IDEAS
Improving digital encoded audio skills

MODULE B

DATA TRANSMISSION AND NETWORKS

VLAD POPESCU

STRUCTURE OF COMMUNICATION NETWORKS

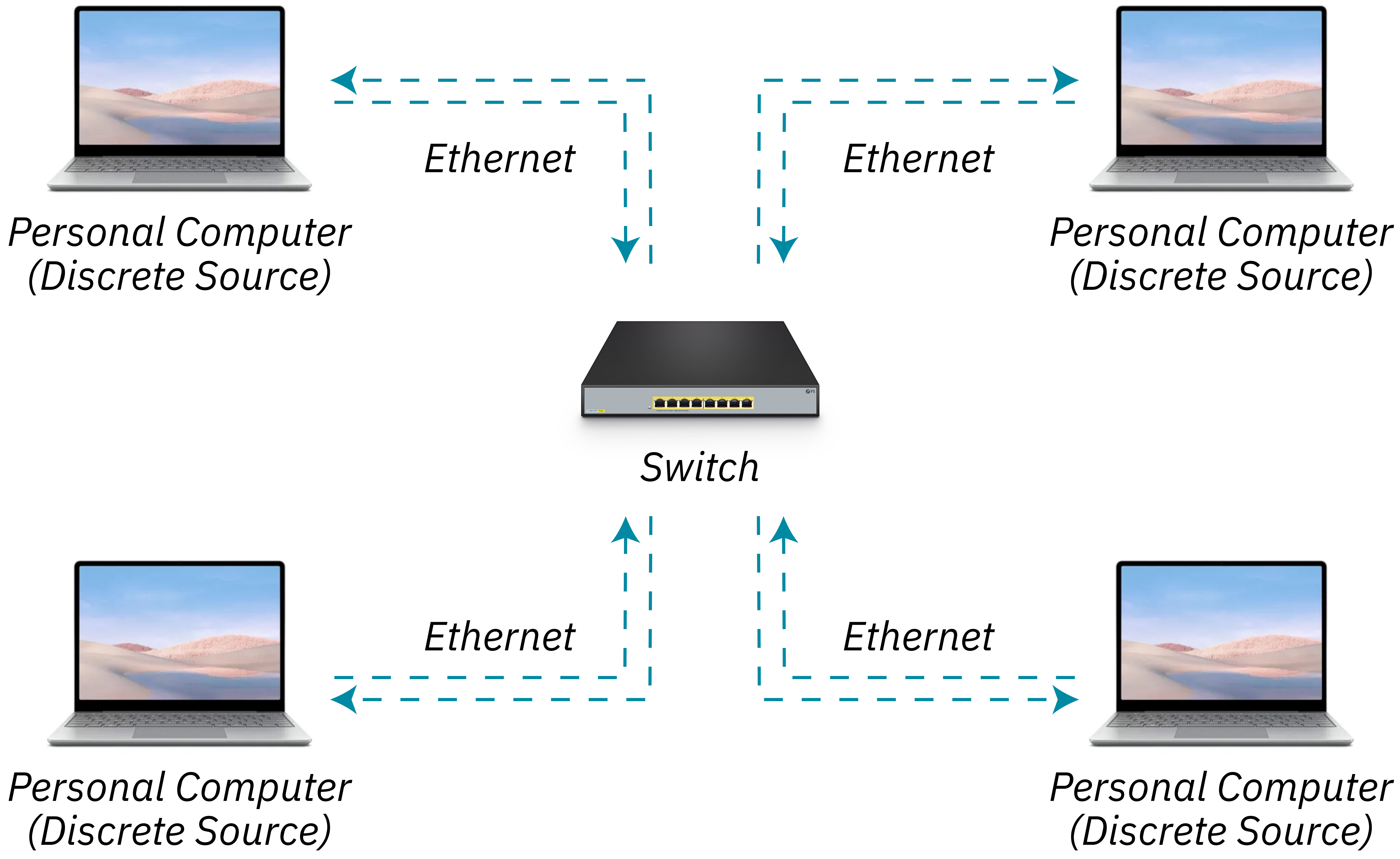
There are two main types of networks used for sending and receiving data in streaming operations: LAN and WAN.

A LAN (Local Area Network) network handles the transmission and reception of data streams within the same physical space, such as a room or an office where two or more devices are connected. Depending on the number of devices, the connection may be direct or switch-based.

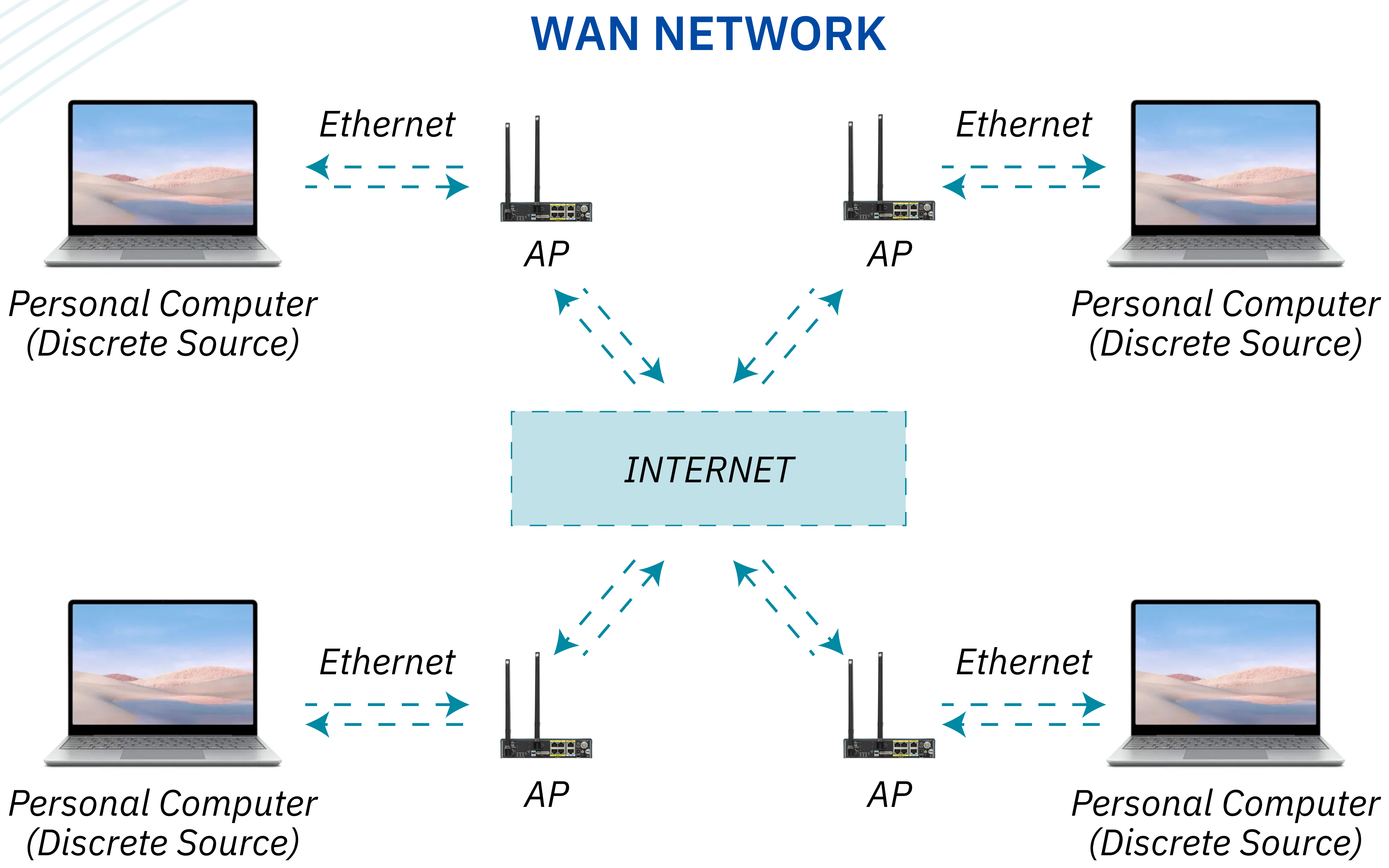
WAN (Wide Area Network) enables the transmission of data streams not only within the same network but also externally by means of an Internet connection. In this type of network, devices are connected using switches and routers

STRUCTURE OF COMMUNICATION NETWORKS

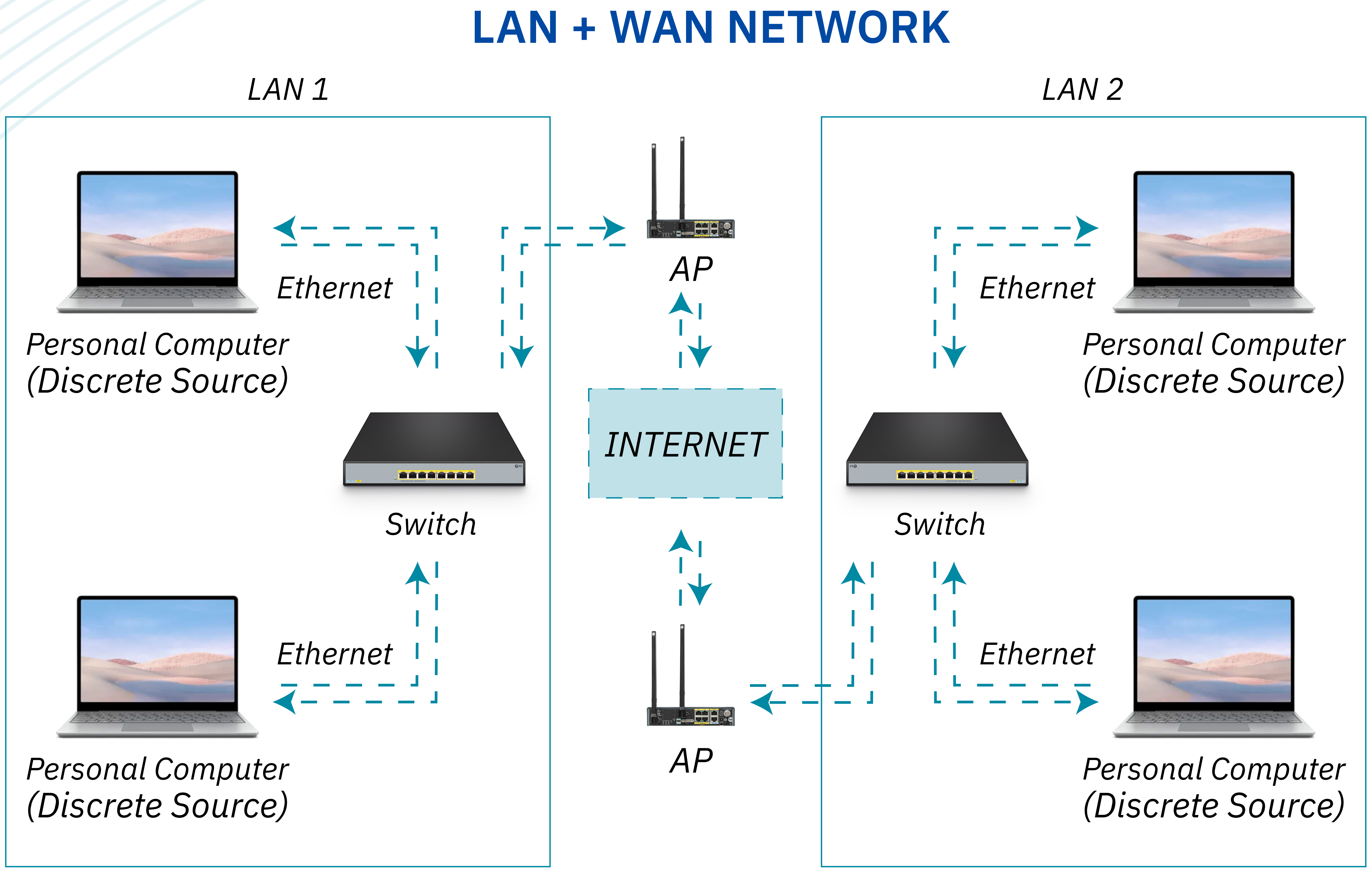
LAN NETWORK



STRUCTURE OF COMMUNICATION NETWORKS



STRUCTURE OF COMMUNICATION NETWORKS



AUDIO & VIDEO ENCODING AND DECODING

The encoding and decoding of audio and video signals are fundamental processes in data streaming, as they ensure efficient data transmission across networks.

