UOSH: Archives+ Audio Digitisation Guide

401: Audio Cassette Digitisation User Guide

This guide is for running a serviced Tascam 122 MkIII cassette deck

via a PrismSound Titan analogue-to-digital converter

into a Windows PC running Wavelab Pro 10. V1 Sep 2023

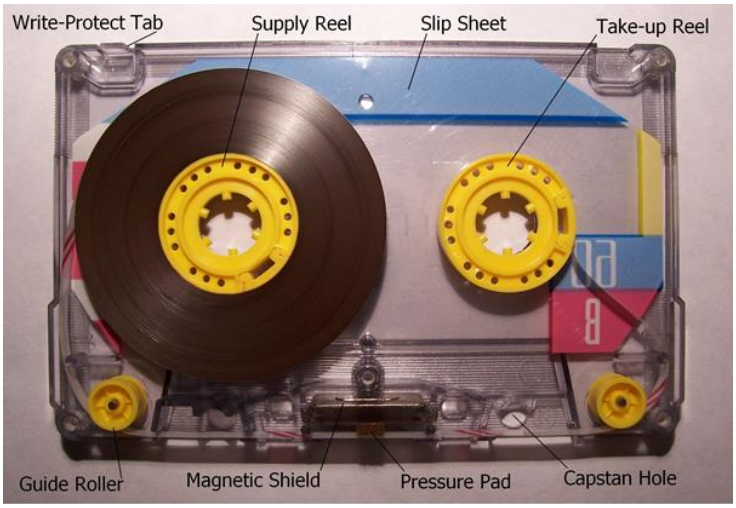
4.1. Identification of Audio Cassettes

[NOTE: Jump to 4.2 to skip the introduction and head straight to step-by- step digitisation].

4.1.1 Types of Audio Cassette

Audio Cassettes come in a variety of types and length including:

C-30 (15 mins per side), C-60 (30 mins per side), C-90 (45 mins per side), C-120 (60 mins per side etc.



From:https://en.wikipedia.org/wiki/Cassette_tape#/media/File:Memorex_Compact_Cassette_opened.jpg

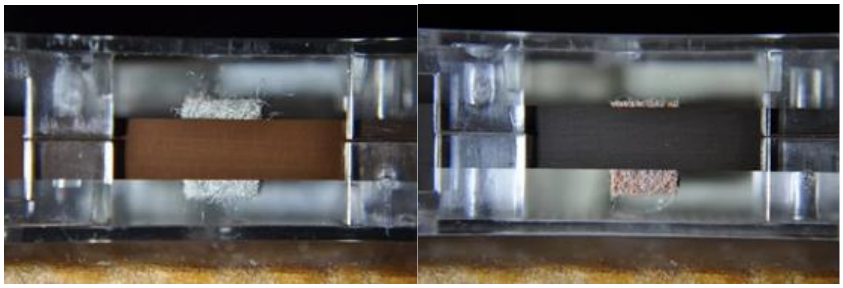
Different types of tape were introduced over the years in the following order:

- Type I: **Ferric Oxide** – had poor high frequency response and excessive tape hiss. This is standard ferric-oxide tape, also referred to as **"normal bias tape"**
- Type II: Chromium Dioxide – better frequency but not as good on low end and output levels. This is called '**chrome**' or CrO2 tape.
- Type III: A Ferric Oxide and Chromium Dioxide mix – slightly improved low frequency response. Called '**ferric chrome**' and now obsolete.
- Type IV: Pure metal particles – increased dynamic range and frequency response but increased head wear. Referred to as a **'metal tape'.**

TOP to BOTTOM:FROM Type 1 Normal, Type 2 Chrome & bottom two Type 4 (one with the safety tabs removed).

From:https://en.wikipedia.org/wiki/Cassette_tape#/media/File:Memorex_Compact_Cassette_opened.jpg

In 1979 notches for automatic tape type recognition were introduced. Most modern cassette recorders sense the tape type by the holes in the back of the housing and adjust bias andequalization to compensate for the differences.



Left image Brown = ferric (TYPE I), Right image Black= chrome (TYPE II)

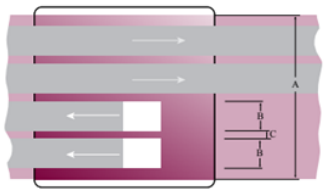
(Image from https://www.instructables.com/id/Cassette-Tape-1101-an-in-depth-look-into-this-an/)

You can also tell the difference between TYPE I (ferric) and TYPE II (chrome) in the colour of the tape itself.

4.1.2 Tape geometry

A stereo Audio Cassette effectively has 4 bands that carry the signal on the outer side of the magnetic tape.

