing tools that work with Fedora-based repositories. Some examples of applications that have been built upon Fedora include library digital collections management, multimedia authoring systems, archival repositories, institutional repositories, and digital libraries for education.

5.2.2.4.3 Massive Data Storage System

To support archival storage in the IU digital preservation repository, our intention is to make use of IU’s Massive Data Storage System (MDSS).⁹⁶ MDSS is a distributed storage service offered by University Information Technology Services to faculty, staff, and graduate students who need large scale nearline storage. The system is based on HPSS (High Performance Storage System)⁹⁷ hierarchical storage management software developed by the US Department of Energy labs and IBM. As a system based on the principle of hierarchical storage management (HSM), data transferred to MDSS initially resides on disk drives, but is quickly migrated to storage in a robotic tape library. The HPSS software manages the disks and tapes as a single logical file system which can be scaled up to store arbitrary amounts of data with minimal expense. The current system at Indiana University is able to store roughly 4.2 petabytes. Data is mirrored between the data centers at IU’s Bloomington and Indianapolis campuses, approximately 60 miles apart from each other, through a dedicated fiber optic link to provide fault tolerance and some level of disaster protection. As storage technologies evolve and new disk or tape storage systems are deployed as part of MDSS, the HPSS software is able to automatically migrate data files to newer storage media while the files remain a part of the same logical file system from the user’s perspective.

95 Fedora. <http://www.fedora.info/>.

96 Indiana University, Distributed Storage Services, “The Indiana University Massive Data Storage System Service” (January 2007), <http://storage.iu.edu/mdss.shtml>.

97 High Performance Storage System (HPSS). <http://www.hpss-collaboration.org/>.

5.2.2.4.4 Preservation Repository

To implement preservation storage in the IU digital preservation repository, we plan to configure Fedora to make use of MDSS as an underlying storage mechanism for large datastreams such as master audio files. We are currently evaluating technical options for implementing this connection. With this integration, we will be able take advantage of Fedora's abilities to manage access control, metadata, and the information about the relationships between the components of a complex digital object (metadata records, audio files, etc.) along with MDSS' large scale storage and mirroring of data across multiple geographic locations.

Beyond the features provided by Fedora and MDSS, additional services will be necessary for this to fully qualify as preservation. Specifically, we will need to implement a preservation integrity service to routinely check files that have been deposited into the repository to make sure that they can be retrieved from MDSS and match their checksums. This will ensure that objects deposited in the repository for preservation have not been intentionally or unintentionally altered or lost over time. We will also need to develop policies and operational and financial plans for the repository to ensure its long-term sustainability and ability to be certified as a trusted digital repository. This will require a concerted joint effort involving the technologists who manage the MDSS, technologists and librarians in the Digital Library Program, and librarians and archivists with preservation expertise from across IU.

6 Preservation Packages and Interchange

6.1 Preservation Overview

Simply put, if every institution's buckets of bits are not only different in character but also not understandable outside their own context, they are idiosyncratic—not interoperable—and true preservation has not occurred. Real preservation depends on the usability and readability of files over an extended period of time and by different technologists at different institutions. Should one institution fail, this type of interchange guarantees preservation by enabling any engineer to access preserved content.

Interoperable files depend upon appropriate metadata in order to ensure readability over time. In particular, descriptive metadata, and administrative technical and digital provenance metadata provide information necessary to identify digital objects, and migrate and preserve them over time. Opaque digital objects are difficult if not impossible to preserve. The development of compatible Submission Information Packages (SIPs) lays the groundwork for defining what constitutes a preservation object.

The Role of Packages in a Preservation Repository

An institution committed to preservation of digital objects over the long term must actively manage those objects in a repository designed with that purpose in mind. A preservation “package” is a representation of the data to be preserved in some sort of managed unit. The OAIS model defines an “information package” as made up of “content information” (the original target of preservation) and “preservation descriptive information” (other information needed to preserve the object, including provenance, context, reference, and fixity information), which are held together with “packaging information” (that which “is used to bind and identify the components of an Information Package”). The information package is then discoverable by external “descriptive information.”⁹⁸ The package by this definition is a conceptual one, with all relevant data logically grouped together for action by a preservation repository. It does not necessarily consist of one single file; in fact, in practice the information needed to make up an information package may be present in a number of different files that must be bound together conceptually for preservation activity.

A preservation repository may represent an information package in several forms at different times in the lifecycle of the digital object. The OAIS model defines a Submission Information Package (SIP), Archival Information Package (AIP), and a Dissemination Information Package (DIP). These packages represent the form in which information to be preserved is delivered to a preservation repository, stored by a repository, and exposed to another entity on request, respectively. The “preservation packages” exchanged between institutions as part of the *Sound Directions* project are DIPs from the originating repository's point of view, and SIPs from the receiving repository's point of view.

The exact format for an information package is not described in the OAIS model. Several XML-based schemas may be used for this purpose: the Metadata Encoding and Transmission

⁹⁸ CCSDS, *OAIS*.

Standard (METS),⁹⁹ the MPEG-21 Digital Item Declaration Language (MPEG21-DIDL),¹⁰⁰ XML Formatted Data Unit (XFDU),¹⁰¹ Instructional Management Systems Content Packaging (IMS CP),¹⁰² and Material eXchange Format (MXF)¹⁰³—just to name a few. Each implementer must determine based on local design considerations which of these best meets the repository's needs.

Within the use of a standard schema such as METS there is a further device for the facilitation of interoperability and exchange—a profile. A profile defines specific limitations and restrictions on the creation of METS documents beyond those present in the Schema. Profiles can aid in the creation of METS documents for submission to a particular institution, in the processing of documents from that institution, or to guide the creation of documents for use with a specific piece of software. Profiles are particularly useful if developed, supported, and implemented by a specific community of stakeholders with common needs. METS profiles can be made widely available by formally registering them through the METS Editorial Board with the Library of Congress.¹⁰⁴

6.2 Recommended Technical Practices

6.2.1 Preservation Packages

6.2.1.1 Best Practices

Best Practice 39: Use METS or another appropriate XML packaging schema in the creation of preservation packages. Use AES31-3 for comprehensively representing digital audio objects within those packages.

6.2.1.2 Rationale

Our goal is to ensure on-going preservation and access of digital audio objects in a library-like, research-centered manner that enables the inevitable file and format migration of content. Many different types of files, including audio, technical metadata, and structural metadata, must be used together to gain a complete set of information needed for digital preservation. At the moment, this suggests a library-oriented packaging schema such as METS to describe the package and provide rudimentary navigation of complex objects. The AES31-3 standard should be used to more comprehensively represent complex digital audio objects that have

⁹⁹ LC, "METS."

¹⁰⁰ International Standard Office, *Information Technology – Multimedia Framework (MPEG-21 – Part 2: Digital Item Declaration*, 2nd ed., ISO/IEC 21000-2:2005(E) (Geneva, Switzerland: ISO/IEC, 1 October 2005). Also available online: [http://standards.iso.org/ittf/PubliclyAvailableStandards/c041112_ISO_IEC_21000-2_2005\(E\).zip](http://standards.iso.org/ittf/PubliclyAvailableStandards/c041112_ISO_IEC_21000-2_2005(E).zip).

¹⁰¹ CSC, NASA, and International, "Getting Started with XFDU API," ver. 1.4.0 (7 July 2006), <http://sindbad.gsfc.nasa.gov/xfdu/xdoc/gettingstarted/gettingstarted.html>;

Consultative Committee for Space Data Systems, *XML Formatted Data Unit (XFDU) Structure and Construction Rules*, White Book CCSDS [no number] (Washington, DC: CCSDS Secretariat, 15 September 2004), <http://sindbad.gsfc.nasa.gov/xfdu/pdfdocs/iprwbv2a.pdf>.

¹⁰² IMS Global Learning Consortium, "Content Packaging Specifications" (3 August 2007), <http://www.imsglobal.org/content/packaging/index.html>.

¹⁰³ Bruce Devlin, "MXF – The Material eXchange Format," *EBU Technical Review*, no. 311 (July 2002), http://www.ebu.ch/en/technical/trev/trev_291-devlin.pdf.

¹⁰⁴ LC, "METS."

multiple, cross-faded overlapping files. METS has a baseline ability to handle timelines. In METS, the destination timeline is only implied, whereas AES31-3 specifically represents a destination timeline. At this time, METS is not supported by audio editing applications—and it is not likely to be, so, no audio application can read a METS <structMap>. AES31-3 comes from the audio community. It is an evolving standard that is widely supported in digital audio applications, and the AES31-3 ADL is a simple, human-readable, non-proprietary text document.

6.2.1.3 Preservation Packages at Harvard

Audio preservation packages destined for our Digital Repository Service consist of the following:

- METS file containing
 - Original Audio Object (core audio—AES-X098-B) metadata XML
 - Process History (digiprov—AES-X098-C) metadata XML
 - Preservation Master Audio Object metadata XML
 - Production Master Audio Object metadata XML
 - Service (Delivery Master) Audio Object metadata XML
 - Preservation Master AES31-3 ADL (base64-encoded) & checksum
 - Production Master AES31-3 ADL (base64-encoded) & checksum
 - Misc (files used in our workflow that in future will be stored in more specific sections of the METS)¹⁰⁵
- Preservation Master BWF files
- Production Master BWF files
- RealAudio Delivery Master files
- SMIL files

It should be noted that we do not include descriptive metadata in the DRS deposit package—except for that which is incidental to the various other metadata documents in the package. By explicit intent, the DRS does not manage descriptive metadata. Descriptive metadata for *Sound Directions* audio packages is discoverable in the Harvard University Library Online Public Access Catalog.

We chose the AES31-3 ADL encoded within the METS file as the definitive representation of the relationships of our digital audio objects' files because AES31-3 is widely supported and easily understood by humans. Even in the absence of a software application to import the ADL, all the information necessary to faithfully represent the complex time-based relationships of files within the object can be extracted by reading and understanding this simple text document. We feel that having such an accessible form of information in the package provides added assurance that the object will be faithfully represented into the future.

In order to construct the deposit package, we run the “makedeposit” tool that automates the process. “Makedeposit” prepares the data, and calls the Dmart tool. Dmart checks the configuration file for the deposit that contains vital information for the process.

¹⁰⁵ Our tools produce ADLs, BWF media and audio object metadata for those BWF files as a result of production and other processes in the preservation workflow. The ADLs and the audio object metadata for the BWF files in addition to marker files are stored in the misc directory as a convenience so that they may be referenced in the process history.

An example Dmart configuration file follows.

```
<!-- this file is used as the example configuration file when distributed -->
<configFile app="Dmart">
  <accessFlag>P</accessFlag>
  <batchSchema>http://hul.harvard.edu/ois/xml/xsd/drs/batch.dtd</batchSchema>
  <billingCode>FHCL.MUSI.MUS_0002</billingCode>
  <emailSuccess>dackerm@fas.harvard.edu, bgordon@fas.harvard.edu</emailSuccess>
  <emailFailure>bgordon@fas.harvard.edu, dackerm@fas.harvard.edu</emailFailure>
  <ownerCode>FHCL.MUSI</ownerCode>
  <service>STREAMING_DELIVERY_SERVICE</service>
  <urnMask>urn-3:fhcl.loeb:sa{n}</urnMask>

  <agentName>Harvard College Eda Kuhn Loeb Music Library</agentName>
  <status>PROD</status>

  <archivalCreator>SONI</archivalCreator>
  <deliverableCreator>PROT</deliverableCreator>
  <productionCreator>SONI</productionCreator>
  <smilCreator>XMLS</smilCreator>
  <waveformCreator>SONI</waveformCreator>

  <saxParser>org.apache.xerces.parsers.SAXParser</saxParser>
  <bufferSize>131072</bufferSize>
</configFile>
```

Dmart does the following:

- Starts constructing the METS file and the batch.xml deposit control file
- Locates the “original” directory, retrieves the original source audio object metadata coreaudio.xml document, adds that data to the <techMD> section of the METS and confirms the copy
- Locates the “archival” directory and adds a <fptr> (file pointer) for the AES31-3 ADL into the <structMap>
- Base64 encodes the AES31-3 ADL
- Adds <techMD> for the digipro.v.xml and confirms the XML copy
- Temporarily ignores the “.dbof.xml” file for the Archival Master BWF file
- Adds <fptr> for Archival Master BWF file
- Revisits the skipped “.dbof.xml” file and adds the override metadata for the BWF file from that override file
- Adds the object for the BWF file to the <fileSec>
- Adds the audio object metadata for the BWF to the <techMD> and confirms the copy of the XML
- Locates the “production” directory and repeats the previous ten step process
- Locates the “deliverable” directory and ignores the “.dbof.xml” file for the Delivery Master streaming media file
- Adds a <fptr> for the Delivery Master File to the <structMap>
- Revisits the skipped “.dbof.xml” file and adds the override metadata for the Delivery Master File from the “.dbof.xml” file
- Adds the object Delivery Master to the <fileSec>
- Adds the URN for the Delivery Master File
- Adds the audio object metadata for the Delivery Master to the <techMD>

- Confirms the XML copy
- Locates the “misc” directory
- Create FContent (wrapper) for the Archival Master audio object XML
- Base64 encode the archival “fix-pan-pathsub-hdrs” ADL
- Base64 encode the archival “fix-pan-pathsub” ADL
- Base64 encode the archival “fix-pan” ADL
- Base64 encode the archival “fix” ADL
- Base64 encode the archival “marker_xml” file
- Base64 encode the archival ADL
- Base64 encode the archival “PQ_Marks_xml” file
- Create FContent (wrapper) for Delivery Master audio object metadata XML file
- Base64 encode Deliverable ADL
- Base64 encode Production ADL
- Create FContent (wrapper) for Workspace Master files
- Base64 encode Workspace-DITHER ADL
- Base64 encode Workspace-SRC ADL
- Finish METS file
- Remove carriage return characters
- Validate METS versus xmlvalidator
- Validate METS versus Metsvalidator
- Add object for “mets.xml”
- Add URN
- Add “IS_PART_OF” relationship between the Archival Master and the “mets.xml”
- Add “IS_PART_OF” relationship between the Production Master and the “mets.xml”
- Add “IS_PART_OF” relationship between the Delivery Master and the “mets.xml”
- Report condition of “batch.xml”

Once Dmart has finished, “makedeposit” calls the “convertsmil” tool that converts the audio references in the SMIL file to use the owner supplied name.

6.2.1.4 Preservation Package Technical Practices at Indiana

Prior to the start of the *Sound Directions* project, IU did not have a preservation repository for audio from the ATM. During the project we created a workflow that made use of the current Fedora-based IU digital repository, which has not yet been integrated with MDSS or augmented with preservation integrity services. We operated under the assumption that adding these features in the future would not fundamentally change the interactions between the ATM and the repository. Our practices for the *Sound Directions* project were therefore built with the exchange needs of the project in mind, rather than necessarily reflecting our ideal local repository practices. Our workflow for this phase of the *Sound Directions* project involved building METS packages with tools located outside the repository, sending these packages to Harvard, then ingesting the same package sent for exchange into our own repository, whereas in the future we intend to first ingest material into our repository, and generate a package for exchange with another institution from the data in the repository. We extended an existing IU-developed content loading tool to accept METS-based Submission Information Packages (SIPs) created from Indiana’s or Harvard’s preservation transfer and documentation processes. This tool loads the package, including all of its component sound and other data files, into the repository, where it is stored for now as a series of Fedora-managed data streams on a local disk.

The IU repository currently keeps the item-centric view of the *Sound Directions* resources needed for the exchange with Harvard, however, as we move forward with content modeling

work for better representing field audio in our repository, it is likely we will add additional information about these objects to the repository that describes the collection as a whole and the relationship of each of its parts. This additional information should better document the archival nature of these materials.

