THE BROADCAST BRIDGE
Connecting IT to Broadcast

Audio For Broadcast

Part 1 - The Theory, The Console, Monitoring & Metering

A Themed Content Collection from The Broadcast Bridge

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Themed Content Collection

Contents

Series Overview

By Tony Orme. Editor at The Broadcast Bridge.

Audio For Broadcast is a Themed Content Collection which serves as an audio course and reference resource for broadcast technologists.

It covers the science and practical applications of all aspects of audio in broadcast. It is not aimed at audio A1's, it is intended as a resource for the 'all-rounder' engineers and operators who encounter and must deal with audio on a day-to-day basis but who are not audio specialists... and everyone who wants to broaden their knowledge of how audio for broadcast works.

Sometimes you don't need to know everything about something. In our frenetic and challenging working lives more and more jobs are multi-skilled and adaptive, and we're often expected to cover more functions than we are comfortable with. We can't all be experts. Sometimes we just need enough knowledge to get the job done.

Broadcast Audio introduces some basic concepts for people who aren't audio people. While the audio signal chain is very different to what it was a generation ago, some fundamentals remain. Things like signal processing haven't changed and still need to be done well, but the series will also look at how we got to where we are.

It will look at how consumer technology and broadcast intent has influenced the production of content, and what that means for the people who work to put everything together. Because you never know when you might be asked to lend a hand.

Audio For Broadcast is a five part series:

Part 1. Theory, The Console, Monitoring & Metering

Future parts due in 2023:

Part 2. Broadcast Audio Capture
- Microphones: Principles & Patterns
- Microphones: Types & Applications
- I/O & Recording Devices

Part 3. Audio Processing Tools
- Dynamics Processors
- Equalizers (EQ)
- Noise Control & Audio Repair

Part 4. Routing, Sync' & Latency
- Routing & Asset Sharing
- Synchronization
- Latency & Delay Compensation

Part 5. Broadcast Audio Workflows
- Traditional Studio Signal Flow
- Outside Broadcast & Distributed Teams
- Cloud Based Audio

Analog Vs Digital

By Kevin Emmott. The Broadcast Bridge.

This is the story of how broadcasters made the shift from analog to digital audio. It is arguably the most emotive shift in broadcast production in recent years, and it has transformed the audio equipment that is used in modern broadcast workflows.

Some people will tell you that once upon a time everything made sense and that everything was analog. Processes were linear; signals would go in, they would get tweaked and grouped, and signals would go out. Everything was routed with physical patch bays and signals could be traced by following an actual cable.

All sound is analog. Sound is just the vibration of air particles; it is a continuous process which allows the ear to pick up every slight change in pressure with no bandwidth limitations. Bandwidth is the range of frequencies covered in a continuous band, and an average human can perceive from 20Hz to 20kHz. But analogue audio isn't bound just by what we can hear – for example, dogs can hear up to 60 kHz – and analogue manipulation of those sounds is just as unconstrained.

Recorded and broadcast sound converts the vibration of air particles into electronic signals for transport and manipulation, and for many decades the broadcast production chain used analogue circuitry to provide a perfect representation of the sound that was captured. Microphones change the sound into an electronic signal, the audio console manipulates that signal across the same frequency range, and then outputs it again.

In the early 1980s the introduction of hybrid broadcast consoles provided the ability to digitally control analogue signal paths – memories, snapshots and automation enabled audio engineers to be more flexible, and while the signal path was still analogue, the seeds were sown. Digital audio for live broadcast wasn't far away, and as digital television became a focus for broadcasters in the 1990s, digital tools became more common.

There Are Only 10 Types Of People In The World

There's a huge amount of technical theory on digital audio, but let's keep things very simple.

Unlike that unconstrained analogue audio, digital audio is an approximation of the sound rather than the full, flowing, continuous range. Sound information is sampled at fixed points on the sound wave, and to ensure a faithful representation the sampled audio bandwidth has to be restricted.

At any moment in time the value of a digital signal can be measured precisely, and digital audio is created by periodically sampling the incoming analogue signal. How often these samples are taken over time is referred to as the sample rate. A recording at one hertz means one sample is taken per second, and sample rates are measured in kHz; CD's use a sample rate of 44.1 kHz, broadcast digital audio tends to operate at 48 kHz, and HD audio is commonly assumed to be 96 kHz (but at any rate must be higher than 44.1 kHz).

Digital systems are binary. They store information in strings of zeros and ones, and it is the job of an analog to digital convertor (an A/D or ADC) to covert that signal. The maximum length of the string for each sample determines the total amount of information that can be stored for the sample – which is referred to as the bit depth or word length. A 16-bit sample can contain 65,536 digits, whereas a 24-bit sample can contain 16,777,216 digits. A 24-bit sample can therefore contain a far higher resolution representation of the audio signal. The main effect of this in digital audio is that a 16-bit sample has a maximum dynamic range of 96dB whereas a 24-bit sample theoretically has a maximum dynamic range of 144dB. In reality, audio converters commonly found in today's technology cannot achieve 144dB dynamic range; 120dB is more realistic with a good quality converter.