Encoding, also known as compression, reduces the size of audio or video files without excessively compromising quality, thus optimising the bandwidth needed for transmission. This becomes particularly important in contexts where bandwidth availability is limited, e.g. internet or wireless networks.

Decoding, on the other hand, restores the original signal upon reception, allowing users to view or listen to content without significant losses.

THE IMPORTANCE OF AUDIO ENCODING

In Module A, we explored how humans perceive sounds differently based on frequency, amplitude, and the time interval between two consecutive sounds.

The approach we are expounding leverages these human perceptual properties and provides a method to avoid storing (and reproducing) anything that would be imperceptible to the majority of human listeners.

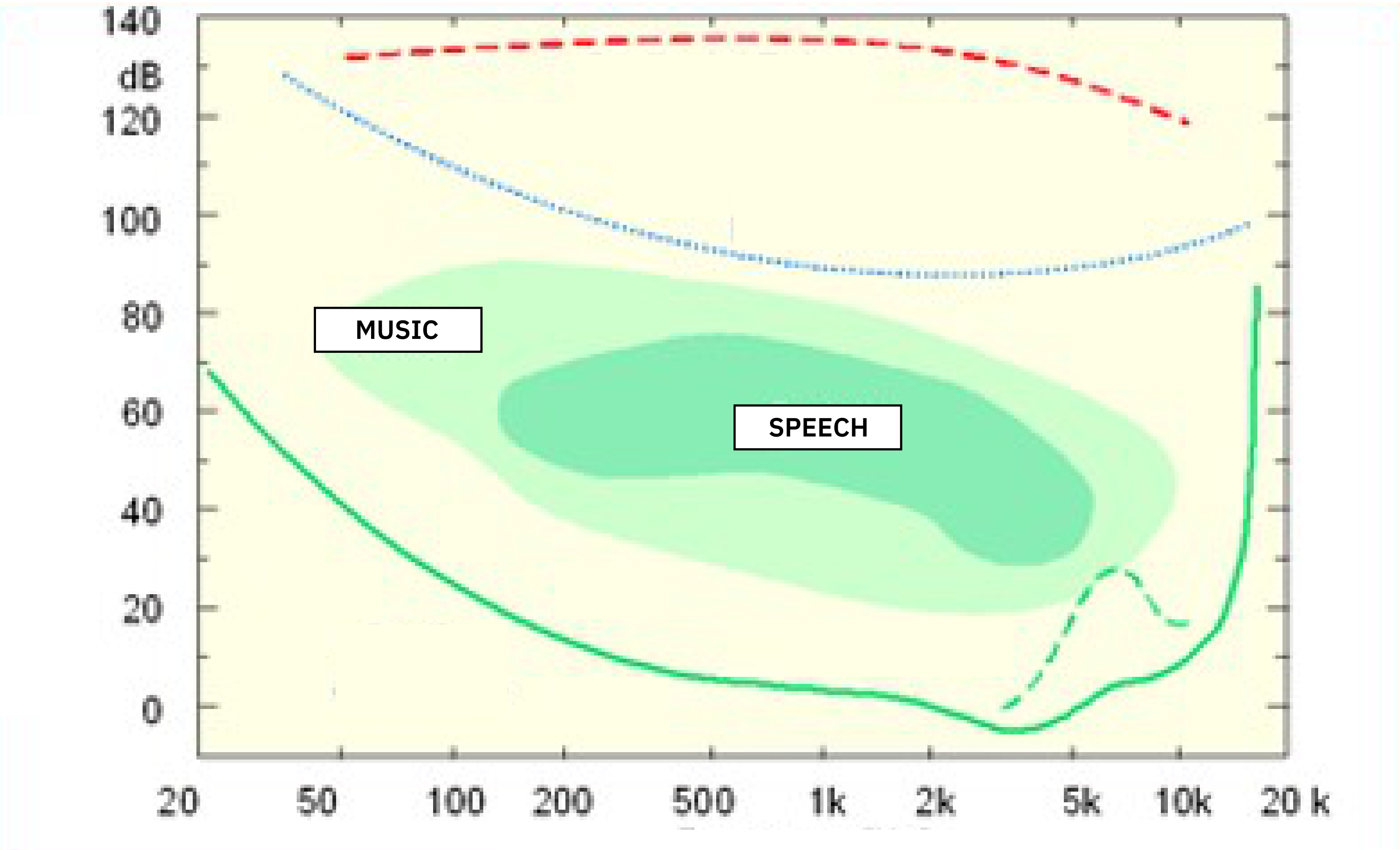
Encoding strategies based on these principles are called psychoacoustic encodings, and most modern advanced encoding technologies make use of these properties.

PSYCHOACOUSTIC ENCODING
- ABSOLUTE THRESHOLD OF HEARING

Humans generally cannot perceive (i.e. hear) sounds below 0 phon. However, the corresponding levels in decibels (dB) vary depending on the frequency. Isophonic curves show that at extremely low or high frequencies, more power is required to exceed the hearing threshold.

- When analysing the sound spectrum, if a frequency is below the threshold of hearing, it can simply be suppressed, given that it would not be perceived by listeners.

- At extreme frequencies, the human ear loses sensitivity and selectivity. Even though such sounds are audible, the information related to them tends to decrease because the ear is not as sensitive to these frequencies.



PSYCHOACOUSTIC ENCODING
- FREQUENCY MASKING

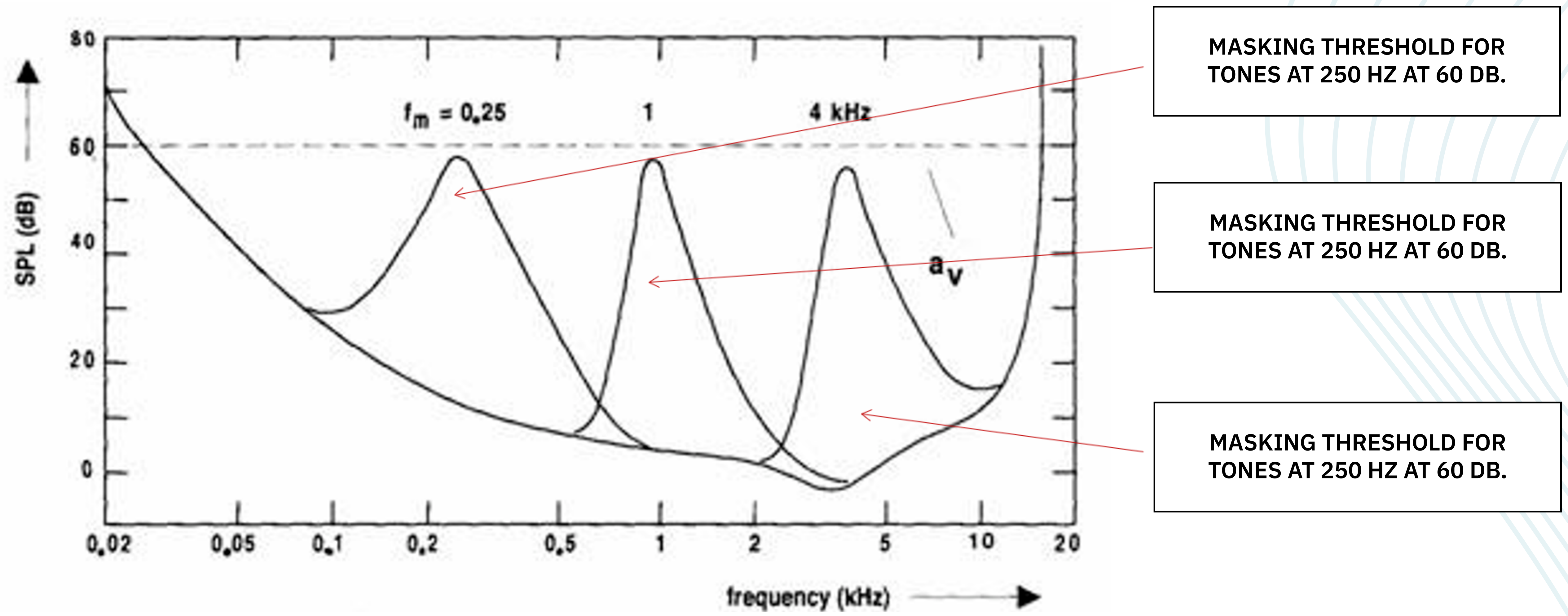
When considering a tone composed of two closely spaced frequencies (e.g., 1000 and 1100 Hz), one might expect the higher-amplitude component to dominate, making the other frequency inaudible. In the context of a pure tone with a specific frequency referred to as the masker tone and another pure tone known as the masked tone, the phenomenon where the presence of the masker tone renders the masked tone inaudible is referred to as frequency masking.

The term frequency masking is employed to describe situations in which the frequencies of the masking signal and the masked signal are similar, and the amplitude of the masked signal is not high enough to be heard.

In practical terms, this allows us to overlook the masked sound for encoding purposes.

PSYCHOACOUSTIC ENCODING
- FREQUENCY MASKING

Given the amplitude of a masker in decibels, it is possible to determine a masking threshold. This threshold is represented by the minimum values (dB) needed for closely-spaced frequencies to be audible. The graph below, for example, illustrates that in the presence of a sound at 1000 Hz at 60 dB, we will not be able to perceive a sound at 2000 Hz if its level is below 10 dB.



PSYCHOACOUSTIC ENCODING

- TEMPORAL MASKING

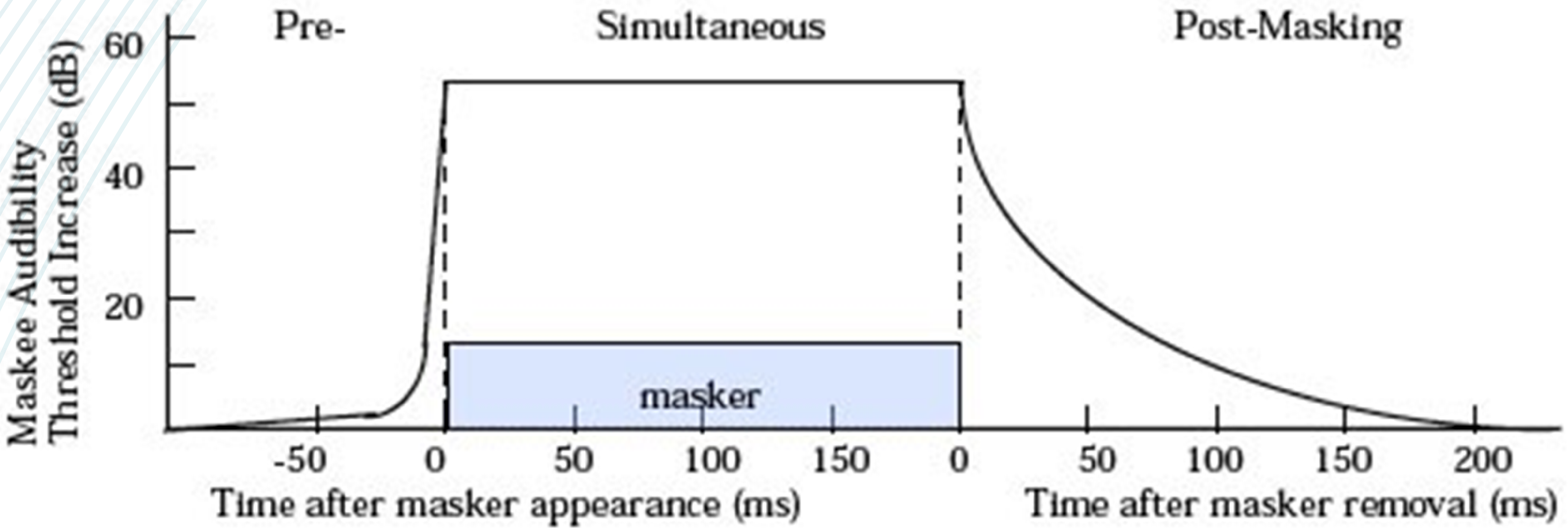
Humans face difficulties when attempting to perceive distinct sounds that are too close together in time. The time required to perceive two tones in sequence depends on both frequency and amplitude.