The two bands at the top (no matter whether playing side A or B) play traveling in the right direction. Look at the image below and imagine flipping it upside down when you turn a tape over in the tapedeck:



	A	B	C
IEC Philips	3.81 mm, (0.15 in)	0.6 mm, (0.02 in).	0.3 mm (0.012 in)

Fig 4 Section 5.4 Stereo Cassette head configuration and dimensions.

From Page 55 TC_04 Second Edition.pdf

The above shows the typical four audio tracks. Side A stereo track has two parts then Side B has two tracks for when you turn the cassette over. There is a small gap where no information is recorded so as to prevent ‘crossover’.

The tape travels at the standard tape speed of 1 7⁄8 ips which is 4.76 cm/s.

4.2 Preparation

4.2.1 Cassette condition

Identify what type of audio cassette you have and if there are any conservation issues first. Does it wind OK? Is the tape look flat? Does the pad behind the tape sit flush? If you have any questions or concerns, please email images to archiveslocalstudies@manchester.gov.uk for advice.

4.2.2 Setting up your equipment

Connect the cassette deck to the Sonifex box using female XLR to TRS leads. Insert the XLR leads into the OUTPUT ports on the back of the cassette deck to the INPUT ports on the converter (black = left channel = top of screen; red = right channel = bottom of screen). Plug in the cassette deck and converter.

Install and run the PrismSound Titan and Wavelab software.

4.2.3 Cleaning

At the start of each day do a thorough clean of all the heads and metal rollers starting left to right to clean the dirt and oxide deposits that have built up. To do this use a lint-free cotton bud soaked in 95+% isopropyl alcohol. Between each tape pass, before digitisation, clean the play head. You may also need to clean the whole tape path between passes depending on how much residue you see appearing after each pass. Always take a look. The curved top side of the playback head must be cleaned if there is any build up.

Image of how to slide the door off the Tascam

Clean the black rubber pinch rollers very gently with Platenclene printer drum cleaner (not alcohol!) about once per week.

Image of how to clean a tape deck playback head with cotton bud.

4.2.2 Playback

Rewind the tape and press the reset but to set the timecode counter to zero.

4.2.3 Dolby Noise reduction

Image of Dolby NR switch

If there is a dolby noise reduction system listed on the tape or listing /case inlay set the dial on the tape player to the correct setting.

IF IN DOUBT LEAVE DOLBY OFF (the use of future plugins like Saturn could in theory apply DNR in edit stage:

For recordings made without Dolby NR, set to "OFF". For recordings made with Dolby B NR, set to "B". For recordings made with Dolby C NR, set to "C". For recordings made with Dolby S NR, set to "S".

4.2.4 Bias tape frequency

The correct bias signal should be automatically identified by the cassette deck using the notches on the cassette housing. You do not need to adjust anything.

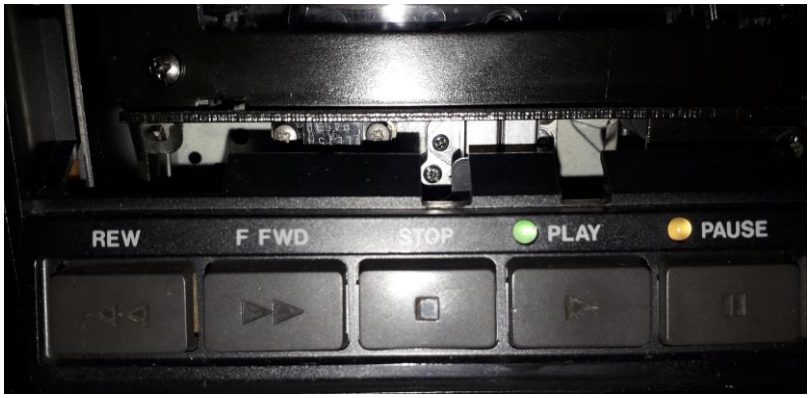
4.2.5 Tape Speed and Pitch

Audio cassette tape travels at the standard tape speed of 1 7⁄8 ips which is 4.76 cm/s. Place the pitch control in the OFF position.

4.2.6 Azimuth

The azimuth is the angle that tape head recorded the original audio signal onto the original audio carrier. To digitise the full range of frequencies from the original recording we need to manually match up the azimuth angle each time so that it matches the true horizontal alignment of the tape path. Sometimes the azimuth can change throughout the recording so we must digitise versions at more than one setting.

Press play on the tape. Once the tape is playing the position of the transport will allow you to enter a small Philips screwdriver into the azimuth modification screw (the bottom screw visible just above the O of the STOP button (as photo below/overleaf).



IMPORTANT - do not push the screwdriver into the hole before/as you press play. This will cause unnecessary pressure on the transport and potentially damage the tape deck. Only adjust azimuth during playback.

