DIGITAL MEDIA

FRAUNHOFER DIGITAL MEDIA ALLIANCE

#WHATSNEXT IN DIGITAL MEDIA

TRENDS AND TECHNOLOGIES IN DIGITAL MEDIA

#WeKnowHow

PREFACE

We cannot change the wind, but we can set the sails differently". (Aristoteles)

The Coronavirus pandemic in 2020 set new priorities in our business, public and private lives. In addition, it teaches us to rethink our usual workflows, adapting them to new requirements and making them more flexible.

In the media industry an unprecedented demand for audiovisual content and communication over the Internet calls for new technologies, codecs, analytics and solutions for crisp images and sounds to enhance our experiences when we chat, discuss or exchange topics or share content. And even more with virtual trade shows, we test how to present new developments to a broader audience.

Fraunhofer Digital Media Alliance accepts the challenge: We provide technological insights via the IBC 365 platform and set up our own virtual booth for IBC 2020. Major topics here are MPEG-H authoring tools to realize immersive and personalized sound adaptable for every surrounding and device. Video codecs and implementation as JPEG XS for transferring up to 8k video over IP in production quality as well as VVC (Versatile Video Codec), the successor of HEVC, for HD/UHD content streaming. Creating free viewpoint volumetric video from real-life scenes is one of the future solutions to go beyond fixed display presentation with static viewpoint.

AI (artificial intelligence) and its power for streaming analytics, coding, and especially for data mining and audiovisual quality control, fake detection and so on, is among the key technologies that will influence the way of content production and transmission.

IBC2020 will show many ways to enhance media workflows even under changing conditions. We hope to see you in our virtual booth at www.ibc2020.digitalmedia.fraunhofer.de and as usual, please enjoy reading our digital trend brochure.

Sincerely, yours

Dr. Siegfried Foessel
Spokesman Fraunhofer Digital Media Alliance

FRAUNHOFER DIGITAL MEDIA ALLIANCE
SOLUTIONS THAT CREATE THE FUTURE OF MEDIA

– Every second bit of the worldwide data traffic can be sent thanks to Fraunhofer technologies.

– mp3, AAC and HE-AAC are today in all consumer electronic devices, PCs and smartphones.

– Nearly 50 percent of the bits transported via the Internet are compressed using the video coding standard H.264/AVC, co-developed by Fraunhofer HHI.

– The Apple and Android ecosystems are based on AAC, for instance music distribution, Facetime, Airplay, or Carplay.

– Half of the world's digital broadcast TV sound and most digital radio systems are enabled by Fraunhofer IIS audio codecs.

– xHE-AAC is natively integrated in Android and standard codec for Digital Radio Mondiale.

– MPEG-H Audio is used in Korean UHDTV since May 2017.

– SpatialSound Wave: First to market in object-based audio.

– Almost every entertainment device uses audio and video codecs of Fraunhofer.

– The easyDCP software suite for movie content mastering and packaging is one of the most reliable tools in the industry.

– AudioID: Pioneer in audio matching since 2002.

– The 3GPP communication codec EVS is in use worldwide, for example in Europe, USA, China, Japan and South Korea.

– World leading in speech recognition transcribing audio and video files in real time with a vocabulary of over 2 million words, processing any voice/dialect.

Fraunhofer IDMT

AUDIO FORENSICS: DETECTING FAKES

The CEO’s voice came through loud and clear: his instructions were to transfer a large sum of money to a certain account. This order, however, was faked – something that was only discovered once it was too late. At Deutsche Welle, the German public international broadcaster, falsifications are set to be exposed more quickly in the future: Within the project “Digger”, Fraunhofer IDMT is integrating its audio forensics technology into DW’s content verification platform “Truly Media”.

“That’s a great quote! But is it real?”– it is not easy for journalists to expose forgeries, as fake audio and video material is becoming increasingly credible. Fakes can be created by means of skilled editing, but also by imitating the speech and intonation of a speaker very convincingly using artificial intelligence.

Researchers at Fraunhofer IDMT are working on uncovering forgeries of both kinds: By analyzing “footprints” that are added during the recording – such as characteristic traces of microphones – or by identifying inconsistencies in the material that result from editing. This is because “every content processing step leaves traces that can be detected with forensic tools,” according to Patrick Aichroth, Group Manager at Fraunhofer IDMT.

In the joint project “Digger”, funded by Google DNI, Fraunhofer IDMT is integrating their audio forensics technologies into the web-based content verification platform “Truly Media”, which is developed by Deutsche Welle and Athens Technology Center. From 2021, with the help of this Fraunhofer IDMT “a detective’s toolkit for fakes,” journalists will be able to carry out analyses of audio material and detect manipulations.