IU preservation packages for *Sound Directions* consist of references to the following, treated as content for preservation:

- Audio files
 - Preservation Master BWF files, with checksums
 - Preservation Master–Intermediate BWF files, with checksums
 - Production Master BWF files, with checksums
- AES31-3 ADL files generated by WaveLab audio editing software for each audio file, with checksum
- ATM “index sheets” in Microsoft Word format with information designed to provide information on the contents of an item to a listener, with checksum

IU preservation packages also contained Preservation Descriptive Information according to the OAIS definition, as follows:

- Technical metadata conforming to the draft AES-X098-B standard for all digital audio files
- Technical metadata conforming to the draft AES-X098-B standard for the physical item from which the digital audio files were created
- Digital provenance metadata conforming to the draft AES-X098-C standard describing all processes performed to create and process the digital audio files
- A collection-level bibliographic record in MARCXML format, generated from the master copy stored in our University OPAC

Most of the content and metadata to be managed as part of the *Sound Directions* project represents a good match with content and metadata we have already planned for as part of our local repository. The AES31-3 ADL file, however, is designed specifically for audio content and is not a format we had previously used within our repository, nor had we to that point explicitly planned for preserving edit decision list files. This file also presents a new challenge to us in that it is not yet encoded in XML. For all of these reasons, the AES31-3 ADL file represented a new class of material for which IU had to develop a preservation management plan. The data contained in the AES31-3 ADL file is essential to understanding the relationships between digital audio files in some situations and, in the IU workflow, it is also the master source of information for markers used to divide the audio content in digital files into meaningful chunks. This marker metadata is carried in a non-standardized section of the ADL in the current version of AES31-3. Due to inconsistent implementation of the current AES31-3 standard in audio editing software, and the proprietary way in which marker metadata is currently carried, the IU *Sound Directions* staff was uncomfortable relying on this format to record this data into the future. We also did not want the repository to necessarily have to understand ADLs and had doubts about our ability to successfully migrate this type of data in the current form into next-generation standards. We therefore decided to parse the ADL to retrieve audio edit decisions and marker information, and duplicate this information elsewhere in the preservation package. Most of this data is expressed in the METS <structMap> in the IU packages, with specific data on fade characteristics represented as additional technical metadata. This duplication of information from the ADL into METS itself provided us with more options for preservation of this information, and created a representation using the METS <structMap> rather than an edit decision list that was closer to how we represent

other formats in our repository, such as video. We plan to revisit this preliminary decision as part of more robust local content modeling for field audio collections and with future versions of the AES31-3 standard, especially if a representation in XML becomes standard. A sample IU METS document may be found in Appendix 4.

While the ATM has defined three different types of master audio files necessary for their workflow and worthy of preservation, not all three exist for every item as separate files. Preservation Master Files always function as carriers of raw data from transfer, Preservation Master–Intermediate Files (when used) always establish the reference timeline, and Production Masters are always used to generate further derivatives. Preservation Master–Intermediate files are only generated when they are needed for technical editing as described in Chapter 3. When Preservation Master–Intermediate files are not generated, the IU preservation package still includes an entry for the file fulfilling that function in the preservation package, but that reference simply points to the file that fulfills this purpose—establishing the reference timeline—which is either the Preservation Master or Production Master. We feel this approach carries with it the benefit of explicitly stating which file fulfills each of these three important purposes, without relying on external knowledge of the ATM’s processing workflow to provide this information.

In the workflow for this phase of *Sound Directions*, the core of the preservation package, with audio file references, audio checksums, AES-X098-B (audio object) metadata, and AES-X098-C (process history) metadata, is generated from our metadata collection tool, ATMC. ATMC exports this data in XML, and then a local script builds the METS package from a variety of distributed parts. This script uses the ATMC-exported XML, parses the ADL to retrieve edit decision list and marker information and include it in the METS <structMap> and <techMD> sections as necessary, and references the Word index sheets with checksums. Finally, it retrieves the collection-level MARC record via Z39.50 from our OPAC, converts it to MARCXML, and combines all of these pieces together into METS.

The sample package used for our exchange was both sent to Harvard and simultaneously ingested into our local Fedora-based repository. Slight modifications were needed to the METS package as part of the repository ingestion process, including changing file references from simple references to a syntax that allows the repository to locate master files in our centrally-managed storage system. The repository then follows its usual ingestion process involving creation of appropriate persistent URLs (PURLs), building Fedora objects as needed, and population of various Fedora data streams.

6.2.2 Interchange

6.2.2.1 Background

Institutions will have different preservation priorities. If true preservation includes the interchange of content, then in order to effectively exchange data and preserve content, institutions must be able to understand data provided by other institutions. One attractive option for doing this is to define a common exchange format, explicitly stating features that must be present according to very strict validity rules, and stating which can vary among institutions. The successful exchange of preservation content is dependent upon many factors—not the least of which is the seemingly simple transfer of data. In practice, digital content and metadata reside and can be delivered in various formats on various file systems. As discovered by the National Digital Information Infrastructure and Preservation Program (NDIIPP) Archive Ingest and Handling Test, of which Harvard University’s Office for Information Systems was a participant, we sometimes cannot assume even the most rudimentary compatibility of files from donor institutions, and we should not impinge upon

an institution's ability to internally manage its content by imposing strict rules of acceptance. Therefore, it is useful to agree upon a basic model and a minimum structure for the data in order to facilitate the exchange, and we should agree upon a set of tools whose versioning is well managed, so that processes using the same tools and semantics are predictable and reproducible.¹⁰⁶

The Preservation Package in Sound Directions

The OAIS model discusses the various information packages as containing or lacking specific types of information. In a production environment, the situation is much more complex. Information as basic as the file type of the primary object of preservation may be represented in different ways by different repositories, each of which may be a reasonable representation of this information. The *Sound Directions* project set out to learn more about which features of a preservation package exchanged between institutions must be consistent, and which could vary, in order for each to preserve the other's content.

In order to learn more about the appropriate balance between a repository's local representation and a common format that may not match either repository's representation, the *Sound Directions* team decided to share preservation packages in the format most convenient for each individual repository. Each institution would then ingest the other's packages into the local repository, interpreting the packages as best they could, and evaluating how well the resulting data in the new repository supports the digital preservation process. Each repository was expected to be able to return a reasonable representation of the data from the preservation package back to the other institution. Both institutions agreed that each preservation package to be exchanged should represent one physical item held by the repository. The project, however, did not specify the exact content that should be included in a preservation package; rather, each institution included the information it deemed to be required of preservation. This approach was necessary given the major differences between the two repositories and the lack of experience of each institution in sharing data in this way. It was our goal that by performing these initial exchanges, we could better understand our institutional differences and therefore better plan for what a more neutral exchange format might look like.

Both of the *Sound Directions* institutions chose to use METS as the wrapper format for preservation packages. This was a fortuitous decision as no major conversion between wrapper formats was required; rather, each institution only had to perform more minor conversion to prepare the other's packages for ingestion into the local repository. More investigation will be needed to determine the feasibility of the transfer of preservation packages between repositories that can disseminate fundamentally different encodings of DIPs.

¹⁰⁶ Stephen Abrams, et al., "Harvard's Perspective on the Archive Ingest and Handling Test," *D-Lib Magazine* 11, no. 12 (December 2005), <http://www.dlib.org/dlib/december05/abrams/12abrams.html>.

6.2.2.2 Interchange at Indiana

At Indiana, our goal in ingesting Harvard content was to adjust the package to resemble ours as closely as possible, so that we could, at least in theory, apply the same preservation services developed for IU content on as much of the Harvard content as possible. For content parts that didn't match any type known to our repository, we will rely on faithfully preserving the bits of the original package so that we can deliver it back to Harvard at any time. We feel this hybrid approach gives us the benefit of being able to apply migration and other preservation services for content types we will already be dealing with, while committing to bit-level preservation for the rest.

The following files from the Harvard packages are ingested and fully managed by the IU digital repository:

- Audio BWF files
- AES31-3 ADL files for BWF masters
- AES-X098-B metadata
- AES-X098-C metadata
- Collection-level MODS

The following files from the Harvard packages do not match the IU repository's internal representation, and are stored for bit-level preservation only:

- All files in the "misc" section of the Harvard METS <fileSec>.
- RealAudio delivery files
- Technical metadata for RealAudio delivery files
- <techMd> sections referencing various "creators"

In order to ingest the Harvard preservation packages into the IU repository, we created a customized ingest process to accommodate differing practices between the institutions. For example, Harvard embedded base64-encoded ADL files inside the METS wrapper, while IU treated these as a file-by-reference. Our ingest process for the Harvard content therefore extracted the ADL from the METS and converted it to a by-reference entry in the <fileSec>. Similar adjustments were made for other variants in institutional practice. The work of creating a customized ingest process required programming, metadata, and audio expertise drawn from five staff members.

6.2.2.3 Interchange at Harvard

Audio preservation interchange packages destined for Indiana University consisted of the following:

- METS file containing
 - MODS XML
 - Original Audio Object (core audio—AES-X098-B) metadata XML
 - Process History (digiprov—AES-X098-C) XML
 - Preservation Master Audio Object metadata XML
 - Production Master Audio Object metadata XML
 - Service (Delivery Master) Audio Object metadata XML
 - Preservation Master AES31-3 ADL(s) (base64-encoded) & checksum
 - Production Master AES31-3 ADL(s) (base64-encoded) & checksum
 - SMIL XML

- Misc (files that in future will be stored in more specific sections of the METS)
 - Preservation Master BWF files
 - Production Master BWF files
 - RealAudio Delivery Master files

There are two differences between the contents of the deposit package we submit to the DRS and that of the contents of the DIP we send to the receiving institution. We include the MODS cataloging metadata in the interchange package for the benefit of the receiving institution—since it is not in our normal deposit packages. Secondly, the SMIL files, which are also not included in our own METS file for deposit, but deposited separately from the METS, were encoded in the METS file for the DIP destined for IU. The reason we do not include the SMIL in the METS for deposit to our own repository is that SMIL files create special relationships that cannot exist without the source file that they reference. That would create an error condition in the deposit process. To prevent that condition, we perform an ordered deposit that ensures that the RealAudio deliverables (the source files for the SMILs) are established in the DRS before the SMIL files are deposited.

Our interchange experience with Indiana University began by encoding the above-mentioned DIP into a zip file, then delivering the zipped DIP via secure-ftp to the IU drop box. IU made their DIP available to us for download as an uncompressed METS document, MS Word documents (as interim format index documents), BWF files, and AES31-3 ADLs. IU had the advantage of using Dmart (DRS METS Archive Tool) to extract the contents of the METS file of our DIP. We had no such tool with which to extract their contents. We extracted their MARC, audio object metadata and digiprov metadata from the METS by hand using a text editor. Had we not known what to expect, extraction could have been problematic.

It was necessary to adjust the file paths in IU's ADLs by hand in order to open them for verification prior to ingesting the package. This is not unusual. The file paths of ADLs always reflect the originating local file paths. This adjustment is a straightforward procedure since the AES31-3 ADL is a simple text document. This is a weakness in the common implementation of AES31-3. It is worth noting that a URI syntax is forthcoming in the next revision of the specification that could be used to avoid this issue by employing relative addressing. It is also possible to store the USID in the ADL source index and use that to verify the correct file has been located when populating the audio project from an ADL document.

We had difficulty with IU's process history (digiprov) document. Their process history was valid according to the schema, but would not open in our APXE application. This was due to some missing reciprocal references (back references) that APXE relies on to build the document. By definition, none of the reciprocal references of a set are optional. This is a discrepancy between the requirements of APXE and the current version of the schema.

When we used Dmart to create a deposit package of the IU content, their Audio Object XML document validated but Dmart failed to find a checksum for any of the BWF files. This was likely due to a bug in Dmart. As a result, Dmart created a batch XML document for the deposit with the missing checksums. We opened the Audio Object, identified the BWF, retrieved the checksums for the BWF files, edited the checksums into the batch XML and attempted to deposit the package. The deposit failed because the checksums from the Audio Object did not match the checksums of the actual BWF files. We reported the mismatch to IU, and they reported that they had mistakenly sent an old batch of checksums, rather than the more recently updated ones. According to IU, this was a result of creating preservation packages by hand using data from two separate administrative units. As mentioned previously, they did not yet have a fully automated tool for preservation package creation.

This is not an unusual problem. In the course of preservation workflows, the <bext> chunk is often edited while the audio data remains intact. This issue is the reason that we designed our scriptable preservation tools to validate the audio data separately from the entire file, and it is why we rely upon automation in building our deposit packages. It is important to note that there were two entirely separate issues relating to the checksums in question. One issue was that the Dmart tool failed to recognize a checksum most likely due to a bug. The other issue was that the checksum failed to validate because the BWF file header had been edited after the creation of the Audio Object metadata. Once we acquired the correct checksums from Indiana, we were able to hand-edit the correct checksums into the batch XML, and proceed with our deposit.

When we ingested IU's content, we did the minimal amount of work necessary to fit their content into our SIP for ingestion into the DRS. We did not conform their production methodology to ours but we did more than bit-level preservation. For example, we did not confirm that their use of timelines matched ours but we did collect all of the DRS-required techMD, and should be able to carry out full preservation on the objects. This does not mean that we should not conform their production methodology to ours. It may well be a better approach to do so. We must discover going forward if the end result is worth the increased cost.

7 Audio Preservation Systems and Workflows

This chapter explores the audio preservation systems and workflows employed at both Harvard University and Indiana University. This chapter also addresses selection for preservation and quality control, two key parts of preservation workflow that are not discussed elsewhere.

7.1 Preservation Overview

Audio Preservation Systems

An audio preservation program may be usefully conceptualized as a system using principles from General Systems Theory. A system may be defined simply as a set of interacting units or elements that form an integrated whole intended to perform some function or attain a goal. Each element ideally contributes to an efficient workflow that maximizes throughput while resulting in high-quality output of preservation-worthy digital objects. Thinking in system terms can aid in evaluating an audio preservation system's design, effectiveness, completeness, behavior, and sustainability. The OAIS reference model provides a useful and detailed examination of the functional entities of a preservation and access system operated by an archive that is charged with preserving content.¹⁰⁷

Selection for Preservation

One part of an audio preservation system that requires significant attention is the task of selecting recordings for preservation. It has been estimated that over 50 million hours of audio recordings exist worldwide; most of them are analog, many are unique, and none reside on permanent carriers.¹⁰⁸ Obsolescence—of playback machines, technical expertise, tools, and audio formats—combined with degradation of carriers represent twin evils that impede archivists' race against time to preserve important holdings.¹⁰⁹ In addition, preservation transfer work can be expensive and not easily afforded by many institutions. Within this context it is easy to understand the need for careful prioritization of preservation work. Selection often consists of an assessment of both research value and preservation condition. The first involves careful evaluation of the depth and breadth of documentation provided by an audio collection, assessing its potential value to researchers both now and in the future. The second requires an analysis of the extent of the risk borne by a collection, including the level of deterioration that is either manifest or expected based on its specific format, storage history, or current condition. Collection managers may also need to take into account political, economic, technical, donor-related, and other issues in making selection decisions. It is important to create a selection process that responds to shifts in priorities as perception of research value and knowledge of preservation condition changes over time. Key tools to aid the selection process for audio recordings include the IASA selection document available on the web, and FACET software and associated documents developed by the *Sound Directions* project.¹¹⁰

¹⁰⁷ CCSDS, *OAIS*, sec. 4 and p. 148.

¹⁰⁸ Schüller, "Preserving the Facts for the Future," 618. See also the survey of the holdings of broadcast archives in Europe completed by the Presto project: Richard Wright and Adrian Williams, *PRESTO—Preservation Technologies for European Broadcast Archives*, PRESTO-W2-BBC-001218 Archive Preservation and Exploitation Requirements (6 June 2001), <http://presto.joanneum.ac.at/Public/D2.pdf>.

¹⁰⁹ IASA, *Selection Criteria*.

