

AUDIO PROCEDURES AND WORKFLOW
FOR THE
UNIVERSITY OF WISCONSIN DIGITAL COLLECTIONS CENTER
(UWDCC)

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TABLE OF CONTENTS

PART I: STANDARDS AND PRACTICES

1.0 INTRODUCTION.....	3
1.1 AUDIO MEDIA.....	3
1.2 ANALOG TO DIGITAL CONVERSION.....	5
1.3 SAMPLE RATES.....	6
1.4 BIT DEPTHS.....	8
1.5 TRANSFER PROCESS.....	9
1.6 PLAYBACK.....	10
1.7 POSSIBLE TRANSFER PROBLEMS.....	10
1.8 FORMATS.....	11
1.9 COMPRESSION.....	13
1.10 DELIVERY.....	14
1.11 PRESERVATION.....	18

PART II: DCG PRACTICES

2.0 SELECTION.....	19
2.1 GUIDELINES.....	20

PART III: WORKFLOW

3.0 CHECK IN.....	20
3.1 WORK SCENE.....	21
3.2 PRE-CAPTURE.....	22
3.3 EXAMPLE OF CURRENT CAPTURE PRACTICES.....	22
3.4 POST-CAPTURE.....	23
3.5 METADATA PRACTICES.....	26

WORKS CITED.....	29
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PART I: STANDARDS AND PRACTICES

1.0 Introduction

What is the importance of digitizing audio, and is digitization necessary? The importance of digitizing analog materials becomes apparent after one analyzes the benefits of digitization. First, digitized audio is easier to handle, manipulate, and deliver to the public than its analog counterparts. Second, content from a fragile storage and delivery analog format can be copied without loss of information. Third, digitization materials lend themselves more conveniently to research and study allowing for quick comparison, searching, and editing within the digital object. Fourth, users can more effectively access audio content in digital form than previously possible with analog collections. For instance, digital audio can be easily converted to another digital format without loss of quality unlike analog formats that degrade with each use and lose quality when copied. Finally, digitization also allows for the preservation of fragile and vulnerable analog materials such as nitrate-based film stock and materials that require special handling, or obsolete playback devices (NINCH, par 6). Thus, the advantages of digitization are numerous.

1.1 Audio Media

Before discussing the process of converting analog audio to digital, it is helpful to become familiar with the different types of audio media that exist. For a listing of specific types of audio media, their properties and the source devices needed for playback, please refer to Figure 1 on page 4. It is also important to keep in mind that for each type of media, different issues may arise during digitization. Therefore, one should

look at the different properties of the media to ascertain if that media will be a good candidate for digitization.

Audio Media	Properties	Source Device needed for playback
Wax or Celluloid Cylinders	1890s & 1900s, up to 5" diameter, 2-4 mins. playing time	Phonograph. See http://www.tinfoil.com for details of digital transfer.
Wire	Magnetic coated wire drums or reels. Invented 1898. Widely used by the US military in WWII. Eclipsed by magnetic tape by the mid 1950s.	Wire Recorder
78 rpm shellac resin discs	1898 to late 1950s, 10"(25cm) and 12"(30cm) most common sizes	Gramophone (wind-up) or Hi-Fi. Gramophone's steel needles need replacing after each side or record played. Hi-Fi needs a 78rpm turntable and a cartridge with a 78 rpm stylus. For best results on modern equipment a phono pre-amplifier is required to correctly equalize the different types of record.
45 rpm and 33 rpm vinyl discs	7" (20cm) single and 12" long play (30cm). Long play (LPs) introduced in 1948, stereo recordings in 1958.	Hi-Fi. Hi-Fi requires turntable with 45 and 33 rpm speeds.
Reel to Reel magnetic tape	½" to ¼" magnetic tape. BASF and AEG developed 6.5mm ferric tape and Magnetophone player in Germany from 1935. Post-war development in USA by Ampex and 3M. Stereo capability from 1949.	Reel to Reel player for appropriate width of tape.
Compact Cassette	Magnetic polyester tape introduced by Philips in 1963.	Hi-Fi. Hi-Fi requires compact cassette player.
Cartridge	¼" magnetic tape. Fidelipac (4-track, devised 1956, released 1962) and Lear (8-track, 1965) cartridge systems.	Despite similarities 4 and 8 track cartridges are not compatible and require separate players. Predominantly used for in-car audio. 4 track unpopular outside of California and Florida.

Figure 1: Audio media, properties, and the source device needed for digitization. (This chart was taken from "NINCH Guide to Good Practice: Audio/Video Capture and Management," <http://www.nyu.edu/its/humanities/ninchguide/VII/>).