DIGGER
deepfake detection

Contact:
Patrick Aichroth
patrick.aichroth@idmt.fraunhofer.de
+49 3677 467-121

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AUTOMATIC ANALYSIS OF TV AND RADIO BROADCASTS

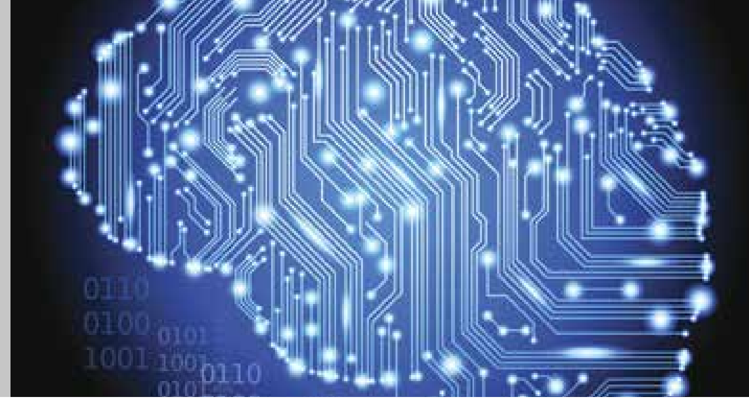
We are all used to matching apps that can help us find out which songs are playing on the radio, but researchers are also working on a similar technology that can help broadcasters and media companies to track metadata and find partial overlaps and duplicates — it's known as partial matching. Patrick Aichroth, Group Manager at Fraunhofer IDMT, explains what it's about.

What does the term "partial matching" mean?
With conventional matching, the goal is typically to identify content, such as a song or a video, using a snippet or sample of that material. With partial matching, on the other hand, your aim is to find any partial overlaps within a dataset, without knowing beforehand whether or where such overlaps exist. Both cases of matching may sound similar, but there are somewhat different technologies involved.

What can this technology do that matching cannot?
While matching enables the analysis of when and how often a piece of content has been broadcast, partial matching can analyze entire TV or radio programs or datasets and identify partial overlaps. This can be used for de-duplication, to automatically propagate metadata within archives, or find out which parts of a preproduced item were actually broadcast, which is useful to create program cue sheets and clear rights. We have already carried out a few test runs with broadcasters and the results have been encouraging.

What does the future hold for partial matching?
We are currently running further tests and will then put the technology into a first product that can be used to detect and localize differences between pre-produced material and broadcasted productions, which should be available by early 2021. Afterwards, we plan to provide further product versions for de-duplication and metadata / rights tracking.

Contact:
Patrick Aichroth
patrick.aichroth@idmt.fraunhofer.de
+49 3677 467-121



ARTIFICIAL INTELLIGENCE AND ITS DRAWBACKS

Does the future belong to artificial intelligence? Can artificial intelligence solve pressing problems facing humanity? It can certainly help – if we recognize the limits of the technology and apply its strengths correctly. The trick is knowing how.

Analyzing radiological images and assisting physicians in evaluating them, recognizing cars in traffic scenarios, even distinguishing dogs from cats – artificial intelligence can do all of this brilliantly. But successes such as these are rapidly raising expectations for this technology.

“Many people see intelligence as a system that thinks for itself and is very smart,” says Hanna Lukashevich, group leader at Fraunhofer IDMT. “But they’re wrong: artificial intelligence can only do what it has been taught in advance – and therefore performs particularly well in well-controlled environments. If, on the other hand, they are applied to versatile content – such as analyzing any kind of audio files with different recording quality and diverse content – this often leads to unexpected effects.”

So is it best to stay away from artificial intelligence?

“Certainly not! However, it is essential to define the appropriate model in advance – e.g. distinguishing between dogs and cats – and to train the model accordingly. If the developer did not include zebras in the definition of the model, the artificial intelligence will not be able to recognize zebras either,” explains Lukashevich. After all, artificial intelligence is one thing above all else: a tool. And just as nobody would think of drilling a hole in a pane of glass with a wall drill, artificial intelligence must also be applied correctly. The fact that it’s not always obvious what the AI model is used for makes the situation more difficult. “The most important thing is that the system is trained with data that is representative for the application,” says Lukashevich. „If, say, a model has only been trained with high-quality audio data, it will later struggle with telephone-quality audio data.“

What if the artificial intelligence just won´t do what you want?

