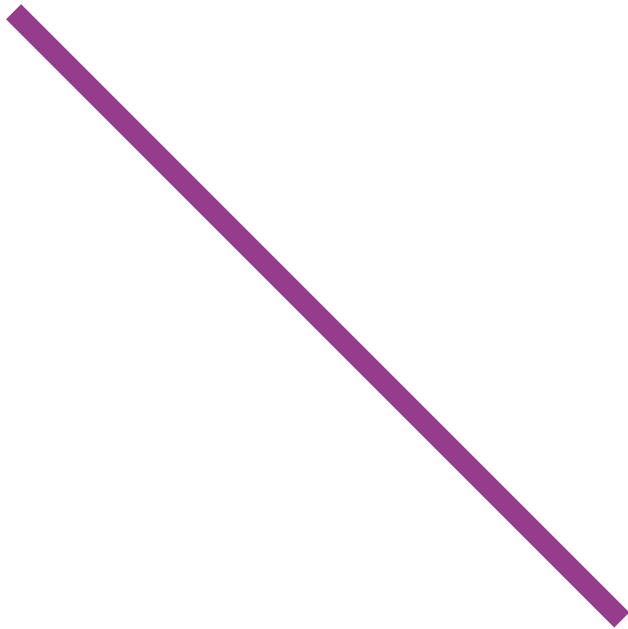


Immersive Audio Series - Part 4



Essential Guide

EG

ESSENTIAL GUIDES

Immersive Audio Essential Guide is a four-part series sponsored by;



GENELEC®



Register today at thebroadcastbridge.com to receive email notification when new Essential Guides like this one are published.

Are You Experienced?

Paul Mac looks at the immersive landscape and examines why the experience is making all the difference.

The rise of immersive audio has seen our industry take a giant leap forward on behalf of the consumer. That leap cannot be measured in numbers of channels and speakers, or resolution, or sample rate, or anything quite so mundane and physical as those things. Immersive audio is a leap forward in experience first, with technology in the wings, providing a whole gang of enablers.

You could argue that the launch of 5.1 into the home got its priorities mixed up. Binding a new experience to additional speakers immediately ruled out participation from a majority of consumers - consumers that were still, for the most part, tied to traditional television services as the focus of that experience. Hindsight is wonderful.

Soon after though, lots of things started to happen. For one, that 'experience' became king in the eyes of all, as alternative devices and platforms became benchmarks for what was possible. Gaming evolved from having to deal with limitations to coping with possibilities - and it went beyond sales of physical gaming products to a broadcast phenomenon all its own.

Broadcasting Changed

The notion of what it was to be a broadcaster, and how to be a broadcaster, also changed. From premium terrestrial and satellite services, to the incredible success of the big subscription streaming services, to one of the biggest broadcasters of them all (with the distinct advantage of having almost zero production costs): YouTube.

Content and its expanding range of outlets has made it clear that the personal experience is a big deal and has real value.

Dolby also taught us a new language. The audio object threw away every speaker and reassembled the sound field according to every point in a three-dimensional space. While channel beds have not disappeared (far from it), objects transcend the notion of simply adding speakers and give content independence from the limitations of the living room, the bedroom, and even headphones.

Then, as it turns out, ambisonics and binaural audio were a fantastic idea after all... After years behind the scenes, YouTube goes and makes ambisonics part of its spec, MPEG-H Audio includes it, and suddenly everyone realises what all the fuss was about. Speaker-agnostic scene-based audio is the perfect complement to an object-based world.

Audio Beam Steering

And speaking of perfect complements - the sound bar is other product genre that has developed to an impressive extent. Use of beam steering principles to solve the problems of rear and height speakers is a very practical enabler. Not necessarily perfect, but a forward step for the all-important experience.

In this series we've covered many of the fringe, not so fringe, and future hopes for immersive audio. The personal HRTF is a special development for such an individualised world. The notion that a game, a video, or a VR experience can be tailored for your own physical self is a 'software advance'. Gamers use headphones a lot already and they won't have to buy a new pair - the enabler in this case is in software, which equates to convenience.

Of course, VR and AR, while still relatively niche, are finding footholds in everything from roaming games to the factory floor. Immersive audio doesn't necessarily have to be about a narrative or creative experience - it can be an augmentation of more functional experiences, to make them simply accurate, or informative.

This multitude of broadcast experiences makes immersive audio normal. It drives up audience expectation.

Consumer Demands

With such a wide range of consumer mediums and increasing demand, it's important that production keeps pace and finds ways of delivering the experience that the consumer clearly wants, and in many cases is prepared to pay for.

In live production, acquisition is key. Clear object channels, unencumbered by the crowd, can become serious enhancements to immersive coverage, even if they're not panned in extremis and sit squarely front and centre. Scene-based crowd and atmospheres are emotive - they can lift a programme to immersive status on their own.

Objects actually increase the value of a skilled engineer and high-quality audio because now there is real design to be done on programme details. It's not only about balance. In the same way, production automation becomes another enabler - freeing up an engineer to mix a show rather than manage it.

