Topic support guide

Cambridge International A Level

Cambridge International AS & A Level Information Technology 9626

For examination from 2017

Topic 10 Sound and video editing

Sub-topic 10a Sound editing

Cambridge Advanced

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Introduction

How to use this guide

The aim of this guide is to facilitate your teaching of Cambridge International AS & A Level Information Technology, syllabus topic 10, Sound and Video editing. The guidance and activities in this resource are designed to help teachers devise programmes of study for sound and video editing that provide teaching time devoted to theory work and opportunities for practical exercises.

Learning objectives

After reading this guide you should be able to teach the following learning objectives:

- trim a sound clip to remove unwanted material
- join together two sound clips
- fade in and fade out a sound clip
- alter the speed of a sound clip
- change the pitch of a sound clip
- add or adjust reverberation
- overdub a sound clip to include a voice over
- export a sound clip in different file formats.

Prior knowledge and preparation

Before you begin teaching this topic:

- Make sure you understand the concepts mentioned in the introduction to the theory section
- Undertake some further study of the concepts using the resources suggested. This will be needed to provide depth to your knowledge and will give you the ability to encourage students to explore these topics further themselves
- Tackle all the tasks and develop some extension material or additional tasks.

1. Key terms

Word/phrase	Meaning
sampling	The conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal).
waveform	A visual representation of an audio signal or recording.
frequency	The number of complete waves or oscillations or cycles occurring in unit time (usually one second).
pitch	The frequency of a sound as perceived by the ear. A high pitch sound corresponds to a high frequency sound wave and a low pitch sound corresponds to a low frequency sound wave.
amplitude	The size of the vibration and this determines how loud the sound is.
fidelity	How close the digitised sound is to the original analog recording.
monophonic recording	Single channel sound.
stereophonic recording	Two (or more) channel sound.

You will also need to become familiar with some commonly used effects like:

- fading in and out
- changing the pitch
- changing the speed
- changing the volume (amplification)
- adding reverberation.

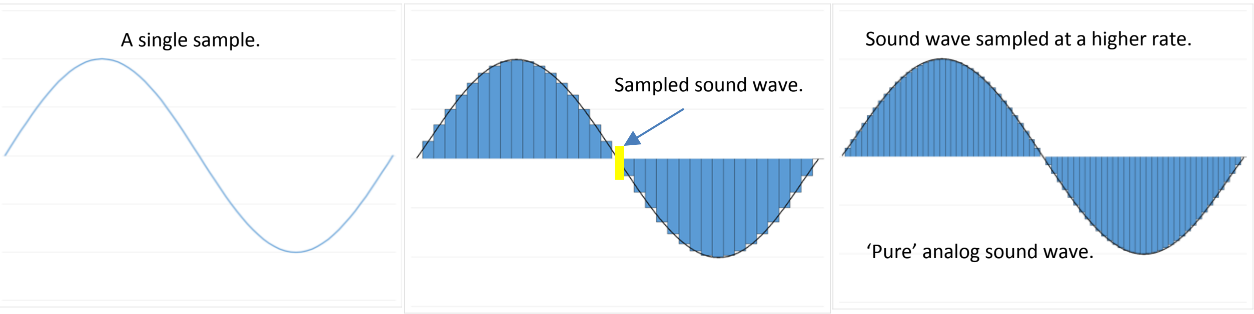
2. Theory

2.1 Introduction

Sounds are transmitted through the air as changes in air pressure. These pressure changes are smooth and continuous. This is analog data. Computers and most other modern audio processing equipment can only use digital data.

2.2 Creating digital data from analog data

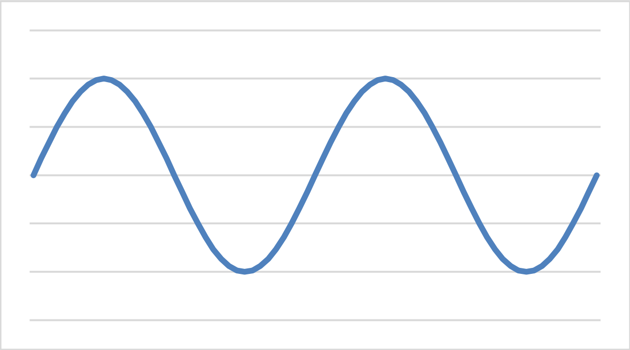
Analog signals are **sampled** to create digital data. The '**sampling rate**' (how many times a second the analog signal is measured), determines the 'fidelity' of the sound and the size of the digital file. Higher sample rates will mean bigger file sizes but it is not just the number of samples that matter. Each single sample has a 'bit depth', which is the number of bits of information in each sample. This directly corresponds to the **resolution** of each sample. Examples of bit depth include Compact Disc Digital Audio, which uses 16 bits per sample and DVD-Audio and Blu-ray Disc, which supports up to 24 bits per sample.