Image of using a small Philips screwdriver to manually adjust the azimuth

Using headphones on both ears, make an assessment whilst gently turning the screwdriver (paying careful attention that left and right channels of the headphones are on the appropriate ears).

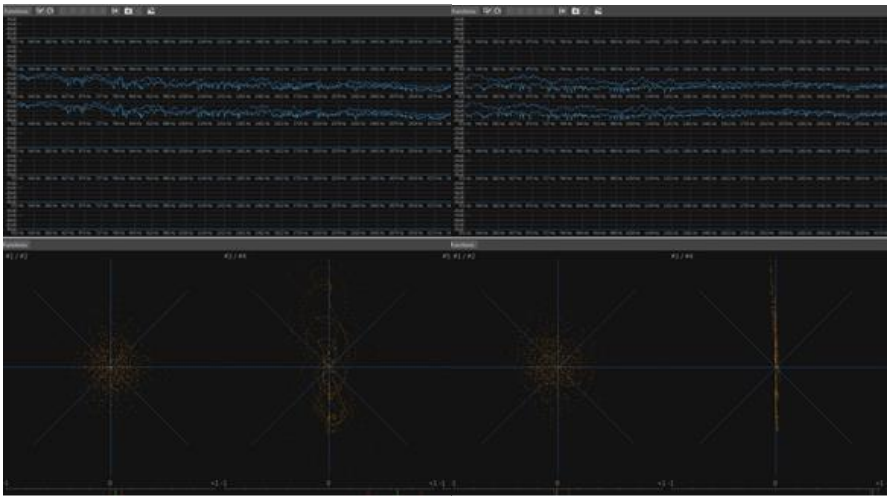
Be careful not to turn the modification dial too much as it will lose its thread and could damage the tape deck.

This process is a bit like the audio equivalent of focusing a camera lens. There will be a sweet spot where you can hear the most high frequencies and hiss. Either side of that spot the sound will grow more muddy and flat. If you're struggling to make a judgement select a part of the part with audience applause or a high frequency instrument like a violin.

Adjust and refine the position listening for when you can hear the most of the recording and the higher frequencies like clear speech, sibilance and hiss. The sound will 'roll off' from the top end frequencies when adjusted to either side of this point and you will hear a muddier sound where the high frequencies are lost.

4.2.7 Adjusting Azimuth (continued) with Wavelab's Phasescope

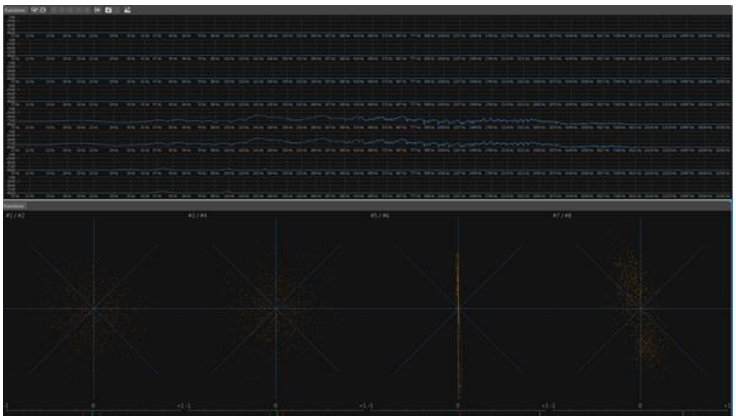
As part of making this decision use the Phasescope monitor on Wavelab:



The easiest way of opening up Wavelab for analysis and recording is to double-click on a wav file on your PC. In Wavelab set your scopes to analyse **Audio Input** using the **ANALYZE** menu tab:

In the **Analyse** menu select **Audio Input** and this will connect the scope to the live feed from the tape decks.

Set up your scopes on Wavelab Pro to show in one full monitor the **Spectrometer** (pictured below in the top half of the monitor) and **Phasescope** (in the bottom half)**:**



The Phasescopes above shows 4 live images, from left to right the two Denon cassette tapes decks on lines 1 & 2 and 3 & 4 and the two Studers on lines 5 & 6 and 7 & 8.

When you make the adjustment on a stereo recording you will see (left image) what looks like wide fuzzy waves around the central vertical line when the azimuth is out of alignment and you are not hearing the higher frequencies (ie. muddy sound).

The right image shows a narrower pattern moving as close to a central line as possible. This is where the azimuth is set as best as possible.