¹¹⁰ IASA, *Selection Criteria*; FACET—the Field Audio Collection Evaluation Tool—will be available as open source software from the *Sound Directions* website later in 2007.

Quality Control and Quality Assurance

Quality control is fundamental to the successful operation of an audio preservation system and must permeate each part. The terms ‘quality control’ and ‘quality assurance’ are often used interchangeably, although their meanings are substantially different. One meaning of control is to make certain or to verify, whereas assurance may be defined as the act of giving confidence. Quality control (QC), then, is the techniques and activities used to ensure that a product or service adheres to a defined set of quality criteria or requirements. Quality assurance (QA) refers to the activities implemented within a system to provide confidence or evidence that a product or service under development or being produced will meet specified requirements.¹¹¹ The first is reactive, and the second is proactive. For example, the procedures and software tools used by an audio engineer to check a digital file fall into the realm of quality control, while the use of an audio engineer within a preservation system for transfer work is quality assurance.

Quality control and quality assurance are both necessary in preservation systems and too often neglected in project planning and implementation, with consequent reductions in the quality of the output. Many things can be done to provide quality assurance that preservation output is both accurate and high quality, such as defining the characteristics of preservation files, using professional equipment, employing professional personnel to play back the recordings, and verifying data integrity with checksums. It is imperative, however, to check and verify the actual output itself before certifying it as preservation-worthy and sending it to long-term preservation storage. It may be useful, conceptually, to distinguish between housekeeping QC (e.g., are all files present and named correctly?) and reproduction QC (e.g., is the quality of the audio acceptable?). Quality control is both expensive and time consuming. One vendor with a robust quality control program estimates that these activities consume 18% of his yearly budget.¹¹² Mistakes and errors are inevitable and it is important to devote sufficient resources to discovering and correcting them.

Quality control must be performed both by technical and/or project staff from either the institution or the vendor. In addition, permanent staff from the organization supplying the content for preservation must be involved in quality control, although possibly at a less technical level. Quality assurance is inherent in the decisions made by institutional management staff or vendor owners in developing preservation systems, programs, and services.

111 See American Society for Quality, “Basic Concepts: Quality Assurance and Quality Control,” <http://www.asq.org/learn-about-quality/quality-assurance-quality-control/overview/overview.html> and “Basic Concepts: Glossary,” <http://www.asq.org/glossary/q.html>.

112 George Blood, “Commentary: Quality Control in Audio Reformatting,” *ARSC Newsletter*, No. 111 (Summer 2006), 7. Also available online: <http://www.arsc-audio.org/nslr111.pdf>.

7.2 Recommended Technical Practices

7.2.1 Selection for Preservation

7.2.1.1 Best Practices

Best Practice 40: *Develop a prioritized list of recordings and/or collections for preservation treatment based on, at a minimum, an analysis of research value and preservation condition.*

7.2.1.2 Rationale

The rationale for this best practice is presented in the preservation overview section, above.

7.2.1.3 Selection at Indiana

For several years the ATM has engaged in a selection process that deeply examines both the research value and preservation condition of its field collections. This process consists of three steps:

1. Assessment of Research Value

As part of the *Sound Directions* project we are building the Cultures in Conflict Digital Archive (CCDA) which will focus on cultural practices that have been threatened or abolished as a result of conflict. Accordingly, our criteria for evaluating the research value and intellectual merit of field collections were couched in terms of conflict. Specifically, we assessed whether a collection

- documented a cultural practice or language that has been lost, endangered, or systematically oppressed as a result of conflict;
- included expressive culture that was directly related to a specific conflict or was tied to a particular event or period before, during or after conflict;
- documented traditions, practices, or languages that have changed significantly as a result of conflict;
- represented cultural practices undergoing active revival;
- provided especially deep or wide documentation of any tradition(s) related to the above criteria.

Points were awarded in each category so that a collection would receive more than five points only in exceptional circumstances. A collection's total score then placed it in one of the following categories:

- 5 points: The collection has exceptional research value as judged by the above criteria
- 4 points: Solid research value
- 3 points: Moderate research value
- 2 points: Minor research value
- 1 point: Minimal or no research value

2. Assessment of Preservation Condition

Over the past three years the ATM's Associate Director for Recording Services, Mike Casey, developed a preservation risk assessment instrument called FACET (Field Audio Collection Evaluation Tool), a point-based software tool for ranking field collections for the degree of deterioration they exhibit and the level of risk they carry. FACET currently assesses the characteristics of, preservation problems with, and modes of deterioration of the following formats: open reel tape (polyester, acetate, paper, and PVC bases), analog audio cassettes, DAT (Digital Audio Tape), wire recordings, lacquer discs, and aluminum discs. This tool helps collection managers construct a list of collections prioritized by the level of risk they represent, enabling informed selection for preservation. FACET was further developed during the *Sound Directions* project through collaboration with a number of audio preservation engineers, archivists, and collection managers. FACET includes an approximately 80-page/35-photo document that details degradation mechanisms and processes for the audio formats covered by the software.¹¹³

At the ATM, data for input into the FACET software was generated from a preservation survey of open reel tape holdings conducted in 2002-03, documentation provided by collectors, and new inspections of collections as needed.

3. Selection Process

Hundreds of ATM collections were assessed for both research value and preservation condition using the procedures described above. The research value and FACET scales carried equal weight and were then combined to provide an overall score that enabled us to generate one ranking for ATM collections that reflected an analysis of both research value and preservation condition. The Director and the ATM Archivist then reviewed and assessed the results in a painstaking process of comparing collections and considering other factors specific to our institution to make final choices.

7.2.1.4 Selection at Harvard

The responsibility for a decision to preserve a collection or an item within a collection rests ultimately with the curator who relies upon the audio engineer to assess the technical attributes of the objects being considered. The research-value-and-object-condition-based principles and criteria we use for analysis resemble those used with, and in, the FACET system. However, since Harvard's Archive of World Music is a relatively small collection compared with Indiana University's Archives of Traditional Music, those decision principles and criteria can be applied manually—rather than through the use of a formal survey tool such as FACET. It is the collaborative effort between curator and engineer that we value most in assuring successful, efficient preservation. Therefore as our systems evolve, we look for ways to facilitate that collaboration.

¹¹³ The FACET software and associated documents will be publicly released as open source software during the fall of 2007.

7.2.2 Quality Control and Quality Assurance

7.2.2.1 Best Practices

Best Practice 41: *Develop a formal, preferably written, plan for both quality control and quality assurance.*

Best Practice 42: *All digital files and all metadata produced as part of the preservation process must be subjected to quality control.*

Best Practice 43: *Permanent staff at the management level from the institution providing the target content for preservation should conduct some quality control.*

Best Practice 44: *Initiate one full round of quality control as soon as possible after beginning a new project, collection, or format, as well as after installing new equipment.*

Best Practice 45: *Match quality control tasks with appropriate personnel by the level of expertise required, and perform these tasks in the appropriate environment with the appropriate tools.*

7.2.2.2 Rationale

A well-defined quality control (QC) program is essential for guaranteeing the quality of the output of both preservation transfer and metadata collection, as discussed above. All data bound for long-term preservation storage must be checked and verified. Some of this verification is undertaken by permanent managerial staff with ultimate responsibility for the outcome of preservation work. These staff members are able to evaluate with larger issues in mind. One particularly important time for QC is at the start of something new, such as when a project is set up and key parameters are established, or any time there is a change in equipment or project specification. It is critical to discover and correct basic mistakes at the beginning before they are propagated through an entire project or collection and a significant amount of work must be re-done. Involving management staff at this stage assures that appropriate high-level decisions are made and helps emphasize for all staff the importance of the QC function. Quality control tasks are best assigned to personnel by the level of expertise that the tasks require. For example, an audio engineer may not be the best person to evaluate metadata while a content specialist may not be the best choice for deep assessment of the audio quality from a preservation transfer. In addition, quality control tasks that involve qualifying audio cannot be effectively performed by staff in a noisy, acoustically challenging environment or with equipment that cannot accurately reproduce the audio content—whereas quantifying QC tasks require a less specialized acoustic environment and rely upon measuring or analysis devices.

7.2.2.3 Background

The distinction between quality control and quality assurance is explored in the preservation overview to this chapter.

Indiana and Harvard share a number of quality assurance functions and procedures. To assure the overall quality of the preservation process we do the following:

- Use professional audio engineers for preservation transfer work
- Use professional equipment and audio engineering procedures in the preservation studio
- Use a test/calibration chain in the preservation studio that is able to verify a number of factors related to audio quality, such as the studio noise floor, for example
- Formally implement relevant standards and best practices in our work
- Maintain and use a document that details the characteristics and uses of each type of file created during the preservation process, including procedures used in creating them
- Maintain and use a document that details quality control procedures for all objects created during the preservation process
- Use checksums for all files
- Use a local (interim) storage strategy that always maintains backup copies of all files

7.2.2.4 Quality Control at Indiana

7.2.2.4.1 Background

At the ATM, the beginning of a new collection triggers a special and particularly intensive round of quality control by the Project Engineer, Project Assistant, and the ATM Associate Director for Recording Services. This round of QC is initiated after the first few recordings—usually three to five—of the collection have been transferred. One goal is to catch and correct basic, repetitive mistakes early so that they do not affect the remainder of the collection as it is transferred. Another goal is to make sure that project parameters are adequate and appropriate. Here are a few issues that have surfaced during this stage of QC:

- A mistake was found in one element of a filename in the Preservation Master Files. It was important to catch this early as renaming a file also means recreating the ADL, re-parsing the file, regenerating the checksum, and deleting and re-entering data into our metadata software
- Discovery of an odd looking and sounding waveform was traced to an incorrect setting in the software from a research project that had concluded the previous day
- Discovery of a tone produced by a pitch pipe on collection recordings generated decisions on what speed to use for playback and how to document our decision

This last issue required the involvement of permanent ATM staff, in this case the Associate Director. The role of this position in the QC program is to look not only for basic mistakes, but to address their causes when appropriate. It is also to verify that appropriate procedures are used and that correct interpretations are made, serving as a check on staff understanding of pertinent issues. Here are a few of the issues that the Associate Director has addressed after QC:

- How to document unusual events, such as sudden level jumps, in the metadata software
- How much unrecorded space/time to leave at the start of a Preservation Master or a Production Master File. How to maintain the reference timeline while addressing this issue
- What information about device settings that are likely to vary from recording to recording within a given collection should be communicated by the engineer to other project staff to facilitate checking of metadata

Quality control procedures are also necessary for assessing the performance of new pieces

of equipment while continuing to verify older units. ATM QC procedures using WaveLab's global analysis tools uncovered occasional, random errors or glitches from one brand of A/D converter that manifested as transient tics or clicks added to the audio file in one channel only. There were typically two or three errors per hour-long file, although in some files there were none. This low number, plus their transient nature, would have made discovery through listening difficult, at least initially.

ATM QC procedures are presented below by the target object. QC tasks within our workflow are presented later in this chapter.

7.2.2.4.2 Digital File QC

For each Preservation Master File, the audio engineer

- visually inspects the waveform representation of the audio in the file, scanning its entirety for anything that looks unusual such as very large peaks or no signal;
- listens to the beginning and end of each file, verifying the start and finish of content;
- checks random places in the middle, listening to around 6-10 minutes of audio in a 30-minute file including a 3-minute continuous stretch, verifying (among other things) completeness of content;
- analyzes the file using the error tab on WaveLab's global analysis tool. Threshold and sensitivity settings in the tool are manipulated to return about 100 points for analysis. The engineer searches for unusual looking things in the waveform, such as spiky and jagged samples, that do not appear as if they could be from a tape machine (for tape transfers) and also looks for events that occur in one channel only.

We do not specifically analyze for clipping as the engineer is attending each transfer and sees clips held on meters as they happen, and restarts the transfer as necessary.

The Project Assistant, using clones of Preservation Master Files

- verifies that all files that should exist according to ATMC (Audio Technical Metadata Collector software) and collection documentation do exist and have been uploaded to the ATM NAS (Network Attached Storage device);
- verifies the checksum;
- listens to a substantial portion of each file, and sometimes nearly all of it, during the process of marking areas of interest. The person in this position, after some training, has the ability to recognize obvious problems, at least, and sometimes more subtle ones;
- notes problems such as extremely low levels, distortion, or other unusual events and checks ATMC to see if these are documented;
- checks all metadata entered into the BWF <bext> chunk;
- checks the filename.

If errors are discovered a report is sent to the engineer.

The Associate Director

- performs QC for a percentage of all files created;¹¹⁴
- follows the audio engineer's procedures for visually inspecting, listening, and analyzing with software;
- checks BWF metadata;
- checks the filename.

7.2.2.4.3 Technical Metadata QC

The audio engineer (who has primary responsibility for entering technical metadata)

- communicates to the Project Assistant the data elements that are likely to vary over a collection. For example: settings on devices in the signal chain such as preamp gain or tonearm tracking force.

This facilitates QC by a non-audio engineer (Project Assistant, below) who may not be able to judge the values for these settings but can note their presence, absence, and repetition over many items.

All records for all audio objects created using our technical metadata software are checked by the Project Assistant who

- matches the data entered to what is known about the source recording and what is heard while listening;
- checks all technical metadata that has been entered for the source audio object, the digital files created from the preservation transfer, and the processing history instances that document this transfer;
- matches the number of Faces (see the discussion of audio object structure in Chapter 4) entered for the audio object to the number of files produced;
- checks that durations are entered for each Face and checks that the data for multiple Faces of one object matches when appropriate;
- checks the signal chain in the processing history instance, focusing on the data elements that are likely to vary as reported by the project engineer, looking for patterns that indicate obvious mistakes—things that are not changing within a collection that probably should and possibly have been forgotten;
- examines notes fields and the evaluation section of ATMC for problems with the source recording or the transfer indicated by the engineer. This provides information on what to expect when listening to the file.

The Associate Director

- examines 20% of all records entered into the metadata collection software;
- searches mainly for errors of interpretation or documentation;
- verifies that required technical metadata for the format and/or collection is in place.

¹¹⁴ The target is currently 20% of all files. Because Phase 1 of *Sound Directions* was an R&D project that required an enormous portion of the Associate Director's time, we are not yet sure if this is attainable under a more normal workflow that does not include R&D work. In the next project we plan to explore the use of statistical analysis tools to help us consider what percentage of files should be checked at this stage.

7.2.2.4.4 Workflow Tasks

The Project Assistant manages an item-level workflow database that matches tasks with personnel and tracks their completion.

7.2.2.4.5 Studio Signal Chain

The following tasks are undertaken to assess the devices in the studio signal chain:

- At the beginning of a new collection or at least once a month, full alignment and frequency response tests are done for the specific tape machines to be used. This data is saved and linked to the collection about to undergo transfer work
- At the start of a new collection, or at least once per month, a basic verification of the entire signal chain is done by sending test tones through the system and looking at channel balance, frequency response after the converter, and other items

7.2.2.5 Quality Control at Harvard

Our workflow affords us the opportunity to perform quality checks at many points and by different sets of ears, eyes and devices. In approximate order of our workflow we do the following:

- Use MRL calibration tapes to verify the performance of our open reel playback machine before a project is started and whenever the head stack is changed
- Use SpectraFoo audio analyzing software during transfers to visually check for aberrations in the signal
- Use the hardware-based Audio Precision audio analyzing system to track down signal chain problems when they are encountered, and to quantify signal-to-noise performance and frequency response when there is a change, when there has been maintenance performed, or when there has been an addition to the equipment in the signal chain
- Our software tools perform message digests on audio data and metadata to ensure their integrity
- Our software application AudioObjectManager will warn in the command line interface of any exceptions during original object metadata creation and will validate the metadata document against a local schema (based upon the as-yet-unreleased draft AES-X098-B)¹¹⁵ when the document is saved
- Our process history software application APXE has a quality assurance tab that records QA information such as checksums or message digests for events, and also will validate the process history against a local schema (based upon the as-yet-unreleased draft AES-X098-C) when the digiprov document is saved
- Use an XML QC viewing tool to compare and examine large amounts of metadata for errors before creation of the deposit package
- Use the process history and tracking sheets to enable any operator to take-up a project and know exactly what has been done up to that point
- Divide the workflow into transfer and archival creation, de-noising, production and deliverable creation, and overall QC and deposit so that more than one person has the opportunity to check another's work
- In the process of creating the deposit package, the Dmart (DRS METS archive tool),

¹¹⁵ Harvard validates its metadata against local copies of schemas based upon early drafts of the emerging AES standards until such time as these standards are published. At that time we will choose a date to implement the published standard for all future work, and all metadata in the DRS created prior to that date will be automatically migrated using a script.