Similar to frequency masking, the tones involved in this phenomenon can be referred to as the masker and the masked tones. Specifically, when the masker tone ceases and the masked tone stops shortly afterward without being heard, it is referred to as temporal masking.

The amplitude difference between two signals is a decisive factor in the occurrence of temporal masking. Characteristic threshold values are determined by the minimum time required to perceive the masked sound on varying the amplitude levels.

PSYCHOACOUSTIC ENCODING
- TEMPORAL MASKING

Characteristic graph of temporal masking between a masking sound at 1000 Hz and 60 dB and a masked sound at 1050 Hz. According to the example, in order to perceive a 1050 Hz sound at 20 dB, we will need to wait at least 50 ms after the end of the masking tone.

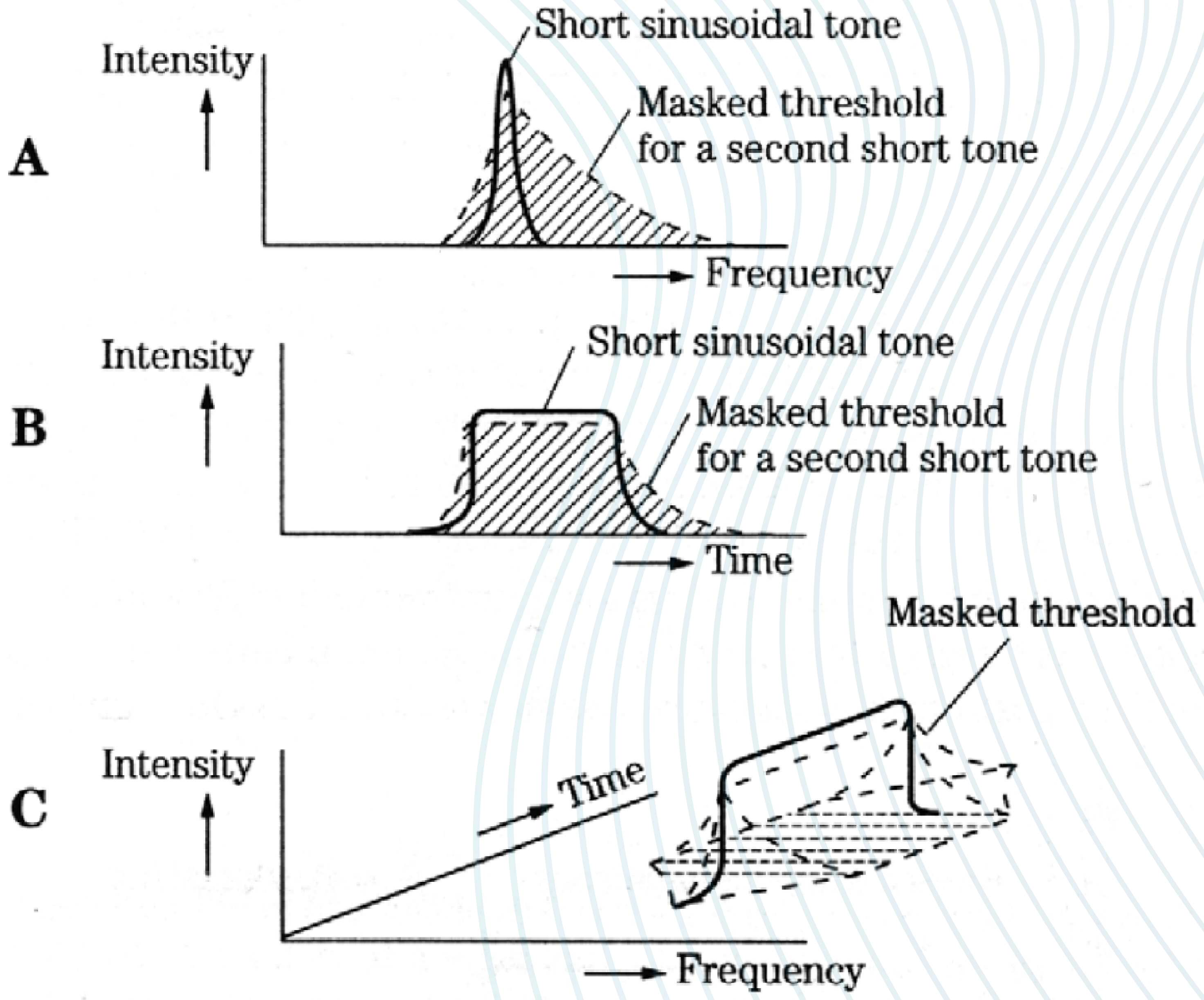


PSYCHOACOUSTIC ENCODING

All the mentioned phenomena, along with a few others, are the basis for the removal of information in advanced encoding and compression algorithms.

- **Frequency masking** is a simultaneous phenomenon, as it involves tones reproduced during the same time intervals.

- **Temporal masking,** on the other hand, occurs at a later stage and is thus considered non-simultaneous. It takes into account the effect of a pure tone on another, after the first tone has ceased and the second is still active.



A. Simultaneous masking.
B. Temporal masking.
C. Combined masking effect in time-frequency.

THE IMPORTANCE OF VIDEO ENCODING

The digital representation of multimedia video content requires a large amount of data.

Despite continuous increases in storage and transmission capabilities, the digitalisation process generates an excessively high volume of data that is incompatible with most current storage and transmission systems. For example, an uncompressed video sequence at Full HD resolution (1080p) can reach over 1.2 Gbps.

Video encoding, or compression, is the process responsible for converting a digital video into a numerical sequence of dimensions which is more suitable for storage and transmission.

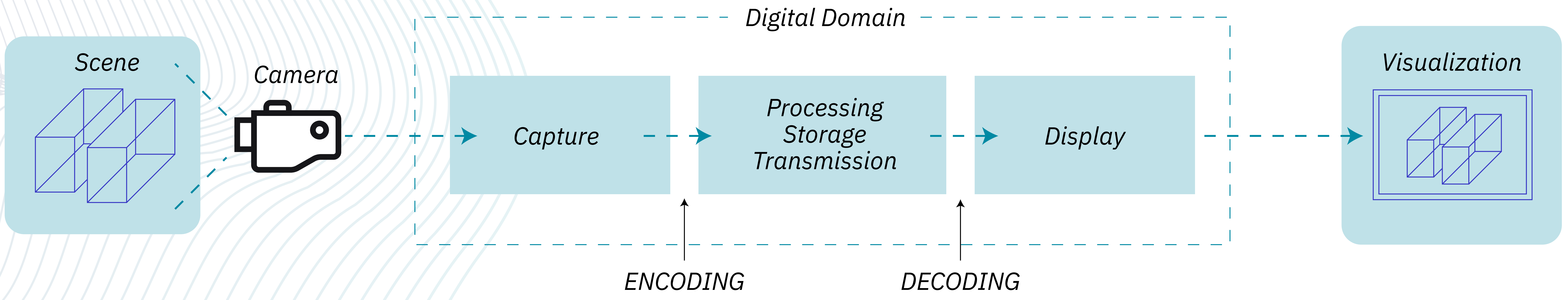
THE IMPORTANCE OF VIDEO ENCODING
- ENCODERS & DECODERS

A video codec is a device or software program used for compressing (encoding) and decompressing (decoding) digital sequences. An encoder-decoder pair is commonly referred to as a CODEC (acronym for enCOder/DECoder).

The term "lossy" describes compression processes that, in order to maximise file size reduction, involve a higher loss of information during compression.

The primary goal of a video coding system is to minimise the volume of processed data whilst simultaneously aiming to maintain an "acceptable" level of quality. To achieve this purpose, specific compression strategies have been designed.

BASIC MODEL OF VIDEO CAPTURE AND REPRODUCTION



VIDEO ENCODING TECHNIQUES

The standard techniques used for image encoding often result in compression ratios of 10:1, but this is not always enough to ensure an optimal video experience.

To overcome this limitation and enhance compression, we can take advantage of the fact that consecutive sequences of frames are often very similar. These similarities, referred to as temporal redundancies, can be encoded as small numerical differences between one sample and the next using an encoding technique known as differential pulse-code modulation (DPCM).

Furthermore, in natural videos, it is common for certain objects or parts of a frame to remain similar across frames, exhibiting only slight movements. This can, therefore, be further exploited to improve differential encoding through the use of specific motion compensation techniques.

VIDEO BUILDING BLOCKS: FRAMES

In a video file, each frame represents a single static image. When rapidly displayed in succession, they create an illusion of motion.

Each frame contains exhaustive information about the image at a specific moment of the recording. The sequence of these frames forms the video, and the frequency at which they are displayed, known as the frame rate, influences the video's perceived smoothness.

Video encoding is responsible for the compression and optimisation of these frames, reducing the amount of data needed to represent each image and ensuring a more efficient transmission across networks.

VIDEO ENCODING TECHNIQUES

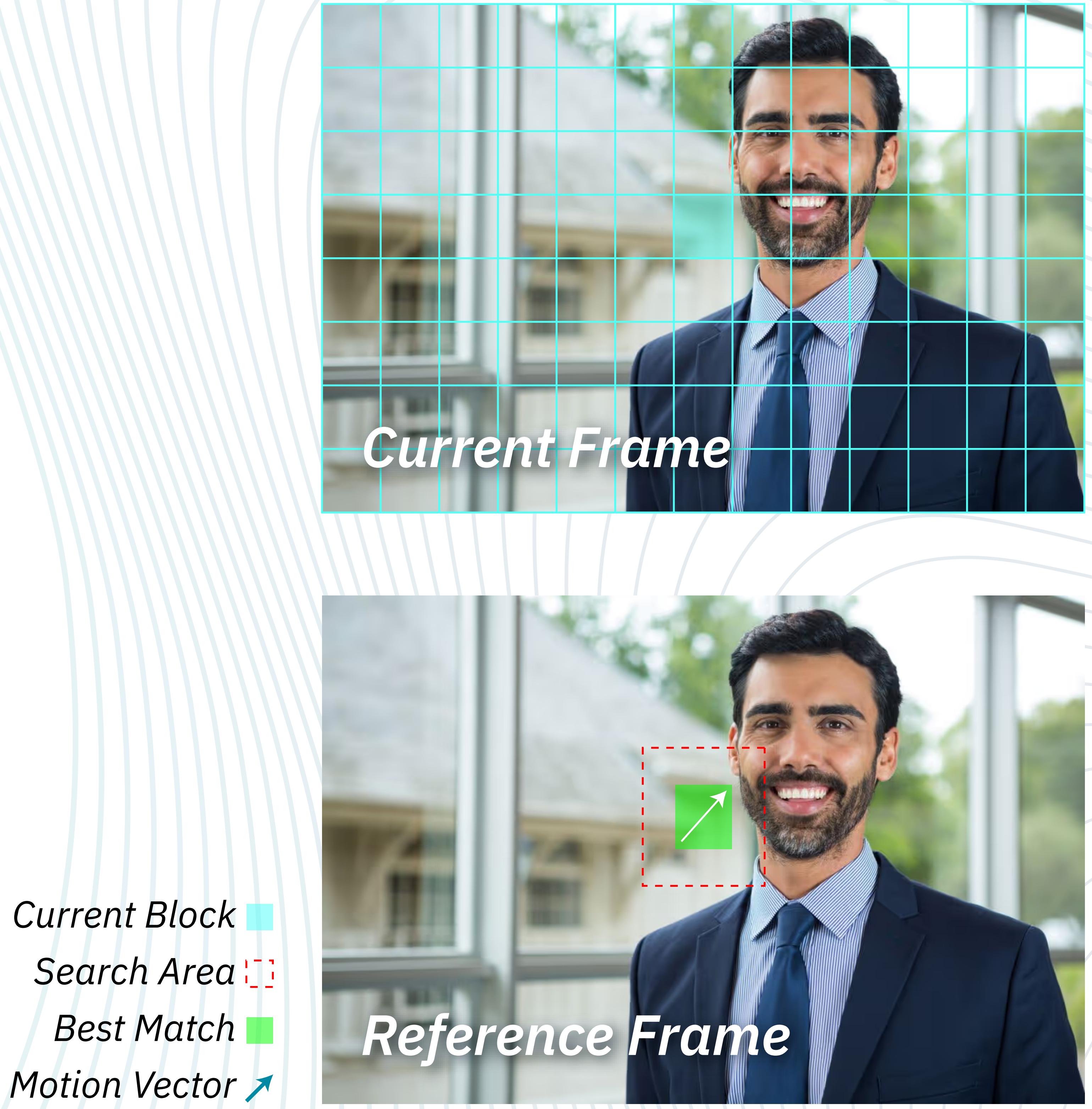
The considerations in the previous slide, which serve as guiding principles for the compression techniques used in all major video coding systems, are integrated and utilised to perform MC-DPCM encoding (Motion-Compensated DPCM). In cases where the elements of a frame are similar but positioned differently compared to previously coded elements, the MC-DPCM technique allows us to encode only the amount of movement in a frame and any small differences in the appearance of the depicted elements.

