AoIP Codec Requirements for SSL Applications

Considerations in the decision-making process

GUESTCOMMENTARY

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An AoIP codec for studio-to-studio link applications must be compatible with all common algorithms, such as MPEG 1/2 Layer 2/3, AAC LC, HE-AACv1&v2, AAC-LD, G.711, G.722, E-aptX and PCM. It should also include a wide range of important features, such as Session Announcement Protocol (SAP) as well as Real-time Transfer Protocol (RTP/RTCP).

Particularly, Session Initiation Protocol (SIP) ensures interoperability between AoIP codecs placed at different studio sites by establishing the connection and negotiating the algorithm.

REQUIREMENTS

In addition, latency control is extremely important for handling sophisticated broadcast environments, such as live broadcasts, audio description or in-ear monitoring. This allows synchronization up to a microsecond. If equipped with GPS or PTP, a sample accurate synchronization is also possible.

To ensure best possible transmission quality, the device should provide functions for perfect redundancy management like dual streaming (IP packet-based redundancy) or exact audio sample synchronization of backup streams on different compression levels. Pro MPEG FEC is a feature for error correction by restoring of lost IP packets.

Due to the different compression levels, it is not possible to achieve the best possible sound with every audio algorithm but it is feasible to find an acceptable quality/cost ratio, depending on the general conditions.

Another important consideration is the type of environment in which one is working. For example, is it a small network, a large broadcaster with subnets and routers, or a complex live application?

If the most economical transfer is necessary, then a codec such as AAC with a high degree of compression should be used. To accomplish this, all imperceptible frequency ranges need to be eliminated from the audio signal.

The Enhanced apt-X codec standard represents an attractive compromise between economy and pure sound. Due to its increased resolution, it is suitable for radio broadcasting. Compressed in accordance to the requirements of the wireless transmission, the audio has hardly any loss as compared to the original track after decompression. This means it is possible to avoid the emergence of cascading effects due to multiple transmissions.

The use of significantly larger bandwidths allows operators to transfer transparent audio bit by bit without any compression by deactivating sound processing. A loss-free transmission of the audio signal is possible over PCM with a very high sample rate (latency below 1 ms). This complies with the requirements of demanding live applications.

With the AES3 format, it is also possible to implement an absolute bit-transparent transmission of AES/EBU signals in studio environments. For this, AoIP codecs should provide IP-based audio network technologies for real-time streaming like Livewire and Dante (proprietary solutions) or Ravenna (license-free solution).

THE FUTURE

Mobile applications in connection with AoIP are becoming increasingly important and are enabling all kind of users to transmit content from almost any environment to the broadcasting studio and sound studio.

In this context, it is important to mention the OPUS codec and x-AAC standards, which are becoming more widespread. The basis for transmissions of this type is an AoIP capable device or server in the broadcasting studio and sound studio.

To meet the upcoming additional requirements, companies should participate actively in development and integration of specifications such as Tech3326 and Audio Contribution Over IP. The purpose of these specifications is to achieve device-compatibility among different manufacturers.