- which is called by the “makedeposit” script, checks for any malformed items
- The DRS loader will reject deposits that do not conform to the batch control document presented for the deposit and will generate error messages that are sent via e-mail to the depositor
 - View the objects in the DRS and confirm that the data is correct
 - If necessary, download the deposited files and compare with originals

7.2.3 Audio Preservation Systems and Workflows

7.2.3.1 Best Practices

Best Practice 46: *Specify the elements in the system designed for audio preservation including the function of each.*

Best Practice 47: *Confirm the quality of the output of the audio preservation system through an assessment of the contribution made by each part.*

7.2.3.2 Rationale

Successful audio preservation requires the completion of many tasks and functions as described in this section, below. These range from planning and selection to long-term storage. Each step contributes to the quality of the final product. An audio preservation system may span units within institutions and may also include other institutions and/or commercial vendors; system elements must be fully functional across these divisions.

7.2.3.3 Audio Preservation System at Indiana

The Archives of Traditional Music built its part of the IU audio preservation system largely from scratch beginning with the *Sound Directions* project. We found it useful during various stages of development to reflect on some of the basic principles by which systems operate, particularly the influence of system parts on the system as a whole, in order to gain insight into the strengths and weaknesses of our efforts. Our focus was usually directed to the “preservation chain” or sequence of actions that are typically employed for audio preservation projects, particularly as they constitute a preservation workflow. Elements in the IU audio preservation chain, organized by the steps within the various stages of a typical preservation project, are presented in Figure 26 (next page).

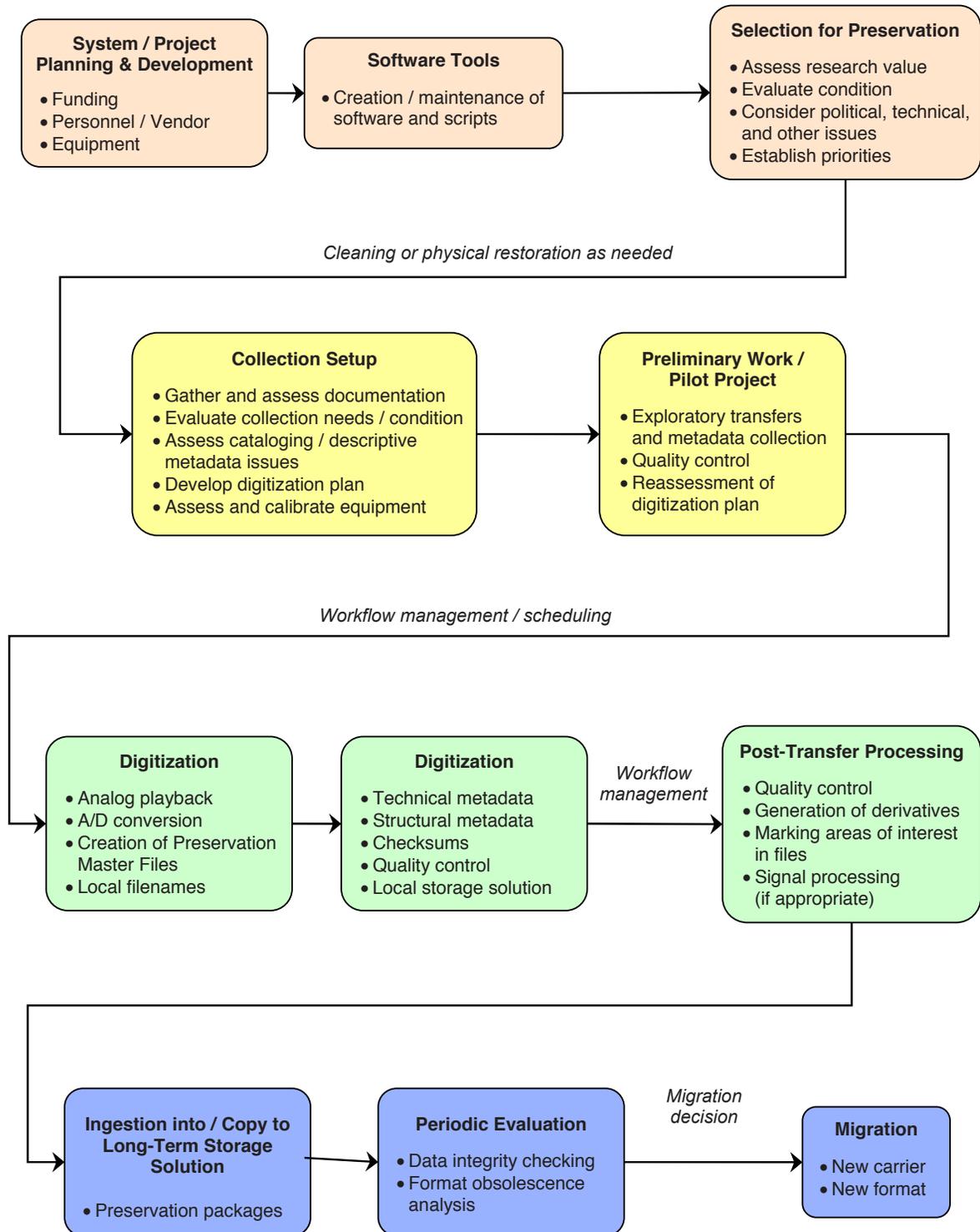


Figure 26: IU audio preservation system “preservation chain”

Below are some of the systems principles that we found especially helpful in our work.¹¹⁶

Input/Output

Systems typically require input from the outside environment to do their work. They take input and operate on it, transforming it into something different which then becomes system output. This process of transformation is necessary to attain system goals and is not at all foreign to audio engineers, computer programmers, and other preservation system technical staff.

We defined the following as necessary inputs for our audio preservation system: funding, planning, equipment, knowledge and experience of system operators, recordings selected for preservation, metadata, and output from software tools. If any of these inputs are missing, or are the result of a low-quality process, our system may degrade or produce inferior output.

Typical outputs from our audio preservation system include preservation-quality digital objects that are stable over time, metadata, digital objects for access, and new digital objects that are the result of a migration.

When input takes the form of analog recordings selected for digitization, then transformation of system input into output is easy to see. An analog recording is sent through a number of steps in our preservation workflow including analog playback, analog-to-digital conversion, and metadata generation, transforming it into a digital file enriched with metadata. Related files and their metadata are then bound together and output as a preservation package, which itself represents a further transformation.

Using these ideas, it is possible to follow any type of input into our system, analyzing the effectiveness with which it is transformed to an output that is consistent with our goals.

Whole/Part

In a system, each element or part influences the whole. It was helpful to analyze the effect that system parts have both on each other and the entire system to gain insight into the operation of our audio preservation system. Here are a few examples to illustrate:

1. If input into our system from the selection process is poor quality, that is, recordings with low research value in stable condition are selected for transfer, then all the subsequent steps in our system (see the diagram above) will be undertaken on low priority content. In the end, the output of the system will not be in priority order resulting in poor use of system resources and the loss of content as recordings degrade before they are digitized.
2. If structural metadata is not adequate, our system will not correctly output related digital files in the proper sequence for end users.
3. If analog playback is compromised, all downstream stages in our system will operate on content that is of less than optimal quality and output will be limited in fidelity, forever. A malfunctioning analog-to-digital converter, which we struggled with early in the project, would yield a similar result. In this situation we were forced to examine how a faulty system element (providing compromised input to downstream stages) affected other parts of the system, namely post-transfer processing and quality control which required additional resources to cope with the problem.

¹¹⁶ These principles are drawn from: Lars Skyttner, *General Systems Theory: An Introduction* (Basingstoke: MacMillan Press, 1996). See page 33 for example.

There are several implications for audio preservation system design and analysis that can be derived from these examples. In general, to use two clichés: if you put garbage in you will get garbage out, and the system is only as strong as its weakest component. More specifically, quality control for each element of the system is necessary to assure that each part contributes optimally to the whole. And, an analysis of what system parts have the greatest impact on the output of the system can be valuable for the allocation of resources. In other words, if funds are limited, where should they be concentrated for maximum benefit to the final product? Are they so limited that preservation-worthy products are not possible? Analysis of questions like these may affect system design, the allocation of resources, and the decision on whether to use vendors for all or part of preservation activity.¹¹⁷

The flip side of this principle is that the system has holistic properties not manifested by any of its parts and the whole may be greater than the sum of its parts. Something distinctive emerges when parts interact. The emergent product may have properties or qualities that do not exist within, or are unique compared to, any one part or partial combination of parts. What emerges from our audio preservation system is a preservation package containing prioritized content residing in a robust storage system, representing the strongest prospect for long-term survival of the content. This stored package, including its on-going value as the best carrier for prioritized content, is not inherent in, and cannot be produced from, any one system element or any partial combination of elements; every individual part of our audio preservation system is needed to produce this specific result.

Division of Labor

In a complex system, specialized units perform specialized functions. Audio preservation systems in the digital age necessarily require greater specialization, and more collaboration among specialists, than their earlier analog counterparts and ours is no exception. An effective preservation system in the analog domain—back when open reel tape served as the target preservation format—might involve only an archivist, an audio engineer, and a few others. The current IU audio preservation system requires the contributions of archivists, librarians, audio engineers, digital librarians, IT specialists, computer programmers, intellectual property experts, content specialists, and others who work collaboratively across disciplines and administrative units at IU. This is a relatively more complex audio preservation system—in the sense that it relies on a greater number of specialists—than existed at IU in the 1980's and 90's. It is also true, however, that the current IU audio preservation system is designed to provide greater functionality and efficiencies than the older analog system. The level of functionality that the OAS model demands, for example, was never realized during the days of analog preservation. Consequently, we are reaping additional benefits from the type of system that is necessary in the digital domain.

Regulation

Elements of a system must be regulated so that system goals are realized. This involves feedback and the detection and correction of deviations. Regulation is achieved in any system, including those built for audio preservation, through quality control and quality assurance. We have learned that these must be part of nearly every system element. Our approach to QC/QA is still evolving, but we have attempted to build these functions into much of our system.

¹¹⁷ We think that, as discussed in earlier sections, in terms of *audio fidelity*, expert staff, analog playback equipment and expertise, and analog-to-digital conversion probably have the greatest impact on the final product. Of course, all steps along the preservation chain are important and many can impact fidelity. In addition, other stages such as appropriate storage and metadata are obviously critical to the long-term viability of the content.

Degradation

All systems age, degrade, and eventually die following the principles of entropy. Systems may, however, import energy from the environment to counter this tendency towards disorder and degradation. We were able to identify a number of ways in which our audio preservation system may deteriorate including aging and obsolescence of equipment, loss of personnel, loss of knowledge at the ATM or DLP, loss of funding, failure to update techniques and procedures, and others. In addition, preserved content in storage must be actively managed to prevent its death through obsolescence, loss of integrity, or loss of understandability. Even a system that receives regular resources will eventually become outmoded and need to be either retired or restructured to remain alive. A preservation system that is created in 2007 is not likely to remain viable in 2017 without substantial refreshment and/or restructuring. Our conclusion is that establishing our audio preservation system is only the beginning, and maintaining it is not a one-time grant-funded endeavor. Rather, our audio preservation system must actively interact with its larger environment—the university, the audio preservation and digital storage professions, and funding agencies—over time to remain operable, adjusting to environmental pressures and seizing opportunities in order to survive.

Redundancy

In a system, small increases in redundancy may yield substantial increases in reliability. Redundancy is obviously critical in the interim and long-term storage subsystems, where multiple copies of files designated for preservation must be kept at all times. It also became clear that redundancy is a feature of quality control procedures in which any given item may be checked multiple times. We realized that because audio preservation systems are charged with a formidable task—the long-term preservation of content—a certain amount of redundancy is not only acceptable, but required, and we built this into our system.

Equifinality

Workflow and procedures used by the Harvard University and Indiana University audio preservation systems are quite different in many areas. In contemplating this we concluded that the principle of equifinality was in operation—open systems may have equally valid alternative ways of attaining the same objectives. Our two systems produce outputs of equivalent quality as discussed throughout this document.

Subsystems

We found it tempting to consider the preservation transfer and long-term storage parts of our preservation operation as separate systems as they reside in different administrative units at IU. However, we felt that it was critical for the health and longevity of the preserved content to evaluate them together as the optimal functioning of all of their components is essential for audio preservation to be realized.

System Function

Systems are designed to perform functions or attain goals through the interaction of their parts. With the above principles in mind, we defined the primary function of our audio preservation system as the following:

Select content for preservation. Preserve target content by producing, in priority order and as optimally as possible, preservation-quality digital objects. Store, provide access to, and verify the integrity of these objects over a long period of time.

The diagram below presents a high-level view of the audio preservation system at Indiana University, organized by unit and staff.



Figure 27: Indiana University audio preservation system by unit/staff responsibilities

The IU audio preservation system consists of both permanent staff (blue) and grant-funded staff (purple). We have learned that ATM permanent staff members must be actively involved in a number of functions including planning, selection, management, and quality control for our operation to be successful. We are also now reliant on grant-funded staff and this poses sustainability issues that must be addressed if ATM collections are to be preserved into the

future. Currently, without grant-funded staff our preservation system will cease functioning. Services provided by Digital Library Program staff (red) are under development for the entire university, not just the ATM.

7.2.3.4 Workflow at Indiana

7.2.3.4.1 Introduction

Workflow within the audio preservation system at IU is characterized by several features worth noting at the outset:

1. Division of Labor

Tasks are assigned to personnel according to training and skill. For example, all activities that require critical listening abilities or deep audio technical skill are handled by the Project Engineer, not the Project Assistant. The Assistant, in turn, is charged with tasks such as creating indexes for preserved content, handling other descriptive metadata, managing workflow, and completing much of the quality control work. These are areas in which audio engineers typically do not have training.¹¹⁸ In addition, the design of preservation packages, management of long-term storage, and other services related to the preservation repository are administered by the IU Digital Library Program, who have primary expertise in these areas. The ATM Associate Director for Recording Services participates in preservation repository development at a high level, representing a unit that generates a significant amount of content for preservation.

2. Automation

The IU audio preservation system is largely not automated at this point in time. Most tasks are completed by hand, with a few exceptions. We developed our system largely from scratch—we did not even have an existing manual system—beginning with the *Sound Directions* project and this represents our current stage of development. While we believe it is useful to prove that solid preservation work can be done in a non-automated environment, as we move forward we expect to increasingly automate quantitative tasks that do not require specific human skill and intervention and that machines can do better. We will especially examine tasks undertaken by the Project Assistant position that may be well suited to automation using scripts.

3. Cost

Not counting our long-term storage subsystem, which is provided by Indiana University, our audio preservation system, particularly at the ATM, was created to fit within the financial limitations of a large grant project.