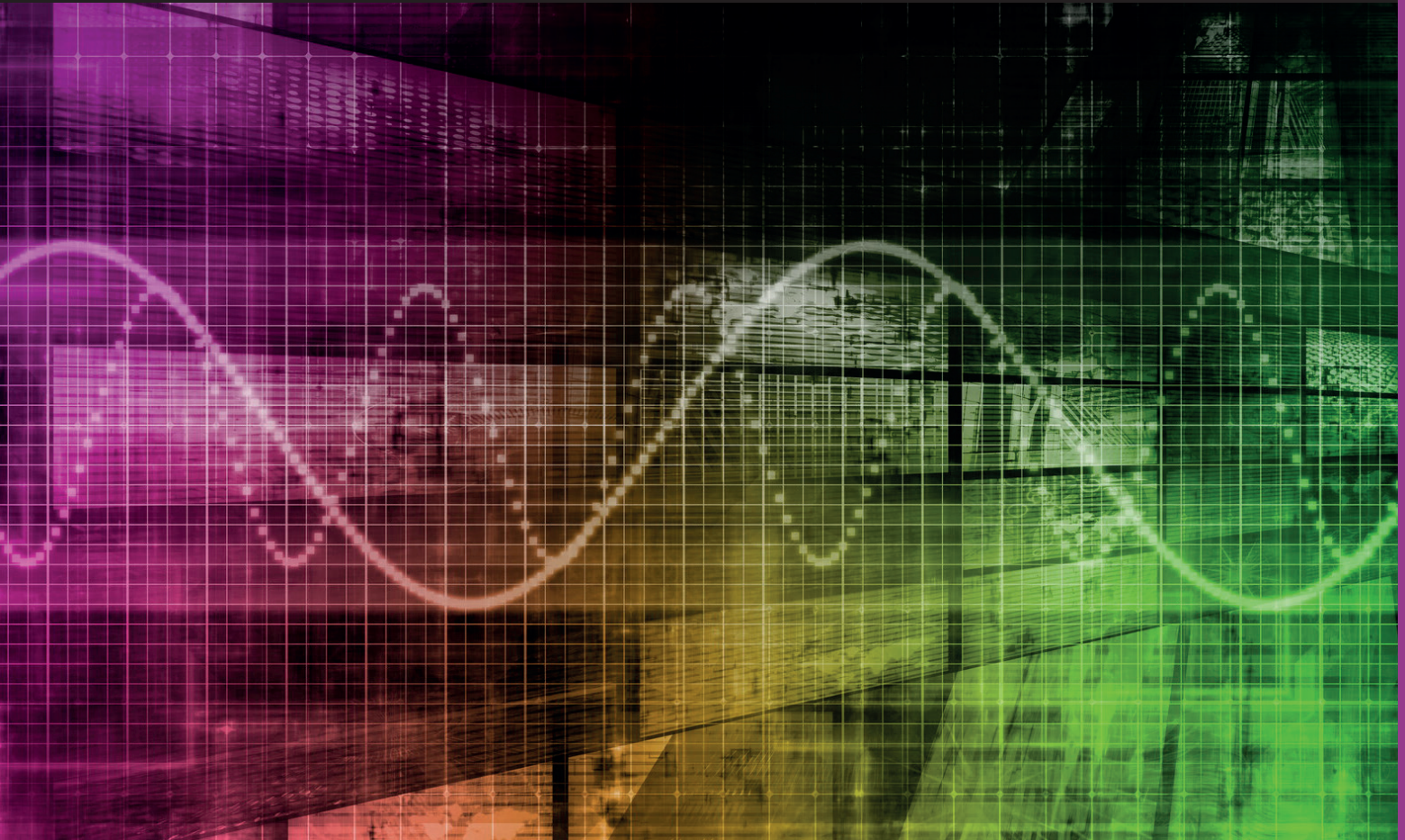
Media Unification

The authoring tools are in place, the workflows are in place, and the standards are in place. Dolby Atmos and MPEG-H audio are already commanding big audiences around the world, and object-based ideas are now uniting cinema, TV, and live production in new and exciting ways. In the studio, IP-based audio transport will inevitably become the norm, and again the lifting of limits on infrastructure simply adds to a long line of enablers.

For the broadcaster, sound engineer, and production professional, the knowledge and skills required to manage an immersive workflow are not so far removed. Additional control and options that directly impact the viewer are evident in the touch screen panner, the new and expanded bus widths, the networked connections, and the authoring GUI.

Immersive audio is so much more than 'better than surround' – it is starting to define a new way of consuming sound and a new role for audio production professionals. It is delivering a personal experience where it's required, accessibility where it's needed, virtual reality where actual reality just isn't cutting it anymore, and much hoped for value in a landscape where new is too often not novel. It's time to be experienced.

Immersive Audio Series



By Paul Mac – Writer, Professional Broadcast Audio

Part 4 - Options And Tools For Production Of Live Immersive Content

Live production of immersive audio formats is all about the metadata. Possibly of primary concern to the normal duties of an audio engineer is that object position / panning is in the domain of the encoder.

Generating channel-based output from the console is of course not a problem, and several consoles manufacturers have been upping their bus widths and monitoring path capabilities to cope with 7.1.4 audio and above, and incorporating 3D panning not just of individual channels, but balance control of incoming stems with height and scene-based formats.

Channel-based audio can be sent straight to the encoder and defined as such in the metadata without having to worry about panning. Also, ambisonics or scene-based audio, has position already included, even though that format is speaker-agnostic.

That leaves objects. One could envisage a flexible scenario where data from the mixing console could control the panning and position in an encoder, but currently that's not a major requirement in the audio chain for live production. Most objects being generated are optional dialogue channels and specific tracked or miscellaneous items such as the ball in a football match, referee audio, and so on.

Separating these out as objects offers flexibility at the mix position for ‘sound design’ decisions and makes them available at the encoder for special treatment. This is eventually realized in the personalized ambitions of the object-based broadcast - object selection and relative level included.

It’s true that currently those objects in live broadcast are mostly front-and-center panned. In audio-follow-video scenarios that’s straight forward. In those wide shots where panning might possibly be an option, the effect of panning a ball, for example, is not necessarily a pleasant or coherent one from the viewer. A screen is wholly in the main field-of-view for the consumer and therefore its content ‘focus’ is absolutely suitable for center panning. Panning outside of that screen position presents conflicting information for primary objects. Atmosphere, crowd noise, and so on - anything that benefits works for panning but might be better presented as a scene or channel-based feed.

Of course, there are creative objectives and particular objects that might benefit from both fixed off-center positions and dynamically panned positions. At the moment, that’s taken care of in the audio encoder.