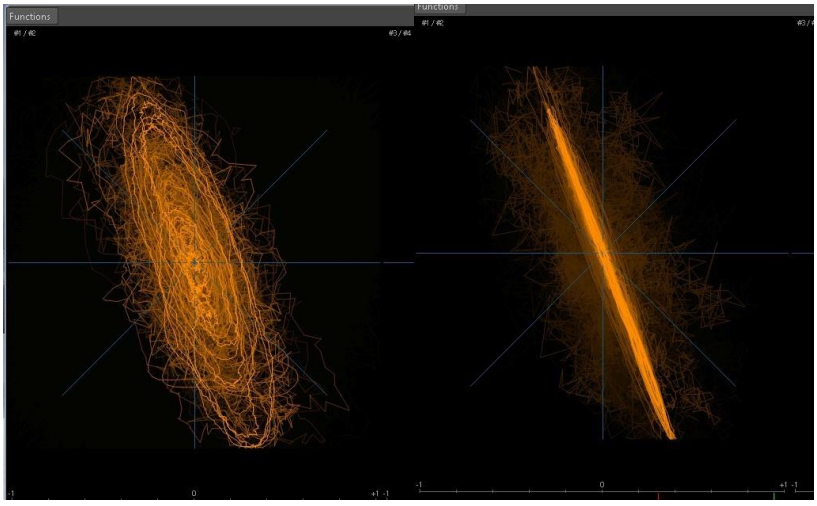
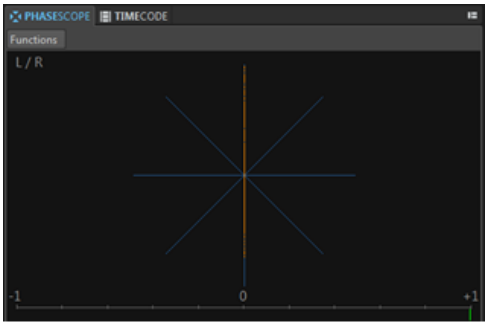


Image on the left is a stereo signal with the azimuth out, and on the right with it correctly set

This image below shows a perfect mono signal where there is the same audio information on both left and right sides. A correctly set azimuth on a stereo recording will never appear as neat a line as this:



Listen to at least 3 points on the tape to make this judgement. Once we have the correct azimuth set up and the tape is rewound back to the start, we are then ready to digitise!

4.2.8 Loudness Levels

Peaks of -12dB are ideal. If your audio is peaking at less than -12dB in Wavelab's loudness meters, you may wish to turn the cassette deck GAIN LEVEL up. However, peaks of more than 0dB are very bad news – this is known as clipping and causes irreparable damage to the digital sound wave.

If your audio is peaking at -6 or above, you can turn the cassette deck OUTPUT LEVEL down. Make a note of any gain dial setting in your digitisation workbook.

And always remember to bring the OUTPUT level back to your preferred default setting before your next tape or recording! And be aware that any gain change settings on your cassette deck will be extremely difficult to replicate if, for example, if you later want to re-do the transfer. It is best to avoid adding gain if at all possible. You can always boost a higher level version for access later.

4.2.9 Frequency and bit rate

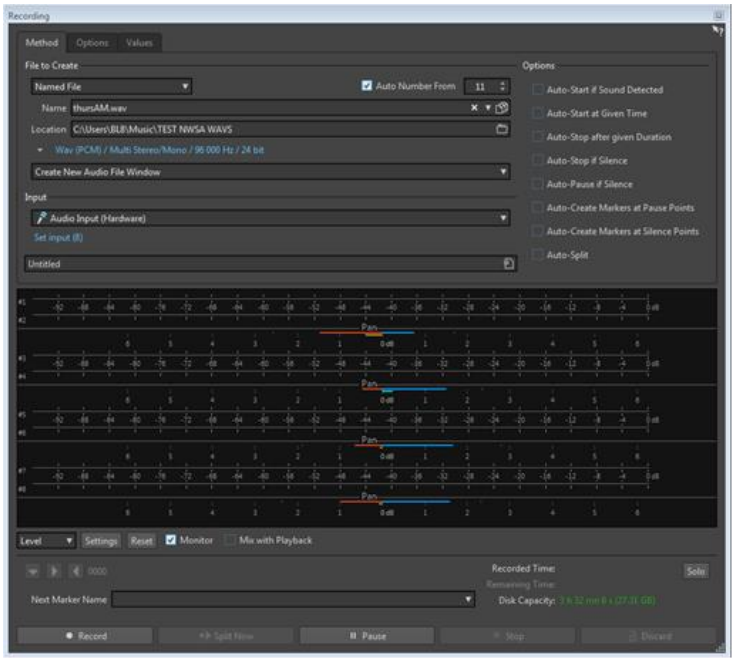
Make sure that your converter and Wavlab are both set to record at 96,000 Hz and 24 bit depth. This means the computer will be taking a sample of the sound from your cassette deck 96,000 times a second. CD quality is 44,100 Hz. Higher bit rates mean bigger digital files and more scope for digital manipulation afterwards.

Bit depth sets the range of data included in each sample. 24 bits gives plenty of scope for the computer to capture enough data from even very quiet recordings.