Then it’s time to adapt the training data. “AI isn’t magic; it’s mostly mathematics,” Lukashevich assures us. “To put it another way, there’s always a reason why something won’t work. By and large, everything can be solved by adapting the AI components to the use case. Or have them adapted: the researchers at Fraunhofer IDMT will be happy to help.”

Contact:
Hanna Lukashevich
hanna.lukashevich@idmt.fraunhofer.de
+49 3677 467-224

MINING PLATFORM: AUTOMATIC MULTIMEDIA CONTENT ANALYSIS AT SCALE

Many valuable data treasures are hidden in media archives. Extracting these treasures is still difficult – manually annotating text, images, and audio and video files with content-related metadata is very laborious, time consuming and can only be created for a limited amount of media files.
Now, however, the Fraunhofer IAIS Mining Platform is capable to analyze almost unlimited quantities of multimedia documents with high precision and flexibility. The automatically generated metadata makes valuable content much more accessible to journalists and other media creators, as well as allowing searching by topic.

A modular and extensible system
The Mining Platform has a modular structure, allowing the integration of almost any metadata extraction method. Components for named entity recognition, keyword extraction, topic modeling, and semantic tagging are already integrated for analyzing text documents.
Spoken material can be automatically transcribed with the high performance Fraunhofer IAIS AudioMining solution. Furthermore, visual related metadata such as faces can be detected in images and videos.

The Fraunhofer team is currently developing and integrating additional analysis services, such as for detecting cuts and scenes, key frame extraction, and object recognition. Methods developed by third-party providers can also be integrated easily on request. It is also possible to train customer-specific models for the existing analysis services, thus replacing the ready-made models – an advantage not offered by other systems so far.

Step-by-step analyses thanks to workflow components
The individual steps involved in analyzing a media file are controlled by a workflow component. This has several benefits: the architecture allows failed analysis steps to be repeated and enables the prioritization of the scheduled analyses. This is helpful when certain media data has priority over other data in the analysis, for example in the case of live events.

The workflow component can also be used to model more complex processing sequences, for example when the output of an analysis service serves as an input for subsequent services.

For example, a video can be analyzed using facial recognition, while audio mining can transcribe the speech captured on the audio track and even recognize the separate speakers.

The metadata – the people recognized in the image, and the speakers recognized – can then be linked together.

The system can, say, detect when a person is visible on screen while also talking. In another step downstream, the transcript can be further analyzed using text mining methods. This means that not only can the system determine when which person can be seen or heard in a video, but also what topics, people, places, or institutions are mentioned.

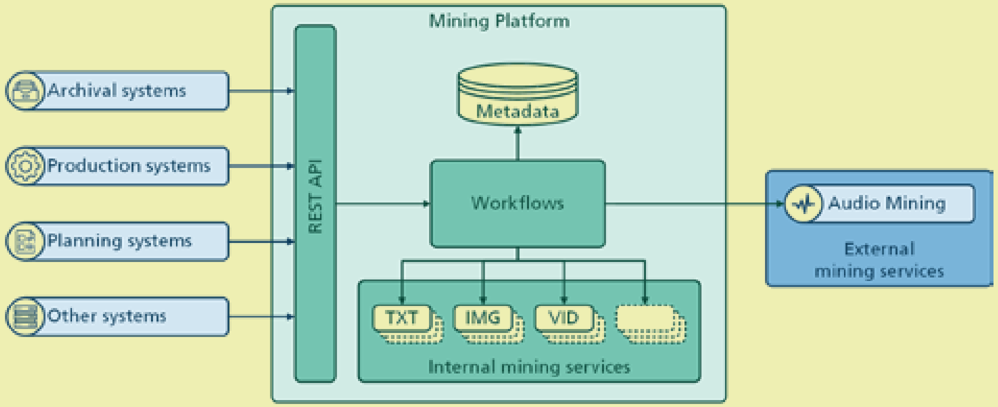
To allow the Mining Platform to scale according to the quantity of media data to be processed, it uses a microservice-based architecture and is operated in a Kubernetes cluster. Thus, the number of instances of the individual analysis services can be adapted to required workload.

Operating the Mining Platform in a Kubernetes cluster also has the advantage that it can be operated in a private cloud or directly on-premises. Cyber sovereignty is guaranteed at all times. Connection via REST interfaces means that the Mining Platform can be easily integrated into other broadcast systems.

Cross-media searching: the Mining Platform successfully developed with and deployed by ARD

The Mining Platform is being developed in a strategic partnership with Germany's public broadcasting network, ARD, where it is being used for applications such as a cross-media search system.