During the execution of MC-DPCM encoding, each frame is divided into blocks, typically with a size of 16x16 pixels. These blocks, called macroblocks (MB), are processed individually.

VIDEO ENCODING TECHNIQUES

For each macroblock within a frame n, a search is conducted for a similar area in frame n−1. This operation is known as motion estimation (ME), and the search can cover the entire frame n−1 (full-scan) or be limited to macroblocks adjacent to the starting position, which are likely to exhibit similarities.

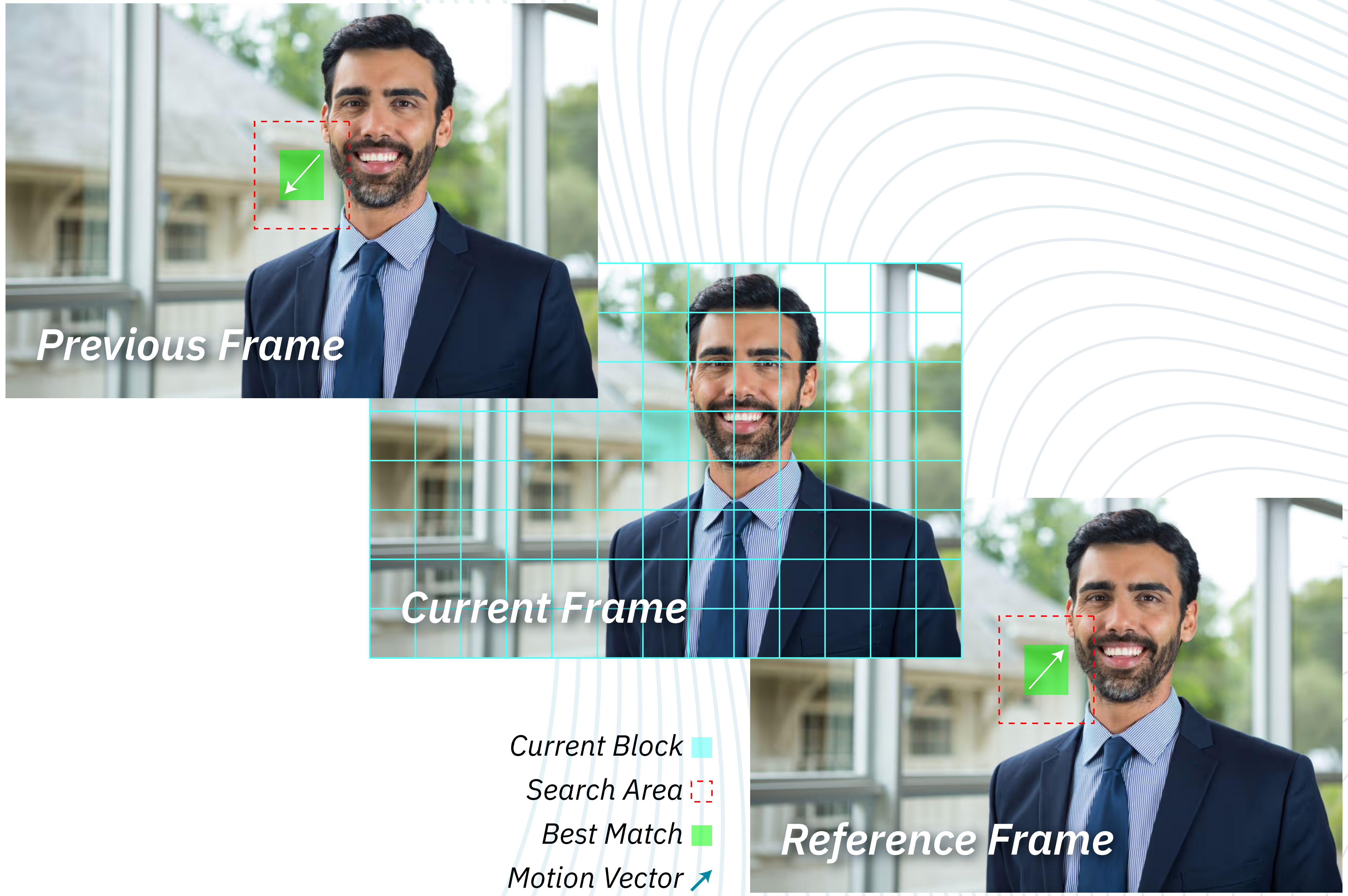
Subsequently, the encoding of values representing the motion of the macroblock in frame n relative to the selected reference area in frame n−1 takes place. The vector indicating the macroblock's motion is called the motion vector (MV). If the macroblock and the selected area exhibit further inconsistencies in their appearance, the differences between them are encoded as the prediction residual.



VIDEO ENCODING TECHNIQUES

If an encoder has access to future frames, it can formulate predictions aimed at encoding any new objects that can appear in the scene.

In this scenario, using both past and future frames, bidirectional predictions are performed. This approach further reduces the prediction residual, ensuring a more accurate prediction.



ADVANCED ENCODING TECHNIQUES

Initial Compression:

We can apply independent encoding to the first frame, using common image compression techniques such as block splitting, chrominance subsampling, DCT, quantisation, zig-zag re-ordering, RLE encoding, and entropy coding.

MC-DPCM

For subsequent frames, we can apply the MC-DPCM algorithm, including motion estimation, motion vector search, and prediction residual calculation. If none of the selected areas is considered adequate, then independent encoding is performed.

WORKING WITH COMPRESSED VIDEO FRAMES

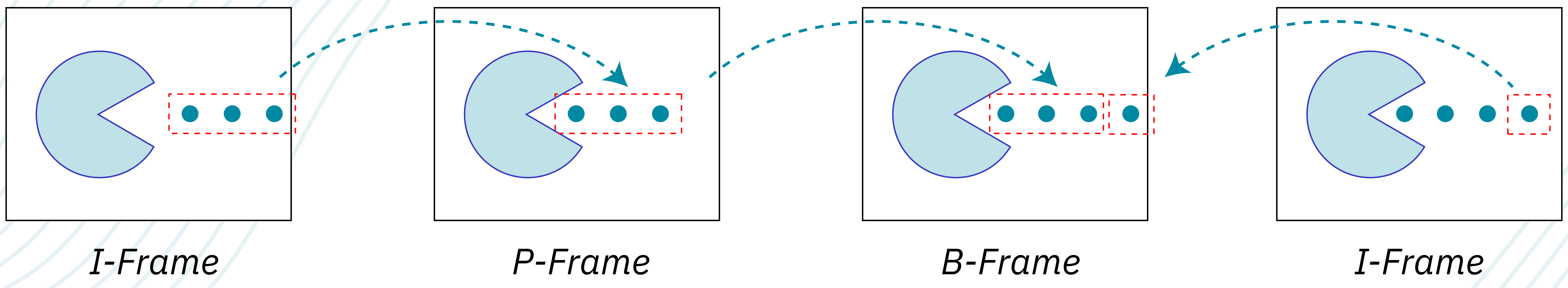
The frames in a compressed video sequence can belong to one of three categories:

- **I (Intra)**: No prediction strategy is used for any of the macroblocks in the frame.
- **P (Predictive)**: Prediction techniques are applied in frame encoding, but only for past references. The macroblocks in the frame can be of type I and P.
- **B (Bi-predictive)**: Both types of prediction are used (past and future references), and the macroblocks can be of type I, P, and B.

WORKING WITH COMPRESSED VIDEO FRAMES

When dealing with B-type frames, it is necessary to encode the reference areas derived from future frames before processing the corresponding B frame. This creates a frame encoding order (codec order) that differs from the natural display order.

Therefore, to ensure proper processing, the encoder must transmit frames to the decoder in the correct order.

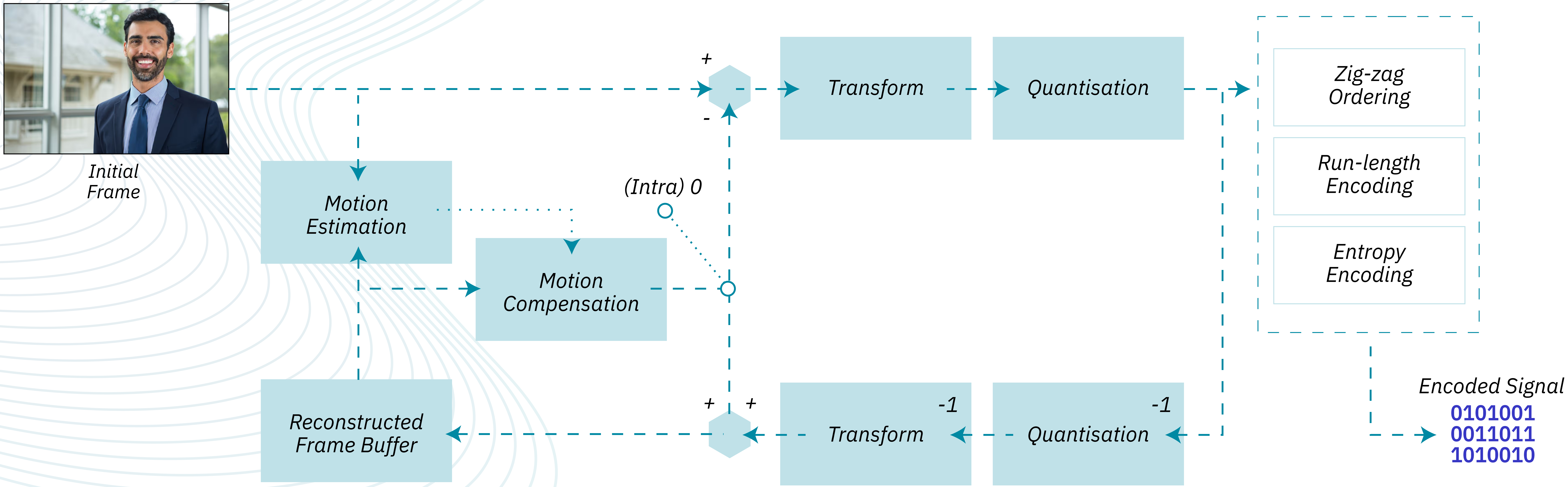


ENCODER STRUCTURE

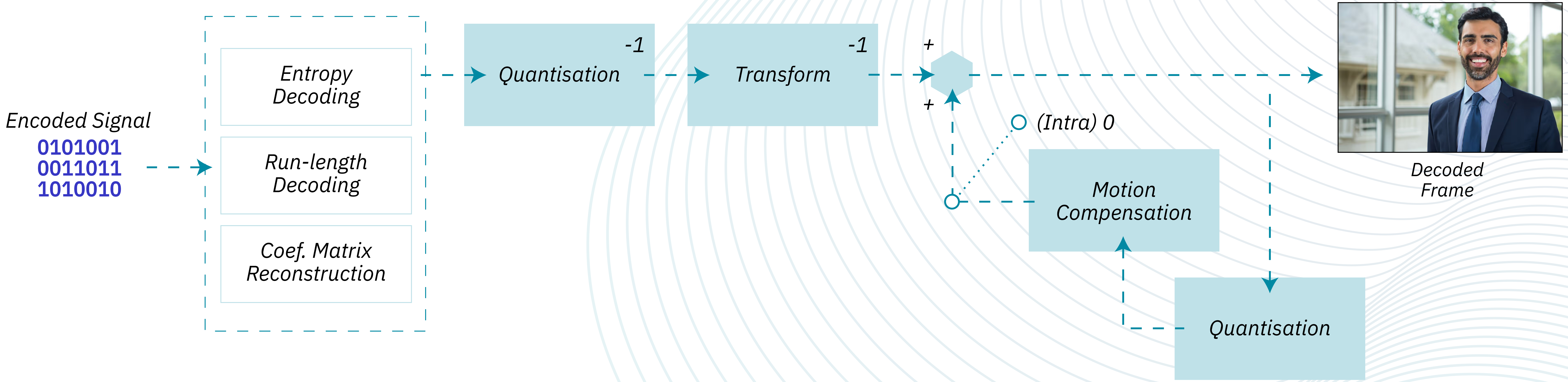
When using DPCM encoding, predictions must be made based on the data processed by the decoder, which is responsible for making predictions based on the reconstructed version of the frame.