1.2 Analog to Digital Conversion

What does audio digitization entail? According to Bartek Plichta and Mark Kornbluh of The Matrix, The Center for Humane Arts, Letters, and Social Sciences Online, “digitization is a process of converting an analog continuous, waveform to digital form by an Analog-to-Digital converter (ADC)” (4). In order to fully understand digitization, it is important to grasp the basics of audio sound and how it can be measured. The Colorado Audio Working Group writes that audio is a “flowing series of (analog) pressure waves” (16). In other words, audio is a sound wave that alternates between high or low pressure creating something the user perceives as sound (Colorado Audio Working Group, 16). The waveform is constantly changing from second to second, and as it changes between two points, it passes through a variety of values in between these two points. When audio is digitized, the pressure of the wave is sampled at very fast time intervals and the amplitude, or wave pressure, at each point sampled is recorded as a number (Colorado Audio Working Group, 16). Use of an ADC allows the audio signal to be sampled at equal spaces on the waveform. The sample value is a number equal to the signal amplitude at the instant the waveform was sampled (Plichta and Kornbluh, 4). By recording many points on the wave, the sound wave can be reconstructed allowing it to be copied into another format. Hence, analog to digital conversion is complete.

Two factors must be considered during the digitization process. The first factor one must bear in mind is the frequency at which a sampling is taken from the audio signal/wave. The sample rate is measured and recorded in kilohertz-- thousands of samples per second (Colorado Audio Working Group, 16). For example, consumer CDs

are recorded at 44.1 kHz allowing each second of audio to be represented as 44,100 separate amplitude measurements. The second factor one must consider is the range of numbers used to record each measurement. That range, or sample size, is measured in bit depth. According to the Colorado Audio Working Group, sample size or bit depth “describes how wide the range of numbers is that is used to represent each amplitude measurement” (16). The more numbers used in the range to measure the sample, the more accurate the recording will be.

If the ADC process is done properly and the recommended sampling frequency (kHz) and range (bit depth) are used, “every detail of the original signal can be captured” (Plichta and Kornbluh, 4). Thus, in effect the “original continuous waveform can be reconstructed exactly and, more importantly, powerful digital signal processing can be applied to the digital representation of the signal” (Plichta and Kornbluh, 4). Digital processing includes such things as editing and manipulating audio files through current software editing programs.

1.3 Sample Rates

Sample rates and bit-depths are two factors that will help determine the quality of the digital capture. The audio group, Sound Devices LLC describes sampling as “expressed in kilohertz per second (kHz) and defines the number of times a second that the analog audio signal has been measured (sampled)” (par. 4). In comparison to digital imaging, sampling rate is analogous with pixels or dots per inch (dpi) (NINCH, par. 37).

When determining sampling rates, experts refer to the Nyquist theorem. The Nyquist theorem states, “the sampling rate must be at least twice the highest frequency of interest to achieve lossless sampling” (Grotke, par. 6). Another way to think about it is

to remember that “the highest frequency pitch that a digital audio sample can hold is one-half of the sampling rate” (Colorado Audio Working Group, 16). The sampling frequency determines the amount or limit of audio frequencies that can be digitally reproduced. Therefore, in practical application, one needs to sample at 40 kHz in order to capture a signal at 20 kHz. Most experts agree that humans cannot hear pitches above 20 kHz, which explains the selection of the sampling rate for consumer CDs of 44 kHz (Colorado Audio Working Group, 16).

In the past, it was found that digital audio files could be recorded in standard 44.1 kHz, 48.1 kHz, 32 kHz, 22 kHz, and 11 kHz at bit rates of 4, 8, 16, and 20 (Colorado Audio Working Group, 22). However, as the 21st century approached, there was a significant move toward capturing audio at 96 kHz. Other institutions find that capturing at 88.2 kHz offers similar results as capturing at 96 kHz, but lends itself toward a more simple conversion to the 44.1 kHz standard (NINCH, par. 37).

If one is digitizing audio primarily for preservation purposes and not merely to be accessed on a day-to-day basis, sample rates as well as bit depths may vary. The Colorado Audio Working Group recommends, “for the purposes of archival audio . . . the CD ‘Red Book’ audio standard of 44.1 kHz and a minimum of 16-bit audio be utilized, both for quality and compatibility” (22). This standard allows the CD copy to be “reproducible” on any CD player whereas use of other formats might result in an acceptable recording, but may not be playable on other formats.

Until recently, CD-quality was the accepted standard. Yet, lately more and more institutions are moving toward capturing at 48 kHz and 16 bit-depth unless originals are of high quality in which case 24-bit depth is used (NINCH, par. 37). The Library of

Congress recommends the sampling frequency be 96 or 48 kHz for the master file, which potentially could replace the analog version for reel-to-reel tapes. As cost for storage space declines, there may be more of a move toward capturing at 96 kHz and 24 bit-depth especially for archival and preservation purposes (NINCH, par. 37).