4. Workflow Database

An Access database, managed by the Project Assistant, helps track the progress of preservation work for each individual recording. This database was designed to track points at which it is possible to stop the workflow and resume later, tasks that are likely to require a reminder to ensure completion, and actions for which we want confirmation of completion. Tasks that

¹¹⁸ We do some cross-training, so that the audio engineer has worked on collection documentation and labeling, for example, and the assistant has learned about preservation studio signal flow and analog playback, and has completed some critical listening exercises. This helps each to understand the other's primary responsibilities. The goal is to foster good communication and increase consistency.

are always done as part of a tight sequence of actions—performed together and unlikely to be forgotten—are not necessarily documented in the database. Rather, these actions are detailed in a workflow document.

The workflow database documents the completion of restoration, transfer, quality control, and post-transfer processing for each recording. It provides reports listing unfinished actions for every staff member, items that are in-progress by collection, and unresolved problems and issues.

7.2.3.4.2 Workflow Stages at IU

System and Project Planning and Development

The ATM Director, working in collaboration with other permanent staff, has primary responsibility for project development and planning, including obtaining funding for preservation work.

Selection for Preservation

As described above, the ATM Director, Archivist, and Associate Director for Recording Services collaborate on selection for preservation.

Software Tools

Software, including scripts, to aid preservation work is created and maintained by the Project Programmer. This position is situated at the Digital Library Program but is supervised jointly by the DLP and the ATM Associate Director. The work of the programmer interacts with all stages of the workflow and is located here for convenience.

Collection Setup/Preliminary Transfers and Metadata Entry

This stage features specific planning for the collection about to be digitized along with the first exploratory transfers and metadata collection.

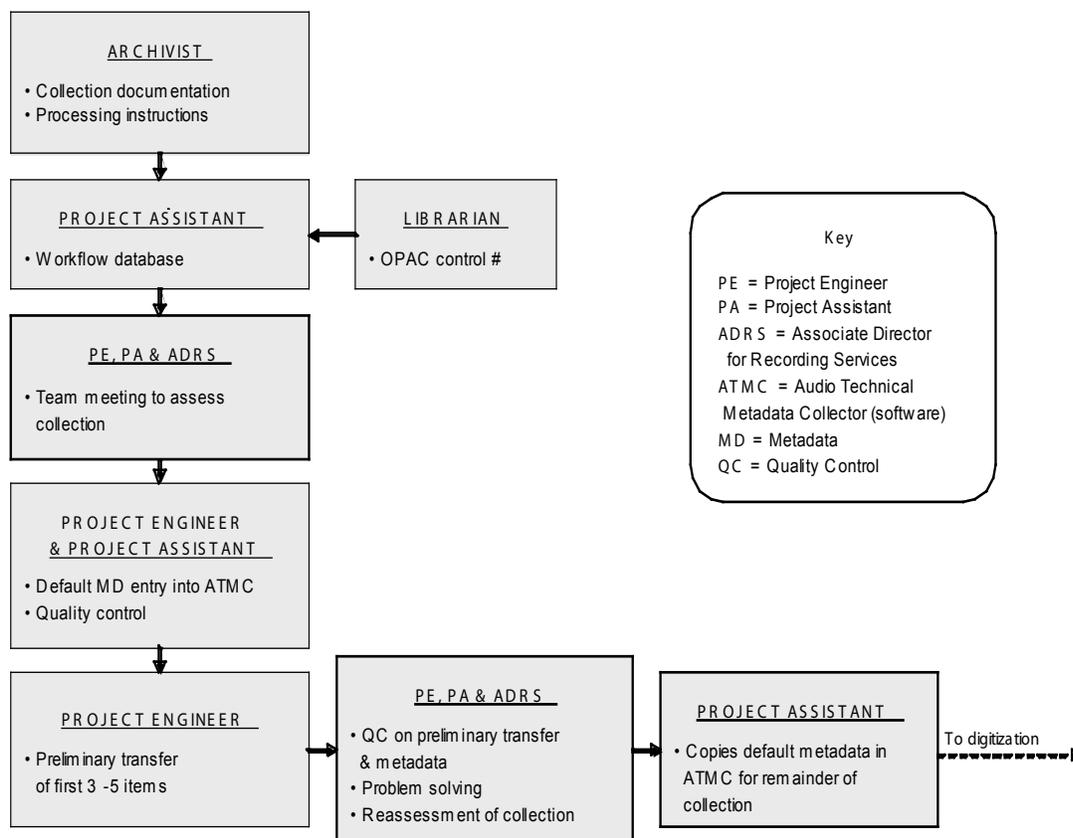


Figure 28: Workflow: collection setup stage

This is a particularly significant workflow step designed to ensure that a collection is transferred according to specification and that errors are discovered and corrected before they propagate through the entire collection. It features these steps:

1. Assessment

The ATM Archivist assesses collection documentation and forwards it to the Project Assistant with processing instructions based on what is known about the collection. A team meeting with the Project Engineer (PE), Project Assistant (PA), and ATM Associate Director (ADR S) explores technical issues or potential/known problems with the audio formats in the collection. At this point a preliminary digitization plan is developed.

2. Preliminary Transfer

The PA enters metadata into ATMC (Audio Technical Metadata Collector software) that we expect to remain consistent throughout the collection. This will become default metadata that will be copied for each recording slated for digitization, eliminating the need to re-enter or copy/paste this data field by field for each new record. The Project Engineer digitizes the first 3-5 recordings in the collection and collects metadata. These transfers are considered exploratory until this workflow stage is successfully completed.

3. Quality Control and Reassessment

An exceptionally intense round of QC by the PE, PA, and ADRS follows that considers the quality of the transfers and the accuracy of the metadata. Problems are addressed, and the digitization plan is altered as necessary.

4. Metadata Creation

The PA copies the approved default metadata, creating a template for each recording that will be digitized. This template is then updated as necessary by the PE and PA during digitization and post-transfer processing.

Cleaning and Physical Restoration

Cleaning of discs is usually handled by ATM Graduate Assistants, trained and supervised by the Project Engineer. Physical restoration—mitigating problems such as tape curling and windowing, for example—is undertaken by the PE.

Digitization (Preservation Transfer)

The Project Engineer is responsible for analog playback, analog-to-digital conversion, creating Preservation Master Files, collecting technical metadata, and entering metadata into the Broadcast Wave file.

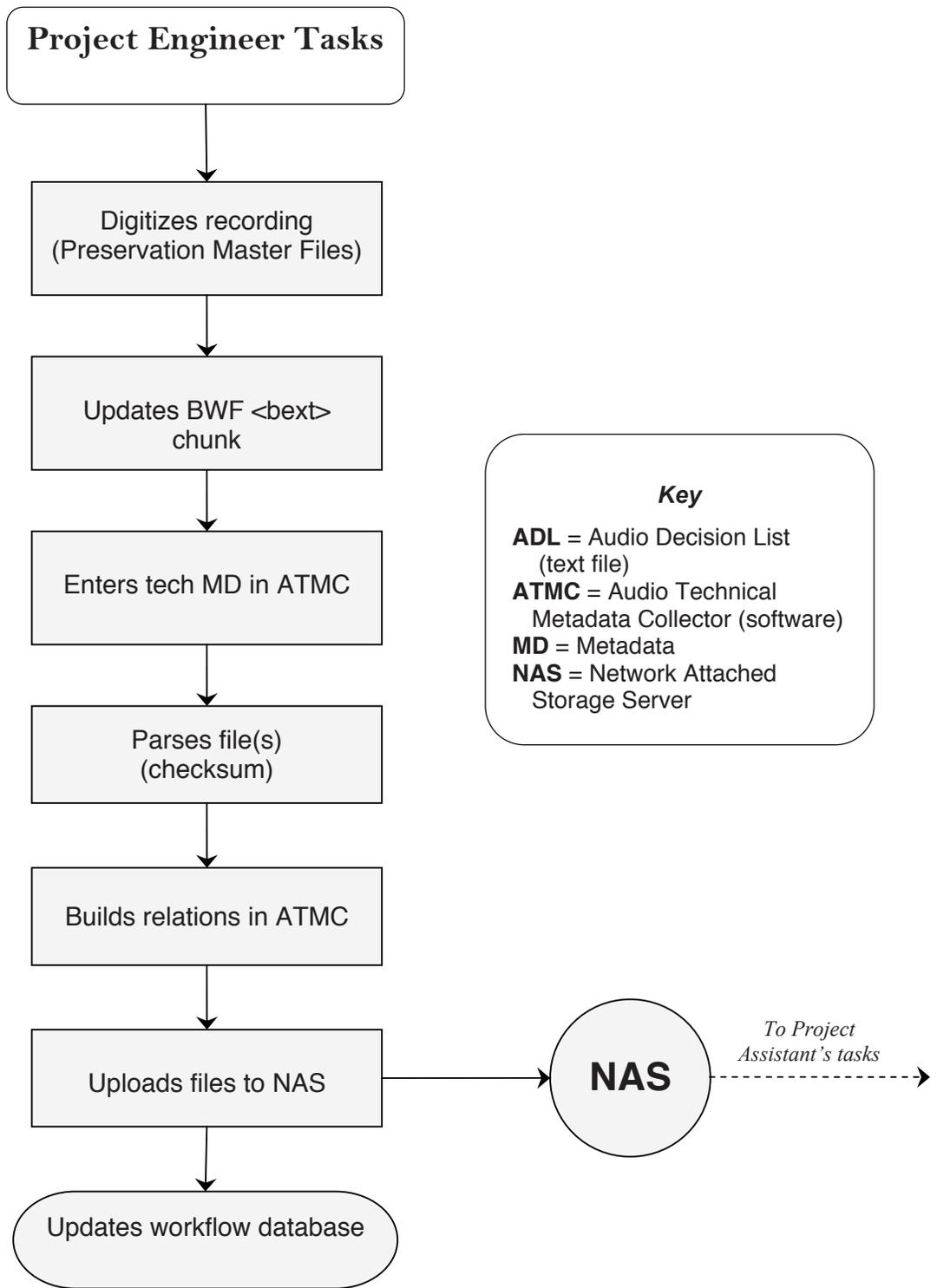


Figure 29: Workflow: Project Engineer

Here are the steps taken by the Project Engineer during this stage of the workflow for transfers of analog tape formats:

1. Set up and align the playback machine
2. Assess the performance of the studio signal chain
3. Preview recording
 - Review collection documentation for this recording
 - Check track configuration using a magnetic viewer
 - Measure the thickness of the tape using a dial caliper
 - Check sound field
 - Inspect for potential preservation problems
 - While fast winding, check the levels in several places on the tape
4. Check settings on the A/D converter, sound card and in WaveLab
5. Adjust azimuth and set levels
6. Begin transfer
7. Enter technical metadata into ATMC during playback. Almost all technical metadata can be entered while the tape is playing
8. Perform QC on the file as described in this section above
9. Update default metadata in the Broadcast Wave <bext> chunk. Most of this metadata is carried from the previous file by WaveLab. Usually only the shelf number and filename must be updated. At the beginning of each day the date must be updated.
10. Save file
11. Edit/update processing history metadata for the transfer event. This includes creating relationships in ATMC by specifying input and output for the event. Most processing history metadata remains unchanged from the previous record; usually, only settings on the playback machine must be updated for tape transfers and settings at the turntable (stylus size, tracking force, anti-skate) for discs
12. Parse the file using ATMC to create a record of the file in the database. Update the collection accession number and file use in the ATMC record. While parsing, ATMC generates a checksum for the file
13. Copy file to the NAS. This is done by batch over lunch or at the end of the day
14. Verify the checksum after transmission to the NAS
15. Update workflow database

This workflow is basically the same for disc transfers, except that a microscope is used as a diagnostic tool and there is a specific set of procedures for turntable setup, problem identification, and stylus selection. Because the playback time of discs is short compared to tapes, and because tracking problems often require engineer intervention, it is usually not possible to enter technical metadata during transfer.

Post-transfer Processing

The Project Assistant is responsible for many quality control tasks, creation of Production Master Files, and marking areas of interest in each file.

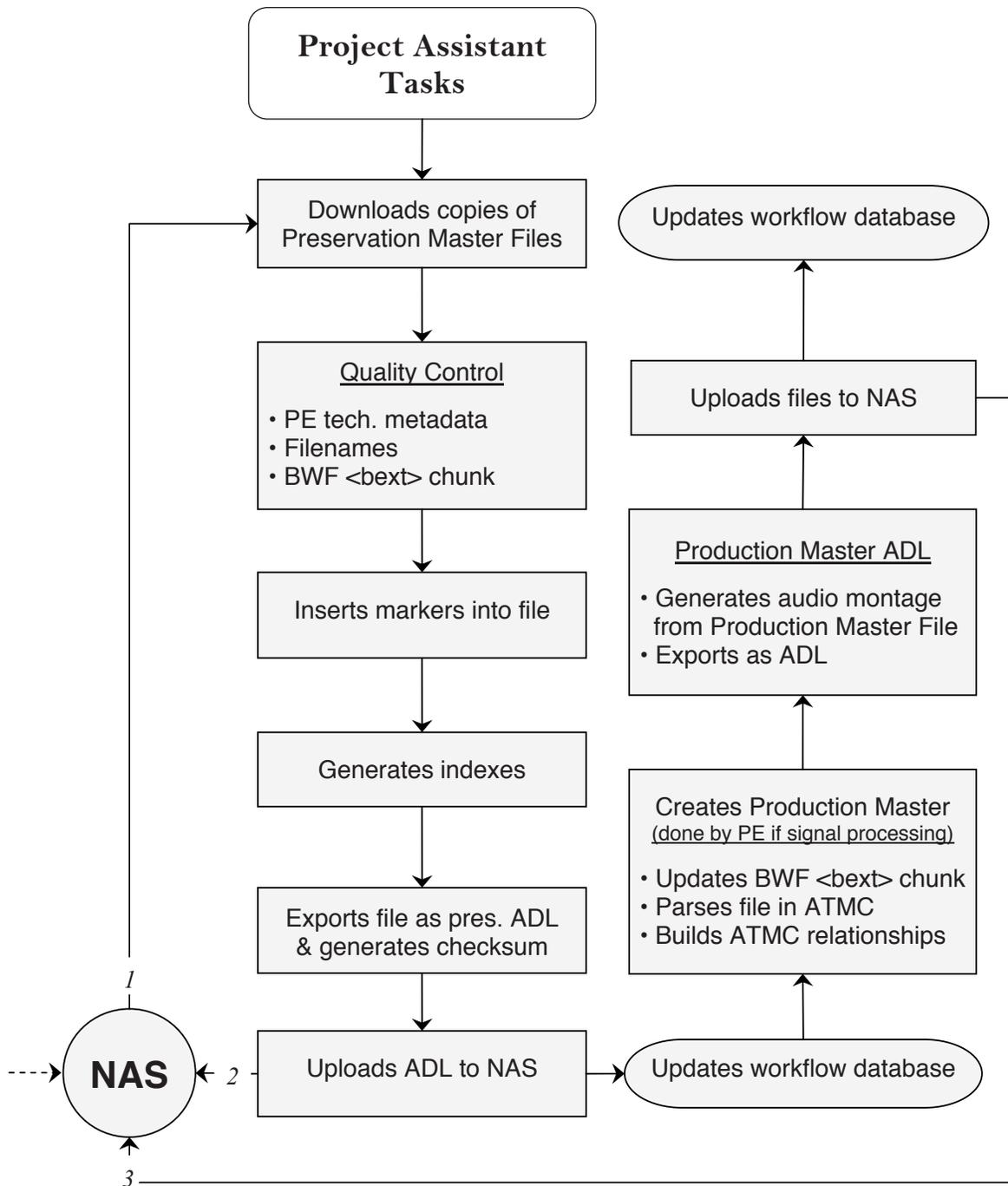


Figure 30: Workflow: Project Assistant

Here are the steps taken by the Project Assistant during this stage:

1. Download copies of Preservation Master Files from the NAS. The PA uses these copies as a reference and the definitive versions remain untouched on the NAS. PA work does *not* update Preservation Masters unless there are errors, which are corrected in the files on the NAS
2. Perform quality control for filenames, metadata in the BWF <bext> chunk, and all technical metadata entered by the PE
3. Mark areas of interest in the digital file and create an index

The file is opened in the WaveLab Montage, which is a non-destructive editing environment. Markers are placed in the file following the collector's documentation and instructions from the ATM Archivist. An AES31-3 ADL is produced that carries this marker metadata; this ADL is uploaded to the NAS where it resides with, and points to, the relevant Preservation Master File, which does not actually contain the markers. A checksum is produced for the ADL file. An ATM index—currently a Word document—that carries further descriptive metadata is produced.