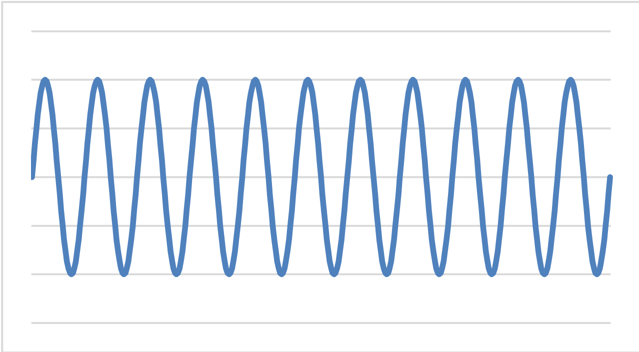
This digitisation is carried out by analog-to-digital Conversion (ADC). After you have edited a digital file, the computer will use a digital-to-analog Conversion (DAC) to convert it to a (relatively) smooth analog waveform that drives your speakers. There is some more important theory work on sampling but these ideas are enough to get us started with the practical work. For all practical work on this topic you will be using digital audio files. Before you start, there are a few other terms like **frequency, pitch** and **amplitude** you need to consider.

2.3 Frequency

The **frequency** is the number of peaks and troughs per second and is given as cycles per second – Hz; pronounced Hertz. The average human ear can hear sounds as low as 20 Hz and as high as 20 kHz (20 000 Hz). This is known as the audible range.



Low frequency

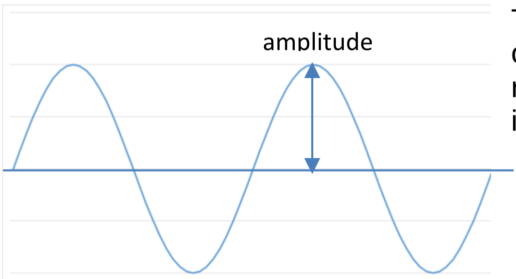


Higher frequency

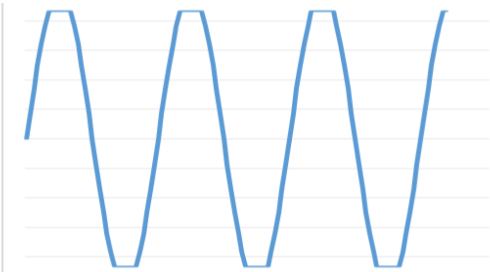
2.4 Pitch

We use the word **pitch** to compare sounds. We perceive higher frequency sounds as higher in pitch. When we work with audio editing software we usually refer to changing the pitch rather than altering the frequency.

2.5 Amplitude



The height of the sound wave reflects the strength or power of the sound. We call this the **amplitude**. Higher amplitudes result in higher volumes. This is why we call a device that increases volume an amplifier.



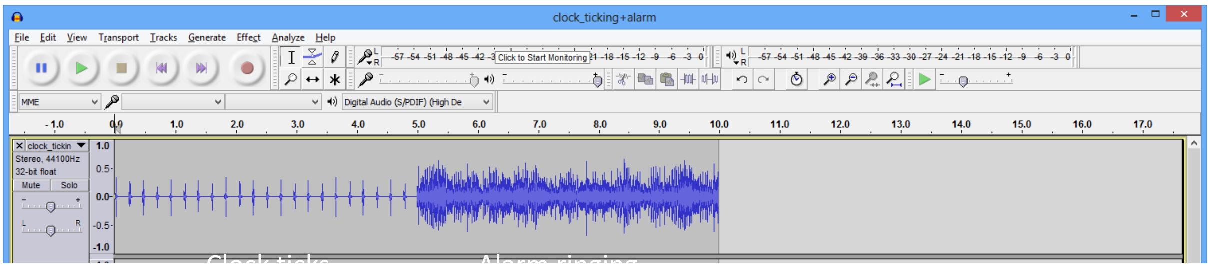
When an amplifier tries to increase the amplitude beyond its limits, the waveform is 'clipped'. **Clipping** produces a distortion, which is one of the 'effects' that rock guitarists often use.

We will use several 'effects' in our audio editing practice.

2.6 Audio editing

What we've seen in the diagrams so far are very simple 'waveforms'. A waveform is a representation of an audio signal or recording. It lets you see the changes in amplitude against time.

Audio editing applications, like Audacity®, use waveforms to show how sound was recorded. If the waveform is low, the volume was probably low. If the waveform fills the track, the volume was probably high. Changes in the waveform can be used to spot when certain parts of the sound occur. In this screenshot of one of the practice files, we can see the clock ticks and when the alarm rings.



When we need to save the file we've been editing we need to decide which format to use. In theory this should be a simple trade-off between the quality of the recording and the size of the file. It is sometimes more important, however, to consider the eventual use and distribution of the recording.