1.4 Bit depths

Bit depth refers to the range of numbers used to represent amplitude measurement (Colorado Audio Working Group, 16). According to Sound Devices LLC, “Bit rate is an exponential measure (exponent of 2), so as bit rate increases, the amount of data [digital audio] increases exponentially” (par. 3). For instance, consider that CD audio uses a bit depth of 16. Therefore, each sample is represented by 2^{16} , which equals 65,536 binary digits. Thus, audio that is sampled at 16-bit is actually being represented by a range of binary digits that stretch from 0 to 65,536. The greater the range of numbers used to represent amplitude measurement, the more accurately represented the sound (Sound Devices LLC, par. 3).

One way bit depth is determined is by looking at the dynamic range. Dynamic range is the range from the softest to the loudest sound a system can reproduce (Digital Audio Primer, par. 18). Humans can hear almost 0 decibels (dB) as the softest sound whereas our pain threshold is approximately 120 dB (Digital Audio Primer, par. 18). When digitizing audio, one must subtract the “noise floor,” amount of noise during “silence,” from the maximum sound output the system is capable of producing (Digital Audio Primer, par. 18) Thus, it is generally accepted that one will have roughly 6 dB of dynamic range for each bit of sample depth. Again, consider the audio CD, which is 16-bit at 44.1 KHz. The audio CD has a dynamic range of 96dB (16 bit X 6dB = 96 dB).

This is one reason why professional audio is usually captured at 24-bit since the dynamic range is 144dB, which is well above the human limit (Digital Audio Primer, par. 18). Overall, when considering bit depth, one must understand the importance of dynamic range and how it relates to the binary digits representing the audio sound. However, there are a number of other factors that play into the final decision of which bit depth to use when digitizing audio such as user needs, space issues, and many more.

Figure 3 illustrates how bit depths directly correspond with sampling range. As mentioned previously, CD quality is the accepted standard, which uses 16-bits for sampling size. Bit depth may vary along with sampling frequency based institutional preferences and specific project needs.

Bit depth	Sampling range
8 bit	0-255
16 bit	0-65,535
24 bit	0-16,777,215

Figure 3: Sampling range represented by bit depth. (Adapted from the Colorado Audio Working Group, 2003).

1.5 Transfer Process

The actual conversion process involves three components (Colorado Audio Working Group, 19). These components are the playback device, the device to convert the analog signal into the digital signal (Analog-to-Digital Converter (ADC)), and the recording device (19). A more common setup involves a computer and computer-based recorder to create a CD. According to the Colorado Audio Working Group, “this system allows for greater flexibility in manipulating the original signal as well as greater flexibility in the CD creation process itself” (19). Such a setup would then utilize audio

software such as Peak, Sound Forge, or Cool Edit Pro, which allow the user to manipulate and edit the signal (19).

1.6 Playback

It is important to select the proper machine for playback. The “proper” machine is determined based on whether the audio is monographic (one track in each direction) or stereo (two tracks in each direction).

Another issue to consider when selecting the appropriate playback machine is speed. Proper speed must be determined. Modern open-reel decks play back at 3 ¾ IPS (inches per second) and 7 ½ IPS. Older tape decks play 7/8 IPS (Colorado Audio Working Group, 21). Cassettes are usually recorded at the same speed of 1 7/8 IPS. However, some half-speed cassettes do exist. (Colorado Audio Working Group, 22).

1.7 Possible Transfer Problems

Several issues may arise when transferring analog sound to digital. It is important to be aware of and to immediately address these issues to avoid damage to the audio media. Common problems are described below.

Archiving vs. Optimization - - When digitizing from analog to digital in archival terms, preserving a near to an exact copy of the original is of utmost importance. When converting from analog to digital in optimization terms, the focus shifts from preserving an exact copy to simply creating a copy that will allow users the utmost ease of access. Creating an exact replica of the original is not as important when optimizing as creating a usable copy for patrons to access easily (Colorado Audio Working Group, 22).

Bleeding tapes/Print through – occurs when data from one part of the tape presses onto another part of a different tape that is stored nearby causing the recording to be unclear and appear to have faint voices in the background. (Colorado Audio Working Group, 5).

Sticky shed syndrome - - exists when a tape has become gummy and exhibits a loud squeal when played requiring that the tape be baked before playing (Grotke, pars. 2, 3).