Contact:
Dr. Jens Fisseler
jens.fisseler@iais.fraunhofer.de
+49 2241 141974



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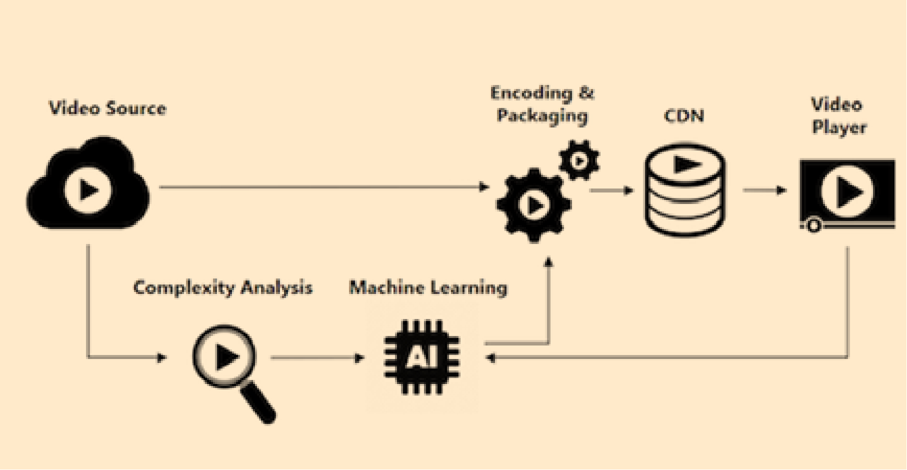
AI-POWERED ENCODING AND OPTIMIZED STREAMING

Nowadays, the world's Internet traffic is predominantly made up of video streaming. In order to support the Internet's limited transmission capacities, videos are compressed during encoding. Reducing video file sizes comes at an expense, however, when disseminating large volumes of streaming video via the Internet, resulting in a loss in quality.
Live streaming, in particular, has special requirements with regards to the video transmission process, and can benefit from intelligent approaches in compressing content in a more efficient manner.

Our FAMIUM Deep Encode solution utilizes artificial intelligence methods for automating per-title encoding for Video on Demand and live streaming.

During a live playout, we analyze the video and collect existing playback metrics in order to predict the upcoming optimal ladder. With the prediction, encoding settings are then adapted accordingly.

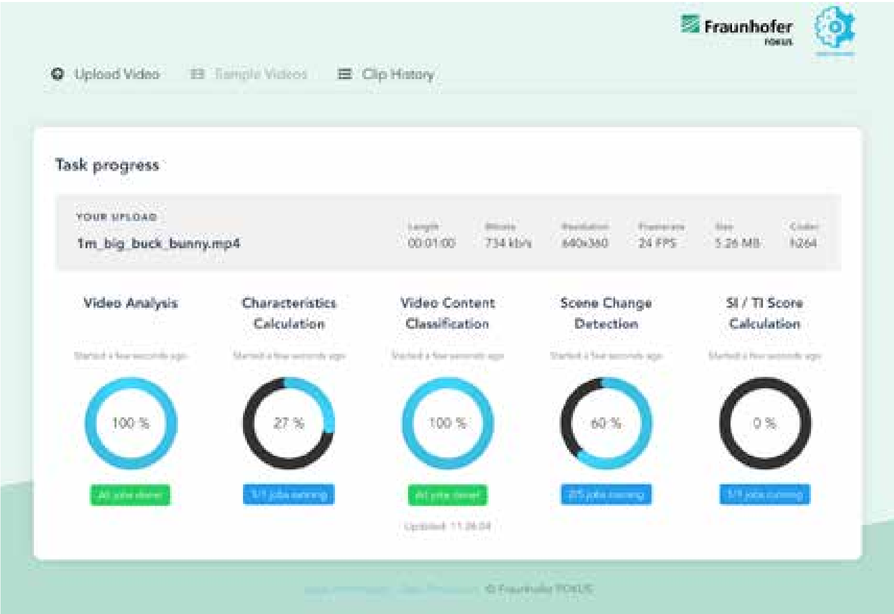
Video analytics is conducted on a per-scene basis in order to adapt settings to the current scene. In comparison to traditional encoding solutions, average storage and transfer volumes are reduced by 30%. This large decrease leads to significant cost savings in the long-run, and improves the overall Quality of Experience for the end user. This FAMIUM Deep Encode solution is codec and format agnostic, and the quality prediction (VMAF) is based on several extracted unique video characteristics.