There are currently two real time encoders available for MPEG-H audio - one from Linear Acoustic and one from Junger Audio. For Dolby Atmos, the live workflow uses the DP590 authoring tool, the DP591 audio encoder, and the DP580 reference decoder.

Dolby Workflow

The three main tools in Dolby Atmos live production are the DP590 authoring tool, the DP591 audio encoder, and the DP580 reference decoder.

The DP590 is the box that generates the metadata for the audio and also takes care of a monitoring path back to the console. It organizes its settings in sessions, which can contain multiple presentations - combinations of audio elements that can ultimately be selected by the consumer in AC-4 and be selected by the engineer for monitoring. The DP590 also provides metering and loudness monitoring facilities.

Audio elements are delivered to both the DP590 and the DP591. The DP591 also receives metadata from the DP590.

The DP591 takes care of encoding the audio elements and the metadata. It’s capable of rendering audio into either the Dolby ED2 extension to the Dolby E format, which includes metadata, or into Dolby Digital Plus with Atmos. It can also transcode ED2 and E to Dolby Digital Plus with Atmos and decode ED2 back to the original channels. In the context of the immersive Dolby Atmos workflow, the resulting encoded stream would be passed onto the broadcast encoder and to the DP580 reference decoder.

The DP580 is a reference decoder and confidence monitor for Dolby AC-4 and Dolby Digital Plus With Atmos content. It enables comprehensive validation of both streams with monitoring, display, and logging of all audio-related metadata and loudness. It can also emulate set-top box devices by using the Extended Display Identification Data (EDID).



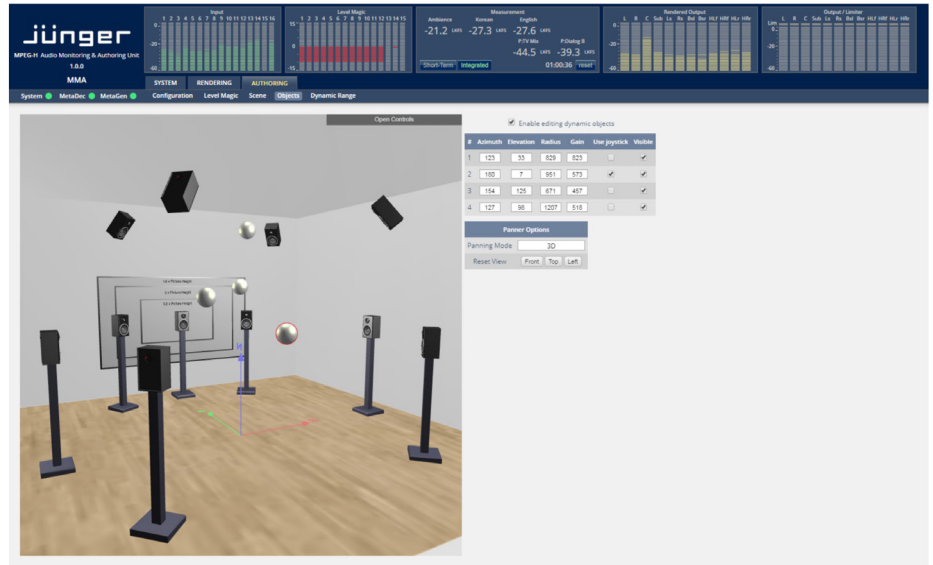
The GUI for Dolby’s DP590 authoring tool showing panning information inside a presentation.

MPEG-H Authoring

There are currently two players in the live MPEG-H authoring market - Jünger and Linear Acoustic. Both units take an all-in-one approach to the workflow, with audio elements as the input, and audio plus metadata as the output, which is then passed to the UHD transmission encoder.

Jünger MMA

Jünger's MPEG-H authoring tool is based around the MMA core processor and can handle up to 15 audio input channels plus the control track and render up to 7.1.4 outputs. It incorporates audio, metadata, and loudness monitoring services, rendering and downmixing for a range of formats, output emulation, dynamic range control (DRC) emulation, 3D object panning, Jünger's own Level Magic™ loudness control, and Ember+ protocol control.



The Jünger MMA GUI with metering and object positions inside the 3D space.

I/O is modular, with a choice of HD (1x 3G SDI) and UHD (4x 3G SDI) interfaces with embedding, de-embedding, and video delay capabilities.

Linear Acoustic AMS

The Linear Acoustic AMS Authoring and Monitoring System incorporates monitoring control, I/O routing, mixing and panning facilities, loudness processing and metering, and full MPEG-H authoring facilities.

AMS is capable of outputting 15-channels of discrete audio with a metadata control track, a 5.1-channel rendered output, 2-channel rendered output, and a dedicated monitoring output, all simultaneously. The loudness control uses Linear Acoustic APTO loudness adaptation technology.



Linear Acoustic's AMS authoring and monitoring system GUI.

Sennheiser

The Sponsors Perspective

The Object Of The Game

Sennheiser introduces its new and innovative approach isolating audio objects in live sports broadcast production with the power of beamforming.



How it fits in - a live broadcast workflow example for the AMBEO Sports Microphone Array, incorporating Lawo's Kick software.

Sports broadcasting is one of the most valuable sectors for paid services worldwide, and also one of the most prolific when it comes to pioneering new, premium viewer experiences. Immersive audio and object-based broadcasts are naturally suited to sports broadcasting as competing sports persons and teams offer obvious points of view, and the creative rewards of reproducing an immerse live atmosphere far exceed the value in immersive audio for many other programme formats.