Warmth - - Analog warmth is the “extra detail” in the recording that is not necessarily audible to the human ear, but does add a certain amount of “body to the original signal, making it sound fuller” (Sound on Sound, par. 2). The online magazine, “Sound on Sound,” states that in comparison, digital audio “has acquired the reputation in many quarters of having a cold, clinical sound” (par. 2). In other words, analog audio can record up to a maximum level with “warmth,” meaning, higher frequency information is left out. Because this information is left out, the audio is distorted. However, it sounds natural. With digital audio, the recording is very precise. Digital recordings measure the exact maximum level without warmth and without distortion, which actually results in a “noisy or unpleasant” sound.

1.8 Formats

According to the “NINCH Guide to Good Practice to Audio/Video Capture and Management,” “the use of standards increases the portability of digital information across hardware platforms, space, and time” (par. 31). Standards and formats can be further classified into two main types: proprietary and nonproprietary.

NINCH also relates that three main formats commonly used today are WAVE (.wav), MPEG 1, Layer 3 (mp3), and streaming formats such as RealAudio (.ra). Please

refer to the following chart for descriptions of these formats. The following chart outlines the common audio formats, extensions, meanings, and, descriptions while including a brief synopsis of the strengths and weaknesses of each format. See Figure 4 on pages 12 and 13.

Figure 4:

Audio Formats:	Extension	Description
Waveform Audio File Format	.wav	Audio file format for Windows developed by Microsoft and IBM. .wav support was built into Windows95 and has essentially become an industry standard since. A variety of applications now support .wav as do additional operating system platforms such as Macintosh.
MPEG Layer 3	.mp3	mp3 files are tightly compressed but preserve the original quality of the recording. These files are small and travel easily via the Internet. These audio files are popular not only because of size and playback quality but also because a wide variety of players are available. mp3 has gained wide recognition because of Napster, a sound application that allowed users to share audio files without charge.
MPEG -7	.mp7	MPEG-7 is an ISO standard developed by MPEG (Moving Picture Experts Group). This committee also developed the standards MPEG-1, MPEG-2, MPEG-4. MPEG-7, formally named "Multimedia Content Description Interface," aims to create a standard for describing the multimedia content data that will support some degree of interpretation of the information's meaning, which can be passed onto, or accessed by, a device or a computer code.
Real Audio	.ra	One of the most common formats especially for web distribution. Real audio is often used for streaming audio.
Audio Interchange File Format	.aiff or .aif	Digital audio file format widely used with Macintosh platform although it is Windows compatible.
SUN Audio	.au, .snd	Mostly found on Unix computers. Specifies an arbitrary sampling rate. Can contain 8, 16, 24 & 32 bit.
PARIS (Professional Audio Recording Integrated System)	.paf	Used with Ensoniq PARIS digital audio editing system. Can contain 8, 16, and 24 bit.
Sound Designer II	.sdi	Originally digital sampling and editing platform. The format is still in use. Used mostly on Macs by professionals. It's a widely accepted standard for transferring audio files between editing software.
IRCAM	.sf	Usually used by academic users. 8 or 16 bit, specifies an arbitrary sampling rate.
	.voc	Older format, .wav files are far more common. Used mostly in IBM machines. It samples in relation to an internal clock. It is not a flexible format.

Musical Instrument Digital Interface	MIDI	Good for instrumental music. The files plays digitally stored samples of instruments, which are located on a sound card.
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Figure 4: Adapted from "NINCH Guide to Good Practice: Audio/Video Capture and Management," <http://www.nyu.edu/its/humanities/ninchguide/VII/> and the Colorado Audio Working Group, 2003).

1.9 Compression

Compression is another factor in the audio digitization process. Because audio file sizes tend to run rather large and take up a lot of space, it is important to conserve space when possible while retaining the best quality information. Compression formats are divided into two types: lossy and lossless. Lossy compression actually removes some information from the original audio file in order to make it easier to compress. Lossy compression is popular because it allows for smaller file sizes and often a user is unable to tell the difference in sound between a lossy compressed file and an uncompressed or lossless compressed file. On the other hand, lossless compression is a method of compressing audio in such a way that no information is lost. It is statistically identical to the original sound file. (Digital Audio Primer, pars. 8,9).

Compression methods are often referred to as Compressor-DECompressors (CODEC). According to the University of Wisconsin’s Division of Information Technology (DoIT), “there are many CODECs and they use a variety of technologies to reduce the file size of the video and audio files while attempting to keep the quality as good as possible” (DoIT, 4). A CODEC uses an algorithm to compress data and then decompress it again. According to the Digital Audio Primer, “some CODECs are implemented in software, some in hardware, and some are limited in their functionality” (par. 5). It is important to note that the process of compressing and decompressing files

is not symmetrical. More specifically, it often takes longer to compress a digital file than to decompress it for playback (Digital Audio Primer, par. 6).

