Smart Audio is the way forward for live broadcast production

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Introduction

Today's broadcast facilities are facing ever-increasing demands on their resources as they strive to keep up with consumers who expect more content, on more devices both where and when they want it. Amongst all the various elements broadcasters have to handle including video, graphics, data, captions and subtitles, the importance of audio should not be underestimated.

Smart Audio

Smart Audio as an element of Smart Production

Smart Production is an evolving strategy. It uses a creative mix of technologies to deliver high quality production results with increased efficiency. It enhances the relationship between dedicated production tools and processes, it will unify the production for different media sectors (radio, television and online) and it requires a global view of production workflows. Smart Audio and the elements that we have introduced so far are a dedicated part of this trend. The introduction of remote production techniques will definitely benefit from Smart Production in general and Smart Audio in detail.

"Smart Audio - Intelligent and complementary audio algorithms that optimize performance for higher efficiency and increased automation."

The Smart Audio campaign aims to focus attention on effective, high quality and, to some degree, automated audio production, particularly in live broadcast environments. Smart Audio means investing in simple, reliable and predictable equipment that can automatically deliver audio content while maintaining the high quality that consumers rightly expect.

The way to efficient and high quality live production

To attract and retain viewers, consistent, stable and coherent audio is a vital requirement. One aspect that is particularly important to pay attention to is speech intelligibility. In today's file based environment, for content being recorded to be stored and subsequently broadcast at a later date, the task of creating good audio, especially dialog, involves many techniques. Appropriate microphone choice and placement, post-production techniques and a myriad of software tools that can analyse and enhance faster than real time.

In a live broadcast and production situation however, things are very different. Think about a breaking news story that needs to be on air as quickly as possible. There is simply no time to use all the techniques used for say a drama production. A feed from an outside broadcast or remote truck needs to be combined with a local studio presenter and perhaps a correspondent reporter speaking from a remote studio. Audio levels cannot be assumed to be consistent with each other and may not be compliant with relevant loudness standards. To ensure that viewers receive a consistent and above all clear, audio experience, it is necessary to find a way of dealing with potential issues in a more efficient way and one that requires minimal intervention and manual control or operation.

The concept of Smart Audio

The answer is to utilize real time processing algorithms that are both intelligent and adaptive. This way, the dynamic structure and characteristics of audio content will be preserved to ensure pleasing results. Devices need to be fully interoperable with others in the broadcast environment and need to seamlessly integrate with both playout automation systems and logging and monitoring processes.

Auto-Level, Auto-Upmix, Auto-EQ, Auto-MIX, Auto-Loudness and Codec System Metadata Management are the core algorithms for building Smart Audio workflows today.

Figure 1. One operator taking control of both video and audio in automated live production

Use cases for Smart Audio

Auto-Level and Auto-Loudness

Automated levelling of individual sources is used to pre-condition the audio before it reaches the output mixing stage. This way, the balance between different audio elements can be set perfectly to create consistent sounding mixes. Adaptive leveller processes will apply source gain control whilst target levels are predefined to maintain the important characteristics of the audio elements. This might be a line level target for incoming lines, effects channels and for speech recordings (microphone inputs) or a much lower level target such as an audio bed for voice over. The summing point will collect all the pre-levelled sources and depending on the presence of parallel audio sources in the final mix, an Auto-Mix function might be advantageous here. As a result, the final mix output is basically defined by the source levelling procedure. At the output stage (after mixing), a final loudness based levelling will be applied.

The loudness control process will balance the audio energy in accordance with the appropriate standards and recommendations and if the pre-levelling is done correctly, then the loudness control can operate in a much more gentle and unobtrusive way.

A workflow like this can also be fully automated. The automation system will start the audio elements and recordings and will control all the hardware devices in the chain including playout servers, studio facilities, mixing desk and Smart Audio devices for levelling and loudness control.

The concept has already been adopted by a number of broadcasters, including Input Media in London and the ARD Tagesschau television prime time news service in Germany where for more than two years, all their live production news shows have been delivered with perfect audio quality and an average program loudness within +/- 0.2 LU of the target of -23LUFS!

Adaptive Auto-EQ

The use of adaptive EQ ensures consistency of spectral balance, and that all important element, speech intelligibility. Spectral Signature™ was primarily designed to provide an automatic tone control for various fields of broadcast applications. Using pre-defined sound fingerprints, also called Signatures, this unique proprietary algorithm is able to automatically match the desired tone color to the sound of the signal source.

In fact, Spectral Signature™ can relieve the sound engineer of the task of controlling the equalizer gains manually as it works like a fully automatic equalizer, consistently reliable and accurate.

Conventional equalizers continuously change the gain characteristics of the processed signal, regardless of whether it is required or not. On the contrary, Spectral Signature™ applies gain changes to the signal in a "smart way". Only really weak frequencies will be increased and only the overloaded ones will be reduced.

A proprietary wideband filtering technique allows an absolutely distortion-free tone control with strong linear phase response.

Because of this, Spectral Signature™ doesn't allow any sharp frequency changes and therefore it is insufficient to completely remove very narrow-band distortions (like fan noises), or undesired tones.