4.3.1 Digitisation - Recording playback

Either open any wav file with Wavelab to see this controller pictured above, then press the record button at the bottom of Wavelab (highlighted yellow above) and the recording menu will appear.

Or just press the shortcut which is the asterix * key (SHIFT + 8)



In the location box, either use G:\ALL UOSH RAW or create a folder to save the recording (raw capture file) and

As well record from 4 tape machines at the same time, it is advisable to use a naming method such as below with (1) being the tape set up in the top Denon tape deck on lines 1 & 2 and (2) for the lower tape deck on lines 3 & 4:

(1)MANOH-01_s1_(2)MANOH-02_s1_(3)RMAN-01_(4)RMAN-02_.wav

Make sure all other non Wavelab operations and software and shutdown including Chrome and any file explorer windows and press record which make the record menu red and the signal start to be recorded.

IMPORTANT: Whilst we are using a Wavelab and Prism Sound set up, leave the PC workstation uninterrupted for this time, not even using the mouse.

Press play on the machines that have all been set up.

When you have finished digitising press stop and the recording will stop.

You will now have 4 files in your folder similar to this:

(1)MANOH-01_(2)MANOH-02_(3)RMAN-01_(4)RMAN-02_-1.wav

(1)MANOH-01_(2)MANOH-02_(3)RMAN-01_(4)RMAN-02_-2.wav

(1)MANOH-01_(2)MANOH-02_(3)RMAN-01_(4)RMAN-02_-3.wav

(1)MANOH-01_(2)MANOH-02_(3)RMAN-01_(4)RMAN-02_-4.wav

-1.wav is the recording of the signal from **Denon** cassette tape deck on **lines 1 & 2.**

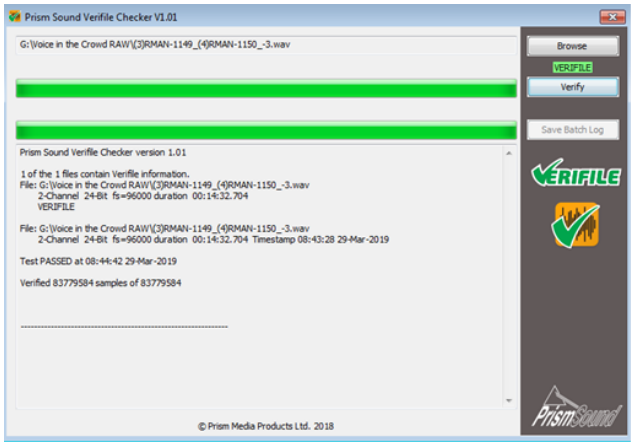
-2.wav is the recording of the signal from **Denon** cassette tape deck on **lines 3 & 4.**

-3.wav is the recording of the signal from **Studer** on **lines 5 & 6.**

-4.wav is the recording of the signal from **Studer** on **lines 7 & 8**

4.3.2 Verifile Checker

Close Wavelab and open **Prism Sound Verifile Checker** (icon with green tick on the toolbar). Drag and drop these 4 files into the checker and click the **'Verify'** button**.**



This software will automatically check for any drop outs where there was a break in the signal.

The files may pass, fail or show red one the verify button as there is no available date to perform this check.

Currently we are having some issues with the set up so if the file fails then use the **Save the batch log** button and so we can keep track of this. Always update the tracking sheet with this too.

If the files failed or have no data for the checks, it doesn't mean that we need to abandon this digitisation pass. Instead we need to perform some additional checks in the next stage of digitisation. Keep the list of timecodes that Verifile Checker lists where there has been a fail and use this to look for specific timecodes where there are likely issues in the Spectrogram view as described in the next step.

4.3.3 QC for dropouts using Spectrogram

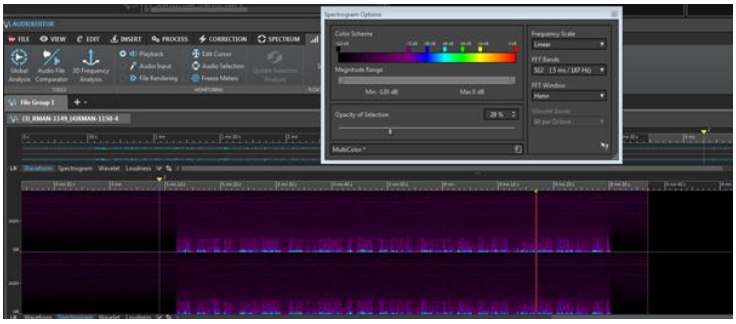
In Wavelab open the raw file that you are checking.

Press ALT+N > click 'Find current peak value' > check this doesn't read 0dB (if it does you have digital clipping and the item will likely require retransferring if this occurs during the recording).