Bit depth should not be confused with bit rate, which refers to the number of bits transmitted per second.

Whatever is left out at this stage can't be added in later, which means any digital signal is only as good as the A/D conversion from the original analogue sound.

In order to recreate the signal it must be passed through a digital to analogue convertor (a D/A or DAC) at the other end. The Nyquist Principle from Swedish engineer Harry Nyquist states that if you sample at twice the maximum frequency of the signal being sampled, the DAC will render an output waveform identical to the input waveform. So, if a sampled audio system is required to carry signals up to what a human can hear - 20kHz - the sampling rate must be at least 40kHz.

This explains why digital audio has to have a restricted bandwidth, and also why those poor dogs aren't enjoying broadcast content as much as they could be.

Digital Broadcast Consoles

Digital audio provided an opportunity for sound designers to achieve much more in broadcast, and early adopters began to install digital broadcast consoles into audio control rooms in the 1990s.

They were flexible, they benefited from cumulative software updates, they were more powerful, and they had huge I/O matrixes. They also simplified installation costs by using less cabling, and once signals had been digitised, they remained in the digital domain throughout the production chain which made for easier integration with digital video systems.

Once in that environment, Digital Signal Processing (DSP) is used to manipulate those signals in real time. By this point the signals are all numbers – ones and zeros – and the DSP manages those all mathematically.

The good thing about DSP is that is it endlessly adaptable, and its firmware can be programmed to do very specific jobs which enable designers to develop new features and build more value into a product in a relatively short space of time.

For many years most commercially available consoles used the same off-the-shelf floating-point chips for DSP, and as capacity increased so did the number of chips required to process channels. As broadcast mixes became more complex more chips were required, with more PCBs, more backplane activity, and greater potential for on-air failure.

As broadcasters prepared for HD, the onset of televised 5.1 surround sound multiplied the number of required audio channels further still. Now, for every two-channel stereo input, broadcasters needed to provide a six channel 5.1 input, with a stereo downmix for legacy formats.

Open The Gates

Conventional DSP systems were limited by the number of signals passing between DSP cards, fighting for space on the backplane along with I/O. They were limited by backplane speed and took up more space.

Broadcast was grateful adopter of Field Programmable Gate Arrays (FPGAs), and broadcasters still use them for DSP processing today. FPGA's are blank chips which provide a canvas to create processing structures to perform very specific tasks. FPGAs meant that processing power could be tailored to exceed the channel numbers which were possible with traditional DSP chips. It was a total step change, a paradigm shift.

It also changed the way people thought about DSP for audio; FPGA's don't come with any limitations on bit-depth, and provided some manufacturers an opportunity to select the number format (the bit depth) to meet the level of performance which was required by the function.

We're Going To Need A Bigger Desk

This increase in capacity in turn drove the design of broadcast audio consoles, as the worksurface became the bottleneck and the ability to control and manage such a huge number of channel inputs became the limiting factor. Digital architecture gave broadcasters the ability to provide more immersive, more involving and more content, and hardware UI design adapted to fit these bigger workloads.

For large-scale broadcast audio processing, FPGAs are still the most efficient way to do things, using either on prem or edge hardware. But as cloud workflows become more acceptable and the benefits become more tangible, expect things to change again.

Analogue audio still has its fans, and as any visit to an online audio forum will show you, they are vociferous. The irony is that most modern output is a combination of the two – even if something has been recorded and mastered in a fully analogue workflow, it's most likely being streamed and listened to in a digital format, at whatever bit rate it has been converted to.

As consumers we've traded quality for convenience, and digital audio has allowed all that to happen.

The Role Of The Mixing Console

By Kevin Emmott. The Broadcast Bridge.

When most people picture an audio control room (ACR), they are most likely picturing the mixing console. It is the mixing console, with all its lights, knobs and fancy sliders, which delivers the wow moment.

The mixing console is the Instagram star and rightly so; it not only has all the bells and whistles, but it literally manages all those bells and whistles too.

Nevertheless, shifts in production workflows are changing this picture. More aspects of live production are being democratised by remote working and by the Cloud, and as more broadcasters embrace distributed workflows, more productions embrace the benefits of flexible working with confidence.

It is in this environment that traditional broadcast mixing console need to keep pace.

Broadcasters As Cartographers

Although broadcasters are already on this road, they are drawing the map as they go. Issues like control latency over Wide Area Networks are largely being managed and the mixing console has already been split into its constituent parts.

The control surface – what everyone thinks of as the mixer – can use on-premise DSP in a traditional way, it can control a DSP engine somewhere else (such as a venue or another broadcast facility), or it can be a hybrid of both.

The control surface no longer needs to be in a studio; it could be anywhere. There's no longer a requirement for a physical surface and control can be on a laptop connected to public internet or using automation to simplify some of the processes.

