It's the Beginning of a Long Journey

Schneider of 2wcom discusses standards as well as tips for ensuring transmission robustness

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Anke Schneider is responsible for sales and corporate communication at 2wcom.

***Radio World:** What is your perspective on AoIP right now?*
Anke Schneider: We believe certain quality requirements are increasing. Compatibility, audio quality, flexibility, simplicity and transmission robustness are the current top buzzwords. Broadcasters need solutions that support studio-to-studio, studio-to-transmitter links and cross-media tasks. The latter also means production concepts for content distribution and storage including both radio and television.

In addition, looking at mixed networks with expanded infrastructures, we have to take into account the main needs of broadcasters. This means also accepting the challenge to harmonize old and new technologies and navigate to the next generation of technology.

***RW:** What important industry questions need to be settled?*
Schneider: Some may think this is pretty clear, but they're wrong. It depends on perspective as well as several parameters, such as big or small radio station; target group to be reached; content monetization; pure studio, pure event or expanded network; and so on.

We have to concentrate on a few common denominators like high-level compatibility and flexibility in application regarding interfaces, even through third-party apps, protocols, codec algorithms and distribution technologies. These are the main issues for broadcasters that we've identified based on discussion with customers and dealers; and in my opinion it's the beginning of a long journey. Right now, several protocols, standards and proprietary solutions are in use, and this makes it sometimes difficult to integrate new devices into existing structures. The crucial question to be answered is how can harmonization be realized.

***RW:** What is the status of efforts to create interoperability and "discovery"? How much closer can we get to true "plug and play"?*
Schneider: The level of effort is really high. Discovery turns out to become more an adventure. Even if your technical concept is well-thought-out and includes all common codecs, protocols and standards, it is trial-and-error due to the challenge of assuring interoperability with third-party applications.

Some manufacturers use a proprietary algorithm — encoding and decoding of an AAC profile can be done successfully, or you have to deploy a variant respectively for a certain external decoder. Several protocols for Audio over IP interoperability are in use like Livewire+, Dante or Ravenna and operate properly in closed networks; but the exchange of data among each other remains a challenge. As a result it is hard work in development to get a plug-and-play experience. But as proven by our team of engineers it is indeed possible: easy transcoding and protocol transforming.

***RW:** What have AoIP "plugfests" taught us?*
Schneider: In these days of fast-paced development in broadcast technology, the best technical solution is worthless unless it's as compatible and interoperable to third-party products as possible.

In everyday operations, most engineers rarely have the time or opportunity to focus on these requirements. The priority is on developing new products or features; testing interoperability is of secondary importance, especially as the technical contact of the relevant manufacturer is not available. Which leads to the question of how to increase compatibility of prototypes without interference

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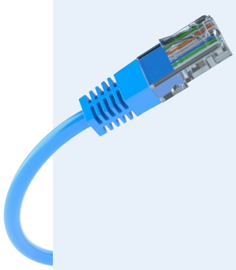
from non-technical, more sales-oriented staff.

The logical answer was to start plugfests — events at which only engineers are allowed to participate, offering a chance for an agile and creative exchange of know-how. This has also ensured that initial fears of industrial espionage have quickly disappeared. In the end, a win for everyone.

RW: *How do AES67 and AES70 complement various AoIP solutions?*

Schneider: Eyeing the established IP network protocols, each offers advantages for certain use cases. For example, Dante or AVB are mostly operated in concert halls or theatres, Ravenna in expanded radio networks, Livewire+ in broadcast studios and SRT to provide content for both video and audio. These protocols are not interoperable to each other, which turns out to be a problem when intercommunication in mixed networks is needed, e.g. Livewire+ and Ravenna.



AES67 and AES70 … take harmonization of data exchange to a higher level but we believe you have to go one step further and provide in addition the following standards: EBU N/ACIP Tech 3326, SMPTE ST 2110, SMPTE ST 2202-7 or NMOS.

AES67 and AES70 are links between and hence they optimize interoperability. AES67 functions like a Babel Fish for different audio protocols and allows us to pass audio between the respective ones, but it does not enable us to discover the devices involved. That is done by the open standard AES70, which offers a control and connecting management of professional audio and AC media network devices regardless of whether the systems are small or big.

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For example EBU N/ACIP Tech 3326 narrowed parameters of existing VoIP protocols, adopted it to Audio over IP, now known as SIP (Session Initiation Protocol) to easily establish connection and negotiate codec algorithms between devices and not only to discover a unit. Moreover, SMPTE 2110 and NMOS turn out to become the successors of AES67 or AES70, as future tasks can also imply management of mixed and asynchronous networks including not only audio but also video.

RW: *What is the most common technical/support question you get from users?*

Schneider: "How can I ensure retransmission robustness and at the same time take into account quality and economical needs?"

This can be done in several steps and depends on the respective parameters of a station or a network. Normally we go in detail with our clients and finally advise best configurations and further aspects to be taken into account. Let me explain by pointing out on three samples.

First, choosing the right codec algorithm: For real-time applications like audio description or live events, audio quality and low latency is mandatory and achievable by choosing PCM or by encoding to E-aptX or Opus. When uploading files to websites transcoding to e.g. all AAC profiles like AAC xHE, Ogg Vorbis and all common MPEG layers ensures bandwidth economy. With cross-media applications in mind, a smart display of broadcast content on a station's website is possible by transcribing the audio in combination with an image generated from the produced video. Both transcoding and transcribing enable radio stations to subsequently provide broadcast content to the audience and store it for future re-usage.

Second, establishing connections between devices in mixed networks: For this purpose, only a device that complies with EBU N/ACIP Tech 3326 should be used to guarantee an automatic connection setup between devices via SIP and to negotiate the suitable codec algorithm or protocol. Personally, I recommend testing the short-listed devices on compatibility to third-party products. We learned manufacturers implement SIP sometimes in a proprietary style, which can turn out to be disruptive for the original aim of interoperability. Looking at day-to-day tasks, a device already providing a SIP phonebook pre-configurable via web interface by operators eases the work of users in the field.

Third, a multi-layer concept ensures transmission robustness: Hardware devices should be equipped with at least two power supplies. Protection against IP packet loss offers Pro-MPEG FEC or Reliable User Datagram Protocol, which is more effective and more economical in means of bandwidth. Further redundancy can be achieved by transmitting two streams or different audio qualities in parallel. In case of interruption of the main stream, IP packet losses can be recovered by picking them from the second stream or the respective receivers switch seamless to the next audio quality. ■