2. Audio and video elements on the web

2.1. Multimedia formats

When preparing multimedia content for web publishing, there are two important factors to consider:

- Content encoding format
- Content container format

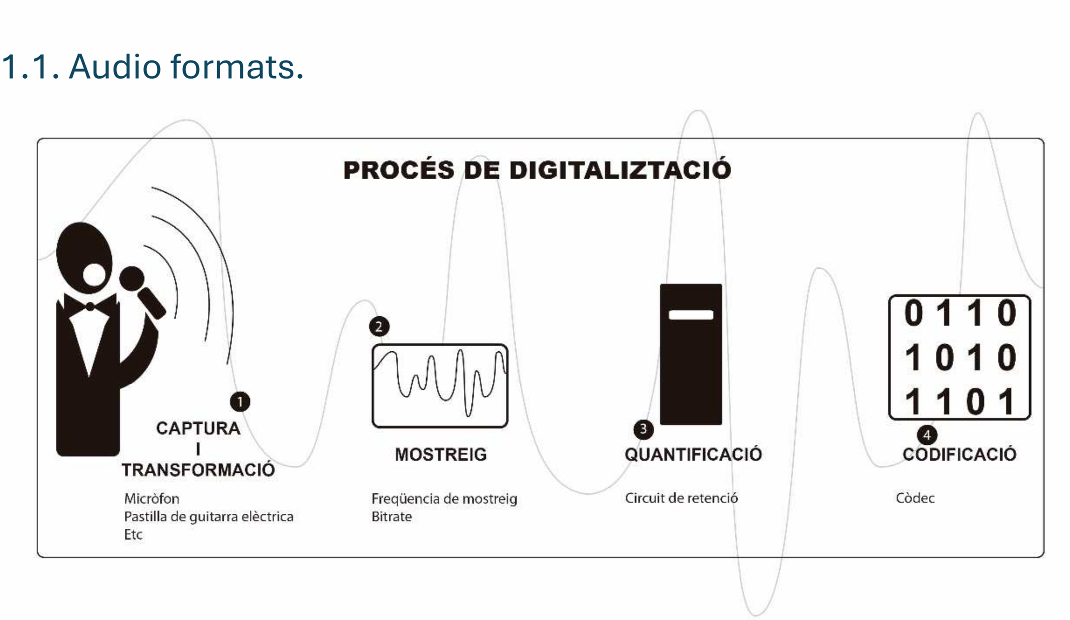
The method of encoding a multimedia element is called a codec, which is short for code / decode.

The codec determines the algorithm used to encode the binary information and the compression algorithm applied, if applicable, as not all codecs apply compression. Some common codecs are MP3, H.264, AAC and Vorbis.

The package that contains the compressed media content (audio and/or video) and its metadata is called a media container
A media container format usually supports more than one type of codec. Package metadata can contain information related to functionality such as:

- Subtitles
- Different audio tracks according to the language
- Audio and video possibilities with variable refresh rates, VFR (Variable Frame Rate)
- Streaming viewing capabilities, to view tailored content that is being downloaded without it being downloaded in its entirety.
- Content structure in chapters and menus

2.1.1. Audio formats.



In sampling (sampling) during the digitization process of an audio source, a sound pressure value that is measured every few microseconds is converted into a digital value (binary information).

There are three parameters that influence the quality of audio sampling:

- **Sampling rate** (how often the signal is measured for one second).
- **Number of channels** (how many locations are used to sample the same signal).
- **Precision of the samples** (number of bits used to store the sample, also called bit depth).

See a diagram of how digital audio sampling works. The number of bits in the sample determines the precision for measuring the vertical axis, while the sampling rate determines the precision for the horizontal axis.

For phone calls, for example, a typical sampling rate is 8 kHz (ie 8,000 times per second), while stereo music quality is 44.1 kHz or 48 kHz. A typical number of channels is 2 (stereo) and a typical bit depth is 8 bits for phone quality and 16 for stereo music quality

Audio codecs are algorithms that encode the audio stream, and like video codecs, there are two types: lossy and lossless.

To save as much bandwidth as possible, they are more interested in lossy audio codecs. Note that the WAVE format (.WAV files) is uncompressed and natively supported in all browsers except IE, so if you need uncompressed audio, this is a good choice.