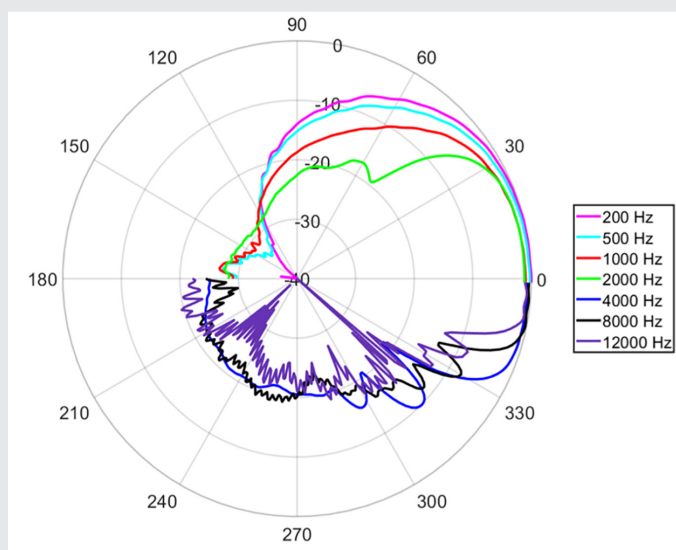
However, even top-tier Immersive sports broadcasts have tended not to embrace all the possibilities of the medium. Rather, a 'pseudo' immersion is constructed due to both the perception of what consumers want and can cope with, and what is practically possible in live production. However, the rise of ambisonic encoding and streaming services pushes immersive audio into new realms, enabled by impressive modern tracking technologies and joined-up audio-follow-camera production automation.

Last year, Sennheiser proposed a new microphone product that embraced the possibilities of beam forming technologies and production automation systems, providing a route to connecting tracking technologies to the microphone. The Sennheiser AMBEO Sports Microphone Array - now refined, field-tested, and ready for deployment - offers unprecedented steerable directionality and off-axis rejection. It's a real opportunity to multiply the impact of live sports. This includes reacting to and reproducing accurate positional information, and also bringing important creative audio opportunities for things like ball impact, competitor effort, in-field dialogue and so on within reach of the live mix engineer – something only previously available to post people and sound designers.

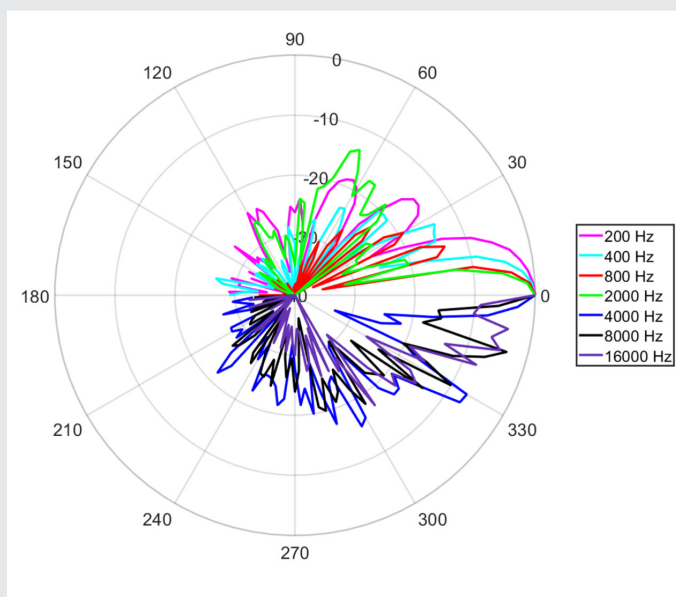
Importantly, the microphone allows many simultaneous beams to be generated from a single array so that many objects can be isolated with a single array. One beam might be following the ball, while another might be focussed on the referee, for example.

The project was initiated to address a long list of issues currently experienced in sports broadcast:

- The sound event may take place at a considerably high distance from any microphone.
- The sounds of interest may be far lower in level than the general ambient level in a stadium.
- The objects creating the sound may be moving at high speed.
- Any processing of the captured sounds must allow for live broadcast of the audio stream and therefore involve no, or conceivably low latency.
- Systems should sustain adverse weather conditions such as rain or wind.
- Systems must withstand mechanical impact such as a ball hitting a microphone.
- Microphones shall not conceal any camera view.
- The frequency spectrum of the sounds of interest ranges from low frequencies below 200Hz up to 5kHz and above, while typical crowd noises cover the same frequencies.
- Depending on the camera view, the sound may need to be panned and played back from a different angle.
- Some sounds may only be of interest if they add information to the visuals, while others (such as the referee's whistle) need to be heard independently of what is shown on screen.



The polar response of a single Sennheiser MKH8070 shotgun microphone.



The polar response of the AMBEO Sports Microphone Array, showing a marked increase in directivity and rear / off-axis rejection.

The solution to these points - the Sennheiser AMBEO Sports Microphone Array - is a 360-degree array of 31 shotgun microphones in the horizontal plane. Sennheiser uses a combination of beamforming algorithms (modal beamforming) paired with detailed modelling of the shotgun response, and its inherent directionality, to achieve a highly directional output that can be steered in any direction via a control link.

The array can achieve 'proper' beamforming over a relatively wide bandwidth and with a level frequency response – from below 200Hz to over 5kHz. Level difference due to distance can be accounted for with a variable gain for each beam.



SENNHEISER

There is no mechanical or motorised movement of the array; it's purely done by processing the individual outputs of all the microphones in the array so that the combined input of the off-axis mics is attenuated, and the on-axis audio is accentuated. Vertical rejection is taken care of by the shotgun mics' physical characteristics due to the usual action of their interference tubes.

The fact that this directionality can be steered 'hands-free' by the algorithm is an important rung on the immersive and object-based ladder.

During the design phase, arrays with other microphone types were trialed and considered, but nothing offered as much off-axis rejection and control as the high-quality shotgun mics. The added benefit of having sufficient rejection in vertical plane without having to make a three-dimensional array was also a bonus and avoids any obstruction issues with the cameras.

Since prototype stage at IBC 2018 Sennheiser has made some big gains with the system especially with the form-factor. The array size has been reduced to a diameter of one meter, which makes it a realistic consideration for pitch-side placement where the larger original could obstruct digital advertising boards.