Overall, the operations performed by the encoder require much more computational resources compared to those performed by the decoder. For this reason, video encoding is an asymmetrical process: it is executed only once by the broadcaster in a broadcast scenario, while decoding is performed multiple times by each individual end user.

ENCODER STRUCTURE



DECODER STRUCTURE



Decoding is a simpler process compared to encoding, and the main operations performed by the decoder, namely those involving the reconstruction of frames, are also carried out within the encoder architecture.

MULTI-PASS ENCODING

If the encoding needs to be performed "offline" – e.g. for storing content locally or streaming on-demand videos – the time required for compression becomes less relevant. In these cases, it is possible to adopt multi-pass encoding strategies, which include the following phases:

- Analysing the characteristics of the video content to be encoded.
- Selecting, based on previously acquired information, the most efficient way to allocate the available bits.
- Performing the encoding with the chosen parameters.

VIDEO ENCODING FORMATS
- H.261

It is the first video coding standard in the H.26x family.

Designed by the ITU-T's Video Coding Experts Group (VCEG), it was released in 1988 and developed for use in ISDN transmissions, where data is processed at multiples of 64 kbit/s. The standard supports two image formats: CIF (luminance: 352x288, chrominance: 176x144) and QCIF (luminance: 176x144, chrominance: 88x72). Both formats use a 4:2:0 subsampling scheme.

It was the first video encoder to ever use macroblocks as processing units.

VIDEO ENCODING FORMATS
- H.261

The main features of the H.261 standard include:

- **Predictions:** based on differences between the current and the previous frame. Both I and P frame types are employed.
- **Transform:** it uses a Discrete Cosine Transform (DCT) with an 8x8 block subdivision. Employed to reduce spatial redundancies.
- **Quantisation**: Allows for optimal compression by accurately representing the coefficients of the DCT.
- **Entropy Encoding**: Utilises Huffman coding (lossless).

H.261 was the first practically established standard and represents a milestone in the development of video coding. All subsequent video compression formats have drawn inspiration from the design of H.261.

VIDEO ENCODING FORMATS
- H.264 / AVC

The Advanced Video Coding standard, also known as H.264 or MPEG-4 Part 10, is a video compression standard developed by the ITU-T's VCEG in collaboration with ISO/IEC JTC1 Moving Picture Experts Group (MPEG). Despite being more than 15 years old, it is currently the most widely used format for recording, compressing, and distributing video content.

The project's goal is to create a standard for delivering a decent video quality at significantly lower encoding rates than previous standards. The first version was released in 2003, and subsequent editions added extensions to its set of features, such as support for resolutions up to 4K (4096×2160) at 60 frames per second (fps). AVC is protected by various patents, and the licence for patented AVC technologies is managed by MPEG LA. The commercial use of this technology is subject to royalties.

VIDEO ENCODING FORMATS
- H.264 / AVC

- **Transform:** this method uses a 4x4 transform, similar to a 4x4 DCT, which enables integer operations, facilitating processing.
- **Predictions**: Prediction modes have been expanded. Each macroblock (MB) uses a greater number of motion vectors (MV), up to 16. Predictions are also extended to I-frame types and use previously coded macroblocks (intra-prediction). Reference areas are no longer limited to the last decoded frame; B-frame types can also be used as a reference, and these references can be from past or future frames.
- **VCL e NAL**: The Video Coding Layer and Network Abstraction Layer are layers introduced in the standard to separate encoding from data transfer.
- **Parameter Sets**: These enable the separate encoding of data, which is particularly useful in the decoding phase. They are part of the NAL elements and can be transmitted out-of-band.
- **Deblocking Filter**: A filter that reduces artefacts of image processing.

  Adaptive Entropy Coding: This adjusts the probability of symbols to be coded based on the occurrences of previous symbols.

VIDEO ENCODING FORMATS

- AV1

To address the challenges posed by licensing conditions related to the use of patented HEVC technologies, several major companies have been making efforts towards new royalty-free video codecs. Following its experience with VP8, Google started the development of VP9 and VP10, Cisco implemented Thor in its video conferencing products, and Xiph, in collaboration with Mozilla, focused on the development of Daala.

These video codecs were showing promising results, but the individual efforts required to make these projects grow were beginning to hinder their development and potential widespread adoption.
To avoid this dispersion of efforts, in 2015, seven major companies (Amazon, Cisco, Google, Intel, Microsoft, Mozilla, and Netflix) decided to combine their patents in the field of video coding and founded the Alliance for Open Media (AOM), a consortium with the goal of promoting the development and adoption of a new joint video codec.

VIDEO ENCODING FORMATS
- AV1

The project created AOMedia Video 1 (AV1), a compression format designed to be:

- **Royalty-free**: Open, interoperable, and with no royalties or license fees for its implementation.
- **Scalable**: Capable of functioning on any modern device and at any bandwidth.
- **Flexible**: Adaptable for both commercial and non-commercial content.
- **Optimized**: Engineered to perform optimally in video streaming services and related applications.

This format provides:

- **Better compression**: Using, at the same perceived quality, fewer bits than previous standards.
- **High-quality real-time video**: Supporting features such as 4K UHD, HDR, and WCG in real-time videos.

VIDEO STREAMING METRICS

The main parameters used to evaluate streaming performance are defined as metrics and serve as the indicators which can assess the quality of a multimedia stream.

The main metrics can be grouped into two categories: those related to Quality of Service (QoS) and those related to Quality of Experience (QoE). The QoE category can be further split into factors related to viewer perception and technical factors.

QUALITY OF SERVICE (QOS)

Quality of Service (QoS) is an objective performance measure for transmission networks and includes important elements such as uptime, probability of interruption, error rates, throughput, and latency.

QoS primarily focuses on the process occurring between an IP and a streaming application, rather than the end users themselves. It is particularly relevant for high-traffic content providers, such as live-streamed sports and musical events, given that lower QoS significantly impacts Quality of Experience (QoE).

According to the CDN distributor Akamai, over 50% of viewers abandon a stream if the content takes more than 5 seconds to load and 76% of viewers will leave a service completely if the streaming experiences frequent buffering.

QUALITY OF SERVICE (QOS)

The most important factors for measuring QoS are:

- Latency
- Buffer latency
- Congestion

- Packet loss/corruption/out of order
- Packet retransmission
- Reordering

- Jittering/packet pacing

QUALITY OF EXPERIENCE (QOE)

Quality of Experience (QoE) refers to how viewers experience live streaming. In particular, it identifies a set of factors that can cause an unsatisfactory viewing experience and ultimately lead to discontinuation of viewing.

These factors can belong to two main categories:

- Perception factors
- Technical factors

QUALITY OF EXPERIENCE (QOE)

- PERCEPTION FACTORS

This set includes components and properties that influence the feelings (perception) of end users of an application, and they are not directly associated with the technical development of live streaming.

For example, various technical issues may contribute to an initial delay, but what the end user will perceive is solely the resulting waiting time.

QUALITY OF EXPERIENCE (QOE)

- **Mean Opinion Score**
- **Waiting time**
  - Initial delay
  - Stalling time
- **Video Adaptation**
  - Switching frequency
  - Amplitude
  - Time in each Layer
- **Video Quality**
  - Spatial resolution
  - Temporal scalability
  - Picture quality
- **Contextrelated factors**
  - Device
  - Content
  - Use

QUALITY OF EXPERIENCE (QOE)

- TECHNICAL FACTORS

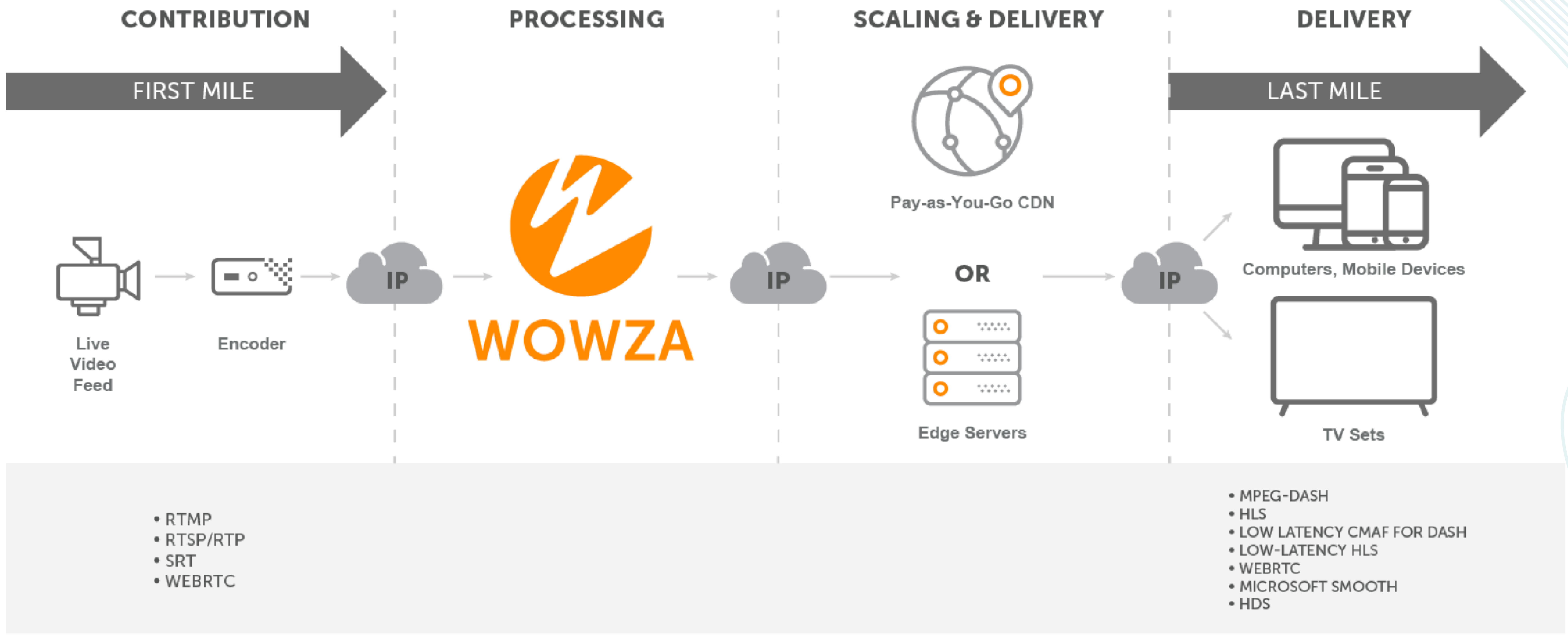
The technical factors that shape viewers' and users' perception play an important role for the perceived QoE of the end product.

The technical factors of QoE include:

- **Server side**
  - Video codec
  - Segment size
- **Client side**
  - Video player
  - Adaptation
  - Video buffer
- **Adaptation logic**
  - QoS monitoring
  - Bit rate estimation
  - Buffer load

LIVE VIDEO STREAMING DELIVERY

LIVE VIDEO OVER IP DELIVERY CHAIN



LIVE STREAMING PROTOCOLS

Protocols for live streaming focus on establishing rules and standardised methods for breaking down video files into smaller segments. This enables quick delivery to the end user. The blocks are then reassembled into a unified video stream on the end device. These protocols primarily operate at the application level but can also be based on various transport sub-protocols, such as TCP/UDP.

Each live streaming protocol may rely on a different transport protocol and therefore employ different strategies for delivering the video stream to the recipient. This results in the generation of a variety of distinct protocols. We will be exploring these in the next slide.