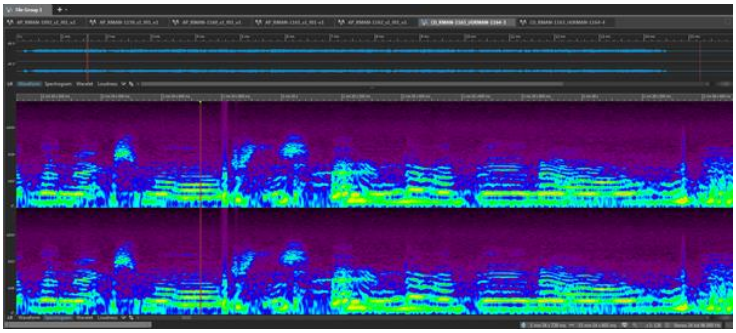
Importantly you only need to redidigise if there is a drop in the recording where the signal is needed. For instance if the Prism Verifile checker picked up drop outs as the start of the tape before the recording truly starts then it isn't an issue as this part of the file is not relevant. Check if this is the case by carrying out the following QC steps.

Select the **Spectrogram** view at the bottom of the timeline (to the right of Waveform)

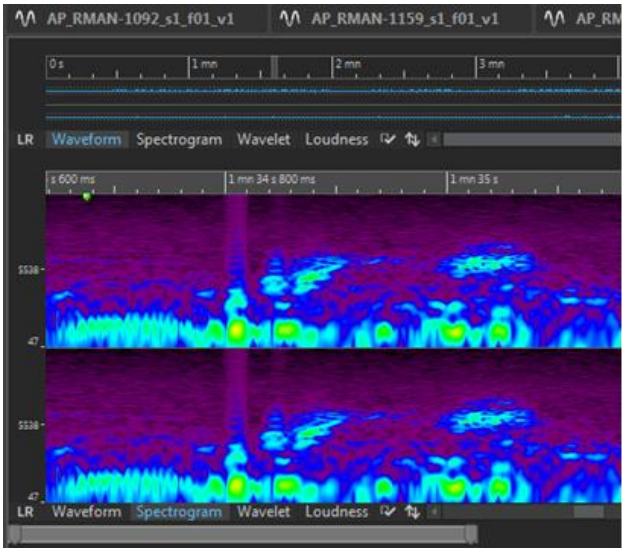
Set the the Spectrogram preferences to those in the following image. You will only need to change this once and Wavelab with automatically remember this setting:



Zoom in and check through the file looking for any abrupt straight lines that signify a digital glitch. The band will appear as a straight vertical line through all frequencies. You may hear a click or a jump in the recording like a small portion of sound is missing. See the vertical line just to the right of the red cursor/playback line below. It travels across all the frequencies shown in the bright colours of the spectrogram:

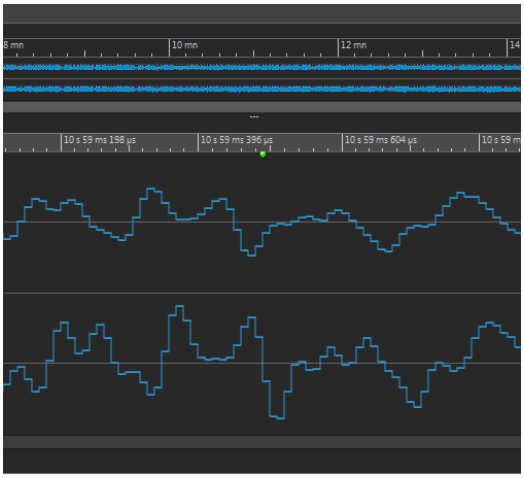


Here it is closer up:

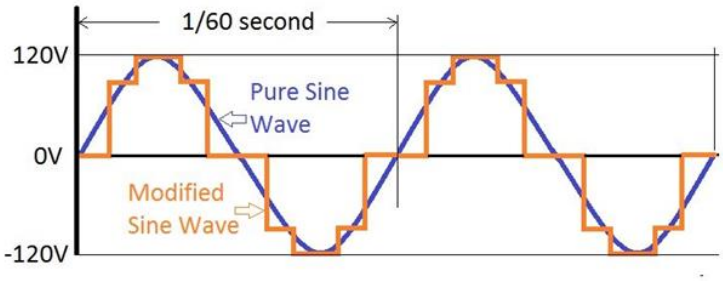


Next to confirm if this is instead a digital glitch introduced during digitisation or an analogue gitch (like an edit) that was inherent in the source we need to switch back to the Waveform view.