And while these workflows all take advantage of tapping into traditional processing engines, the hardware which drives the control surface, the continued development and adoption of Cloud-based, scalable microservices will devolve these workflows even more.

Same Old, Same Old

All that said, the audio requirements of live production are unlikely to change.

Despite shifts to distributed working, the broadcast audio console is the hub in the ACR. It manages every audio input and every audio output, and feeds into other equipment such as comms systems, and that will still be true irrespective of whether it is physically central or whether it exists on the edges. Wherever the Operator is based, they will still have the same fundamental responsibilities.

All audio inputs and outputs will still need to be managed; complex and dedicated comms systems will still have to be arranged; all audio signals will still need to be processed appropriately and mixed together in an engaging way; multiple transmission formats will still need to be mixed; outputs will still need to be monitored for intelligibility and international compliance; latency will still have to be mitigated; and most importantly, the whole thing will still need to be guaranteed to stay on air, whatever the connectivity or geography.

Let's break these down.

Comms & Monitoring

Ensuring that everybody in the production can hear the audio elements of the production that they need to be aware of, and that individuals can communicate easily with the rest of the team are both crucial to the role of audio in broadcast production - which is why they get their own dedicated articles in this series.

Sounding Good

Now everyone in the production is on the same page – thanks comms! - the mixing console turns its attention to managing all the live signals and processing them appropriately so that an audience has a clear narrative to follow.

This is more akin to a traditional mixing console – in fact, it's what all mixing consoles are designed to do. There are lots of things which affect how incoming sources sound, from the acoustics of the venue to microphone choice and placement. These are all important parts of the planning process which can be mitigated, but in live broadcast there are things which cannot be controlled.

Again, specialist broadcast consoles have features to help with this. For example, high input headroom is an important design principle to counter unexpected and unplanned peaks. Multiple insert points in the signal chain are necessary to introduce external processing, and being able to introduce it at multiple points - whether it's pre or post-EQ, or pre or post-fader - will affect the end result, so these also need to be considered.

Increasingly, with more pressure on mix operators to mix to multiple output formats, broadcast consoles also need to have assistive applications which can do some of the heavy lifting to allow operators to concentrate on the craft.

Autofader features - or audio-follows-video - allow faders to be opened and closed automatically through GPIO triggers and are well-served in environments where audio sources have to change alongside different camera shots. This is especially popular in fast-paced motorsports where trackside cameras need matching audio to tell the story.

Automix systems, which date back to Dan Dugan's pioneering automatic mixing system launched in 1974, is often employed on a broadcast mixer to automatically mix the levels of a selection of channels to keep the overall level of the mix constant and ensure a consistent ambient/background noise level.

The Last Leg (And All The Other Legs)

In addition to managing all the comms, managing all the incoming signals, and mixing it all together, the sound desk Operator also prepares the show for transmission.

Transmission outputs used to be simple, with a mono and stereo feed covering all the bases. The last 20 years has seen rapid development with 5.1 surround, immersive formats and individual audio objects all on the table. The uptake in consumer devices which support spatial audio, like Apple's AirPod Pro and 3D capable soundbars, means that Next Generation Audio (NGA) content like immersive audio and personalisation is heaping more responsibility on the Operator; this is another area where broadcast consoles can help pick up the slack.

NGA is an umbrella term which covers technologies and ideas like immersive and object-based audio (OBA), a technique which encodes audio objects with accompanying metadata to describe what it is and how it contributes to a mix.

Audio objects can contribute to personalisation and accessibility features in consumer equipment and allow the contribution of certain objects to be modified by the viewer. While it's still early days, real-time transport for this is possible through Serial ADM, a metadata format which can be used for live production.

Immersive sound on the other hand, is already gaining ground, with mixes for live sporting events adding more crowd ambience in the height channels and immersive stings being inserted to add height to replay graphics and in-game statistics.

Broadcast consoles can help with automatic upmixing to multi-channel formats, whether that is for live transmission, or for multitrack ingest into asset management systems for archiving or repurposing.

And of course, they all need to be powerful enough to cope with all the extra stems and be able to flex to cope with the demands of any production schedule.

What's The Delay?

Delay is an important consideration in a broadcast console, and this can be complicated further when production workflows are geographically distributed. Artificial delay is necessary to compensate for a variety of sync issues, such as video processing delays. Most broadcast consoles will have multiple points where delay can be inserted to bring things back into line.

As workflows become more geographically diverse, and incorporate more varied environments like on-premise, off-premise and cloud processing, sync will need to be continually assessed and regulated.

Ready For Anything

We often say the broadcast industry is in transition. In fact, it's always been in transition.

The adoption of IP infrastructures and a renewed focus on remote working and virtualization has changed how we look at broadcast infrastructures, but in essence the same challenges need to be met and the fundamentals of those workflows remain unchanged.

Control is still paramount, reliability and redundancy are still key components, and an operator still needs to manage all the audio in a simple and ergonomic way where they have access to every parameter and can adapt on the fly wherever the broadcast demands.

However, that is done and in whatever context that is, the mixing console will continue to be the technology which holds everything together.