LIVE STREAMING PROTOCOLS

Traditional protocols based on classic TCP/UDP:

- RTMP
- RTSP/RTP

Adaptation protocols based on HTTP:

- HLS
- LLHLS
- DASH
- LLDASH

Latest generation protocols based on advanced TCP/UDP:

- SRT
- WebRTC
- RIST

COMPARING PROTOCOLS

1 RTMP, RTSP e HLS

In a study conducted on mobile networks, RTMP, RTSP, and HLS protocols were tested and compared for IPTV usage. The test video was in a 720p format at 30 FPS, with a bit rate of 8000 kbps and H.264 encoding. The HLS protocol was also evaluated at lower quality resolutions such as 240p, 360p, and 480p, with bit rates of 5600 kbps, 3400 kbps, and 200 kbps, respectively. The authors travelled by car through various network coverages, including 2G, 3G, and 4G, to examine the behaviour of each protocol.

The results indicated that, in this context, HLS is the most suitable protocol in terms of stability, given that it was the only one not to experience disruptions, thanks to its adaptive nature.

COMPARING PROTOCOLS

2 RTMP e protocolli HTTP

Another study examined the impact of network parameters on RTMP and HTTP protocols by transmitting three different video sequences over a controlled network. The videos collected from the client were evaluated based on subjective parameters, measured using Mean Opinion Score (MOS), following the International Telecommunication Union (ITU) standard.

- The RTMP protocol obtains a higher score than HTTP when packet loss ranges from 0% to 5%. However, as packet loss increases, HTTP exhibits slightly superior performance compared to RTMP.
- Under conditions of very low jitter variation, users gave higher scores to RTMP streaming videos than to HTTP video sequences. Conversely, with high jitter variations, HTTP outperforms RTMP.

COMPARING PROTOCOLS

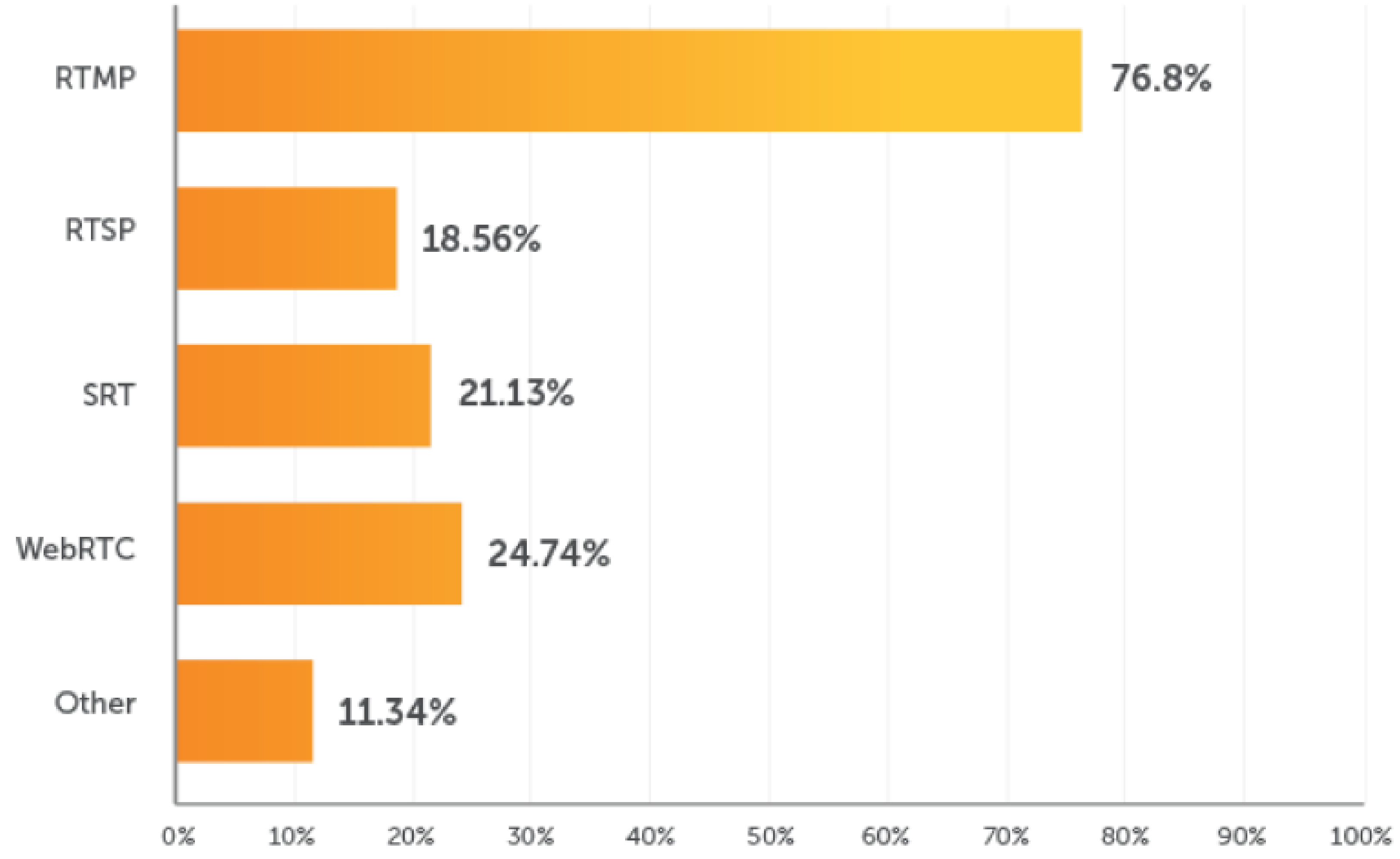
3 RTMP e SRT

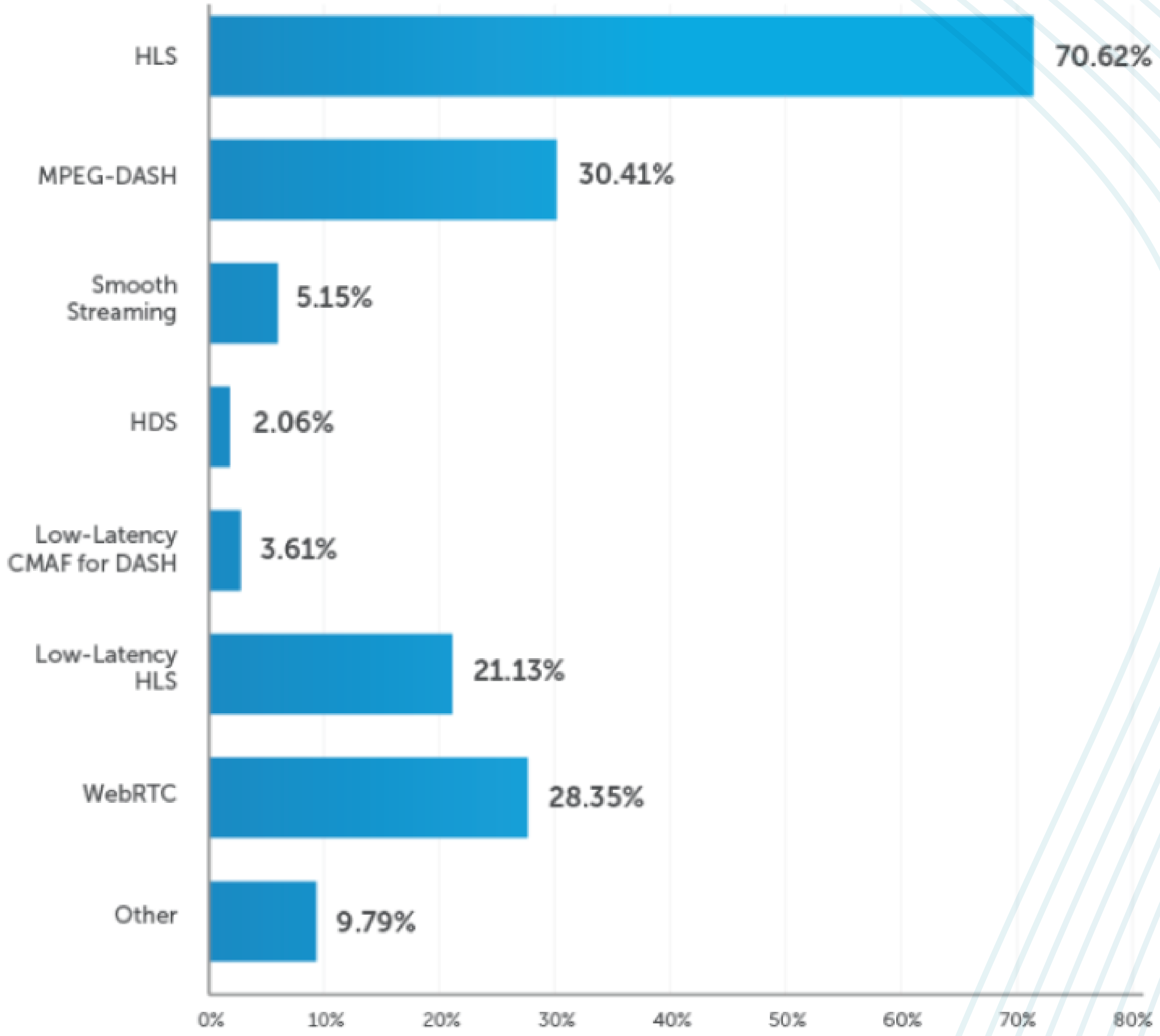
One final study focuses on the comparison between RTMP and SRT protocols, examining end-to-end latency and maximum transferable bit rate on public networks. To assess latency, the authors measured the delay in a stream sent to servers located in different locations using the two distinct protocols.

The results show that SRT encounters no issues in streaming up to 20 Mbps in any region worldwide. RTMP shows good performance when the sender and receiver are in the same continent, in this case North America. In the tests, it was possible to send up to 20 Mbps from Redmond to California or Virginia using RTMP; however, streaming to Europe and Australia was not possible at speeds over 2 Mbps.

USER PREFERENCES

The Wowza 2021 Video Streaming Latency Report shows which protocols are most popular among service providers for both ingest and egress processes.



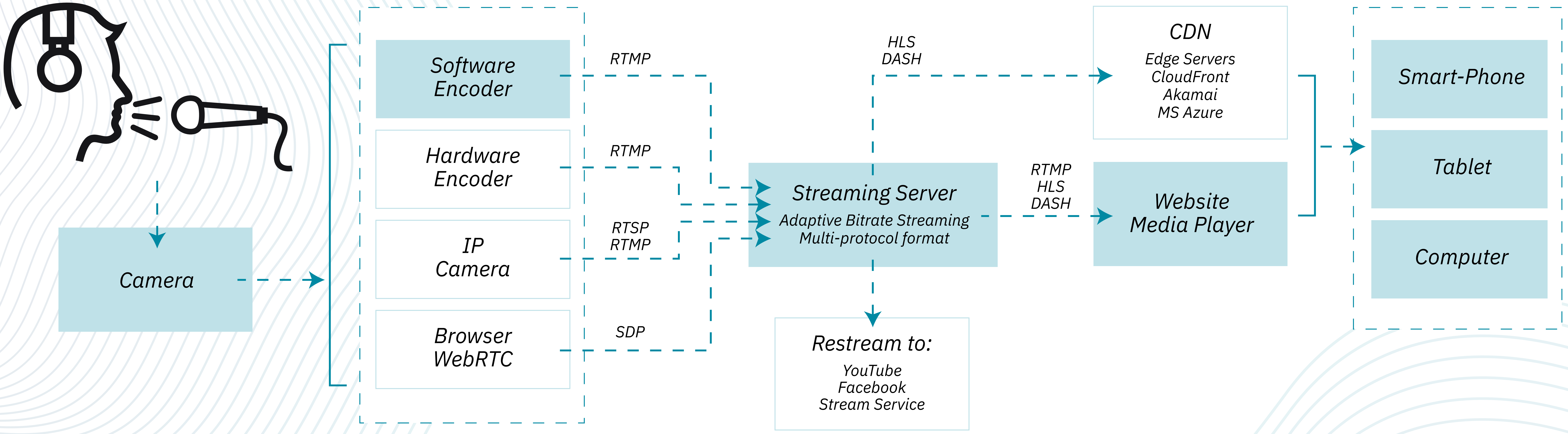


LIVE STREAMING ARCHITECTURE

Live streaming can be categorised into the following stages:

- **creation of multimedia content**
- **data processing**
- **distribution of streaming data**
- **reproduction of streaming content**

LIVE STREAMING ARCHITECTURE



LIVE STREAMING ARCHITECTURE

1 **Content Creation**:

The creation of multimedia content begins with a source that generates an audio and video stream – this can be, for example, a camera, a microphone, or another capturing device for multimedia. The source produces what is called the original video stream (RAW), which is then routed to the next component in the chain.