4. Create Production Master File

The Montage file is rendered with markers to create the Production Master. An ADL that carries marker metadata is produced for this file.

5. Collect technical metadata for Production Master File

The PA enters metadata into the BWF <bext> chunk for this file. The newly-created Production Master is parsed by ATMC to create a record of this file in the database and an ATMC processing history instance is created for this event. This process also generates a checksum for the file.

6. Quality control for digital files

The PA opens the ATMC relationships graph for the source analog recording to confirm that all files are present and accurately related to their parents and children.

7. Copy Production Master and ADL files to the NAS and verify the checksums

8. Update workflow database

Additional Quality Control

The Associate Director for Recording Services undertakes quality control work for selected parts of the collection before work on it is considered complete. This is described in the section on quality control, above.

Ingestion into Preservation Repository

The physical location for the digital objects created in our workflow is as follows:

- An Oracle database on a Digital Library Program server contains XML representations of the audio object metadata, digital provenance metadata, and relationship information
- The NAS storage device at the ATM contains the audio files, ADL files, MD5 checksum files, and other miscellaneous files

Both of these storage mechanisms are backed up to Indiana University's Massive Data Storage Service (MDSS) regularly.

However, over the course of the *Sound Directions* grant, we have begun to work with the DLP project to develop a preservation repository. This project is an attempt to streamline and standardize the way digital objects are stored and preserved, by using the Fedora digital object repository system. Therefore, we have developed methods for integrating the digital objects created using our workflow with the Fedora system.

At the current time, ingestion of ATM-generated digital objects into Fedora is a partially manual process, as is described below:

- The ATM Associate Director gathers the audio files, ADL files, MD5 files, and any other relevant files
- Audio object and process history (digiprov) files are exported in XML format from ATMC and placed with the other digital files
- All of these files are then ingested into Fedora using ingestion scripts

Note that in the future, ingestion should happen automatically. We hope to remove the manual process of gathering files, and have links between ATMC and Fedora to enable files to be ingested automatically.

7.2.3.4.3 Workflow Variations

Signal Processing

If we choose to signal process a collection, this work is completed by the Project Engineer, who then is responsible for creating the Production Master Files rather than the Project Assistant.

One Face = Multiple Digital Files

Most of the time, in our experience so far, just one digital file is required for one Face of a source recording and no editing is needed. In this case, the source timeline becomes the reference timeline that is carried by all manifestations of the content.

However, if the transfer process results in multiple digital files representing one Face of an analog source recording, the Project Engineer produces the ADL that edits these files together in their correct position on the destination (reference) timeline. The Project Assistant opens this ADL to begin work, later replacing it with an updated ADL that carries marker metadata as well as edit information. The PA's workflow also includes one extra step: the diff command

in the program Cygwin¹¹⁹ is used to compare the original versus the updated ADL. This is done to ensure that the PE's edit points were not inadvertently moved. Accidental moving of the edits will show up when the two ADLs are compared.

Preservation Master-Intermediate Files

As discussed above, Preservation Master-Intermediate Files are stand-ins for the Preservation Master Files. The PA will perform QC on the Preservation Masters, but will begin the normal workflow on the Preservation Master-Intermediates.

7.2.3.4.4 Simultaneous Transfers

The Archives of Traditional Music at Indiana University holds over 110,000 recordings, almost all of them analog. With a sound archive of this size, preservation is partly a race against time due to format deterioration and obsolescence. Even without these factors, not all ATM recordings will be digitized anytime soon due to the sheer numbers. It is against this backdrop that we began as part of this project to explore in a preliminary way methods to increase throughput within our preservation system. There are a number of ways to boost the number of recordings transferred within a given time period; we chose to research one small part of one procedure—simultaneous transfers attended by one engineer. Our objective was not only to increase our numbers per unit of time, but to attain a high level of confidence that we were still outputting preservation-quality digital files. In many ways this is a matter of risk management: can the potential loss be quantified and is it acceptable? One strategy is to define the acceptable risk level, perhaps from an analysis of the value of the content or the stability of the format or specific carriers. From that point forward the primary goal is to maintain risk equilibrium at that level. When risk is increased, additional quality control measures are needed to regain equilibrium.¹²⁰

Our goal, with the limited time and resources we had left in this first phase of *Sound Directions*, was simply to begin the exploration of increasing throughput that would continue with the next project. Our assessment of risk level has not yet reached the level of sophistication outlined in the paragraph above. Our research focused only on the simultaneous transfer of two or three cassettes with hardware and software already in use for the project. We do not believe that field discs lend themselves to this type of process as they usually require constant engineer attendance due to their fragility and the prevalence of problems encountered. We did not explore simultaneous transfer for open reel tape or other strategies for increasing throughput; these will be researched under the next grant project that begins June, 2007. Cassettes were selected for simultaneous transfer based on the apparent absence of preservation problems. Any items that appeared to have problems or unusual characteristics were directed to a single transfer process. Cassettes were not mixed across collections, that is, simultaneous transfer was done in batches by collection. The collections we used to explore this process contained spoken word content in a non-English language.

119 Cygwin is a Linux-like environment for Windows. Cygwin tools are ports of GNU development tools for Windows that run because of the Cygwin library that provides the UNIX system environment these programs expect. See <http://cygwin.com/>.

120 This strategy comes from Chris Lacinak, "Quality Control: Evaluating the Digital Surrogate" (paper presented at NARA 21st Annual Preservation Conference, *Managing the Intangible: Creating Storing and Retrieving Digital Surrogates of Historical Materials*, Adelphi, Maryland, April 30, 2007). Accessible through the AVPS site under <http://www.avpreserve.com/resources.html>.

7.2.3.4.4.1 Methods

One instance of WaveLab

One instance of WaveLab can accommodate more than two streams of audio by using the ASIO driver. We attempted to transfer three mono cassettes at once, resulting in three simultaneous streams into the computer. While this functionally worked, we discovered errors in the digital files that appeared random. Obviously, multiple channels are often recorded natively into one computer so there is something about our setup—software, converter, or computer—that made this unworkable. We did not troubleshoot this very far as there are other disadvantages to this approach: the transfer session must end completely when all cassettes are finished. Preparation time for all recordings to be transferred in the next session must then take place during this downtime when no transfers are running. Also, filenames must be edited after recording as it is not possible to assign separate filenames to the individual streams—WaveLab simply adds a number to identify each stream. And, if cassettes are different lengths there will be varying amounts of silence in the files that must be addressed. Overall, this approach seemed far less efficient than the method described below.

Multiple instances of WaveLab using the Overlap Method¹²¹

It is also possible to open multiple instances of WaveLab, each fed from either a separate converter or from separate channels on a multi-channel converter. In our case, we had access only to separate two-channel converters and we used the AES/EBU output of each converter patched directly into the soundcard, bypassing the need to use the ASIO driver. Using this setup makes it possible to use a very efficient method of overlapping one transfer with another. Continuously rolling from one transfer into the next works like this:

- Prepare first cassette for transfer
- Begin first transfer and monitor first 5 minutes
- Prepare second cassette for transfer while the first continues
- Begin transfer of second cassette and monitor for first 5 minutes
- Monitor both cassette transfers at the same time, one from each speaker
- Monitor just the first cassette as it ends
- Prepare third cassette for transfer while the second continues
- Begin transfer of third cassette and monitor for first 5 minutes
- Monitor second and third cassettes at the same time
- Monitor just the second cassette as it ends
- Prepare fourth cassette for transfer while the third continues
- Continue until time for a break

Using this procedure, the Project Engineer would typically create and name files for an entire day, then devote as much of the next day as needed to entering technical metadata and parsing files. For each cassette the PE notes on paper technical metadata elements that change from one to another such as tape brand or repro level, as well as sonic events heard during monitoring. These notes are used when entering metadata the next day. We discovered that it was not wise to try to enter metadata into ATMC during transfer: the PE was busy enough monitoring multiple sources and adding metadata entry provided too much opportunity for error.

We made use of various physical strategies to keep the multiple transfers in order and quickly

¹²¹ We learned the overlap method from George Blood of the Safe Sound Archive. We also benefited greatly from several discussions with him about his simultaneous transfer workflow.

identifiable. For example:

- Cassette machines were labeled number 1 and number 2; number 1 was on top with number 2 underneath it (left and right would also work)
- Cassette machine 1 was always monitored from the left speaker; cassette machine 2 from the right
- Instance 1 of WaveLab was placed in the left computer monitor and routed to the left audio speaker; instance 2 of WaveLab was on the right of both
- Transfer from cassette machine 1 was routed to the top meter in SpectraFoo, while machine 2 was routed to the bottom

This facilitated the identification of problems through both listening and watching meters. It also aided the changeover from one cassette to the next, including tracking the end of a tape.

The use of a simultaneous transfer method was noted in the digital provenance metadata for each digital file. The purpose of the transfer event was noted as: “Simultaneous digitization of multiple analog source recordings to create Preservation Master Files.” In addition, a notes field was used to record similar information, including a statement that the transfers were semi-monitored.

We tried the overlap method using three cassettes and three instances of WaveLab but were unable to get all three instances working at the same time. It appears that WaveLab will not record in three separate, simultaneous instances.

There are other methods that can be used for simultaneous transfer including the use of multi-track recording software. This might solve some of the problems listed above. We did not have this software available and, in any case, felt that it was valuable to explore this procedure using WaveLab. We are also aware of at least one operation that uses complete, separate signal chains for each transfer to keep everything straight.¹²² While this strategy may have potential for us, we do not have the equipment necessary to test this method.

7.2.3.4.4.2 Assessment

Using the overlap method yielded a significant increase in productivity. Not only were two sources undergoing transfer at most times, there was always at least one stream of audio running even while preparing the next tape for transfer. With this method it doesn't matter if the lengths of tapes vary or if some are recorded on only one side.

Our quality control procedures, which were already quite robust, remained the same although we are not sure that we adequately mitigated the additional risk from semi-monitored transfers. Even though portions of each tape are not monitored by the engineer, and other parts are monitored in tandem with another tape, the Project Assistant normally listens through much of the transferred content while marking areas of interest in the files. This provides an additional check already built into our system. Generally, we did not find significantly more metadata entry errors than usual and we felt that this was largely due to devoting a large block of time to transfer, followed by a dedicated block of time for metadata collection.

We were able to identify some trade-offs using this process: evaluation of sonic problems that already exist on the recording was not as complete as with fully-monitored transfers. We consider this metadata as “extra” anyway. Time stamps for data entry are not fully accurate

¹²² This is done at Safe Sound Archive.

with metadata entry occurring the next day. In addition, it takes a fairly major aural event to catch the engineer's attention when monitoring two transfers at the same time. Transient clicks or tics may not be noticed.

We also identified other problems: ongoing simultaneous transfer results in a blizzard of files sent downstream to the Project Assistant and an instant, quickly growing backlog. We have yet to fully resolve this issue, although potential strategies include using additional, temporary staff to assist during this process or postponing the work of marking areas of interest in files until later, perhaps not until they are accessed by a researcher. We also felt that PE monitoring of two transfers at once was workable as long as the language was *not* native to the engineer. We speculate that listening to multiple transfers in the engineer's native language might be more difficult although possibly akin to attending a party. Monitoring two transfers of music content at the same time might not be workable. It may be possible in these cases to use sequential monitoring of sources, moving regularly from one to the other.

Overall, our research indicated that simultaneous transfer of two cassettes using the overlap method is manageable, carries low risk, and increases throughput, as long as the source recordings are not experiencing preservation problems. Preliminary results indicate that simultaneous transfer of two cassettes requires 36% less engineer time than single transfer. We were not able to find qualitative differences between digital files from multiple transfer versus single transfer processes. We did find an unexpected benefit from the simultaneous transfer procedure: there is typically significant downtime for the engineer during transfer of a single source recording without problems. The PE reported that running two sources at once kept him fully engaged in the transfer process, and that separating audio transfer from metadata collection enabled more sustained focus on each, leading to greater efficiency. This, of course, may vary from person to person.

7.2.3.5 Workflow at Harvard

When Harvard started work on this project, our workflow was almost entirely a manual process carried out by audio engineers. The only automated process in place at the start of the grant was use of the Dmart tool for building batch deposits for the DRS loader. This tool relies upon naming conventions and business rules that place specific files in predefined locations on the file system to signal their role in the project. As we approached the task of building the capability to automate our workflow, we had to produce a result that would be compatible with the business rule expectations of the Dmart tool.

At the beginning of the process we evaluated what we were currently doing, what we liked about our approach and what we did not like. We also made a list of the things we wished we could be doing, but which were too cumbersome to do by hand. The result was a starting place for developing the automation tools designed by lead engineer David Ackerman and programmer Robert La Ferla that became the Harvard *Sound Directions* Toolkit.

The toolkit was designed to consist of a large number of command line interface tools, each of which was targeted to a very specific application. Command line tools are initiated by typing commands directly into a terminal window and specifying an input file, an output file and the appropriate options. We think of these tools as the pieces of a jigsaw puzzle with very regular shapes. The idea is that through scripting these can be pieced together into multiple configurations to generate different automated workflows.

We divided our workflow into four major components. We did this by looking at the logical break points in the workflow where human interaction might be desirable. In short, a person

does a preservation transfer, and then an automated process is run to build the archival elements for the deposit. There is then an opportunity for an engineer to signal process the material for the Production Master. Then an automated process handles the routine work of building the Production Master that will be deposited. After which, an automated process is run to build the delivery files for the deposit, and finally, after a person checks the results of the work, the deposit is generated.

The development was carried out in an iterative stepwise process. We started with the development of our “makedeliverable” workflow script. Once we had a skeletal working script we went through multiple iterations of testing and enhancement of the script. Once we were satisfied with that script we went through the same process for developing our “makeproduction” and then “makearchival” and “makedeposit” scripts. Once all four scripts were working, an additional round of testing was necessary to ensure that the four were working together as a reliable system.

The investment in time to build the automated processes was well worth the expenditure. These tools have revolutionized our preservation workflow and have liberated our audio engineers from the drudgery of repetitive tasks. The result is that we are now able to easily generate deposit packages that conform to our expectations with reduced opportunity for error. This process allows the audio engineer to focus their attention on the task of getting the best playback, and in the evaluation of the object undergoing preservation, instead of on the task of generating endless XML documents by hand.

7.2.3.5.1 Selection

Curatorial staff determine the selection of objects for preservation and submit a preservation request to Audio Preservation Services. The request identifies the object by title, persistent identifier (normally the call number or shelf number), format, curatorial responsibility, condition and handling restrictions, whether or not it is to be deposited in the DRS, the DRS owner code, the billing code, desired formats, level of preservation (either full preservation metadata or digital copy only), access permission level (Public, Restricted, or Staff Only), and workflow status. Currently, each request is a single paper document. However, we have begun development of a database with which to replace this document and track objects through the entire preservation workflow.

If the preservation request is one of many in a collection and there are no priorities specified for those requests, then we perform a triage and prioritize preservation on the basis of physical condition (greatest degree of deterioration first), then tape base (acetate first, then polyester) or disc formulation (lacquer, then vinyl). Object condition permitting, we prefer to organize transfers by format to reduce setup time.

7.2.3.5.2 Open Reel Tape Transfer

This example is from the preservation of a quarter-inch open reel tape from the James Rubin Collection of South Indian Classical Music.