It just might look at bit different.

Monitoring, Mix Minus & Comms

By Kevin Emmott. The Broadcast Bridge.

In parallel to its role entertaining the audience, audio is the central nervous system of all broadcast infrastructure, ensuring everybody can hear what is happening and communicate with each other - without it everything rapidly breaks down.

Peter Drucker, the founding father of modern management studies, very wisely said that “the most important thing in communication is hearing what isn’t said.”

He’s right of course.

But in a broadcast environment the most important thing in communication is, in fact, not that. It’s hearing what is said. Communication is the single most important aspect of any production. Good intercoms are too often taken for granted but without them a broadcast studio is made up of groups of people frantically waving at one another off camera.

Intercoms and signal monitoring are less about what a signal sounds like and more about how those signals are managed, shared and distributed between all the people who play very specific roles in getting a programme to air.

Intercoms are partly about guaranteeing a quality of service, partly about ensuring everyone is where they need to be and partly about making sure everyone knows what’s going on.

Let’s Talk

People in a production environment are scattered and reliable communications keep everyone on top of their game. This is even more important when productions are split across multiple sites or have people on location who don’t have access to a screen.

The most casual of glances at the credits on a live show illustrates how many people need to be kept across the live production; talent, camera operators, sound assistants, studio technicians, floor managers, directors, lighting engineers, guests, runners, show callers and more all need to know what’s going on.

If a camera goes down, how does the talent know to look into another? How does a reporter on an airstrip in Dubai know when to answer a question? Or what that question is?

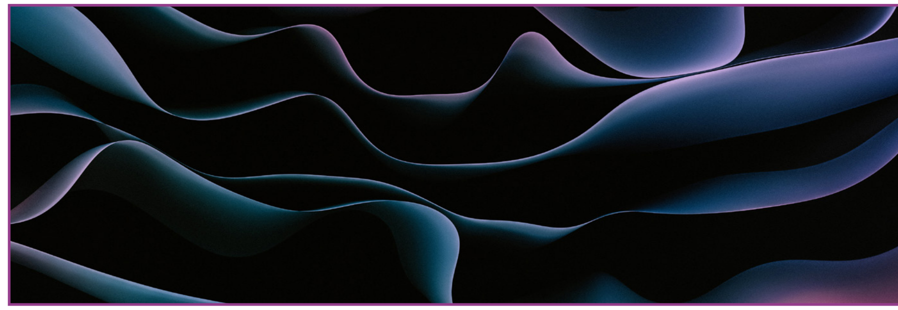
Some people will need to hear everything while others will need more exclusive communication links. The Director might need a two-way conversation with the Producer, or with a Production Assistant who is looking after a special guest. Some people will need discreet in-ear monitors (IEMs), some crew members will need two-way radios, and the talent or the audience might need foldback speakers. They might be wired, they might be wireless; they might warrant private intercoms from A to B, or they might need intercoms from A to B, C, D and E.

For this reason, intercom systems have to be extremely flexible. Helpfully they fall into three basic classifications: partyline (sometimes referred to as simply PL), matrix and wireless communications.

Unhelpfully, most broadcast infrastructures will use all three at the same time.

Party Time On The Partyline

A partyline is no place for secrets. They are two-wire networks which use standard three pin XLR cables to allow full duplex communications, which is a fancy way of saying that multiple people can talk at the same time and everyone can hear everyone else. This makes them useful for communicating to groups of people who all have to be across the same thing, such as camera, audio or lighting crews.



They employ distributed amplifiers so they can be used with Y-cable splitters with no loss to the signal, they can be daisy-chained together and compact intercom panels can be located in various locations around a production infrastructure. This makes them extremely scalable and adaptable to different broadcast environments.

Partylines are quick to set up and useful to keep groups informed, providing two-way communication (talk as well as listen) over a single cable between all end points with everyone involved in the same conversation. If you talk on a partyline everyone hears you.

Let’s Get Moving

Partyline endpoints can also be accessed using wireless intercom belt-packs and headsets to provide more freedom of movement. This can be useful if your role isn’t in a static location and you are required to go wherever the production demands are, or if you are on location at a sports event or outside broadcast. Belt-packs can also be daisy-chained together in the same fashion as wired intercoms.

Handheld radios are the other main method of wireless communications on set and are often used to provide communications across larger sites for people who require the freedom to roam. They are cheap and they are very flexible; they can be set up in a variety of ways, ranging from a simple radio to radio configurations to wider connectivity via radio interface units.

Radio to radio arrangements use the same radio frequency for transmit (talk) and receive (listen) and can only communicate in one direction at a time; this means that if you are talking you can’t listen at the same time, and it also locks out other radios from communicating. This is known as a simplex connection, and it how a standard two-way walkie talkie operates.

But handheld radios can also be connected to the wider network, including a partyline, and can even be set up to use different frequencies for talk and listen; this is known as semi-duplex. This means that they can always receive messages regardless of whether another is transmitting, which means that key personnel can always be heard, and on every radio.