The system has been tested with the German and Spanish football leagues, as well as in the US on American Football and basketball. For football specifically, trials have shown the most effective deployment to be two arrays behind each goal, and one array opposite the centre-line camera and microphone. Sennheiser expects a unit to be commercially available in 2020, initially aimed at the football broadcast market. In the meantime, it continues to show the technology off, coupled with Lawo's innovative Kick system, at the major broadcast shows, including IBC 2019.

The hope is that object tracking technologies and protocols such as Dolby's ED-2, which enables object-audio metadata to be carried along with the audio, will eventually bring all aspects on a live object-based production together, moving fully positional tracking data around, along with live-generated audio objects and that production automation will become the engineer's third-hand in an object-based future.

Genelec

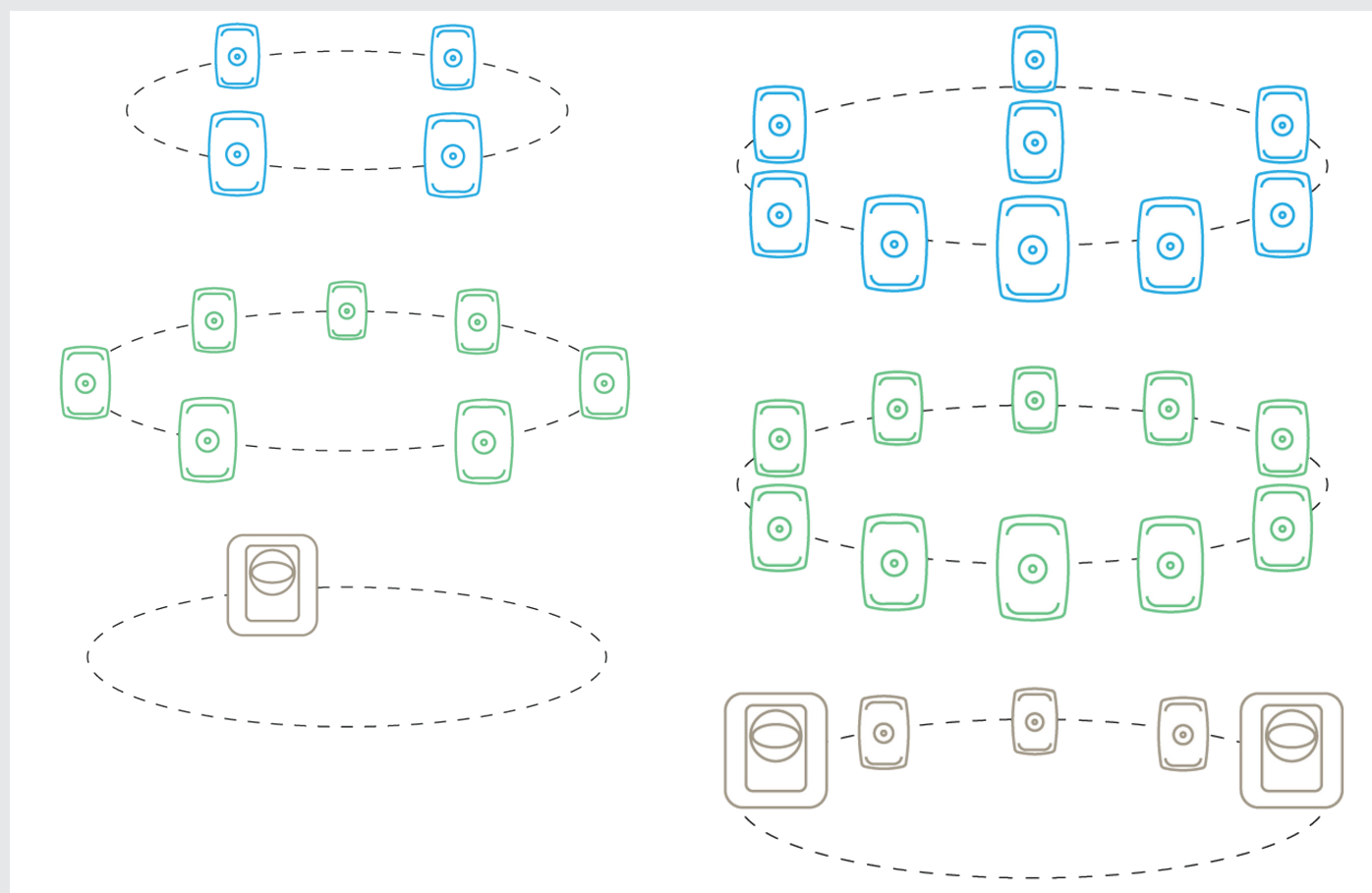
The Sponsors Perspective

Immersive Monitoring

Genelec Senior Technologist Thomas Lund moves the monitoring discussion on to the practical considerations for immersive audio, wherever you are.

The previous article concluded how an in-room immersive monitoring system generally is the best choice for ensuring good translation, not only to other loudspeakers and rooms, but also to soundbars and to personalised immersive headphones in the future. This article examines more practically what it takes to design, setup, and control an immersive monitoring system.

Immersive production in 2019 generally falls into three categories: 1) sound for picture and a large audience, 2) sound for picture and a small audience, 3) sound for pleasure and a small audience.



Two of the most commonly used ITU-R BS.2159 immersive system configurations: 7.1.4 (or 11.1) on the left, and 22.2 on the right.

While everybody knows how engaging immersive cinema can be - and ambitious sports, drama from HBO, Netflix etc. has started to envelop us sometimes even better at home - the third type is also becoming more popular: Stunning immersive recordings of just music. Such production is now happening at a number of diverse locations: Austria, China, Germany, Japan, Korea, Norway, UK, and counting.

