

## Shakespeare's Globe Archives (March 2016) Case Study: Digitising our moving image collection

### **Why the project was happening**

Our moving image collection is a unique record of our performances here at Shakespeare's Globe Theatre. From our first archive recordings of the Prologue Season in 1996 until our current season we continue recording today in both our outdoor and indoor theatre. We have captured virtually all our productions since the beginning, recording several performances of each play from multiple angles to reflect the development of each production. We don't edit our recordings and we use multiple static cameras, each placed at different angles to the stage, the intention being to create an objective view from different vantage points. The quality of our recordings is widely varied, we use the light available on stage and our best recordings are generally our matinees during daytime hours. Because of our equity agreement our archive recordings can only be viewed in house and are mainly accessed in the Library and Archive on desktop terminals via DVDs. From 1996-2007 we recorded onto VHS and SVHS creating a collection of nearly 2000 VHS and SVHS tapes.

### **What we did**

I started work here in 2011 and I made DVDs of each of the VHS and SVHS tapes creating access copies of our performance archive. To make the DVDs I used 3 x Panasonic DMR-EZ48V VHS/ DVD recorder that had been purchased by the Globe for the project before I had arrived. In addition I bought a couple of second hand SVHS players (one professional player with built in Time Base Corrector: Panasonic AG7650 Video Cassette Player) to enable better quality playback of the SVHS tapes and to generally improve the playback quality.

I encoded the DVDs to MPEG4 using the WinX DVD Ripper software (Video Codec: H264-MPEG4 AVC(Part10)(avc1) and AudioCodec: MPEG AAC Audio (mp4a)). These files are stored on a small server (5TB).

In 2008-2009 the Globe recorded their performances onto approx 800 DVDs, each play spanning over 3 DVDs. I encoded these as well to MPEG4 using the same WinX DVD Ripper software and the files are stored on an archive server.

From 2010 onwards our recordings are created as digital files, originally as wmv files and now as mov files (ProRes Lt). I encode all files to MPEG4 to create viewing copies and we save both the originals and the MPEG4 files on our archive servers which are presently being backed-up onto LTO6.

It is now possible to view our entire collection of moving image recordings dating back from 1996 to today. In future we would like to have a selection of our original VHS/SVHS collection digitised to an archival standard. We want to view and select the best quality recordings and then have these digitised to a higher standard than the files I created from the DVDs.

## What's next?

### Audio digitisation

I am currently digitising our Audio collection of lectures and talks from our education department dating back to 1987. The recordings are made on DAT and audio cassettes. For the audio digitisation I am using a Mac Pro 5.1 (with Quad-Core Intel Xeon Processor) with 12 GB Memory.

We already had a TASCAM DA-20MII DAT player/recorder and we bought a new TEAC Double Auto Reverse Cassette Deck W-890R MKII. I had purchased a second-hand audio cassette deck on Ebay that had Dolby facilities but that sadly broke down. We were unable to purchase a new Cassette Deck with Dolby facilities within our price range (approx £200), so settled on the TEAC W-890R MKII which has no Dolby facilities but is otherwise a good quality cassette player.

The cassette deck is connected via its LINE OUT to the analogue LINE INPUT (1/8 inch Jack) of the Mac. The digital output of the DAT player is connected to the S/PDIF Optical Audio Input of the Mac.

I am using Audacity 2.1.2 (free, open source, cross-platform software for recording and editing sounds) for capturing the audio and doing simple edits and processing. I save our digital audio files as wav and mp3 files. Wav (Waveform Audio File Format) is Microsoft's native uncompressed audio format for Windows and is our preservation copy and mp3 is the compressed viewing copy.

### Sample rate and bit depth for DAT conversion

Digital Audio Tapes (DAT) are magnetic tape cassettes that store audio in a digital manner. DATs use lossless data compression which makes a lossless transfer possible. When doing a digital-digital conversion it is important to match the original sampling rate and sampling size. "All audio transferred from digital carriers should retain the sample rate and bit depth (word length) of the original digital audio".<sup>1</sup>

Sampling at a higher rate or depth may introduce noise to the file.

Digital Audio Tapes allow four different sampling rates: 32 KHZ at 12 bits quantization, 32 kHz, 44.1kHz & 48kHz at 16 bits. Some DATs could record a bit-depth of 24 to 98kHz but this was rare.

The digital files created from the DAT recordings should be captured at their original settings, which is likely to be 44.1 or 48 kHz and 16-bit depth.

Most of our DAT recordings were made at 44.1kHz at 16 bits so I select the project rate of 44.100Hz in Audacity when creating 16bit Wav and mp3 files.

### Sample rate and bit depth for audio cassette digitisation

Our audio cassette recordings are mainly speech, lectures and talks. The recordings were made on limited-bandwidth analogue equipment that was not capable of capturing any of the high frequency information in the first place. The audio bandwidth of old cassettes will struggle to reach 20kHz.

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<sup>1</sup> Endangered Archives Programme Guidelines for the preservation of sound recordings (British Library)  
[http://eap.bl.uk/downloads/guidelines\\_sound.pdf](http://eap.bl.uk/downloads/guidelines_sound.pdf)

According to the Nyquist Theorem, a principle that audio engineers follow in the digitization of analogue signals, the sampling rate must be at least 2 x highest frequency or twice the highest analogue frequency component.

“Audio sampling is based on the Nyquist-Shannon sampling theorem, which concludes that for accurate signal reconstruction the sampling frequency should be at least twice the bandwidth of the source signal”.<sup>2</sup>

The sample rate (frequency) is the rate at which the samples are captured or played back, measured in Hertz (Hz). It is thought that the “maximum essential frequency audible to the human ear is approximately 20KHz. ...therefore in audio systems the minimum sampling rate needs to be approximately 40KHz”<sup>3</sup>. If the sampling frequency is not at least twice the bandwidth of the source signal then aliasing (distortion) can occur.

We’ve decided to digitise our audio cassettes at 16-bit/44.1kHz. I feel that this will capture all there is to capture, and provide easy compatibility with whatever storage or replay format we are likely to want to use.

The ‘Endangered Archives Programme Guidelines for the preservation of sound recordings’ (British Library) recommends that “All audio transferred from analogue carriers should be digitised with a minimum sample rate of 48 kHz (96 kHz recommended), and a 24 bit word length.”<sup>4</sup>

Mathilde Blum

Shakespeare’s Globe Archives Recordings Co-ordinator

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<sup>2</sup> “An Introduction to Media”, <http://www.jiscdigitalmedia.ac.uk/audio/advice/an-introduction-to-digital-audio/>

<sup>3</sup> “An Introduction to Digital Audio”, <http://www.jiscdigitalmedia.ac.uk/audio/advice/an-introduction-to-digital-audio/>

<sup>4</sup> “Endangered Archives Programme Guidelines for the preservation of sound recordings”, [http://eap.bl.uk/downloads/guidelines\\_sound.pdf](http://eap.bl.uk/downloads/guidelines_sound.pdf)