In The Matrix

A matrix interface system is more complex. In the matrix, everything goes and matrix interface panels do it all. They are compact panels which can be rack mounted or located on a desktop to provide flexible access to individuals or groups of people scattered around the broadcast environment.

A matrix intercom system allows cross communication with many different communication tools. They allow for direct communications between panels as well as to partyline and wireless communications. Buttons can be programmed to talk or listen to anything connected to the system and they manage all intercoms including GPIOs and access to interruptible foldback mixes (IFBs).

Can I Just Interrupt Here?

IFB mixes are central to broadcast intercoms. They are commonly known as a mix-minus because they enable a full mix to be sent to multiple listeners but minus their own input. This is really useful, especially for crew members and talent who are on location at a venue or on an outside broadcast. It's vital because it allows them to hear the broadcast audio without hearing their own delayed voice, which will inevitably be delayed due to the distances involved making the round trip to the studio and back. For example, a field reporter on a news item needs to hear the programme output while communicating with the news anchor, and mix-minus removes the reporter's own (delayed) voice from the mix.

The mix minus bus is so useful that broadcast consoles always have mix minus busses built in, which means a sound operator doesn't have to build time-consuming multitrack mixes to achieve the same result – it's one button press away, for absolutely anyone who needs it.

What Piffle!

There are other handy features built into the broadcast console to ensure we all keep talking. PFL stands for pre-fade listen and is a monitoring function you will find on even the tiniest of broadcast desks. The PFL button creates an audible check on an integrated speaker, a headphone output or the main monitor speakers which enables an operator to check an incoming feed before it is fed into the mix when the fader is opened.

It's a useful confidence monitor to pre-hear an outside source to make sure it's still live, as well as an opportunity to check its signal level so that the right amount of gain can be applied on the channel input so it works with the rest of the mix.

Feeding The Beast

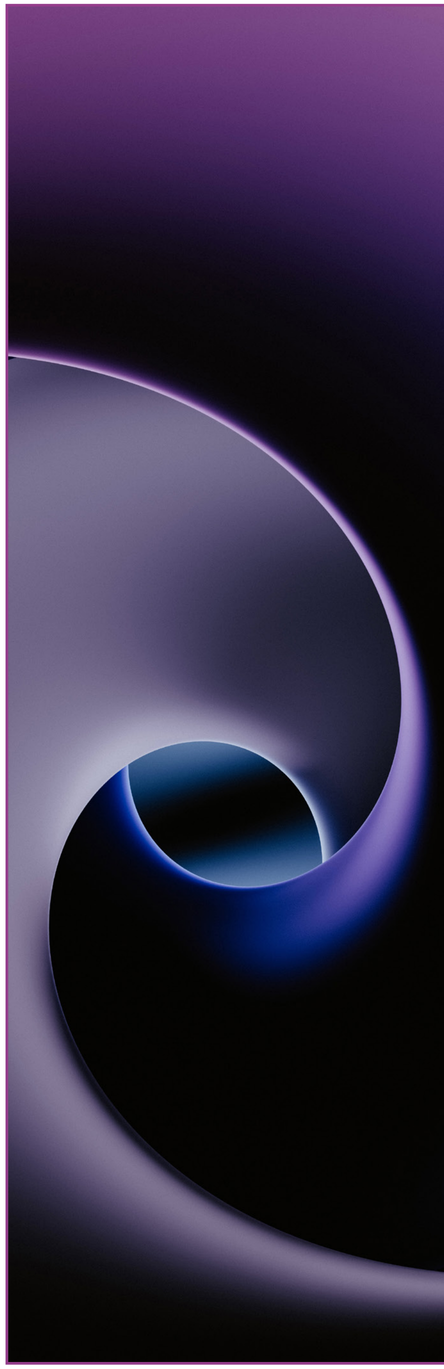
The mixing console also plays another key role. In fact, it is central to everything.

In addition to managing the broadcast mix, it is stuffed to the brim with every single audio signal flowing through the production, making it ideally placed to generate all the audio stems to feed into the intercom system.

But it is the intercom system which provides the fabric, managing the flow of those signals and creating the interrupt when that all-important IFB goes to the talent's earpiece. It is the intercom system which provides the ability to talk to anyone else in the production environment, and depending on how things are set up, does so for anyone with access to a panel, belt-pack or radio.

Many intercom systems now utilise standard IP protocols like AES67 or Dante, which simplifies connectivity and squeezes more signals down a single cable, while more adaptability is being created with initiatives like TFT touchscreens, programmable channels and Bluetooth connectivity, but the principal of broadcast communications haven't changed.

Reliable communication is essential to guarantee the smooth operation of every broadcast, at all levels, and in this case that still means hearing everything that is said.



Flexible Comms For Tech Teams & Creatives

By Karsten Konrad. Head of Intercom Product Management Riedel Communications.

How to design flexible comms systems which satisfy the requirements of both technical engineering teams, and creatives who need simple transparent operation.