Morten Lindberg from 2L puts it this way: “There is no method available today to reproduce the exact perception of attending a live performance. That leaves us with the art of illusion when it comes to recording music. We should create the sonic experience that emotionally moves the listener to a better place. Immersive audio is a completely new conception of the musical experience. Recorded music is no longer a matter of a fixed one- or two-dimensional setting, but rather a three-dimensional enveloping situation.”

System Design

Before designing an immersive listening room, consider which of the three above scenarios you wish to cover. The biggest difference is between targeting a large or a small audience. The former is generally based on “shower type” top layer loudspeakers and irregular listening distances, while the other two situations are better served using an ITU-R BS.2159 configuration.

In BS.2159, the ideal is an equidistant setup of all monitors, though impossible placement may be compensated by delay-alignment instead. At the design-stage, listening level and system headroom needs consideration. EBU R128 includes a simple and elegant requirement: Each monitor should generate 73dB SPL at the listening location, for a test signal input of -23 LUFS. With 20dB of headroom, each monitor should therefore be able to deliver a minimum of 93dB SPL.

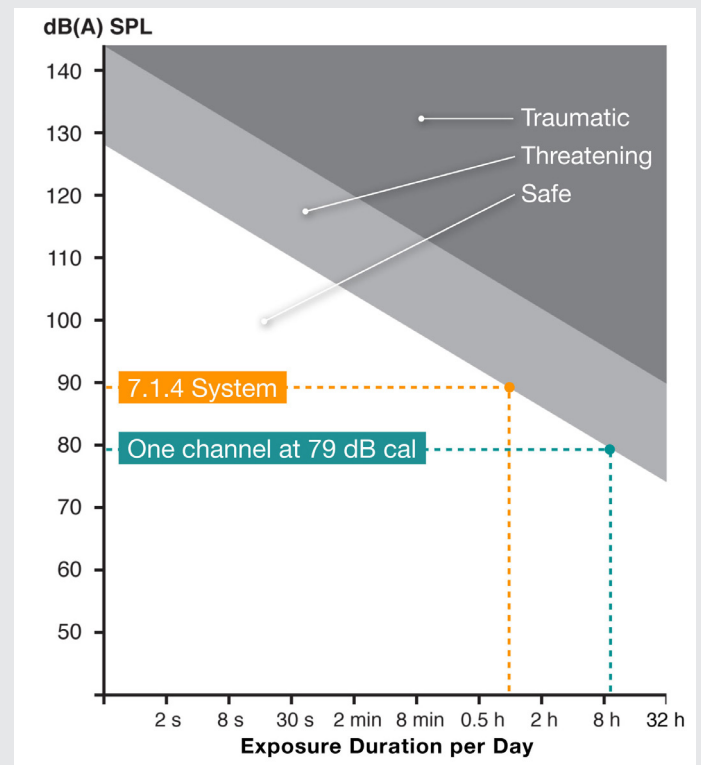
ATSC A/85 specifies listening level based on room size, from 76dB SPL to 82dB SPL in rooms from below 40m3 to 560m3, i.e. requiring between 96dB and 102dB SPL per monitor.

For a theatrical monitoring system, SPL requirements as per SMPTE RP 200 are even higher, with standard operating level of front channels at 85dB SPL. Dolby has created a useful DARDT tool for design of Atmos theatrical rooms, and a DARDT HE tool for design of Atmos home entertainment rooms. Both tools inform whether a certain monitor is of sufficient capacity, taking room and distances into account. The list of monitors includes many Genelec types, and models from other vendors, too.

Low frequency (LF) reproduction depends on application. Sound for picture makes use of a dedicated sub channel with 10dB of extra SPL capacity, called LFE, which is rarely used in pristine music production. In all applications, however, bass management may be applied, thus relieving the main monitors of LF duty to increase system headroom.

Considering pristine music, bass management is based on at least two subwoofers at different locations in order not to compromise imaging and envelopment. The same concept should ideally be applied in immersive sound for picture, additional to a subwoofer used for LFE reproduction.

Finally, when planning the listening level, the risk of hearing loss (HL) also needs to be taken into account. Sound pressure level (SPL) is a measure of sound power, while the more relevant metric in prevention of HL is sound energy, i.e. SPL with time. A-weighted sound pressure integrated with time is called sound exposure, and health requirements based on contemporary research recommend a sound exposure of not much more than 80dB per day (8 hours); which is the same as 83 dB for 4 hours or 86 dB for two hours etc.



Graph showing range of exposure levels with standard 7.1.4 system levels indicated.

That equal energy principle also applies when installing several loudspeakers in a room: For a certain calibration level, each doubling of loudspeakers tends to increase exposure by 3dB. Working for a day with 7.1.4. content may therefore easily give 10 dB more exposure than the calibration level suggests, so check your daily sound exposure once in a while to remain on the safe side.



Genelec’s GLM monitoring calibration system can help set up large immersive loudspeaker systems and also act as the monitoring controller.

System Calibration

There are six steps to credible immersive production monitoring:

1. Select and optimize the room.
2. Optimize placement of the monitors.
3. In-situ frequency response calibration.
4. Time of flight (delay) compensation.
5. Trimming of spectral balance.
6. Calibration of listening level.

High quality monitors should have been individually calibrated at the factory for a flat on-axis frequency response in an anechoic chamber, but the response is not the same once a real room is used. Furthermore, the response varies dramatically depending on placement in that room. Considering typical positioning of immersive monitors, there may often be frequency response differences of 18dB or more between devices of the same type. Such differences are reduced in step 3 above.

The goal is a flat direct-sound frequency response from each monitor, that can be measured objectively using a microphone at the listening position. In case the room is not well acoustically treated, it may be indicated instead to take measurements at three or four locations, 10-25 cm around the main listening position. More information can be found in [1].