If our object is a quarter-inch, open reel tape, we determine track configuration with the aid of a magnetic viewer. The proper head stack is installed in the Studer A807 open reel tape player, and the transport and heads are cleaned and then demagnetized if needed. The tape is played briefly to confirm tape speed by listening to the built-in tape monitor speaker. The outputs of the Studer are patched to the inputs of the Prism AD-2 analog-to-digital converter for A/D conversion. The output of the AD-2 is then routed to both the G5, so that we may use SpectraFoo for analysis, and to the DAW for recording. Once SpectraFoo is running, we

align azimuth, then playback EQ using the reference tones on the subject tape, if any. If there are no reference tones on the subject tape, we calibrate azimuth and playback EQ using the appropriate MRL reference tape. Normally after calibration we do not change the output gain of the tape machine, but document the reference fluxivity in the process history, adjust the input sensitivity of the A/D converter and document that adjustment. Then we attempt to optimize azimuth alignment for the subject tape using a suitable portion of the tape's program.

Once playback is optimized, we mount an NTFS volume on the SAN using the SAN client on the Pyramix PC. We create a file structure for the project such as:

```
C:\Volumes\V13\DRS_DEPOSITS\Rubin\Deposit_AWM_RL_15465\AWM_RL_15465\workspace
```

In that workspace directory we create a simple, text document that we label "project.prop." This is a configuration file that contains the data necessary for our tools to create metadata for preservation parts and production deliverables.

project.prop file example:

```
title="Rubins tape 86-5, Music Academy (Madras) 8am Dec 27 1986"  
shelfnum="AWM_RL_15465"  
soundfield="Stereo"  
disposition="Deposited in DRS"  
copyright="© President and Fellows of Harvard College Library"  
config="/Users/gordon/drs_dmart-1.1.7/conf/dmart.conf_Rubin.  
example"  
access="R"
```

Using the Macintosh G4 or G5 and the SAN client, we mount an HFS+ volume on the SAN and create a similar directory structure to the NTFS volume such as:

```
/Volumes/V02/DRS_DEPOSITS/Rubin/Deposit_AWM_RL_15465/AWM_RL_15465/original
```

Using the application "AudioObjectManager," we create the original audio object metadata document labeled "coreaudio.xml" in the "original" directory, and begin to customize and populate its fields which we update as data becomes available through the completion of the transfer. However, there is much data to enter from the request form, and from observation and measurement of the object before we begin the transfer.

We create another directory on the HFS+ volume called:

```
/Volumes/V02/DRS_DEPOSITS/Rubin/Deposit_AWM_RL_15465/digiprov/
```

In this directory we create a process history document using the "APXE" application. We enter our first process events which will be the transfer and export of the two sides, then add the proper device chain, and configure and connect the devices for the event using the APXE device chain patchbay.

Next we create a 96 kHz, 24 bit Pyramix project called "AWM_RL_15465_Archival_Side_1," and mount the directory "workspace" in Pyramix. The "workspace" directory will contain the recorded audio files and Pyramix project files. We set-up the Pyramix CD/SACD tab for

the creation of PQ markers during recording.¹²³

We optimize the input sensitivity of the Prism AD-2 so that average program peaks as viewed in the Pyramix VU-meter display are approximately -12 to -9 dBFS and highest peaks are no more than -3 dBFS with the VU-meter display set to 0VU at -16 dBFS. We have found that the Prism AD-2 sounds best at that level.

We advance the cursor three frames past midnight to avoid creating files that span the zero hour, initiate record in Pyramix, and begin playback of the tape. When we encounter the beginning of program, we insert a CD Track Start Marker and name it appropriately with the performance information if it is known. We do this for each performance as they are encountered during record. After recording, we will fine-tune start-mark placement. If we encounter noise or areas that might be of particular interest in production such as drastic level changes, we indicate them by placing non-PQ markers at the start of each incidence and naming those markers appropriately. If we encounter tape playback problems such that we must stop and restart the tape, we advance the cursor three seconds past the end of the recorded material, re-cue the tape to a point at least thirty seconds prior to the trouble spot, or far enough to contain sufficient context for editing, re-initiate record in Pyramix and resume playback of the tape. We proceed this way until we reach the leader or end of the tape and then stop recording. At the end of program we place a CD Stop Marker.

Ideally, the point at which tape playback began in the recording would be at the zero hour and not before. We avoided spanning zero hour in recording, but now we must adjust the beginning of the digitized tape playback to zero by deleting the blank part of the file and rippling the remainder to the left to zero. Then we apply the correct time stamp to the moved file by rendering it in place. The next task is to edit this newly trimmed and placed file together with the next file in the recording (if there is one) which also will need its time stamp corrected once the edit is done. To do the edit we find and mark the edit point on both files, then select the area between that point on both files and do another delete-and-ripple. If there are only two files in the recording, we proceed to the end of the second file, place the cursor after the point where the tape had ended, set selection start to the cursor and delete from the cursor to the right. Then we select that newly moved and edited second file and adjust its time stamp by rendering the file in place. If there were more than two files in the recording, we would proceed by editing and rendering for each one. We save our project after each step, and now would be a good time to do just that.

Now that editing is complete, we fine-tune the performances' CD start markers and ensure that they are properly labeled.

At this point we can determine the start time and duration of the Face and Region(s) and enter that data in the coreaudio.xml document.

Pyramix can export projects in a number of formats. We use the AES31-3 export to produce a preservation ADL and media set. We use the XML export to produce a marker document. To export the AES31-3 ADL and media, we create a directory named C:\Volumes\V13\DRS_DEPOSITS\Rubin\Deposit_AWM_RL_15465\AWM_RL_15465\pre-archival. From the Pyramix project AWM_RL_15465_Archival_Side_1, we export the entire project as AES31-3 into the pre-archival directory we just created, and name the export with the same project name. We then export the project as XML into the same pre-archival directory, and name the export as "AWM_RL_15465_Archival_Side_1_PQ_Marks." This produces a marker

¹²³ We have chosen to use standard CD PQ start and stop markers to delineate performances or areas of interest because PQ markers are widely used in the audio industry and they are exportable and importable in Pyramix as XML. They are not intended for CD creation unless a preservation request includes a CD access copy.

document for the project.

Before we close the side one project, we prepare a project for side two, continuing on the same timeline, by placing a marker three seconds past the end of side one, deleting the side one audio clips from the timeline, changing the project settings to reflect the values for side two project name and record filename, creating a new Disc in the CD/SACD tab with the correct info for side two and deleting the side one disc from the tab, then *saving the project* as “AWM_RL_15465_Archival_Side_2.” This newly created project will contain no audio, but will have a marker on the project timeline at which side two audio should start, and will be ready to create CD markers for side two. The original side one project still exists intact since we *saved the modified project* as a different project name. The transfer of side two proceeds similarly to side one, except that side two starts later on the same timeline. Again we export the AES31-3 ADL and media, and the XML PQ markers as we did for side one into the pre-archival directory.

At the completion of the side two transfer and edit/render, we can determine the start time and duration of that Face and Region or Regions, and enter that data into the coreaudio.xml document.

Once side two has been exported to the pre-archival directory, we run the first custom software tool in the process, called “makearchival.” Table 8 describes the actions performed by makearchival, whose ultimate products are pristine Archival Masters and accompanying metadata untouched by the operator.

Preliminary Processes

The script “makearchival” looks for a pre-archival directory. If there is none, the script exits. If there is a pre-archival directory, the script looks for a project config file (project.prop). If there is none, the script exits. If there is a project config file, the script continues.

The script looks for an archival directory. If there is no archival directory, it creates one.

The script looks for a misc directory. If there is none, it creates one.

The script creates a list of all ADLs in the pre-archival directory and validates that the ADLs’ filename syntax conforms to HCL-APS conventions. If they do not conform, the script warns and exits. If they do conform, the script checks the property file for a shelf number. If there is no shelf number the script warns and exits. If there is a shelf number in the property file, the script continues.

The script locates the Pyramix XML marker file in the pre-archival directory, copies the marker file to the misc directory and then validates the file copy using the MD5 algorithm.

The script calls the tool “convertmarkers” that converts the Pyramix marker file to the HCL-APS marker format XML document. This is a stopgap measure in our workflow until the new version of AES31-3, which provides a standardized way to store marker metadata, is adopted by manufacturers. Here is an example of the interim HCL-APS format:

```
<?xml version="1.0" encoding="UTF-8" ?>
  <hclAudioMarkerDocumentElementType xmlns="http://www.fas.harvard.edu/
    hclAudioMarker" xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" x
    si:type="hclAudioMarkerDocumentElementType">
    <projectSampleRate>44100</projectSampleRate>
    <marker markerNum="1" markerName="Duo sonata in A major op 162 - Schubert"
      markerType="PQ_START" samplePosition="222264" performanceTitle=""
      performancePerformer="" performanceComposer="" performanceArranger=""
      />
    <marker markerNum="2" markerName="A life - Ernst Bacon" markerType="PQ_START"
      samplePosition="59434452" performanceTitle="" performancePerformer=""
      performanceComposer="" performanceArranger="" />
    <marker markerNum="3" markerName="String quintet in C minor K 406 - Mozart"
      markerType="PQ_START" samplePosition="144971988" performanceTitle=""
      performancePerformer="" performanceComposer="" performanceArranger=""
      />
    <marker markerNum="3" markerName="String quintet in C minor K 406 - Mozart
      Stop" markerType="PQ_END" samplePosition="217078428" performanceTitle=""
      performancePerformer="" performanceComposer="" performanceArranger=""
      />
  </hclAudioMarkerDocumentElementType>
```

Events Recorded in Process History (Digital Provenance)

For each ADL in the pre-archival directory, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
adlfix	Complete the <SOURCE_INDEX> by retrieving all the optional information fields in the BWF <bext> chunk such as the USID, start time and duration and the description, and then entering them into the source index. Completes the remarks for each entry in the event list. Write process event (to document the operation).	Input pre-archival AES31-3 ADL. Output fixed (-fix) ADL to misc directory.
addpanentries	Add a <PAN_LIST> to ADL by determining the number of sources and assigning them to: Center (0.0) if there is only one, assigning the odd numbered source to Left (-100.0) and the even to Right (+100.0) if there are two. If there is already a pan list, the script exits stating pan list exists—no operations performed. Write process event.	Input -fix ADL in misc directory. Output -fix-pan ADL to misc directory.
adlpathsub	Replace pre-archival <SOURCE_INDEX> URL path with archival path in ADL. Write process event.	Input -fix-pan ADL in misc directory. Output -fix-pan-pathsub ADL to misc.
editadlheader	Set project title. Set project notes. Write process event.	Input -fix-pan-pathsub ADL in misc. Output -fix-pan-pathsub-headers ADL to misc directory.
addmarkers	Retrieve markers from input HCL-APS marker file and adds them in a <MARK_LIST> to the ADL. Write process event.	Input -fix-pan-pathsub-hdrs ADL in misc directory. Input HCL-APS marker file. Output Archival ADL to archival directory.

Table 8: makearchival script: ADL actions

For each BWF file in each of the ADLs in the pre-archival directory, the script does the following:

Tool Called	Actions Taken	Input/Output Media
cp	Copy one of the BWF media from pre-archival to archival directory. Write process event.	Input pre-archival BWF file. Output archival BWF file.
cmpcksum	Perform MD5 checksum on file 1 audio data only. Perform MD5 checksum on file 2 audio data only and compare. Write process event.	Input file 1 audio data only. Input file 2 audio data only. Output checksum file 1. Output checksum file 2. Output true or false value on compare.

Table 9: makearchival script: BWF actions

If the original recording required noise removal or any other process before making deliverables, such as consolidation of multiple archival files in the timeline into one continuous file, then we would create a Pyramix project in the workspace for that purpose and import the archival ADLs into that project. The resulting processed project would be exported to a pre-production directory where we would use the software tool “makeproduction” to create Production Masters from the pre-production exports. If the original recording required neither de-noising nor consolidation of files, then we would run “makeproduction” immediately after “makearchival” without needing to create a pre-production directory. The following table describes the actions performed by the “makeproduction” script.

Preliminary Processes

The script “makeproduction” looks for a pre-production directory. If there is no pre-production directory, the script looks for the archival directory. If there is no archival directory, the script warns and exits. If there is an archival directory, the script continues.

The script checks for a project config file. If there is no project config file, the script warns and exits. If there is a project config file, the script continues.

The script looks for a production directory. If there is none, the script creates one and continues.

The script looks for a workspace directory. If there is none, the script creates one and continues.

The script looks for a SRC directory within the workspace directory. If there is none, the script creates one and continues.

The script looks for a DITHER directory within the workspace directory. If there is none, the script creates one and continues.

The script gets each ADL from the archival directory and checks to see if they contain markers. If they do not, the script gets the marker info from the respective marker files and adds those markers to the ADLs. The output ADLs are saved to the misc directory. If the ADLs do contain markers, the script continues.

Events Recorded in Process History (Digital Provenance)

For each ADL in the archival directory, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
cp	Copy Archival ADL to misc directory. Write process event.	Input Archival ADL. Output Archival ADL to misc directory.
adlsrc	Sample rate convert the Archival ADL in misc directory to 44.1 kHz and save result as Workspace-SRC ADL in misc directory. Write process event.	Input Archival ADL in misc directory. Output Workspace-SRC ADL to misc directory.
editadlheader	Set the project title in Workspace ADL header to “_Workspace-SRC_” Write process event.	Input Workspace-SRC ADL in misc directory. Output Workspace-SRC ADL to misc directory.
adlpathsub	Replace each BWF unique ID in Workspace-SRC ADL to match respective converted BWF file ID. Write process event.	Input Workspace-SRC ADL in misc directory. Output Workspace-SRC ADL to misc directory.
adlpathsub	Replace role specifier “_AM_” in Workspace-SRC ADL source filenames with “_WM_” (Workspace Master) designation. Write process event.	Input Workspace-SRC ADL in misc directory. Output Workspace-SRC ADL to misc directory.
adlpathsub	Replace role “Archival” in the ADL name with “Workspace-SRC.” Write process event.	Input Workspace-SRC ADL in misc directory. Output Workspace-SRC ADL to misc directory.
adlpathsub	Replace “/archival/” with “/workspace/SRC/” in the Workspace-SRC ADL path. Write process event.	Input Workspace-SRC ADL in misc directory. Output Workspace-SRC ADL to misc directory.
adlpathsub	Replace “/pre-production/” with “/workspace/SRC/” in the Workspace-SRC ADL path.	Input Workspace-SRC ADL in misc directory. Output Workspace-SRC ADL to misc directory.

Table 10: makeproduction script: archival or pre-production directory ADL actions

For each BWF file in each ADL, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
Resampler	Sample rate convert 96 kHz BWF archival file to 44.1 kHz BWF Workspace-SRC file. Write Process event.	Input 96 kHz BWF archival media file. Output 44.1 kHz BWF Workspace-SRC media file.
MBITPlus	Change bit depth of the Workspace-SRC WM BWF file from 24 bits to 16 bits using iZotope 64-bit dither processing and save in misc directory. Write processing event.	Input 24 bit WM BWF file in misc directory. Output 16-bit WM BWF file to misc directory.
bwavedit	Set the <bext> chunk description of the Workspace Master (_WM_) BWF to “_PM_” and save in production directory. Write process event.	Input the Workspace Master BWF in the misc directory. Output the Production Master BWF to the production directory.
cmpchecksum	Compare MD5 checksum of the audio data of the Production Master BWF with the checksum of the audio data of the Workspace Master BWF. Write process event.	Input file 1 audio data only. Input file 2 audio data only. Output checksum file 1. Output checksum file 2. Output true or false value upon compare.
bwavedit	Set the <bext> chunk description of the Workspace Master BWF’s role specifier to “_PM_”. Write process event.	Input Workspace Master BWF in misc directory. Output Production Master BWF to the production directory.
cmpchecksum	Compare MD5 checksum of the audio data of the Workspace Master BWF with the checksum of the audio data of the Production Master BWF. Write process event.	Input file 1 audio data only. Input file 2 audio data only. Output checksum file 1. Output checksum file 2. Output true or false value upon compare.