A broadcast comms network is the central nervous system of any broadcast infrastructure. It not only ensures that messages are freely carried between the control room and the rest of the production, but it controls every function, and coordinates everything to work as a single organism.

Like a central nervous system, it does all this in the background. You don't even notice it is there and that's how it should be. Comms has no ego, it is invisible; and yet is has been setting the pace in networked broadcast infrastructures since the very first televised broadcast.

The Big Shift

Early analogue comms systems developed into with 2- and 4-wire connectivity which provided two-way communication, and then Partylines made it easy to talk to multiple people at the same time. This covered all the bases for many years, but the rapid growth in digital infrastructures has changed the game for everyone.

Digital signals like MADI and AES3 were a big change and offered superior audio quality and speech intelligibility, while VoIP offered the ability to transmit audio data via long distance connections and integration into telephone systems. The introduction of AES67 and SMPTE 2110 revolutionised how we communicate and what we are now able to achieve within real-time networks.

Riedel has been an early adopter throughout every paradigm shift, building up our knowledge and experience with all these emerging technologies, and blazing the trail to the future. This bank of knowledge makes us the perfect partner for anyone eager to move to state-of-the-art technologies.

Today, IP provides our users with increased flexibility, greater efficiency, and huge scalability. It provides our support teams with the ability to diagnose and fix issues remotely. But more importantly it provides Riedel with the tools to continue to drive the evolving broadcast landscape.

User Types

Post-Covid, workflows are inherently more complex. Remote production, distributed working and cloud resources are all being utilised in different ways to achieve more efficient programming. Customers are working with workflows in a hundred different ways but while everything has changed around us, one thing remains the same: reliable comms are as important as ever.

We don't need to ask what the end user wants because they want the same thing they have always wanted; reliable and flexible comms which keep their teams in touch with each other. But perhaps we should be asking who the end user is.

The emergence of distributed production models and the democratization of broadcast workflows to users outside of traditional engineering teams has created new user types. We cater to users who are hands on with the practicalities of comms management, and we also cater to users who view comms purely as a creative tool. Both are equally embedded in the broadcast environment, and both have different responsibilities and motivations. With different sets of needs we can categorise them as: the technical customer and the creative customer. Riedel is all about creating a focus for both of these users.

The Technical Customer

Technical customers are usually engineers. Their responsibility is to plan the whole system and manage its deployment. System designs need to be scalable; they need to flex with requirements and they need to handle multiple signal types in a transparent manner.

There are practical considerations. Where are all the endpoints? Are they fixed or on the move? How many are remote? What are the restrictions on access? Who needs to talk and listen, and who just needs to listen? What are the plans for future expansion? Are there any links to outside sources? How easy it is to apply firmware and software upgrades?

Manufacturers like Riedel have dealt with these issues for many years, but future-proofing modern comms systems isn't just about choosing the right hardware. It is about ensuring that your existing architecture is capable of doing the heavy lifting when required and can be updated when the time is right with no on-air disruption, and no additional investment.

For network designers, IP provides a lot of this out of the box. SMPTE 2110 and NMOS-compliant equipment delivers interoperability between a range of devices and services, and allows equipment like cameras, editing systems and control panels to seamlessly communicate with each other.

Guaranteeing interoperable connectivity requires strict adherence to standards, and while Riedel has been SMPTE 2110-compliant from the start for the encoding, transport and synchronisation of media streams, its support of NMOS for discovery and management, GPIO functionality and channel-level operations guarantees a better fit into dynamic broadcast workflows. Wider adoption of specifications like NMOS IS-04, IS-05, IS-07 and IS-08 give engineers the confidence that we are all striving for the same goals.

Scaling Up

IP makes scaling up a network infrastructure based on actual production requirements much more efficient, with no need for physical rewiring or patching. IP-based infrastructures can also leverage existing network infrastructures and COTS equipment, making them more affordable and accessible.

In the past, people had to choose hardware with inflexible port counts, or had to connect multiple frames into one net. Artist-1024, with its networked backbone, can easily scale from 16 to 1024 ports with its flexible licensing model. And trunking technology allows systems to grow beyond 6000 ports and allows interconnection across continents, if required.

Successful implementations like this still require careful planning and robust network management, but the benefits make IP an increasingly attractive choice for all modern broadcast infrastructures. Greenfield builds are now designing for IP from the ground up, while more and more existing production environments are switching over.

Adapting The Future

Software-defined hardware can also promote flexibility and future-proof systems for whatever is coming. Riedel pioneered this approach in 2009 with MediorNet and there are now millions of MediorNet SDI and IP I/Os in daily operation all over the world. In addition, initiatives like Riedel's UIC-128 (universal interface card) for the Artist-1024 matrix platform can help minimise future hardware investments. Each high-density UIC provides up to 128 ports per card and can be reprogrammed to switch between SMPTE 2110-30/31, MADI or router/processor/Artist fibre.