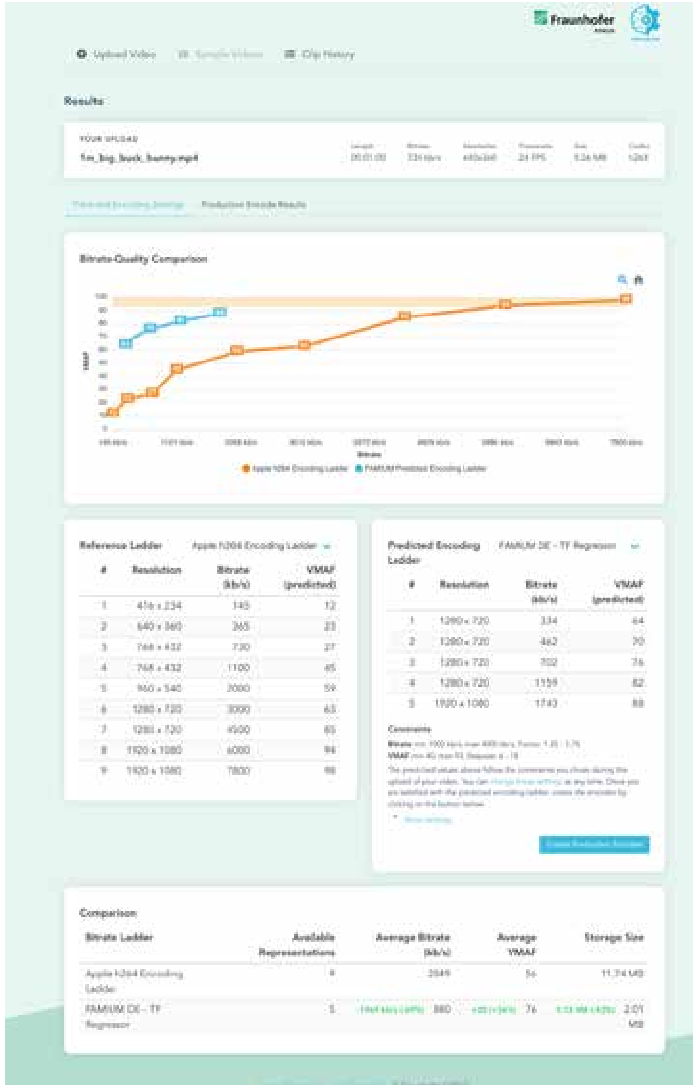
Key Benefits:

- Supports VoD and live linear content
- Encoding and streaming workflow is dynamically optimized by collecting metrics with a feedback loop Format-agnostic with established video codecs
- Prediction of video quality metrics (VMAF) without the need of test encodes

Contact:
Robert Seeliger
robert.seeliger@fokus.fraunhofer.de
+49 30 3463 7262



Automatic analysis, classification and
metric calculation of video content by
Deep Encode



Predicted Encoding Settings and optimal Bitrate Ladder

LOWEST LATENCY FOR VIDEO OVER IP TRANSMISSION JPEG XS

The standardized JPEG XS codec was developed to handle media workflows from acquisition to distribution by using Ethernet settings and infrastructure only. Until very recently, digital image transmission for production and contribution could be done only by using specific interfaces such as SDI, IEEE1394, or CameraLink. However, with the availability of higher bandwidth of Ethernet interfaces, the handling of highest-quality images over internet protocol (IP) in local and wide area networks was required and JPEG XS is a codec enabling these requirements.

An update to the new video production codec for professional video

A low compression of up to 10:1 allows near-transparent transmission.
JPEG XS – developed to offer lowest l atency for multiple encoding-decoding cycles and moderate computational resource requirements while preserving image quality at the highest level – fulfills these demands to facilitate production/ contribution settings, even for 4 and 8k images.

The core coding system of JPEG XS was standardized in ISO at the end of 2018 as ISO/IEC 21122-1, the remaining parts in 2019.

What is available for industry applications today are the compression of RGB and YCbCr images in 444 and 422 sampling formats with up to 12 bits per component for broadcast and prosumer use cases. Some smaller extensions, like compression of 420 sampling formats and lossless compression, are under development.

Integration of JPEG XS into cameras and image sensors

The current standardization activity is a big step forward to enable JPEG-XS for compression of RAW Bayer image data. During this JPEG XS development phase, a PSNR gain of 5 dB in coding efficiency could be achieved and will be included in a new amendment.