Table 11: makeproduction script: BWF actions

For each ADL in the Workspace directory, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
adlfix	Fix source index of Workspace-SRC ADL, fix remarks in Workspace-SRC ADL and save as Workspace-DITHER ADL.	Input Workspace-SRC ADL in misc directory. Output Workspace-DITHER ADL to misc directory.
adlpathsub	Replace BWF unique ID in Workspace-DITHER ADL to match converted BWF file ID. Write process event.	Input Workspace-DITHER ADL in misc directory. Output Workspace-DITHER ADL to misc directory.
adlpathsub	Replace "Workspace-SRC" in the Workspace-DITHER ADL with "Workspace-DITHER" in the Workspace-DITHER ADL. Write process event.	Input Workspace-DITHER ADL in misc directory. Output Workspace-DITHER ADL to misc directory.
adlpathsub	Replace "/workspace/SRC/" in the Workspace-DITHER ADL with "/workspace/DITHER/". Write process event.	Input Workspace-DITHER ADL in the misc directory. Output Workspace-DITHER ADL to the misc directory.
adlfix	Correct the source index and remarks in Workspace-SRC ADL to reflect dither operation and save as Workspace-DITHER ADL.	Input Workspace-SRC ADL in misc directory. Output Workspace-DITHER ADL to misc directory.
editadlheader	Set project title of Workspace-DITHER ADL to "_Workspace-DITHER_". Write process event.	Input Workspace-DITHER ADL in misc directory. Output Workspace-DITHER ADL to misc directory.
adlpathsub	Replace the Workspace-DITHER ADL's BWF's role specifier "_WM_" in the misc directory with "_PM_" and save as Production ADL in the misc directory. Write process event.	Input Workspace-DITHER ADL in misc directory. Output Production ADL to misc directory.

adlpathsub	Replace “_Workspace-DITHER_” in the name of the Production ADL in the misc directory with “_Production_”. Write process event.	Input Production ADL in misc directory. Output Production ADL to misc directory.
adlpathsub	Replace path “/workspace/DITHER/” in the Production ADL of the misc directory with “/production/”. Write process event.	Input Production ADL in the misc directory. Output Production ADL to the misc directory.
adlpathsub	Replace Production ADLs BWF’s unique ID from the dither operation with the unique ID for the Production ADL’s dithered BWF. Write process event.	Input Production ADL in the misc directory. Output Production ADL to the misc directory.
adlpathsub	Replace the unique ID of the Workspace Master BWF Production ADL with the unique ID of the Production Master BWF in the Production ADL. Write process event.	Input Production ADL in misc directory. Output Production ADL to misc directory.
editadlheader	Set project title role of the Production ADL in the misc directory to “_Production_”. Write process event.	Input Production ADL in misc directory. Output Production ADL to misc directory.
adlfix	Fix source index and remarks in Production ADL of the misc directory. Save resulting Production ADL to production directory.	Input Production ADL from misc directory. Output Production ADL to production directory.

Table 12: makeproduction script: workspace directory ADL actions

Once “makeproduction” has been run, we run the “makedeliverable” script to create deliverables as RealAudio files and SMIL files. SMIL, or Synchronized Multimedia Integration Language, is a W3C recommended XML markup language for describing multimedia presentations. SMIL files present performances within our RealAudio streaming media files. The benefit of using SMIL to navigate media is that an unlimited number of SMIL files can be created to serve different access purposes for the same media file. Therefore, future

customized presentations of that media will not require additional labor to remake production parts or deliverables. Note that the creation of streaming media requires an interleaved source file for stereo sources. Therefore we interleave the BWF files and update the ADL source index and edit list accordingly.

Preliminary Processes

The script “makedeliverable” checks to see if there is a production folder. If there is none, it warns and exits. If there is a production folder, it continues.

The script looks for a project config file. If there is none, it warns and exits. If there is one, it continues.

The script looks for a misc directory. If there is none, it creates one. If there is one, it continues.

The script looks for a deliverable directory. If there is none, it creates one. If there is one, it continues.

The script looks for a deliverable/smil directory. If there is one, the script deletes it and creates an empty deliverable/smil directory.

The script looks for an INT directory in the workspace directory. If there is none, it creates one. If there is one, it continues.

Events Recorded in Process History (Digital Provenance)

For each ADL in the production directory, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
adlinterleaver	Interleave all multi-channel BWF file representations in the ADL. Create a tmp directory. Create a batch.xml file in the tmp directory. Create an interleaved Deliverable ADL in the misc directory. Create a Deliverable Media directory in the INT directory. Write process event.	Input a Production ADL in the production directory. Output a Deliverable ADL to the misc directory.
adlpathsub	Replace the BWF role specifier “_PM_” with “_DM_”. Write process event.	Input a Deliverable ADL in misc directory. Output a Deliverable ADL to misc directory.
editadlheader	Set project title of Deliverable ADL to “_Deliverable_”. Write process event.	Input a Deliverable ADL in misc directory. Output a Deliverable ADL to misc directory.

Table 13: makedeliverable script: production directory ADL actions

For each multichannel BWF file in each ADL in the production directory, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
interleaver	Interleave multi-channel BWF file. Write process event.	Input Left channel BWF file in production directory. Input Right channel BWF file in production directory. Output interleaved BWF file to workspace/INT directory.
bwavedit	Set BWF file <bext> chunk description's role specifier to "_DM_". Write process event.	Input interleaved BWF file in INT directory. Output interleaved BWF file to INT directory.
cmpchksum	Compare MD5 checksum of input BWF files audio data only to output BWF file audio data only. Write process event.	Input file 1 audio data only. Input file 2 audio data only. Output checksum file 1. Output checksum file 2. Output true or false value upon compare.

Table 14: makedeliverable script: production directory BWF actions

For each Deliverable ADL, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
adlfix	Correct source index and remarks of ADL to reflect change in <bext> chunk description's role specifier.	Input Deliverable ADL in misc directory. Output Deliverable ADL to misc directory.

Table 15: makedeliverable script: misc directory ADL actions

For each interleaved BWF, the script performs the following:

Tool Called	Actions Taken	Input/Output Media
Real Producer	Encode interleaved BWF as RealAudio and save to deliverable/performances directory. Write process event.	Input BWF file in INT directory. Output RealAudio file to deliverable/performances directory.
Real rmeditor	Create audio object metadata for RealAudio file based upon project config file. Sets the download and copy restriction flag. Write process event.	Input RealAudio file in performances directory. Output RealAudio file audio object metadata in performances directory.

Table 16: makedeliverable script: misc directory BWF actions

Upon completion of RealAudio metadata, the script runs the `adltosmil` tool on each Deliverable ADL—once to create a single SMIL file representing all performances of a single side, and once to create a SMIL file for each performance for the side.

The next step is to refresh the NTFS volume on the PC and re-mount it on the G4 as read-only, then create symbolic links from the files on the NTFS volume to the HFS+ volume using the G4. These symbolic links, or symlinks, allow us to test deliverables and build a deposit package on the HFS+ volumes from linked files on the NTFS volume without having to move or copy files. We test the RealAudio media and SMIL files before proceeding further.

Each critical event in the preservation workflow is documented in a digiprov document. The pre-archival transfer and pre-production processes are manually entered by the operator. At each automated stage in the workflow, such as archival, production and deliverable, process events are automatically entered into specific digiprov documents. Upon creation of the deposit package, all digiprov documents are merged into a single document in the archival directory for deposit.

Our deposit package creation process is automated by the script “`makedeposit`,” which is designed specifically to suit our particular workflow and data environment. Such a script would be different for another institution with a different workflow and environment. The “`makedeposit`” script’s general description follows.

The “`makedeposit`” script tests the source paths for valid symbolic links, warns if a link does not point to a file, and asks if all network volumes are mounted. The script checks for a project config file. If it finds none, the script exits. If it finds one, it continues.

The script checks for a deliverable directory. If it finds none, the script exits. If it finds one, it continues.

The script generates DBOF files (Dmart Batch Override Files). These files allow the DRS batch metadata for submission to be overridden for each content file. This enables one to add or change metadata values for the generated batch.xml file on a per-audio-file basis. For instance, one may specify a different DRS owner supplied name, or local ID, than the default which is in the file path and name.¹²⁴

The script merges the multiple process histories into a single digipro document and copies the document to the archival folder as digipro.xml.

The script moves non-Dmart directories to a temporary area.

The script converts the marker.xml and PQ_Marks.xml files to marker_xml and PQ_Marks_xml respectively in order to overcome a bug in Dmart.

The script runs the Dmart tool. Dmart creates a batch.xml and a mets.xml file for the deposit.

The script restores the non-Dmart directories from the temporary area.

The script creates a convertedsmil directory and runs the “convertsmil” tool which changes the SMIL files’ source from a relative path, to the owner supplied name of the deposited RealAudio deliverable file. The script also creates a SMIL batch file.

At this point, the deposit package is complete. We examine the Dmart processes in the terminal to see that Dmart properly included all data in the METS file. In the batch file we check the owner supplied name, start time and duration of the Preservation and Production Masters, and confirm the access permissions and the URNs for deliverables. Once QC checks of the package have been made, we run a script called “aggregatedeposit” that moves the deposit data into a STAGE directory so that the RealAudio deliverables get deposited just before their related SMIL files, and then we use secure FTP to transfer the contents of the STAGE directory to the DRS loader.

Upon successful deposit, an e-mail notification listing all of the items deposited (including URNs for the deposited SMIL files) is sent to the depositor and curatorial contact. The curatorial staff can then enter these links in the online catalog for access. If there is a problem with the deposit, the DRS sends an error report e-mail to the depositor that contains information useful in finding and correcting the offending data.

124 Office of Information Systems, “DRS METS Archive Tool (Dmart) for Audio Deposit,” President and Fellows of Harvard College (rev. December 22, 2005), <http://hul.harvard.edu/ois/systems/drs/dmart/current/>.

8 Summary of Best Practices

Personnel and Equipment for Preservation Transfer

Best Practice 1: Use audio engineers and technicians with solid technical skills and well-developed critical listening abilities at points in the preservation transfer workflow where their skill is required.

Best Practice 2: Perform preservation transfers in an appropriately designed, critical listening environment. If such a space is not available, choose a room that is quiet and is removed from other work areas and traffic, and be acutely aware of its sonic deficiencies.

Best Practice 3: Route the signal from the playback machine to the analog-to-digital converter using the cleanest, most direct signal path possible.

Best Practice 4: Design the monitoring chain to allow instant comparison of the signal from the playback machine to the signal that has passed through the analog-to-digital converter.

Best Practice 5: Preservation studios must include test/calibration equipment to test and monitor the transfer chain itself for noise as well as to test individual components for performance. During transfer, the test/calibration equipment shall not be inserted between the playback machine and the recorder.

Digital Files

Best Practice 6: Use the Broadcast Wave Format for the preservation of audio.

Best Practice 7: Use the <bext> chunk of the Broadcast Wave Format for metadata that is needed to interpret the contents of a file in the absence of accompanying metadata. Do not use it as the authoritative source or for metadata that may change over time.

Best Practice 8: Include the local name for the file in the Description field.

Best Practice 9: Use the Time Reference field to provide a time stamp showing the location of the file on a reference or destination timeline.

Best Practice 10: Clearly define the purpose(s) of every digital file surrogate created for preservation or access. This metadata must be preserved so that files are managed in a way that is appropriate to their defined role and value in the archive.

Best Practice 11: Develop specific, written guidelines on the characteristics of primary preservation surrogates, and the procedures used to produce them, within the context of the institution's workflow.

Best Practice 12: Develop written guidelines on handling both technical and content edits.

Best Practice 13: Use ASCII letters (a-z), ASCII digits (0-9), underscores and hyphens, and be aware of the implications of using any other characters in filenames.

Best Practice 14: Do not use diacritic marks or any non-printing characters.

Best Practice 15: Reserve the period (full stop) for the file extension at the end of the filename.

Best Practice 16: Do not use values in file elements that might change over time.

Best Practice 17: The first element should identify the unit that created the file.

Best Practice 18: Make filename elements more detailed and/or specific as they are read from left to right.

Best Practice 19: Identifiers used in the filename should correspond to those used with physical objects and in existing catalogs.

Best Practice 20: Include a sequence indicator in the filename.

Best Practice 21: For derivative files, use the same name as the master file with the addition of an element that indicates the derivative's type.

Best Practice 22: Generate a checksum for every file that has enduring value. This includes both audio and non-audio files created during the preservation process.

Best Practice 23: Generate a checksum as soon as possible after a file is created—usually after basic work with the file is completed.

Best Practice 24: Consider the checksum value critical technical metadata. Store the value in the system used for other technical metadata, with backup copies kept in multiple physical locations.

Best Practice 25: Verify the checksum before trusting any file that has been moved, copied, or had its header edited.

Metadata

Best Practice 26: Validate all generated metadata against the schema of a published standard, or against a copy of a locally agreed upon schema.

Best Practice 27: Generate valid audio object (technical) metadata for all physical and digital audio objects in the preservation workflow.

Best Practice 28: Generate valid digital provenance (process history) metadata that describes each process event in the preservation workflow.

Best Practice 29: If a transfer must be stopped and restarted, resume in a new digital file that contains overlaps in content. Use an AES31-3 ADL to document the edits needed to present seamless audio as originally recorded on the source audio object.

Best Practice 30: Maintain a common timeline that references the Preservation (Archival) Master and all further file manifestations of the content.

Storage

Best Practice 31: Generate backup copies of all preservation files as soon as possible after creation, and preferably at the end of each complete cycle of work during which new data would normally be created.

Best Practice 32: Verify that backup copies are all unchanged by re-generating message digest (checksum) values before trusting that the copies are valid.

Best Practice 33: Store backup copies on a separate device, in a physically separate location from the original files.

Best Practice 34: Use mirroring techniques for redundancy of online preservation and access storage, and migrate the storage environment as technology changes.

Best Practice 35: Use off-site data tape for near-line storage and tape clones with periodic media refresh.

Best Practice 36: Regenerate message digest (checksum) values periodically, and when accessing files, to verify that all copies are unchanged.

Best Practice 37: Implement systems that generate periodic reports about the condition of stored objects, and allow for ad hoc reporting of those conditions such as preservation risk factors and confidence levels.

Best Practice 38: Monitor digital audio formats, the technical environment in which they are used, and the service requirements of the user community. Look for usability threats or opportunities and implement an appropriate preservation action plan.

Preservation Packages and Interchange

Best Practice 39: Use METS or another appropriate XML packaging schema in the creation of preservation packages. Use AES31-3 for comprehensively representing digital audio objects within those packages.

Audio Preservation Systems and Workflows

Best Practice 40: Develop a prioritized list of recordings and/or collections for preservation treatment based on, at a minimum, an analysis of research value and preservation condition.

Best Practice 41: Develop a formal, preferably written, plan for both quality control and quality assurance.

Best Practice 42: All digital files and all metadata produced as part of the preservation process must be subjected to quality control.

Best Practice 43: Permanent staff at the management level from the institution providing the target content for preservation should conduct some quality control.

Best Practice 44: Initiate one full round of quality control as soon as possible after beginning a new project, collection, or format, as well as after installing new equipment.

Best Practice 45: Match quality control tasks with appropriate personnel by the level of expertise required, and perform these tasks in the appropriate environment with the appropriate tools.

Best Practice 46: Specify the elements in the system designed for audio preservation including the function of each.

Best Practice 47: Confirm the quality of the output of the audio preservation system through an assessment of the contribution made by each part.