Meanwhile, Riedel's SmartPanel combines an intercom panel, router control panel and audio monitor on one device, which not only takes up less space and consumes less power, but also provides technical engineers with the ability to adapt it to different use cases when the production demands them. Cost of ownership is lower from day one and over time the saving on power consumption can be significant. Crucially, it also saves on switch ports, which is especially important in IP environments.

The Creative Customer

Riedel's SmartPanel concept also speaks directly to our creative customer. Live production environments demand real-time decision-making. SmartPanels help encourage greater user focus by combining several functions in one, easy-to-operate piece of hardware.

When we address the creative customer, we are addressing the person who most needs the system to be invisible. As workflows have evolved, so has the shift of power. IP has facilitated more remote production and collaborative working to allow content creation to come from multiple locations. This not only reduces travel and setup costs, but it promotes wider collaboration between teams across different regions by delivering all necessary bandwidth and low-latency comms required for real-time production.

In the battle for eyeballs across more OTT and OTA channels, creative collaboration and input between teams is key. Implementation is key; the Creative customer may not be technical and may have little interest in how the system operates, but they do need it to be as seamless as possible. And everyone wants to see happy internal customers.

An Eye On The UI

This is where Riedel makes a difference. In fact, Riedel has an entire team devoted to it – our dedicated User Interface team works hand-in-hand with users to understand and to anticipate changes in workflows.

Every project and every broadcaster has a different way of working, but building devices collaboratively with the customer helps Riedel get the most from changing environments. Riedel is a highly flexible comms solution because at Riedel, flexibility is designed in from the start.

Bolero was designed around this principal. With a possible 250 beltpacks and 100 antennas in a single deployment, Bolero sets the standard for wireless intercoms: it has six full-duplex keys, uses Riedel's DECT receiver technology allowing an installation in the most difficult environment and uses a high-clarity voice codec to increase beltpack to antenna density. NFC means no registration headaches; licence free, just touch the beltpack to the antenna for quick deployment.

The SmartPanel encourages distraction-free focus so creatives can stay on task, while Riedel's Artist-1024 licensing scheme enables licences to be quickly switched from frame to frame to deploy in different configurations for bigger, one-off events. With no requirement to buy big up-front to meet the demands of the biggest possible event, flexible licence packs mean that creatives can stay in touch while keeping costs low.

Quiet Performance

The business of content is changing, and the roles within it are changing, but comms is still key – it is the unsung hero of award-winning content. Riedel's job is to keep everyone talking and to adapt to whatever the production looks like, wherever it is, and however many people are involved.

And to do it quietly.

Metering

By Kevin Emmott. The Broadcast Bridge.

The capacity to accurately measure what is happening within an audio signal in terms of level, frequency, phase and loudness brings the solidity of science to knowing all is as it should be.

There are hundreds of audio tools which can help simplify broadcast workflows and sweeten signals. Probably thousands. But the most important audio tools in your arsenal aren't plug-ins, processors or analysers. You don't need to find valuable rack space to house them. In fact, you already own them.

You will know if your mix sounds good because you will hear it with your trusty, old-fashioned, analogue ears. If it sounds good, it's a good mix, but that doesn't mean that you can't use some extra help.

There are many reasons why you might need a clearer view of what's going on and metering can provide it: you'll need it for confidence monitoring, to ensure QoS, to verify downmixes, or to check you have an active signal from a remote source. In a digital architecture you need reassurance that you have enough headroom to avoid distortion of the signal as well as guarantee your loudness levels adhere to legal requirements.

You've only got two ears. You can't be across everything.

The Objective View

Metering makes use of objective data to enable an audio operator to ensure everything is as it should be and it does so at every link in the broadcast chain from input through to output.

Output metering is often about compliance and loudness, while metering within the chain helps ensure operators manage gain staging and signal to noise ratios (more on this in a later article).

Confidence metering is important throughout. At the input stage it's good to check that you have audio where you expect it. For example, a sound operator at the console is more likely to be listening to an output mix rather than an individual input, so mixing consoles have an input bargraph to visually illustrate a live outside source with a PFL (Pre Fade Listen) button as an auditory double check.

Broadcast infrastructures will also have confidence monitors at input and output points throughout the chain to check audio levels and improve efficiency. These meters will gauge levels and also enable engineers to identify where any issues might arise and guarantee QoS.

Phase meters ensure you have phases aligned and show the phase relationship between the left and right channels of a stereo signal on a scale of -1 to +1, where +1 is completely correlated and -1 is completely out of phase. This is important at the channel input level to avoid issues further down the mix.

All these build confidence and help guarantee quality, but what are they measuring in the first place?

Decibels

Decibels (dB's) are useful because they represent sound in the same way that humans perceive it. Human folks perceive loudness logarithmically rather than in a linear fashion, so if the amplitude of a signal is increased by 6dB it will be double the original amplitude. Another 6dB increase will at four times the original amplitude.

Meters measure decibel levels, but not necessarily in the way you might think. Decibels do not describe an absolute level but provide a means of comparing the amplitude of one audio signal with another, so they need a reference point to function.