Many CODECs are proprietary. It is important to keep in mind that if files are compressed with proprietary hardware, the appropriate software must be obtained to play the file on the computer. According to DoIT, some popular compression software tools today are as follows:

1. Discrete Cleaner 5.0
2. Quick Time Pro
3. RealProducer - Mac
4. Helix Producer Pro – PC
5. Windows Media Encoder

1.10 Delivery

Once audio has been digitized it needs to be converted into a deliverable format for public access. There are several good delivery methods available today, which are briefly discussed below. The best delivery options will be based on current technology and the ways in which each institution can best meet the needs of their users.

Four types of delivery methods are discussed in the “Audio Video Series: Delivering Multimedia Content to Students” provided by DoIT. The following common delivery methods as well as certain strengths and weaknesses are listed below.

1. Downloaded delivery - - files are downloaded to a user’s computer.

Strengths

- a. The quality of downloaded media is not affected by connection speed or bandwidth whereas the quality of streamed media is dependent on the connection speed.
- b. Users have the option to listen to media as it is being downloaded. This option is known as progressive download. However, if the players do not keep up with the download, play may be choppy.
- c. Once the files are downloaded, the user can listen to the files repeatedly at any time.

Weaknesses

- a. Download time can be time-consuming especially for users with a 56k or slower modem. (i.e. A 20-30 mb clip can take hours for a 56k modem user to download.)
 - b. Because the files are downloaded to a user's computer, they can easily be distributed to others as the user so desires, which may lead to copyright infringement. (i.e. Napster).
 - c. Downloaded files consume large amounts of space on the hard drive.
2. Streamed delivery - - files are delivered with no waiting time. Content is "streamed" or sent immediately across the internet to the user's computer and is deleted as soon as it plays.

Strengths

- d. Streamed audio is delivered with no waiting time. Although an initial synchronizing and buffering time usually occurs, sound can usually be heard within 30 seconds.
- e. The streaming server can create multiple versions of content based on the connection speed.
- f. Streaming allows for access control since the audio files are deleted immediately after they are played and cannot be downloaded to a user's computer.
- g. Streamed files do not take up space on the hard drive.

Weaknesses

- a. Streamed audio is delivered based on "connection capacity," which means multiple audio versions for different connection speeds are recommended otherwise users will have dropped frames and/or no audio (DoIT, 2).
- b. If multiple versions of audio are created, sometimes the quality of the streamed audio is especially poor for slower connections.
- c. Because streamed audio has a built-in buffering and synchronization that occurs with each media file, every time a new section or file is listened to, the server must rebuffer and synchronize to the computer.
- d. Because streamed files reside on another server, reference or pointer files are needed to get to these files causing more steps in the workflow.

- e. An Internet connection is required to listen to streamed audio.
3. CD-ROM delivery - - files are written to a CD-ROM and delivered to the users.

Strengths

- h. Writing audio to CD-ROM ensures that the user has access to a high quality version of sound.
- i. An Internet connection is not required for any steps in the process.

Weaknesses

- a. Content that is written to CD-ROM is hard to update and deliver at regular intervals.
- b. Because content is housed on a CD-ROM, it is difficult to “embed media inside content. Users may need to find the files associated with each class and view them outside of the normal Web content.”
(DoIT, 3).

4. DVD delivery - - files are written to a DVD.

Strengths

- a. DVD authoring software can be utilized to create an interface to present the material.
- b. DVDs provide high quality video media as well as audio.

Weaknesses

- a. According to DoIT, “the install base of DVD players is not large enough to assume that all of your [users] will have access to one” (3).

- b. DVD authoring is “cost-prohibitive.” Presently, DVDs cost \$10 per DVD while CDs are usually less than \$1 per CD (DoIT, 3).

1.11 Preservation

According to the Library of Congress (LC), “electronic formatting has been an inescapable part of audio and video preservation programs for several decades” (par. 4). In order to retain content, sound recordings have been copied from one tape to the next over time. Due to the short life span of tape media, preservation professionals have searched for various venues that will preserve sound. One such venue is digital preservation. Presently, many tape media and formats traditionally used for audio reformatting activities are no longer available (LC, par. 8). LC notes, “the condition of tape recordings from the 1960s and 1970s has reached a near-crisis state” (par. 8). Yet, at the same time, technology in the digital sphere has been steadily developing suggesting “the current environment is conducive to the adoption of computer-based digital reformatting” (LC, par. 8).