After having achieved a flat in-room frequency response, it is time for a final trim, based on actual listening. For instance, monitors with highly controlled directivity, such as Genelec’s, generate more uniform and neutral reflections than monitors of less refined design. It takes a human to really compute direct sound with reflections, and frequency response trimming should be performed at a controlled listening level to reduce sensory variation. Depending on listening level, the difference in human sensitivity between 1kHz and 100 Hz can vary by 12dB or more (ISO 226). For that reason, subjective trimming should also be performed at a listening level no lower than 80dB(C), or the subject may not be able to sense the lowest frequencies at all.

Expert software like Genelec’s GLM application can assist with all steps in the list above, and by itself take care on steps 3-6. Once setup and calibration has been done, GLM doubles as the monitor controller needed in immersive production with solo and mutes, calibrated level, bass management, and switching between loudspeaker setups having up to 72 channels.

References:

[1] A. Mäkivirta & T. Lund, “Is single microphone position enough for immersive system equalization and level calibration in production monitoring?” in proceedings of Tonmeister-tagung, Cologne (2018).

Lawo

The Sponsors Perspective

Production Automation For Immersive Audio

Lawo’s Christian Struck looks at the potential for production automation in immersive sports broadcasting, and how it can help move towards a personalized, object-based experience.



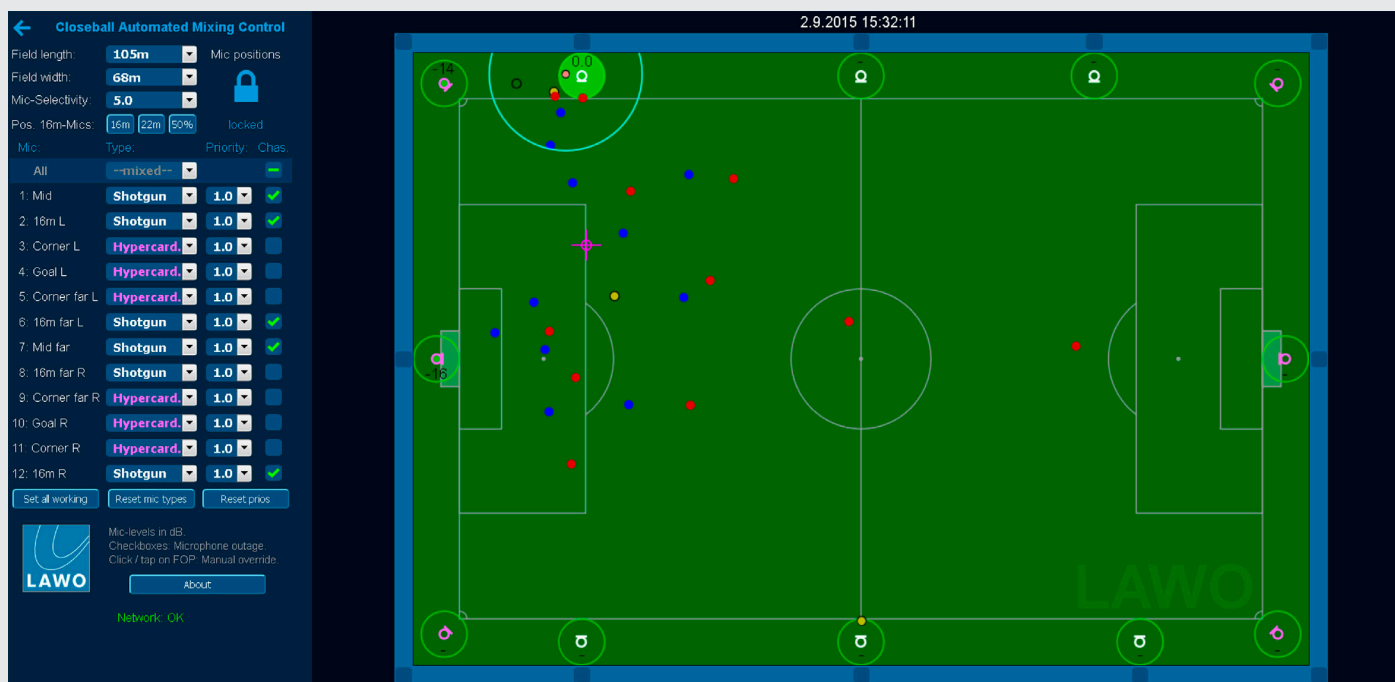
Live sports broadcasting is being transformed with tracking technologies.

Although they hardly ever say so, most A1s consider themselves artists. And art, it has been argued for centuries, is a matter of contrast. In the case of immersive, next-generation (NGA), audio, contrast results from the enveloping ambience with the odd “sonic movement” and the stability of a point of reference. In live sports productions, this sonic anchor is usually provided by the “point of interest” sounds: ball kicks, referees, player discussions, coach instructions, etc.

I Get A KICK Out Of You

The more an A1 is expected to deliver in a fast-paced production environment, the more they need all the help they can get.

In sports productions, one important step in this direction has been the introduction of Lawo’s KICK software for tracking-based automated close-ball miking.



The Lawo Kick GUI showing microphone positions around the field of play.

The fear that this would allow broadcasters to save on manpower was quickly dispelled: the A1 is still firmly in place, taking care of the same core tasks and new aspects brought about by new technological developments and ever-higher expectations.

Perfectly integrated into Lawo’s mc²-series mixing consoles while also available for mixers by other manufacturers, KICK is automated audio mixing technology for live sports events where a ball is used. Interfaced to camera- or transponder-based player/ball tracking systems, it guarantees a consistent, fully automated, high-quality, close-ball audio mix, and more.

KICK is a timeless mix assistant that delivers in any production scenario one throws at it. It takes care of the basic work: following the ball and all-important events on and off the pitch, leaving operators more room to attend to other aspects: intercom communication, monitoring various NGA presentations, refining the sound, and so on.

KICK’s close-ball mixing algorithm produces more precise crossfades than a human operator ever could: no level jumps, no audible fades among microphone channels, and so on. It’s main fortes for object-based production scenarios are the precision and flawless channel balancing that create a pro-grade experience even when listened to in isolation.