In relation to audio, you can currently enjoy a video with six or more speakers in spectacular surround sound, but that doesn't mean visitors to a web page can. Most content on the web is mono or stereo, and a typical smartphone or mobile device only offers stereo output anyway. However, it is possible to create Ogg Vorbis and MPEG AAC files with six or more channels and play them in a browser with surround sound, as long as surround is actually available to the web browser and operating system.

To transmit audio through a data communication medium, such as the Internet, another important parameter in relation to playback quality is the **bitrate**.

Bit rate is the frequency with which data, described in bit form, is stored or transmitted in a specific period of time (usually in seconds).

The bit rate measures how many thousands of 1's and 0's are transferred to a device per second. For example, a bit rate of 1 kbps (kilobits per second) means that 1,000 bits, or kilobits (kb), move from the server to the audio player every second.

Bit rates (kbps) can range from 32, 64, 128, 256, and up to 320 kbps. There is no direct relationship between bitrate and quality perceptible to the human ear. After a certain point, increasing the bitrate only increases the file size with a marginally noticeable increase in audio quality. For example, a 128 Kbps file sounds much better than a 64 Kbps file. But the audio quality is not doubled by doubling the rate to 256 Kbps.

Parameter	Effect on quality	Effect on size
Lossless compression	Maximum fidelity	A compression ratio higher than 40-50% cannot be achieved.
Lossy compression	Loss of fidelity with respect to the original. The more compression, the more loss.	Compression rate up to 80-95%.
Number of channels	The number of channels affects the perception of the directionality of the sound, but not its quality.	Each additional channel increases the size of the audio file.
Sampling frequency	The more samples available per second, the higher the fidelity of the digitized sound compared to the original.	Increasing the sample rate increases the size of the audio file.
Quality	The more quality, the more loyalty.	The higher the quality, the larger the file. The growth ratio varies by codec.
bit rate	The higher the rate, the more quality it presents.	The higher the rate, the bigger the file.
Audio frequency bandwidth	If there are sounds in the discarded frequencies, loss of quality can be perceived.	Removing frequencies means less data to encode, and this results in smaller files.

The most common formats for audio content are:

- **MP3 (MPEG-1 Audio Layer 3)** is both a container and a codec. The files have the extension .mp3. It is a de facto standard and its use is widespread on the Internet. It can use bitrates from 128 to 320kbps. It can use variable or fixed bitrate.
- **WAV**: It is also a container and a codec at the same time. Since it does not incorporate compression, it is only suitable for small sounds such as effects, alerts or the like.
- **FLAC** is a lossless container and codec with compression. It has smaller sizes than WAV, but since it is lossless, for high bit rates it takes up much more space than MP3. It is freely distributed.
- **OGG Container + Vorbis Audio Codec (commonly known as Ogg Vorbis)**: Uses the .ogg or oga extension. It is a free format.
- **MPEG-4 container + AAC codec**. It is better known as MPEG-4 audio. It uses the m4a extension. At the same bitrate, it achieves better quality and takes up less space than MP3.
- **WebM container + Vorbis codec**. It is the WebM format that only contains audio. It uses the .webm extension.
- **WebM container + Opus codec**. Opus is a newer standard, open source, completely free and very versatile. It has been developed by Xiph.org, the creators of Vorbis. It is standardized with the RFC6716 definition of the IETF (Internet Engineering Task Force).

2.1.2. Video formats.

Digital video consists of a series of frames that, when played back sequentially, generate an animation or moving image.

The frame is each of the individual images that make up a moving image or video. The speed at which digital videos play is measured in frames per second (FPS).

Frames per second (FPS) is the unit of measurement that indicates the reproduction frequency of each of the images that make up the digital video sequence. Indicates the number of images that appear in a single second.

Video resolution is the size in pixels of each of the frames that make up the video.