LIVE STREAMING ARCHITECTURE

2 **Elaboration**:

In streaming, the processing phase usually begins directly at the source device. Here, using a software or hardware encoder, compression algorithms are applied to the data that leaves the source. This operation reduces bandwidth consumption.

The stream is segmented into packets to compensate for and better adapt to the varying speed of the network. Each packet is marked with a timestamp that the receiver can use to correctly rearrange the received packets, given that they are not received in the order needed for playback. The encoder also takes care of transmitting the data to the processing server via an ingest protocol, such as RTMP.

LIVE STREAMING ARCHITECTURE

The processing server is responsible for transcoding the streamed data. It generates video streams with different resolutions and bit rates, thus enabling Adaptive Bit rate (ABR) streaming. In this way, clients will require lower bandwidth to receive streams, thus preventing potential interruptions during the streaming experience. The server also manages transmission to the distribution network following output protocols such as RTMP, HLS, or MPEG-DASH.

Another operation performed by the processing server involves sending the streaming flow to another processing server on a different streaming platform, enabling restreaming on various services using a single outgoing connection from the source.

LIVE STREAMING ARCHITECTURE

3 **Distribution**:

Today, the distribution of streaming data is mainly carried out via Content Delivery Networks (CDN). 'CDN' refers to a group of servers distributed strategically across the internet that work together to ensure the fast transmission of online content.

In addition to enhancing access speed to specific content, a CDN can also provide additional security services, helping to protect against potential malicious attacks.

LIVE STREAMING ARCHITECTURE

4 **Receiving Data**:

The quality of an incoming/received streaming flow is a key factor in ensuring a smooth and satisfactory experience for end users. In the context of video streaming, bit rate management plays a fundamental role in the effective transmission of data.

Bit rate indicates the amount of data transmitted during a specific time interval; it is expressed in bits per second (bps). In streaming, the bit rate is directly proportional to the quality of the video: the higher the bit rate, the better the image quality. Therefore, when designing a video streaming system, it is crucial to also consider what the available bandwidth could be.

STREAMING PLATFORMS

Vimeo offers streaming and video hosting services.

The core functionalities include data analytics and insights, a video management platform, privacy checks, and more.

To access basic live streaming features, users need to subscribe to a Premium plan. Vimeo provides technical support by email for hosting plans. For live customer assistance, users must subscribe to a custom Premium or Vimeo OTT plan. Live support is part of the higher-budget option on the Vimeo platform.

STREAMING PLATFORMS

Features:

- Video management service
- High-quality videos and streams
- Large storage capacity
- Privacy options
- Upload from anywhere

Pros:

- Unlimited events and spectators
- No in-stream advertisements
- Sleek, professional appearance
- Detailed analytics and insights reports
- User-friendly
- Pay-per-view plans

Cons:

- Generates less traffic than other similar platforms
- No video delivery in China

STREAMING PLATFORMS

Restream

Restream is a platform for live video broadcasting directly within a browser. It is a popular choice in various sectors including gaming, technology, public administration, sports, media, and music.

Although it is primarily a multi-streaming platform, Restream also offers support for collaborative in-browser streaming alternatives.

STREAMING PLATFORMS

Restream

Features:

- In-browser streaming
- Multi-streaming
- Engagement-boosting features
- Support for peer-to-peer streaming
- Tools for audience interaction
- Brand customisation
- Scheduled events

Pros:

- Easy to use
- Collaboration tools
- On-brand streaming
- Audience engagement tools

Cons:

- Not a dedicated streaming platform
- In-browser streaming is secondary to live hosting

STREAMING PLATFORMS

YouTube Live is a free video streaming platform designed for users with little to no prior broadcasting or streaming experience.

YouTube was primarily intended to serve as a video sharing platform. While it does support both VOD and live streaming, its popularity derives mostly from its on-demand video content.

STREAMING PLATFORMS

YouTube

Features:

- Monetisation for users with large followings
- No notable security features
- Popular among consumers
- Powered by Google
- Easy to share videos
- No white-label capabilities

Pros:

- Free
- Easy to use (for viewers and streamers/broadcasters)
- Live hosting and Video On Demand (VOD)

Cons:

- Limitations on live streaming
- Harsh restrictions on content
- Platform owns partial rights to content

STREAMING PLATFORMS

Facebook is a social media platform that offers tools for uploading and sharing videos within the online community.

To maximise the benefits of using Facebook Live, users can integrate this platform with a professional live streaming service: this way, broadcasters can multicast content on other sites.

When streaming on Facebook Live, users can reach existing Facebook audiences.

STREAMING PLATFORMS

Features:

- Easy video sharing
- Live commenting and reacting
- Paid ads
- Business tools

Pros:

- Free
- Well-known platform
- Live chat boosts engagement
- Leverages existing following

Cons:

- Limited monetisation options
- Time limits for live-streamed videos
- Facebook Live branding cannot be removed

STREAMING PLATFORMS

The majority of TikTok users are “Gen Z-ers” – this means that most viewers on the platform are aged 25 and younger. This platform is therefore best suited for businesses with a target audience in that age group.

While TikTok does offer some live streaming tools, the platform is mainly used for sharing and watching short video clips on-demand. Supported video lengths include 15 seconds, 60 seconds, and 3 minutes (180 seconds).

STREAMING PLATFORMS

Features:

Fast sharing of on-demand content
Restricted target audience
Familiar content and formats

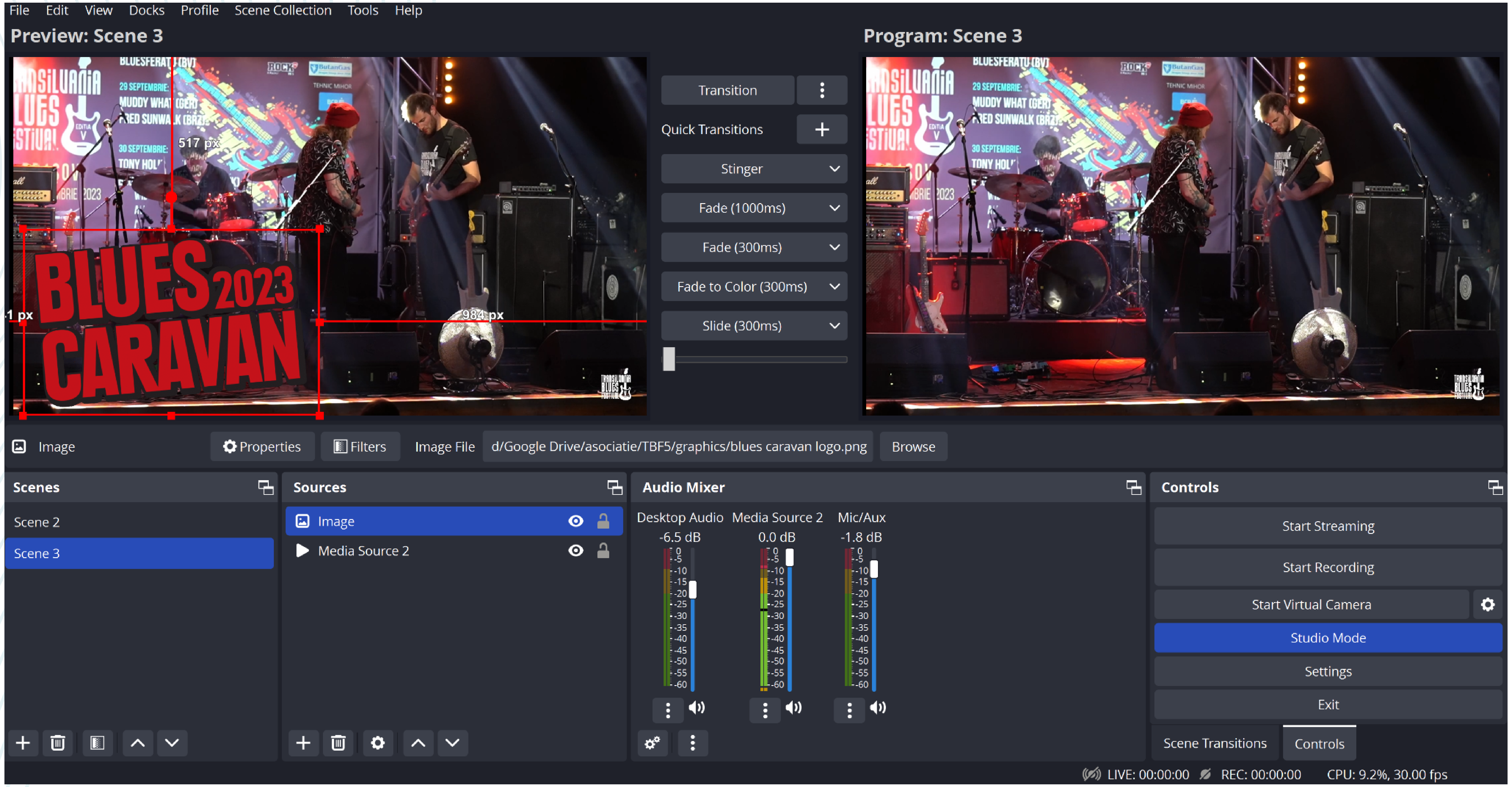
Pros:

Free
Virality algorithms
Suitable for fast growth

Cons:

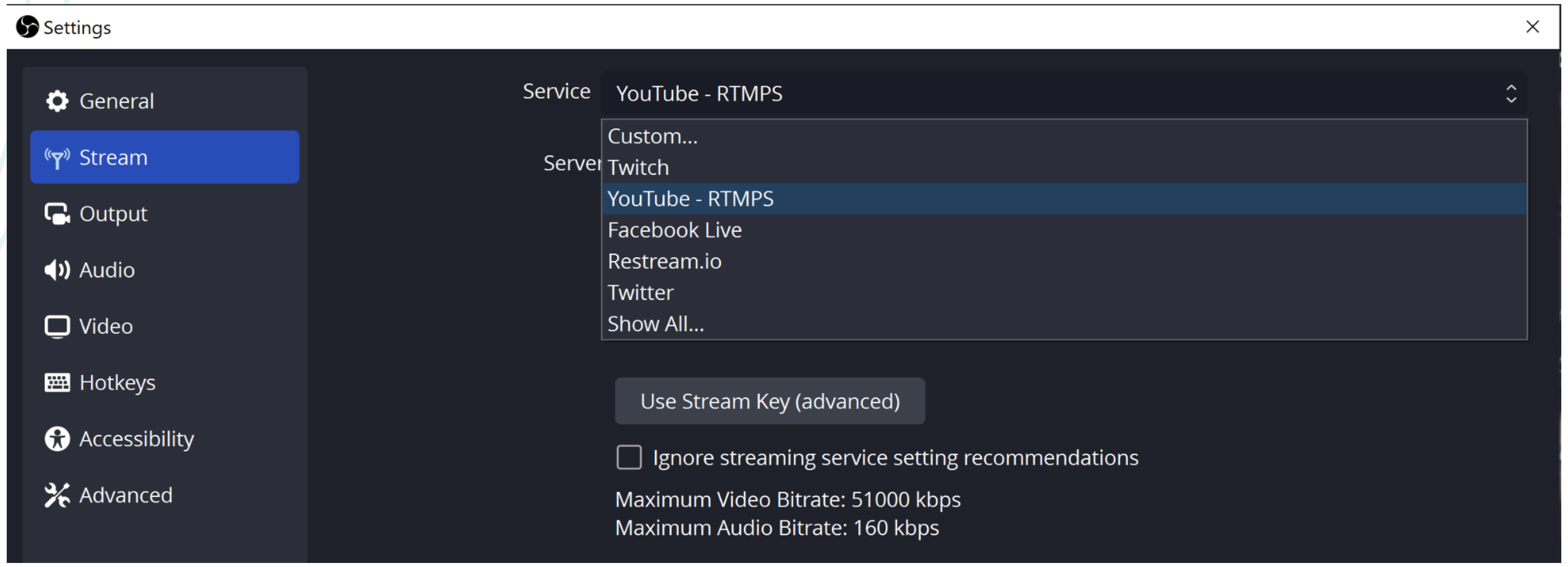
Extremely limited profit/monetisation options
180-second limit for on-demand videos
TikTok branding cannot be removed
Ongoing privacy issues and concerns
Following needed for live streams