2.6.1 Audio formats

There are three major groups of audio formats:

- uncompressed
- lossless compression
- lossy compression.

Uncompressed formats like .WAV give pure digital sound as recorded. However, the files can be very large. Five minutes of WAV audio can need 40 to 50 mb of storage.

Files saved with Lossless compression like .FLAC files are still the sound as recorded but the compression algorithm allows some parts of the file to be stored as coded data, saving some file space.

Files saved with lossy compression like .MP3 have some audio information removed and the data is simplified. This does result in some loss of quality but clever algorithms only remove the parts of the sound that have the least effect on the sound we can hear.

The compression of a recording and the decompression for playing back a recording is done by CODECs (coder–decoder). CODECs for the common audio formats are usually automatically installed in media applications and extra CODECs for other file types can be downloaded and installed as required.

3. Audio editing tasks

Theme – clock work

These tasks are designed to be undertaken as a learning process covering most of the objectives listed above. Learners should be encouraged to use the tasks to explore the menu items and tools available in the audio editing software.

Often there is more than one way to satisfy the requirements of the task. At first, the exercises should be about exploring a variety of options and not about determining the most efficient methods. It is recommended therefore, that learners begin the tasks without access to the tutorial material, which you downloaded with this guide.

Task 1	**Task 2**
(1) Extract five seconds of the ticking clock sound from the **Clock_tick.wav** file. Export the file in **.mp3** format at 128 kbps. (2) Trim the **Alarm_ring.wav** file to be five seconds long. Export the file in **.mp3** format at 128 kbps. (3) Create a single file of the clock ticks followed by the alarm ringing (10 seconds). Export the file in **.mp3** format and save as **Timer1.mp3**.	(1) Remove the leading noise and trim the **Winding_clock.wav** file to five seconds. Export the file in **.mp3** format at 128 kbps. (2) Add the sound to the beginning of the **Timer1** file created in task 1. (3) Fade in the volume of the alarm-ringing section to full volume over the first three seconds. Export the file in **.mp3** format and save as **Timer2.mp3**.
Task3 (1) Open the **Timer2** file. (2) Slow the ticking-clock section by 50%. (3) Re-trim the ticking section to five seconds. (4) Decrease the pitch of the alarm ring to 700 Hz. Export the file in **.mp3** format and save as **Timer3.mp3**.	**Task 4** (1) Open the **Wake_Up.wav** file. (2) Change this sound file to be monophonic. (3) Now amplify the track to the maximum value possible without clipping. (4) Add reverberation to the track. (Choose a preset of a large room or equivalent if available.) Export the file in **.mp3** format at 128 kbps. (5) Overdub the **Wake Up** sound file with the **Timer3** file to start when the alarm starts ringing. Repeat the **Wake Up** clip for the length of the alarm ringing. Export the file in **.mp3** format and save as **Timer4.mp3**.
Sound files provided: Ticking clock sound (**Clock_tick.wav**); Alarm ringing sound (**Alarm_ring.wav**); winding a clock sound (**Winding_Clock.wav**), voice sound saying ‘wake up’ (**Wake_Up.wav**). The files are included in the zip file with this guide.	

4. Further resources

Free sound files

https://www.freesound.org/

Register for a free account to search for and download sound files.

http://www.freesfx.co.uk/

Register for a free account to search for and download sound effects (mp3 only).

http://soundbible.com/royalty-free-sounds-1.html

Search for royalty free files – no registration needed.

Free editing tutorials

http://manual.Audacityteam.org/o/man/tutorials.html

The official Audacity manual tutorial files. (Well-explained example tasks; text only, but just as useful as many video tutorials.)

http://www.freeAudacitytutorials.com/category/beginner-tutorials/

Excellent video tutorials

A YouTube search will bring up many excellent tutorials, for example:
https://www.youtube.com/results?search_query=Audacity+tutorials+for+beginners

Audacity: Complete Tutorial Guide for Beginners by David Taylor would be a particularly good starting point.

5. Further study

It is recommended that learners be encouraged to research the following questions independently.

1. For a digital audio recording, explain what is meant by:
   a. the sample rate
   b. the bit depth
   c. the bit rate.

2. Give an example of when a low sampling rate is acceptable for audio transmission or recording.

3. Explain the difference between stereo and mono audio.

4. For each of the following describe the effect on the size of the digital audio file:
   a. recording with an increased sample rate
   b. converting a stereo recording to mono
   c. saving the file in .mp3 format.

5. Explain the function of a CODEC.

6. Give an example recording format for each of the following:
   a. an uncompressed audio file
   b. an audio file recorded with lossless compression
   c. an audio file recorded with lossy compression.

7. Describe an advantage and a disadvantage for both lossless and lossy compression.