While digital preservation seems like a viable option, there are several issues that should be considered. Concerns associated with digital preservation include the need for content to be migrated to new systems and media periodically as technology changes. Periodic migration is time-consuming and costly. A further issue outlined by LC is that “some complex data forms may not be migratable and, in order to continue their use, the...obsolete systems or environments in which they function must be emulated” (par. 6). Emulation poses problems again in areas of cost as well as access issues. Thus, while acknowledging a great interest in digitizing audio, LC remains cautious in relying on digitization as a means of preservation (par. 8).

Because digitization is not currently in a stage that allows for optimal preservation, it is necessary to retain copies of the original item in analog form.

PART II: DCG PRACTICES

2.0 Selection.

Perhaps the single most important step in the selection process has to do with the subject specialist. It is important to identify a subject specialist who will in turn identify the items that will be captured. The Digital Content Group (DCG) does not decide which items will or will not be digitized. Rather, the DCG can only make recommendations based on the physical quality of the material. For instance, the DCG will be able to ascertain if the original item will be a good candidate for digitization based on its current physical condition. According to Preservation Microfilming: A Guide for Librarians and Archivists, “decisions to [digitize] or not to [digitize], to retain or discard the original after [digitization], and so on, must be based on the [subject specialist’s] knowledge of the subject, the language, and scholarship in the field” (Fox, 76).

Once a subject specialist has identified materials for selection, he or she must go through the proposal process outlined by the DCG. According to “Project Development,” “The process consists of two steps culminating in a brief Memorandum of Understanding. Initially, in this form, you are asked to provide the UWDCAC/DSC with general project management information and to write a synopsis of your idea for submission to the appropriate steering committee. After an initial review by UWDC staff to ensure that the project can be undertaken in accordance with UWDCAC/DSC expectations, you will be asked to develop a detailed proposal assessment and cost estimate form for UWDCAC/DSC review. Your idea must have the support of your

library director (or designee) to be considered. The UWDCAC/DSC weighs various criteria in determining which projects to fund and/or develop” (University of Wisconsin Digital Collection Center, screen 1).¹

Issues that both the subject specialist and the DCG should consider are listed under “Guidelines.”

2.1 Guidelines:²

1. What are the physical characteristics of the tape? (i.e. wax cylinder, reel-to-reel, speed at which the original object was recorded) Consider also such things as length, format, content, condition, age, and quality.
2. Will an attempt be made to preserve the original tape after duplication? Consider appropriate format for archival master.
3. Who will test the tape for sound quality? Will digitization be used as an enhancement quality?
4. What is the time frame?
5. How is the project being funded?
6. Who will be responsible at different stages of production?
7. How will the institution perform the actual transfer?

PART III: WORKFLOW

3.0 Check in

The following steps are to be taken when checking audio formats into the DCG:

1. Review and check in the original material with the inventory record from the

¹ University of Wisconsin Digital Collections Center (<http://uwdcc.library.wisc.edu/develop.html>).

² Selection guidelines are adapted from the Colorado Digitization Project Digital Audio Guidelines, May 2003.

owning institution. (i.e. The African audio has its own record keeping system.)

2. DCG staff will assign an internal ID to audio material.
3. Information will be entered on an internal tracking sheet that has been designed to reflect information needed to track the workflow of the reformatting of the original item. This sheet will be adapted for specific project needs.
4. Audiotapes will be stored in one of two places. Depending on the size of the collection, it may be stored in Special Collections on the 9th floor of Memorial Library. If the collection is smaller, it may be stored in the locked file cabinet located in the DCG.

3.1 Work Scene

Because the DCG is growing and changing with technology, there are several possible scenarios that could occur when it comes to the work scene. The following scenarios are examples of past, present, and future work scenes.

3.11 Scenario 1: One full time equivalency (FTE) Research Intern will supervise 2 student audio technicians. The student technicians will be handling the recorder workstation in real time with one tape having two sides with each side having 6.5-7 hours of recorded sound totaling 13.5 – 14 hours per single tape. This will take approximately 16-22 hours of work time per tape for duplicating/converting the analog tape to a digital format. This time will include the dedication of one workstation that will be monitored by one student technician switching with another student technician throughout the week to maintain an 18-hour workweek.

3.12 Scenario 2: Playback of the original tape at 2 or more times the speed of the original. One tape with seven hours of recorded sound on each side representing 14 hours could be captured in double speed by one technician using one deck in about 9.5 hours of work time. (The tapes we are currently working on have approximately 3 hours on each side. Later in the projects, we expect to encounter tapes with 6 to 6 ½ hours per side. These will be played back at 4 times the original speed.)