Auto-EQ like this offers live sound control "on the fly", and is ideal for situations that do not allow for proper post-processing. It also provides improved quality of "unmanned" voice recordings as the spectral balance between human voice, the chosen microphone and the acoustic environment can be equalized in real time to achieve the aim of a consistent tonal balance. The use of adaptive Auto-EQ is also another technique to ensure the all important speech intelligibility.

Auto-Upmix

Sudden and unexpected switching between surround and stereo audio during program content changes (e.g. commercials) can be very annoying for television viewers. In addition, switching artefacts in the decoder, loudness jumps or video sync. shift during encoding format changes can be caused by audio format changes during production.

A sophisticated Upmix algorithm will continuously analyze the input stereo mix and will extract diffuse, ambient sound components and place them in the rear surround channels while highly localized, direct sound components will be mixed to the center channel.

The Upmix algorithm must always deliver downmix-compatible surround sound. The surround image should be stable, offering adequate spatial perception without artificial reverberation or hall effects. No signal transformation artefacts or phasing effects should be noticeable by using linear-phase signal processing. If the Upmix algorithm can achieve this level of performance, it can be used for Auto-Upmix - fully-automatic operation with format auto-detection and programmed cross-fade operation.

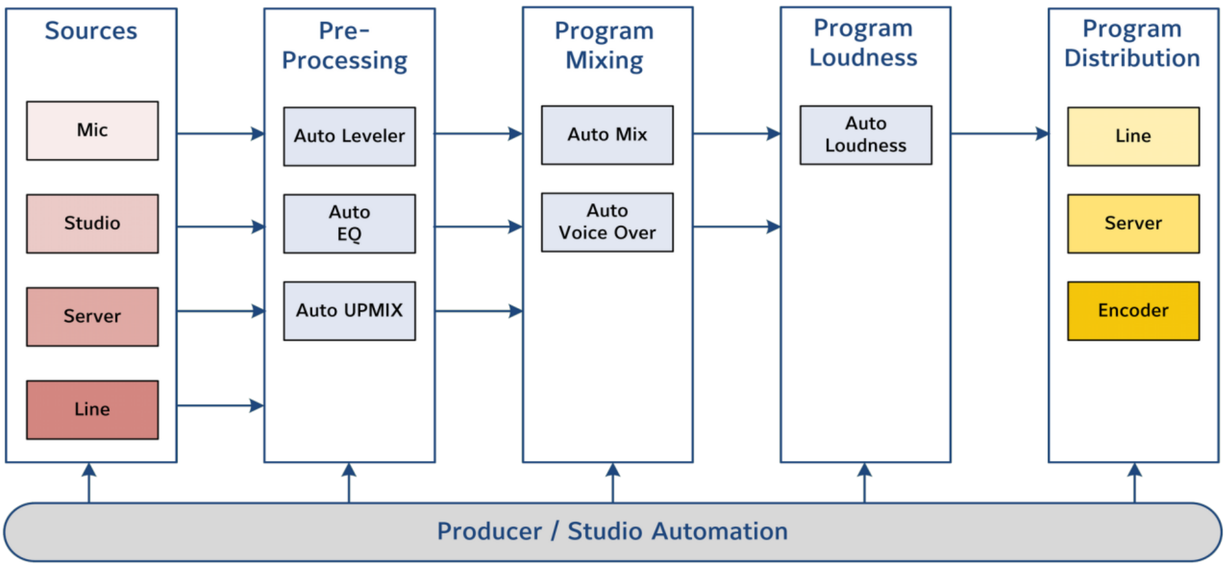

Figure 2. Complex Smart Audio production workflow, matching a lot of popular production layouts (newsroom show, variety shows, sports live, radio show, clip show).

Other Smart Audio procedures

As already described, auto surround Upmix can be used to maintain a constant surround experience independent of the format of the source audio. The use of adaptive Auto-EQ is another technique to ensure the consistency of spectral balance and that all important speech intelligibility.

Figure 3 Voice Over booth with no fader controls required. Simplified operation can be managed by the presenters.

The ability to interface with Audio over IP networks just by plugging in a single cable is another step forward to Smart Audio.

Conclusion

This paper has described how intelligent and adaptive audio algorithms can be used to create Smart Audio workflows. Auto-Level, Auto-Upmix, Auto-EQ, Auto-MIX, Auto-Loudness and Codec System Metadata Management are the core algorithms for building Smart Audio techniques today. Audio over IP protocols along with Next Generation Audio codec systems are additional elements that will be part of future audio production structures.

Smart Audio is a part of Smart Media Production and is an important emerging strategy. Using a creative mix of technologies to achieve high quality production results with increased efficiency is an increasingly common requirement for all media production, especially in live broadcast environments such as television and radio.

About Jünger Audio

Established in Berlin in 1990, Jünger Audio specializes in the design and manufacture of highest quality digital audio dynamics processors. Jünger Audio has developed a unique range of digital processors that are designed to meet the precise needs of the professional audio market. All Jünger Audio products are easy to operate and are developed and manufactured in-house, ensuring that the highest standards are maintained throughout. Jünger Audio's customers includes the world's top radio and TV broadcasters, IPTV providers, music recording studios and audio post production facilities.

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