In the **Waveform** view, zoom in and look for any right angles in the sound wave stepping:

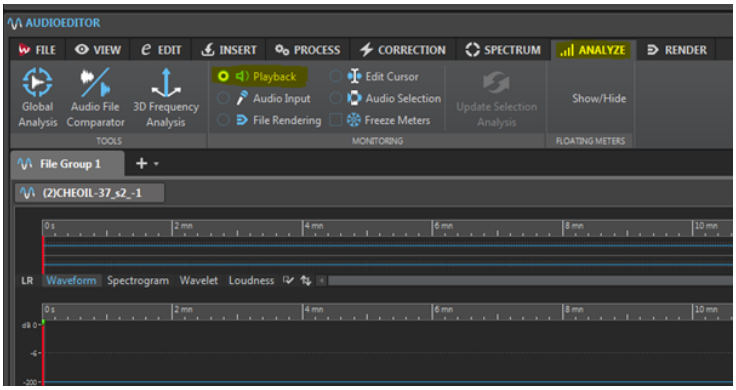


A pure analogue sound wave will have a natural curve and any blocking/stepping is an error that has been digitally introduced and therefore modified that sound wave.

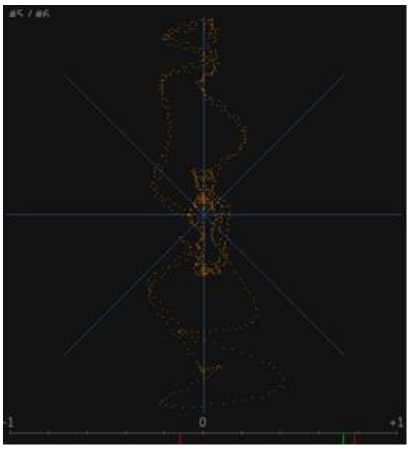


4.3.4 QC for Azimuth

Spot check points in the recording, especially where you can see there have been changes for instance a difference song or performance. In the **Analyse** menu set the **Phasecope** monitor to **Audio Playback**:



Looking at the Phasecope check weather the azimuth drifts too much at points in the recording



Make a judgement if this was as accurate you could set the azimuth at the time or if you need to record another version of the recording with the azimuth at a different setting.

4.4 Editing & Saving Preservation Files

4.4.1 Editing

Once you have a file recording an accurate signal of the recording, in Wavelab open this raw file so that you can edit it.

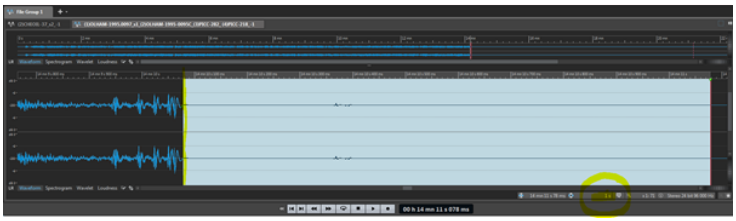
In the waveform on the bottom half of the screen click **CTRL +** ↓ **DOWN ARROW** and this will make the whole waveform for the length of the file fit the bottom of the screen.

For creating preservation files the only intervention required is for 'topping' and 'tailing 'each file. No other adjustments should be made including correcting any errors or low amplitude

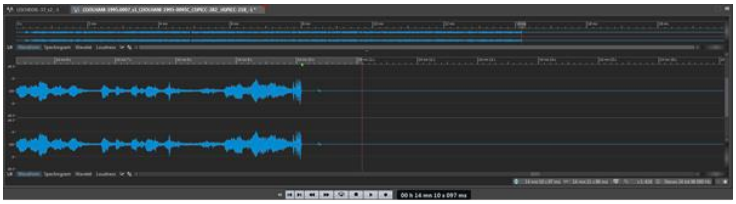
Add min 1 second of silence to the start and end of the recording.

Adding a 'tail'

To do this at the end ('tail'), find the last part of the audio signal for the recording (this might be the end of a fade out). Click the playhaead/cursor at this point and drag a highlighted section of the waveform forwards (towards the end of the file) to highlight 1 sec of audio. Then click just to the right of this section (to deselect it) at this 1 sec ahead point.



Then click **SHIFT + END** which selects the area from the play backhead to the end of the file, then press **DELETE.** This removes the end of the file that is not needed and leaves 1 second of audio after the signal truly ends. Playback to test this and observe the timecode.