When KICK was released in 2015, next-generation audio was still a long way off. Its underlying principles, however, are still fresh and relevant today.

More To The Point

What does the above have to do with immersive or next-generation audio? After all, most A1s subscribe to the point of view that sound effects (ball kicks, referee whistle-blowing, tackles, arguments on and off the field, etc.) benefit from a stationary (phantom) center placement in an immersive audio mix to avoid confusion and listening fatigue.

For NGA, the close-ball/close-action signal constitutes a separate sound object of choice whose level will be adjustable by listeners in search of a personalized experience. This sounds like one of the most important features of next-generation audio productions.

Since viewers are (or will soon be) able to also adjust the commentary level, this may lead to situations where one essential element providing sonic anchoring in an immersive ambience context is no longer there. This makes it all the more important to leave the sound effects object at the center.

The Importance Of Being Perfect

As it can be assumed that consumers will be free to adjust sound FX objects to taste, the sound quality of these objects becomes critical. Even slight phasing problems, level jumps, and minor flaws become annoying when the sound effects are turned up and when no other signal (commentary, say) is there to partially mask such imperfections.

In the absence of a software like KICK, it is very likely that certain field-of-play microphones are unintentionally left open. Viewers/listeners are bound to notice spill from the crowd noises—which should be confined to an immersive object—and the resulting sonic imbalance. Even the best manual crossfades tend to produce level jumps that adversely affect the NGA experience.

Lastly, given the likely possibility of listening to the sound FX object in isolation, its sound needs to be continuous and, above all, organic.

Proponents of other solutions argue that the NGA experience would be even more satisfying if close-ball miking, referee whistling and other sonic action close-ups were divided into separate audio objects. It remains to be seen whether this indeed provides added value for viewers at home, or just makes personalization unnecessarily complex.

Without going into too much detail, the difference between a live-sound approach like KICK and a solution based on snippets and samples is that the live-sound solution is continuous and coherent in itself.

While personalization has not yet been fully embraced by the industry, the KICK software is already available and fully functional in personalized NGA scenarios.

Right On Track

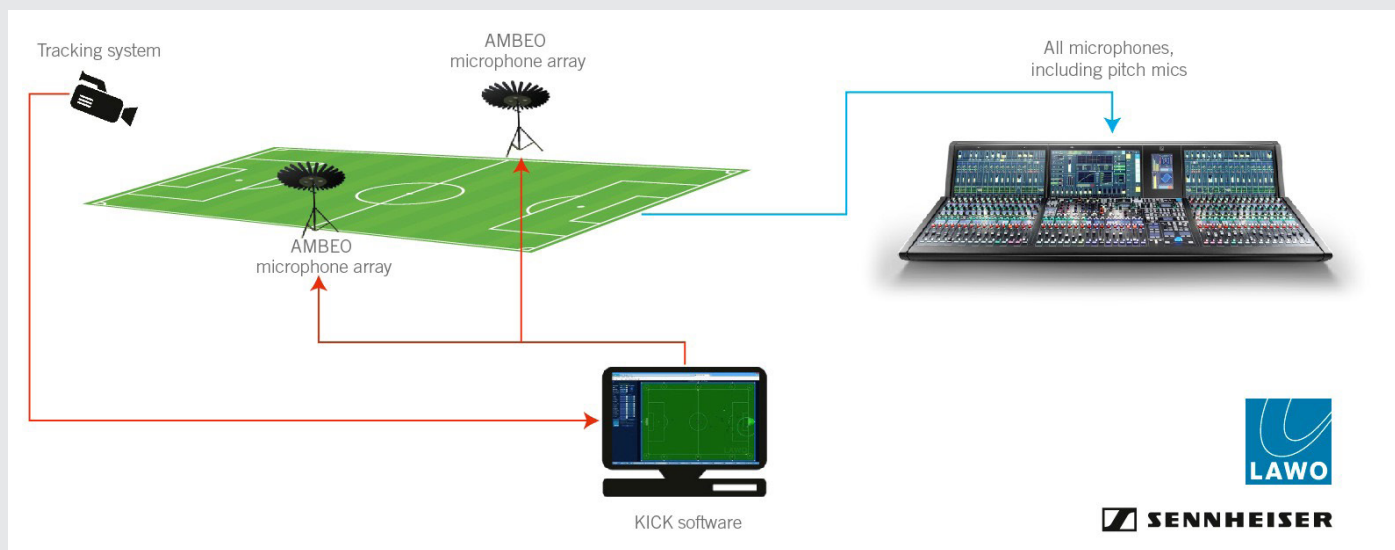
Based on the information provided by ChyronHego’s camera-based TRACAB camera tracking system, KICK knows what happens, and where it happens, and where the microphones used to capture such events are located. The software computes the angle of relevant microphones with respect to an event.

A recent development has been to use KICK in combination with Sennheiser’s AMBEO Ultimate Kick microphone array prototype for object-based mixing.

Based on the information coming from KICK, the microphones in the AMBEO array can be software-controlled to form capture beams (where several microphones of the array work together) that result in even better directivity than can be obtained with shotgun microphones on rotating, motorized stands.

The processors used to control the AMBEO array microphones require specific information for generating the right kind of beam in the relevant direction. This information is provided by Lawo’s KICK system.

Based on the tests performed by Sennheiser and Lawo, it is safe to say that close-ball miking is about to make another significant leap forward, which is good news for sports broadcasters. Stay tuned for more news in this respect. Its relevance for next-generation should be obvious by now.



Kick brings together camera, microphones, and console in a single live production workflow.

Find Out More

For more information and access to white papers, case studies and essential guides please visit:

thebroadcastbridge.com

WP

WHITE PAPERS

EG

ESSENTIAL GUIDES



MEDIA

CS

CASE STUDIES

09/2019

Sponsored by Lawo, Genelec and Sennheiser



GENELEC®