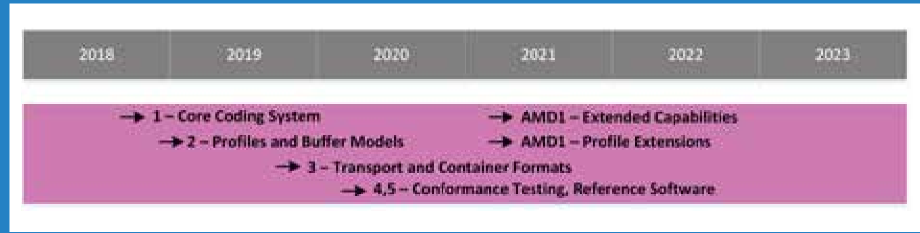
This allows the industry to integrate the codec into today's cameras and image sensors. It offers the use of the codec in the complete production pipeline – from the image capturing to the distribution encoder. It facilitates the use of the codec in other use cases, like integration in cameras for machine vision, automotive, or high quality surveillance, too.

JPEG XS already exists as transport and file formats, like RTP, MPEG2-TS, JXS, MP4, and HEIF. The standardization of JPEG XS inside the MXF file container is under progress in SMPTE under the item ST 2124. With these activities, a complete suite of formats is now available for JPEG XS allowing the transport and storage of this format in the postproduction workflow.

JPEG XS SDK available

Fraunhofer IIS offers development kits for CPU and GPU usage, as well as consulting projects for integration into products to the industry. Initial implementations for JPEG XS were carried out successfully, even in 8k.

Dr. Siegfried Foessel
siegfried.foessel@iis.fraunhofer.de
+49 9131 776 5140



JPEG XS – Advantages in brief

Professional formats: Support of RGB/444, RGBA/4444, YCbCr 444/422, YCbCrA 4444/42224 and YCbCr 420 image formats of up to 12 bits per color component sample precision with the option to extend it to 16 bits in the future

Highest fidelity: Visually lossless, i.e. no visible degradation, even over multiple compression/ decompression cycles. The latter is known under the term "multi-generation robustness".

Low latency: Maximum 32 lines end-to-end (compression-decompression) latency in hardware, in special modes even subline-latency.

Bitrate: Exact bitrate allocation per frame slice avoiding data overshooting

Parallelism: Support for multiple platforms e.g. FPGA, ASIC, GPU and CPU, by high degree of parallelism

High Performance: Real-time software implementation capability for 4k/60p formats on today's standard computers.

Low complexity: defined as a maximum percentage of a specific low-cost FPGA.
No external frame buffer required in embedded applications. In particular, individual frames shall be decoded independently

Typical compression ratios: 2:1 – 10:1

Transport/ Container	Type	Description – Main purpose	Extension
RTP	RTP Payload Format for JPEG XS (IETF draft)	IP based transport	
MPEG2-TS	ISO/IEC 13818-1:2019 AMD1:2020	Carriage of associated CMAF boxes for audio-visual elementary streams and JPEG XS in MPEG-2 TS	
Video over IP	SMPTE 2110-22:2019	Encapsulation of compressed video streams in SMPTE 2110 as RTP stream	
JXS	JPEG XS file format (defined in ISO/IEC 21122-3 Annex A)	For storing of single images JPEG 2000 syntax based	.jxs
MP4	ISO Base Media File format (ISOBMFF) (defined in ISO/IEC 21122-3 Annex B)	For storing of video ISOBMFF syntax based	.mp4
HEIF	High Efficiency Image File Format (defined in ISO/IEC 21122-3 Annex C)	For storing of mixed image and video content	.heif
MXF	SMPTE 2124 (FCD)	For storing of video MXF syntax based	.mxf
Notes: Status is as of 01.June.2020			

Transport and File formats for JPEG XS

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VVC BRINGS VIDEO COMPRESSION TO NEW LEVELS

Is it possible to stream high quality video despite having a poor internet connection? With the newly released (July 2020) Versatile Video Coding (VVC) standard, now it is! Benjamin Bross, head of the Video Coding Systems group at Fraunhofer HHI and editor of VVC, gives us the inside scoop.

Fraunhofer HHI played a key role in developing the new H.266/VVC standard.
Mr. Bross, what is special about this new standard?
Compared to its predecessor (H.265/HEVC), VVC reduces the required bit rate by 50%. In other words, it is much more efficient at compressing data. Therefore, even with a slow Internet connection, VVC enables the streaming and viewing of high-definition video. VVC also benefits smartphone users by reducing the consumption of data volume when streaming video.

So, the main benefit of VVC is the efficiency at compressing data?
This is an important benefit, but not the only one. VVC also offers flexibility, as reflected in its name ("versatile"). It is the first standard that is suitable for a range of applications: from game videos or virtual reality to video conferencing and teaching. The last two have been recently proven to be of critical importance in face of a pandemic. Each application has specific requirements for the compression algorithms, which were taken into account in the development of VVC.