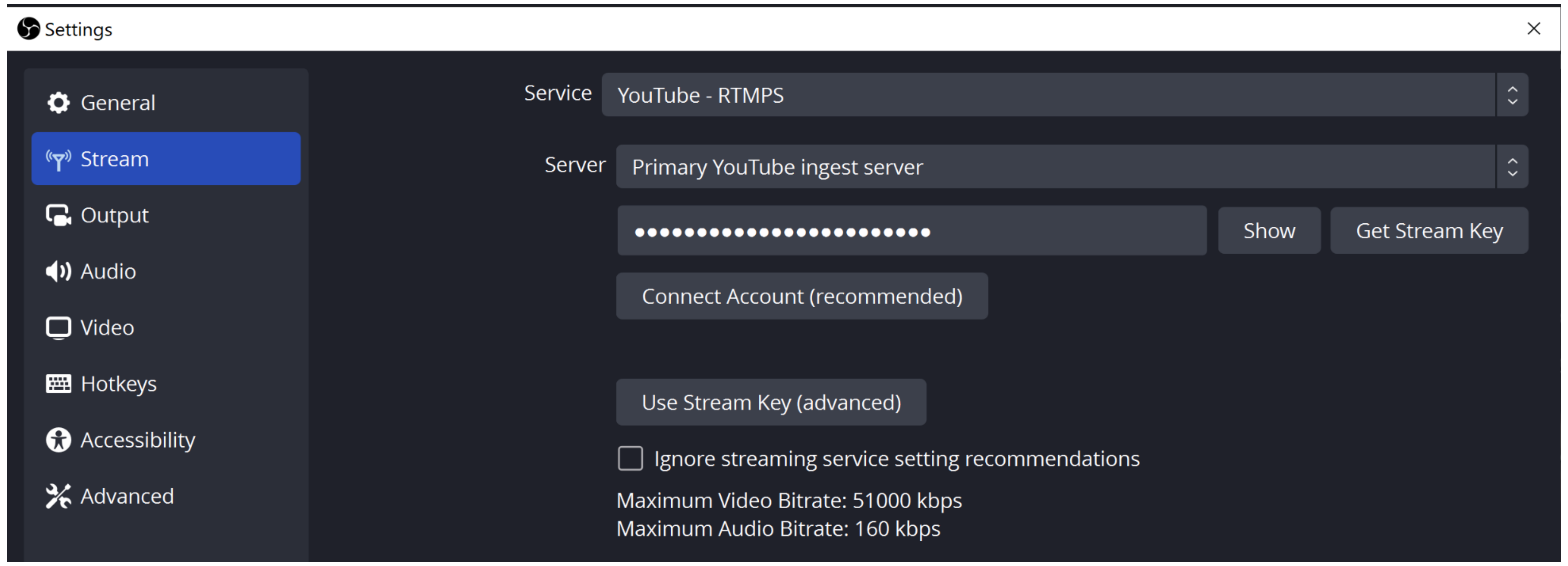
OPEN BROADCASTER SOFTWARE (OBS)



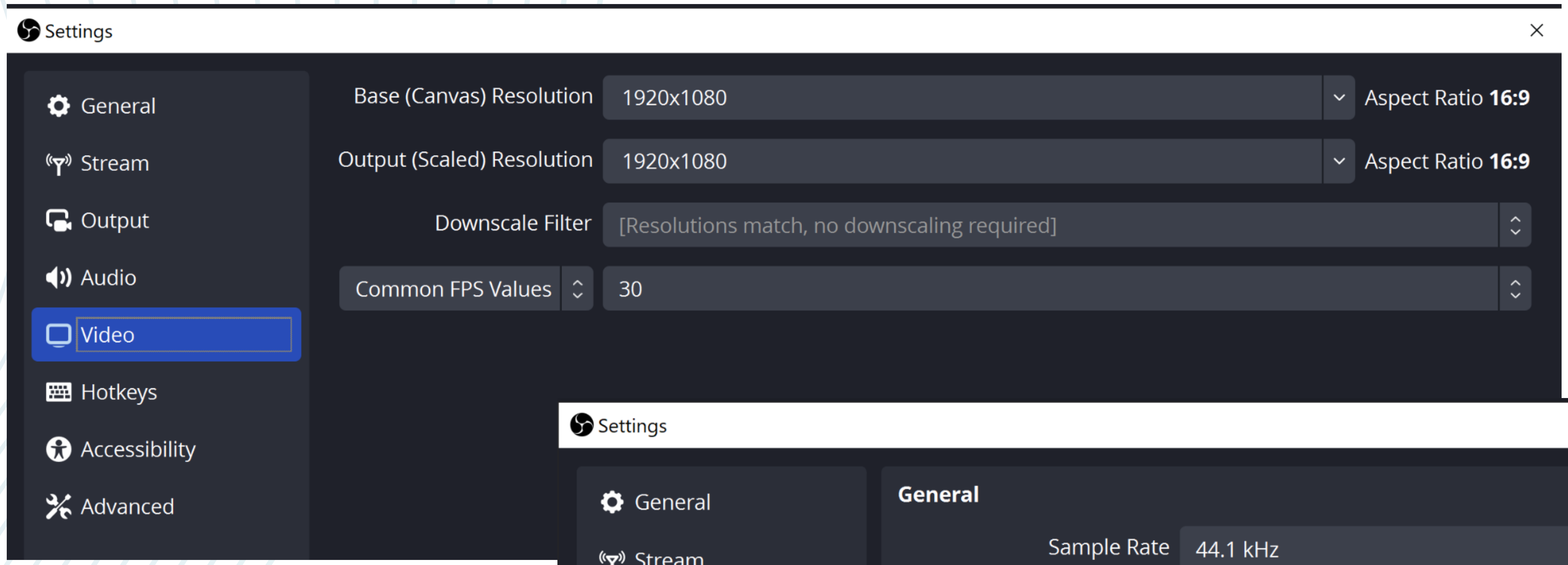
OPEN BROADCASTER SOFTWARE (OBS)

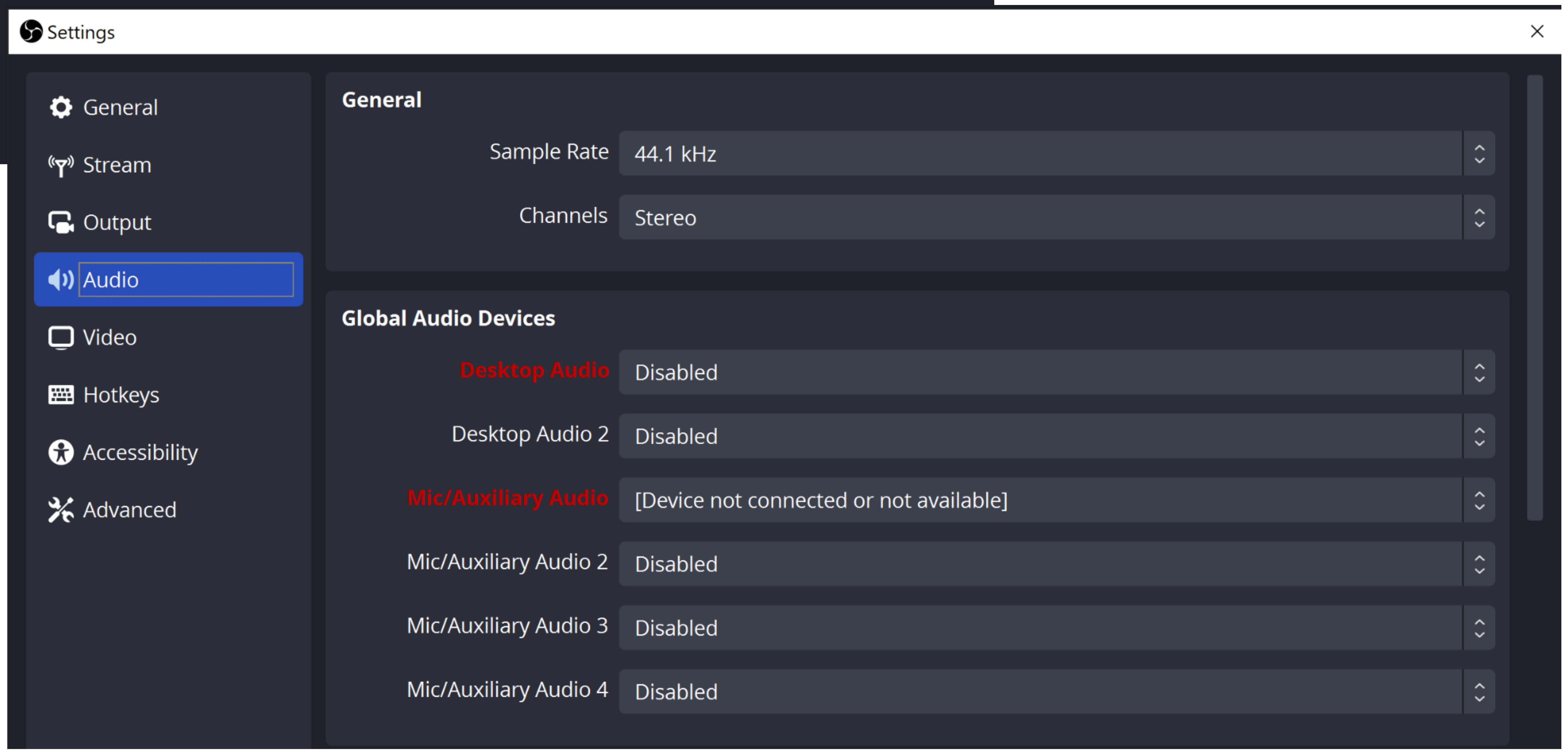
- STREAMING SERVER CONNECTION





OPEN BROADCASTER SOFTWARE (OBS)
- SETTINGS





MULTI-PLATFORM STREAMING

Multi-streaming techniques are driving the expansion of the streaming process to include multiple systems and platforms. By using such methods, it becomes possible to simultaneously broadcast content to various user categories, rather than being limited to sharing content to the audience of a single platform.

Multi-streaming is also referred to as "simulcasting," "multicasting," or "multi-destination streaming." These terms all describe the same concept of streaming to multiple destinations (e.g. platforms) simultaneously.

OBS does not support multi-streaming; for this technology, a multi-platform streaming service, such as Restream.io, or a dedicated custom server is required. Users can choose between a streaming service or a custom server in the program’s dropdown settings menu.

OPEN BROADCASTER SOFTWARE (OBS)
- BANDWIDTH

OBS uses the best open-source video encoding library, x264, for video encoding. This can result in high CPU usage, and other programs running on the computer might experience reductions in performance when OBS is open and active, especially if the program settings exceed the computer's hardware capabilities.

In certain cases, OBS may show an "Encoding overloaded!" message on the status bar, indicating that the computer cannot encode the video quickly enough to maintain the desired settings. This may cause the video to freeze after a few seconds and it may lead to periodic stuttering in the video signal.

OPEN BROADCASTER SOFTWARE (OBS) - TROUBLESHOOTING

1

Reduce Output Resolution

The resolution of a streamed video is the aspect that most significantly affects CPU usage levels. For instance, a 1080p resolution has more than twice the number of pixels in each frame than a 720p format. As a consequence, CPU usage increases. The most common way to reduce CPU load is, therefore, by lowering resolution.

Downscaling settings can be adjusted in Settings > Video > Resolution. The base resolution (canvas) can be kept the same to ensure the layout remains unchanged. The various downscaling filters (bilinear, bicubic, and Lanczos) simply correspond to the different algorithms used to reduce the image. Bilinear filtering is faster and requires fewer resources but does not provide excellent image quality, whereas the Lanczos filter is more resource-heavy but outputs an end product with visibly better quality.

OPEN BROADCASTER SOFTWARE (OBS)
- TROUBLESHOOTING

2

Reduce Frame rate

When streaming at more than 30 frames per second (fps), it is recommended to lower the frame rate to 30 fps. This lightens CPU load.

3

Cahnge x264 preset

The x264 video encoder has various "presets" that can alter video quality and CPU usage levels. The default setting in OBS is very fast and provides the best balance between CPU usage and video quality. This setting can be modified in Settings > Output. Although the image may appear slightly more blocky or pixelated, this setting allows users to maintain their desired resolution and frame rate.

VLAD POPESCU
Streaming technician