Format	Resolution	Description
SD- PAL	720×576	European analog system, now obsolete.
SD-NTSC	720×480	Analogue system in America and Asia, no longer in use.
DVD	740×480	DVD disc format.
DVD-HD	1280×720	Attempt at high-definition DVD that did not gain much popularity.
HDReady (720p)	1280×720	High resolution video, very popular on digital television. Good quality with moderate weight, which can be distributed over the Internet easily.
Full HD (1080p)	1920×1080	As bandwidths have increased, it has started to become one of the most popular formats. It is used to stream movies in high resolution. It is also the standard format for Blu-ray media.
2K (UHD)	2040×1080	It is mainly used in digital cinema, as it is considered to be the most similar digital

		quality to the analogue 16mm celluloid format.
4K (UHDV)	3840×2160	It is the new generation of high resolution standard. It has a 16:9 aspect ratio and has been adopted for digital TV.
Full 4K	4096×2160	It is the format adopted for high-resolution digital cinema as it adapts well to the 17:9 standard used in cinema.
8K	7680×4320	It is the format of the future. Similarly to 4K, we have 8K UHDV (7680×2160) with a 16:9 ratio, and Full 8K (8192×4320), with a 17:9 ratio, designed for digital cinema.
Quad HD (1440p)	2560×1440	Non-standard resolution format. Halfway between FullHD (720p) and UHD (1080p). Despite not being a standard, it has had some acceptance in high-end mobile devices.

Bit rate measures the amount of bits that are transmitted and processed per unit of time. It is measured in bits per second (bps), and we will usually see it measured with the units kbps (kilobits per second) and Mbps (megabits per second).

When video is encoded for broadcast over the Internet, encoders use different types of compression. They have different algorithms in relation to the bit rate:

- CBR (Constant Bitrate): The bitrate remains constant, sacrificing video quality if necessary.
- VBR (Variable Bitrate): The video quality remains constant, causing the bitrate to fluctuate to suit the needs of the video.
- Capped VBR: Same as VBR, but with a maximum bitrate cap (None).

https://support.google.com/youtube/answer/1722171
https://help.twitch.tv/s/article/broadcasting-guidelines?language=en_US

Container / codec	Description	Extension	Type
MPEG4 / H.264/H265 (vídeo) + AAC (àudio)	Usually called MPEG4. H.264 is a high-quality codec, but with a paid proprietary license. All browsers that support the HTML5 video element can play MPEG4 files with the H.264 codec. There is a later version of this codec, H.265, also known as HEVC (High Efficiency Video Coding), which is partially supported in	.mp4 o .m4v	audio/mp4 video/mp4 application/mp4

	Chrome and Edge and fully supported in Safari 16.1. Firefox does not yet support this format.		
Matroska / x264 (vídeo) + FLAC (àudio)	The Matroska container format is open source under the GNU LGPL license. It has support for subtitles and stereoscopic video, among other features. It tends to have videos encoded with the free x264 library, licensed under the GNU-GPL, which allows encoding video in H.264 format. The FLAC audio codec (Free Lossless Audio Codec) allows you to compress audio without loss, maintaining a very good compression	.mkv	video/x-matroska audio/x-matroska

	rate and sound quality.		
WebM / VP8/VP9 (vídeo) + Vorbis / Opus (àudio)	The WebM container is also open source and free. Originally designed for use with the VP8 and Vorbis codecs, also open source. The VP8 codec was released by Google after acquiring the company that had developed it and has subsequently implemented a new version, VP9, which lowers the bitrate and maintains the quality. The Vorbis codec provides lossy compression and discards frequencies that the human ear does not perceive, in a similar way to what the old mp3 codec did. It also	.webm	video/webm audio/webm

	supports the Opus audio format.		
Ogg + Theora (vídeo) + Vorbis / Opus (àudio)	Free and open container format, maintained by the Xiph.org foundation. The Theora codec implements lossy compression. Ogg also supports FLAC and Opus encoded audio. This format has been replaced by the better quality WebM.	.ogg	video/ogg audio/ogg application/ogg

2.2. Audio elements

2.2.1. Integrating audio into a web page

The HTML <audio> element allows audio content to be embedded in a web page.

It can contain multiple <source> elements with different possible audio sources, so browsers can choose what they think is most appropriate.