On a broadcast console, the reference point is at 0dB and is clearly marked on the fader. This is called 'Unity Gain', and at unity the output signal is exactly the same level as the input signal. Pushing the fader above 0dB will increase the level of the signal, while pulling the fader below 0dB will reduce (or attenuate) the signal.

Broadcasters need to measure these levels for two reasons; firstly, to maximise the integrity of the signal, particularly when it is mixed with others where relative levels are important, and secondly to ensure consistency across a station's output in accordance with ratified standards.

There are a variety of reference levels but in practical terms broadcast environments are more concerned with measuring dBFS, or 'decibels relative to Full Scale'.

Meters provide the tools to keep track of this although different meters do this in different ways.

Ballistics

Ballistics is a fancy way to describe a meter's reaction time. Meters don't instantly react to the level of the audio signal which is feeding them, they rise and fall to illustrate those levels. These changes are referred to as attack (rise) and decay (fall).

In a broadcast environment there are two common meter types; VU (Volume Unit) and PPM (Peak Program Meters). When fed a signal with a constant level such as tone they both display exactly the same level, but live TV sound doesn't deal with constants; Live broadcast audio is by nature unpredictable and transient.

VU meters struggle with this as they track the average volume level of an audio signal. They have a much slower attack and decay (around 300ms) and while they are too slow to react to fast peaks hey do a fabulous job of representing average audio levels.

Due to the unpredictable nature of the content a broadcast meter is more concerned with measuring peaks than measuring the average, and one way to track those pesky peaks is to use a meter which reacts faster to the sound. PPM meters have a much faster attack time (less than 10ms), which means they are much better at displaying quick, transient peaks in audio such as the sound of a gunshot or a drum hit. Because peak level meters react faster to these transients rather than an average they also tend to read several dB's higher than a VU.

In fact, transients in broadcast can be so fast that bargraph meters on broadcast consoles will have a peak-hold where the peak level remains displayed whilst the rest of the bargraph decays so operators can keep track.

There's A Limit

It is useful for meters to display peaks to ensure signal integrity because it's those peaks and transients that are the most common culprits of distortion. Remember dBFS? Digital audio has an absolute level of 0dBFS, so dBFS values are always less than or equal to zero. At 0dBFS the audio runs out of quantising levels and distorts (also known as 'clipping').

Mixers need to ensure that the combined signal doesn't overload (i.e. go above 0dBFS), and digital signals are referenced from this point down simply because a 0dBFS signal already contains the maximum amount of digital information it can deal with without distorting.

Reference Levels

How broadcasters visualise the meters and how much headroom they have to play with until it hits that limit is ratified in different territories with different reference levels. First issued in 1992, the European Broadcasting Union's (EBU) R68-2000 Alignment level in digital audio production recommends a standard alignment for professional equipment where 0dBu is equal to -18dBFS. This gives operators 18dBFS of headroom to play with before the meter shows clipping. Meanwhile, the US uses SMPTE's RP155 standard where +4dBu equals -20dBFS, which provides 6dB more headroom.

More headroom gives broadcast mixers more flexibility where the inputs are metered at the input, and broadcast consoles have to be able to adapt their reference levels depending on location.

Lots Of LUFS

Another vital meter is loudness. In 2010 the EBU introduced its Loudness Recommendation EBU R128 and the US Congress passed the Commercial Advertisement Loudness Mitigation (CALM) Act. Both are based on the International Telecommunication Union's BS.1770-3 recommendation which was introduced as a universal standard to measure perceived loudness over time.

Loudness regulations were introduced to keep the volume of broadcast output and the volume of advertising consistent. Loudness metering extends the capabilities of VU or PPM meters and is measured in LUFS ('Loudness Unit Full Scale'). In the simplest of terms it measures a broadcasts relative loudness over time. Because they're referenced to full scale where 0dBFS is the maximum, readings are always negative.

BS.1770 loudness meters measure momentary (average loudness over the last 400ms), short term (averages over three seconds) and integrated (average over the complete program so far). The latter is used to ascertain whether the program passes international guidelines. It also measures the maximum True Peak, which is similar to a peak level measurement but also considers inter-sample peaks.

Broadcast standards around the world are actually very similar and are used to help normalize audio across all streaming platforms too. The main profiles for live OTA broadcast are EBU R128 (working to a Target Loudness (LUFS) of -23 and a Max True Peak of -1), and the ATSC A/85 profile in the US and Canada (working to -24 and -1 respectively).

Loudness meters provide a way to monitor and regulate average loudness levels over the duration of a broadcast and are usually part of the meter bridge on a broadcast console. If the transmission output runs too loud, then broadcasters are obliged to compress that output to fit, so mix operators are at pains to adhere to it to avoid on-air compression.

Ears Are Not Enough

Ears are a wonderful thing, but with so much to keep across, the two of them aren't enough to measure what is happening within an audio signal in terms of level, phase and loudness. Metering brings the solidity of science to ensuring everything is as it should be, and it means that you can use your ears to concentrate on more creative endeavours.