What is the current status of the standard?
In July, the standard was finalized by ITU-T and ISO/IEC. We plan to present the first efficient implementations in software at the IBC trade fair.
The necessary hardware (e.g., chips that reduce battery consumption on smartphones) should be available in 2021.

Maria Ott
maria.ott@hhi.fraunhofer.de
+49 30 31002-862
Benjamin Bross
benjamin.bross@hhi.fraunhofer.de

DECEPTIVELY LIFELIKE: REALISTIC VIRTUAL PEOPLE!

Your favorite sports star comes into your living room to give you private lessons? Mixed Reality makes it possible: The chosen person is inserted into the real environment through mixed-reality glasses. Dr. Oliver Schreer, Wieland Morgenstern, and Dr. Cornelius Hellge from Fraunhofer HHI explain the tricks that make virtual humans look lifelike.

Why don't you let us in on the secret of making avatars look realistic?
Dr. Oliver Schreer: Until now, avatars have been created using computer animations – but these often seem artificial. So we've taken a different approach. We place an actor or an actress in the middle of a rotunda and film him or her with 32 cameras, calculate the 3D information, and use this to create a 3D model. The result is a dynamic mesh sequence, known as a volumetric video.
Wieland Morgenstern: In the next step, we animate this volumetric video with a high level of detail – we use model-based animation to change the actor or actress so that, say, his or her gaze can follow the user. This makes the contact more personal.

Are special end devices needed?
Dr. Cornelius Hellge: If these photorealistic volumetric videos are rendered on the end device, the bit rates and the demands on the graphics cards are too high. Together with Deutsche Telekom, we are therefore moving the rendering to the cloud – so only a normal 2D video stream is transmitted to the end device. This 2D video stream runs on all standard devices and on all browsers.

Maria Ott,
+49 30 31002-862
maria.ott@hhi.fraunhofer.de
Oliver Schreer
oliver.schreer@hhi.fraunhofer.de


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LC3/LC3PLUS: A PLUS IN AUDIO QUALITY AND TRANSMISSION ROBUSTNESS FOR WIRELESS ACCESSORIES

LC3 / LC3plus (short for Low Complexity Communication Codec) was developed by Fraunhofer IIS and Ericsson. The new audio codec resolves fundamental shortcomings present in today's wireless communication platforms, such as Bluetooth and Digital Enhanced Cordless Telecommunications (DECT). Its operation modes range from medium bit rates for optimal voice transmission to high bit rates for high-resolution music streaming services. At the same time, the codec operates at low latency, low computational complexity and low memory footprint.

While LC3 is the audio codec for Bluetooth LE Audio, a new audio architecture designed to boost the performance of Bluetooth Audio, the superset LC3plus was standardized in 2019 as ETSI TS 103 634 and is included in the 2019 DECT standard. This makes LC3plus the only open standardized high-resolution audio codec, reducing dependencies on specific proprietary vendor solutions. LC3 and LC3plus cover the same bases, as in high speech and audio quality for wireless audio accessories, as well as a reduction in the required bit rate compared to other state-of-the-art technologies. The latter helps prolong battery life and paves the way for smaller products. Low latency, low complexity and low memory requirements are also characteristics of both codec variants.

What puts the "plus" in LC3plus?

Among other things, higher robustness against transmission errors, even lower encoding delay, and the ability to play back high-resolution audio quality. In addition to applications and devices based on the 2019 DECT and ETSI TS 103 634 standards, LC3plus is also suitable for Bluetooth. This enables manufacturers to deliver the benefits of LC3plus on Bluetooth-based wireless accessories, such as headsets, headphones and earbuds.

A plus in user experience

With the introduction of the 3GPP Enhanced Voice Services (EVS) audio codec for VoLTE came super wideband (SWB) audio quality for mobile phones. Users feel as though they are in the same room as the person they are talking to. Now, LC3plus makes the equivalent of EVS available for the landline network, enabling users of VoIP applications and DECT telephones to also share in the feeling that the conversation is really taking place face to face.

LC3plus can also minimize disruptions during phone calls: the codec is extremely robust when it comes to voice packet loss and bit errors. In overloaded VoIP channels, the redundant transmission of LC3plus voice data ensures verifiably more stable phone calls. For DECT telephones, the inherent tools for forward error correction in LC3plus were again specially adapted to exploit typical characteristics of DECT links. This significantly improves call quality compared to previous codecs: uninterrupted calls can be made even when the handset is far away from the base station.