Note: DCG recorders are capable of the following speeds: 15/16, 1 7/8, 3 ¾, and 7 ½

Example: 3 ¾ quadruples the speed so seven hours of recorded tape is reformatted in 1 ¾ hours.

3.2 Pre-Capture

The type of project being digitized by the DCG determines the bit-depth and samples per second used for capture. For instance, if the project is primarily an access project, the audio may be digitized at standard CD quality. However, if the project is a preservation driven project, the audio may be digitized at 48 kHz and 24 bit-depth. (These figures may vary as standards change over time.) Although it may seem that there is not a vast difference between the two areas of capture with each project, information is captured in the most compatible way. For instance, there is a different standard for every situation in which digital sound is used – primarily speaking in terms of samples per second.

3.3 Example of Current Capture Practices at the DCG

For the Harold Scheub African Storytelling tapes (Scheub project), the DCG captured at 24-bit depth to reduce distortion and 96,000 samples per second or 96 kHz. Once again, the bit-depth and sampling rate will vary according to the needs and

demands of each project. For the Scheub project, the DCG recorded at a faster rate than the original rate of the object to save time during capture. After capture, it was then necessary to change the rate of playback from 96,000 to 48,000 in order to keep the sound from being distorted. Therefore, when the DCG captures sound at 96 kHz, the audio technician must tell the computer to treat the sound as if it were captured at 48 kHz for playback.

3.4 Post-Capture

After the analog audio waves have been successfully digitized, the files are edited and saved in minor ways before uploading to the server. See the following set of steps for workflow instructions.

1. Briefly listen to the digital copy of the audio to ascertain if any noise or other interference has distorted the audio. If so, the tape must be re-recorded.
2. Delete empty frames at the beginning and end of the audio wave form using Peak software.
3. Save the files on the Macintosh hard drive. See the Audio Procedures guide for instructions on editing and saving files using Peak software. Both Macintosh computers (A and B) can hold up to 28 GB in the Project Files drive. It is time to transfer files when only 3 GB remain free in the Project Files drive. (**Note** *Although it has been done before, it is unknown how often or how advisable it is to regularly defragment the Macs.*)
4. Transfer the files from the audio Macintosh machines (Audio Macs) to the Macintosh running OS 10 (UNIX driven operating system). The computer currently running OS 10 is FTHORES. FTHORES is connected to

the Audio Macs through a direct fire wire interface that uses FTHORES as the target drive and plugs into either Audio Mac. The fire wire is used because all networking components have been stripped off the Audio Macs to ensure minimal interference during the digitization process. FTHORES has two 80 GB drives for storing audio files until they are uploaded to the server.

IF THIS IS A PRESERVATION PROJECT, THE FOLLOWING STEPS MUST OCCUR:

5. From FTHORES, transfer files to the server (LIBELLA). A directory (*/db/stage*) is set up on LIBELLA to receive the audio files. The files are uploaded in batches of 30 GB because 30 GB is the amount that will fit on the Digital Linear Tape (DLT) drive that the audio files are being backed up to.
6. Tell the Library Technology Group (LTG) that 30 GB of files are now uploaded on the server.
7. LTG informs DoIT that files are ready to be backed up. DoIT backs up the files making three (DLT) tapes with identical information. One copy is kept in remote storage, one copy is stored on site in LTG in an 18-hour fire safe, and one copy is kept at DoIT. The copy kept at DoIT will be sent to remote storage as well once it has been determined that none of the files are immediately needed.
8. DoIT informs LTG that files have been backed up to DLT.
9. LTG deletes the files from LIBELLA leaving the directory empty and ready for uploading another 30 GB as necessary.

10. In the future, if the DCG decides to make this material available for access rather than preservation only, files can easily be recovered from DLT. Deterioration of DLT has not yet been addressed within the LTG since the tapes have just been created, but a check/refresh schedule will be implemented once the hardware environment has been expanded and configured.

IF THIS IS AN ACCESS PROJECT, THE FOLLOWING STEPS MUST OCCUR:

5. Utilize the Cleaner software, a batch-processing tool that allows .aiff audio files to be converted into streaming audio. The files will be compressed into REAL audio streaming files for access.
6. Contact LTG to set up the directory on the server (GLS-NT3), which will hold the streaming audio.
7. The streaming audio files are then transferred to the streaming server (GLS-NT3). The metafiles are kept on the DCG server (Libtext) and serve to point to the exact location of the streamed audio on GLS-NT3. (A **metafile** serves to redirect the browser to the actual audio file. The metafile, a **.ram** file uses rtsp (Real time protocol instead of http protocol to link to the .ram file on GLS-NT3).
8. A back up file of the audio will be created and stored on CD within the DCG if the DCG suspects they will need a readily available copy of the audio on hand. CDs would not be used as an acceptable form of preservation or storage for the audio files. Their intended use would only to provide the DCG with immediate access to the audio files. The CDs used by the DCG will hold up to 650 MB. Naming schemes follow a “logical” pattern. Names may be

assigned according to Project_ID, Folder name, Original or Derivative, and disc number. (i.e. Africana_Jesuit01_Orig_001) In the future, there may be a move to store sound files for immediate access on DVDs rather than CDs.