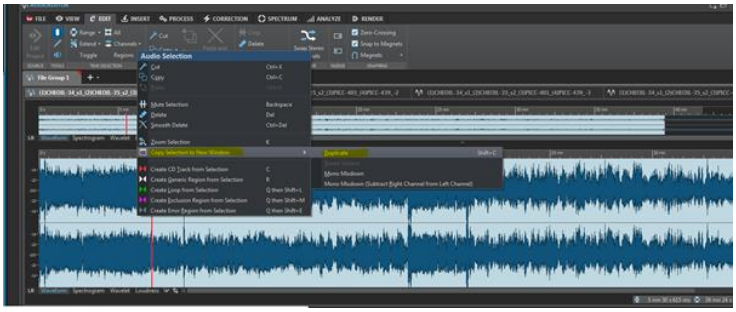
Adding a ‘top’

To do this at the start (‘head’), find the first part of the audio signal for the recording (this might be the start of a fade in). Click the playhaead/cursor at this point and drag a highlighted section of the waveform backwards (towards the start of the file) to highlight 1 sec of audio. Then click just to the left of this section (to deselect it) at this 1 sec back point.

Then click **SHIFT + HOME** which selects the area from the play backhead to the start of the file, then press **DELETE.** This removes the start of the file that is not needed and leaves 1 second of audio before the sinal truly starts. Playback to test this and observe the timecode.

[NOTE: most cassettes are in stereo but it is common for the channels to replicate much the same thing, at different levels. This is because many home-made recordings used internal microphones, pointing vaguely in the direction of speakers if you are lucky, rather than lapel directional mics. Keep both channels as-is in your digital master file but be aware that you may wish to experiment by removing one channel to improve audibility for listeners further down the line].

4.4.2 Adding metadata



Add the name of who carried out the digitisation in the metadata menu under audio engineer. This will automatically populate with the saved setting.

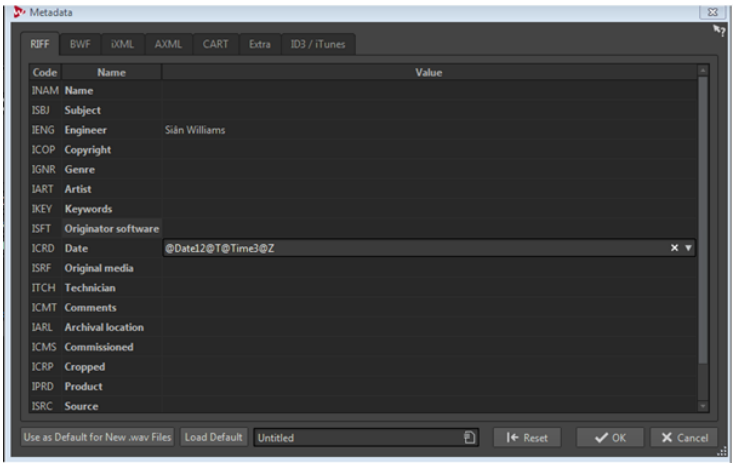


Image of the Metadata menu in Wavelab

Name of Engineer

@Date12@T@Time3@Z

Originator software: @WaveLabAndVersion@

4.4.3 Saving

The raw recording is already a 96kHz 24 bit WAV file. So you can simply ‘FILE’ > ‘SAVE AS’ and pick or create an appropriate folder within:

G:\\UOSH PRESERVATION

[Note: It is important that you use and do not alter this folder name as there are various batch scripts that mirror and backup the files to network storage within]

The is an example of file naming protocol e.g.

AP_RMAN-1092_s1_f01_v1.wav

AP = Archives Plus (Originator)

RMAN-1092= Example of shelfmark with each element separated by a hyphen and number (replace this).

_s1 = underscore side 1 (pad with a zero only if born digital).

_f01= underscore file number 1 (padded with a zero always).

_v1 = underscore version 1

.wav = Audio Cassettes will be saved to a 96kHz, 24 bit wave file.

This is the typical last step complete as you have now produced a preservation file of the digitised recording.

Editing different versions of the files

Changes of Azimuth and Tape Speed (Objective = _f02…)

If a full side of a recording requires different setting of azimuth (all tape) or tape speed (Open Reel only) then you will need to make separate files to do this. It isn't always necessary to digitise the full side again, just the section(s) that need to be digitised at a different setting.

These are **objective** decisions as the recording would simply be wrong if not recorded as the right speed. (more that one file number)

Changes of…. (Subjective = _v2…)

If there are more than one version because of **subjective** changes of set up such as EQ or stylus choice (vinyl). As a rule for any other reason other than azimuth and tape speed, then multiple versions of the file may be needed.

Commented [DG1]: See minor changes and comments in 301.

4.4.4 Recording Process Metadata

Think about how you would like to record what we call the 'process' metadata on transfers and who's doing them. You might want to keep a written notebook and later copy it contents into a shared spreadsheet. We record: the date, reference number, format, staff member, number of sides, any conservation issues, issues with sound quality or azimuth, serial number of machine used, any digital dropouts, number of digital files, filenames, locations of saved and backed up files.

This is useful further down the line for checking whether sound issues are in-source, for checking if we missed any sides, or for identifying which transfers were made using faulty machines, etc. In an ideal world this metadata would be recorded in a collections management system alongside the digital files.