It's not only for cordless telephony that LC3plus offers significant improvements – thanks to its high-resolution audio mode, users can now enjoy music streaming with high sampling rates and wide dynamic range via wireless accessories such as headphones or speakers without any loss of quality. This makes LC3plus the ideal codec for these applications.

Mandy Garcia
amm-info@iis.fraunhofer.de
+49 9131 776-6178

360 REALITY AUDIO IMMERSIVE MUSIC SERVICES AND DEVICES ANNOUNCED WITH MPEG-H

The MPEG-H immersive audio format, which powers 360 Reality Audio, makes it possible for artists and music creators to produce a spatial musical experience. When listeners hear content produced in the 360 Reality Audio music format, they experience an immersion into sound that transports them onstage with their favorite artists. You can hear the music as the artist intended it during production.

Playback of 360 Reality Audio content can now be enjoyed on mobile devices on the go and at home. Being able to enjoy popular recording artists' latest immersive music mixes in any environment and on many 3D audio enabled devices will provide audiences with a seamless immersive experience.

"We are very pleased that Sony has chosen MPEG-H Audio as the distribution format for 360 Reality Audio content. The creative possibilities this offers the music industry and the new level of music experience this provides to consumers are breathtaking and we are proud to be part of it," said Dr. Bernhard Grill, Director of Fraunhofer IIS.

"The future of music is here and immersive. With Fraunhofer, we found an industry-leading innovation partner. The open standardized MPEG-H audio codec enables us to offer 360 Reality Audio music on a growing number of streaming services and many playback platforms from various manufacturers," said Yoshinori Matsumoto, Director and Deputy President, Sony Home Entertainment and Sound Products Inc.

Thanks to a cooperation between Fraunhofer IIS and Sony the format will be compliant with MPEG-H 3D Audio, an international open audio standard, and optimized for music streaming.
As licensors, Sony and Fraunhofer IIS will continue working to expand the available content library, participating streaming services and compatible audio devices moving forward.

About MPEG-H Audio
MPEG-H, substantially developed by Fraunhofer IIS, is the industry's most advanced audio system, supporting both immersive sound and the ability for users to adjust elements in the audio to their preferences. MPEG-H has been on the air since 2017 on all TV networks in South Korea under the new ATSC 3.0 standard, and it has been selected for new broadcast standards to be launched in China and Brazil. Fraunhofer offers MPEG-H software implementations for many popular CPU, SoC and DSP platforms. It is widely deployed today in TV sets, premium soundbars and high-end smart speakers.

Mandy Garcia
amm-info@iis.fraunhofer.de
+49 9131 776-6178

FRAUNHOFER DIGITAL MEDIA ALLIANCE

As an one-stop competence center for digital media we provide for our customers scientific know-how and the development of solutions that can be integrated in workflows and optimize process steps.

The members of the Digital Media network are actively working in renowned organizations and bodies like International Standardization Organization ISO, ISDCF (Inter-Society Digital Cinema Forum), SMPTE (Society for Motion Picture and Television Engineers), FKTG (German Society for Broadcast and Motion Picture), and in the EDCF (European Digital Cinema Forum).

Fraunhofer institutes in the Digital Media Alliance jointly offer innovative solutions and products for the transition to the digital movie and media world of tomorrow. The Institutes in the Alliance are available as renowned contacts and partners for all of the digital topics connected to digital media, digital movies, and standardization, as well as new cinematography, audio, and projection technologies, post-production, distribution, and archiving. The goal of the Fraunhofer Digital Media Alliance is to quickly and easily help find the right contacts, partners, and suitable technology.

The Fraunhofer Institute members are
- Digital Media Technologie IDMT, Ilmenau
- Integrated Circuits IIS, Erlangen
- Telecommunications, Heinrich-Hertz-Institut HHI, Berlin
- Open Communication Systems FOKUS, Berlin
- Guest: Intelligent Analysis and Information Systems IAIS, St. Augustin

Contact
Fraunhofer Digital Media Alliance
Angela Raguse M.A.
Phone +49 9131 776-5105
alliance-dc@iis.fraunhofer.de
www.digitalmedia.fraunhofer.de

Publication Information

Fraunhofer Digital Media Alliance
c/o Fraunhofer Institute for
Integrated Circuits IIS

Am Wolfsmantel 33
91058 Erlangen, Germany

Concept and Editor
Angela Raguse
Fraunhofer Digital Media Alliance

Editors
Dr. Janine van Ackeren,
Romana Redtenbacher,
Robert Seeliger

Layout and production
Ariane Ritter

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