3.5 Metadata practices

Last, but not least, the DCG uses standard Dublin Core fields as well as some locally created metadata fields to capture metadata for audio files. Up to 27 Dublin Core fields can be used for metadata capture, but fields for each project are usually determined by project owners/content providers and the DCG. The DCG will meet with project owners/content providers to determine what information is available with the audio file and what should be included in the metadata according to the project owner/content provider's specifications. This data is then mapped into Dublin Core format. For one audio project, the Belgian-American Research collection, the following Dublin Core Fields were used. (DC denotes Dublin Core.)

Dublin Core Fields

DC_Title - - The name given to the Resource by the Creator or Publisher. If unknown, a descriptive title may be assigned by staff.

DC_Title_Other - - A title other than the main title, such as a translated title or a variant of the title.

DC_Creator - - The person(s) or organization(s) primarily responsible for the intellectual content of the Resource.

DC_Contributor - - The person(s) or organization(s) in addition to the Creator who have made significant intellectual contributions to the Resource but whose contribution is secondary to that of the Creator.

DC_Subject - - The terms, phrases, or classifications used to provide topical access to the Resource.

DC_Description - - A textual description of the content of the Resource.

DC_Publisher - - The entity responsible for making the Resource available in its present form, such as a corporate publisher, a university department, or a cultural institution.

DC_Date - - The date the Resource was made available in its present form.

DC_Type - - The nature or genre of the content of the Resource, such as text, image, physical object, or collection.

DC_Format - - The physical or digital manifestation of the Resource.

DC_Identifier - - An unambiguous reference to the Resource within a given context.

DC_Identifier_LocalID - - Identifier for the Resource; must be unique at least within the scope of the collection.

DC_Source - - A reference to a Resource from which the present Resource is derived.

DC_Language - - The language of the intellectual content of the Resource.

DC_Relation - - A reference to a related Resource.

DC_Relation_OtherFormat - - A reference to a related Resource. The described Resource contains the same intellectual content as the referenced Resource, but is presented in another format.

DC_Relation_IsPartOf - - A reference to a related Resource. The described Resource is a physical or logical part of the referenced Resource.

DC_Relation_HasPart - - A reference to a related Resource. The described Resource is a physical or logical part of the referenced Resource.

DC_Coverage - - The spatial location or temporal duration characteristic of the Resource.

DC_Rights - - Information about rights held in and over the Resource.

DC_Rights_Ownership - - Information about owner of rights held in and over the Resource.

DC_Rights_Terms - - Information about the terms and conditions for use of the Resource or Media Object.

The DCG also uses several locally created fields. These fields are unique to the DCG's specific metadata needs. See below.

Locally Created Fields

Note - - Information about the Resource (or its surrogates) not intended for Resource discovery. Use only for internal notes not appropriate for other elements.

Submitter - - The personal or corporate name of the submitting agency.

Mediastring - - Digital surrogate for Resource. This information links the Resource metadata record and the digital surrogate. See example below.

MediaRights_Terms - - Structural and administrative information pertaining to a digital representation of the Resource. This information is often provided by DCG.

Update - - This locally created field contains the date that the metadata record for the Resource was most recently updated. Updates include corrections to titles, authors, etc.

Note The mediastring contains specific metadata about the audio file. The media string serves to point the user to the sound file and is delimited with semi-colons. See the following example for a breakdown:

belgian/belgian1/ap1a12;1;r;audio/x-pn-realaudio;

1. **belgian/belgian1;** - - refers to the directory structure or where the sound file lives.
2. **ap1a12;** - - is the file name.
3. **1;** - - tells the audio client that there is only one sound file.
4. **r;** - - denotes the presence of a “reference” file or a .ram metafile. (The URL for the metafile is displayed on the web browser). The metafile has its own URL that the web client is directed to. However, the metafile also contains a URL within it. When the web client comes to the second URL embedded within the metafile, it automatically launches the audio client. Once launched, the audio client will automatically follow this second URL to where the actual sound file is stored and begin playing it.
5. **audio/x-pn-realaudio;** - - denotes the format of the file. This segment of the mediastring is the MIME (Multi-purpose Internet Mail Extension) type for the file that contains the base media type followed by the more specific media type.

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