There are a few options that can be enabled directly with HTML attributes:

- autoplay The sound will play as soon as possible.
- controls Playback controls will be displayed.
- loop The sound will play continuously.
- muted The sound will start muted.
- src For only one audio format, this attribute can be used instead of the source tag.

```
<audio controls autoplay loop>
  <source src="https://www.w3schools.com/tags/horse.mp3"
type="audio/mp3">
  <source src="https://www.w3schools.com/tags/horse.ogg"
type="audio/ogg">
  Ho sento, el teu navegador no suporta àudio d'HTML5.
   <p>
    Pots descarregar el fitxer MP3 <a
href="https://www.w3schools.com/tags/horse.mp3">MP3</a> o
l'OGG
    <a
href="https://www.w3schools.com/tags/horse.ogg">OGG</a>.
  </p>
</audio>
```

https://codepen.io/iocdawm9/pen/LYrYWPq.

2.2.2. Audio editing tools

- The sound or music is in free or Creative Commons licensed repositories. In these cases it is necessary to take the same precautions in relation to copyright as when incorporating images into projects.
- The sound or music is provided by the company or the customer.

Audacity
Allows audio recording from multiple sources. It can process and convert all types of audio files, adding effects such as normalization, trimming and fading (fade-out). It allows you to work with multiple tracks and make mixes.

Adobe Audition
It is a professional sound studio that allows multitrack digital audio editing. It has support for surround sound and a multitude of filters and effects.

FFmpeg
It is a set of utilities and multimedia libraries that allow decoding, encoding, transcoding, multiplexing, demultiplexing, transmitting, filtering and playing virtually any format. It supports from the oldest to the latest formats. It's also highly portable: works on almost any platform.

VLC
It is a very popular media player that also has audio and video format conversion functionalities.

2.3. Video elements

2.3.1. Integrating video into a web page

The <video> element allows you to embed a video into a web page.

```
<video
src="https://upload.wikimedia.org/wikipedia/commons/1/18/Big_
Buck_Bunny_Trailer_1080p.ogv" width="640" height="480"
poster="https://peach.blender.org//wp-content/themes/bf-
bunny/graphics/header1.jpg" controls>
        El vostre navegador no suporta l'element HTML5 vídeo.
Podeu descarregar-lo <a
href="http://distribution.bbb3d.renderfarming.net/video/mp4/b
bb_sunflower_2160p_60fps_normal.mp4">aquí</a>.
</video>
```

width/height
Controls the size of the playback window. These parameters used to be reported in pixels, but responsive design uses alternatives with relative sizes.

Poster
Provides the location of an image that is displayed in slideshow mode before the video starts playing.

Controls

Show playback controls. Custom playback controls can be created using CSS and JavaScript.

autoplay

Allow the video to play as soon as possible. Many browsers ignore this option if you don't combine sound deactivation with the muted parameter.

Loop

Allows you to play the video continuously.

Muted

Makes the video play muted.

Src

If we only need one video format, this attribute can be used instead of the source tag.

2.3.2. Video manipulation and editing tools

- Video recording software: Allows you to record a video.
- Video encoding and converting software.
- Basic editing software:
- Video editing studios:

Avidemux

Free video editor designed for simple trimming, filtering and encoding tasks. It supports many file types, including AVI, DVD-compatible MPEG files, MP4 and ASF, using a variety of codecs. Tasks can be automated through projects, batch process and includes scripting functionality.

FFmpeg

Set of utilities and multimedia libraries that allow decoding, encoding, transcoding, multiplexing, demultiplexing, transmitting, filtering and playing virtually any format. It supports from older to newer formats. It's also highly portable – it works on almost any platform.

Shotcut

Free video editing studio that supports many video formats because it uses the FFmpeg library. It supports 4K resolution, incorporates multiple audio filters, image and sound transitions, video filters and integrates access to freely licensed resources such as images, animations and sounds.

Kdenlive

Video editing and editing application with features such as: multi-channel video/audio editing, support for multiple formats thanks to FFmpeg, highly configurable interface (themes, keyboard shortcuts), integrated video titling tool, many effects and transitions, add- ons and available resources.

Adobe Premiere Pro

Professional video editing studio with 4k support, multitude of filters, advanced subtitling tools, effect libraries like animated titling. It also incorporates many collaborative work tools specific to work teams.

Filmora

Very easy to use app, practical for simple videos, with a smooth learning curve. It incorporates many preset effects that make the video publishing process quick and easy. It incorporates an extensive library of sound effects, titles and copyright-free music/sounds.