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TECHNICAL
REVIEW

LiveIP: a practical exploration

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Foreword

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Abstract

Over the last few years we have seen a change in the way media is being consumed. We have seen new ways to reach audiences as well as a multitude of devices coming into the market. The Internet and the increasing (mobile) broadband connectivity have changed the expectations of consumers who now want content to be available at any time, at any place on any device.

The phenomenal success of Internet technology in recent years is leading broadcasters to adopt new ways of working. Almost every part of a broadcaster's production chain has now evolved into an IT- and IP-based infrastructure. Today, only one part of the production chain still relies on dedicated SDI based networks: this is live production.

The LiveIP Project is a pragmatic exploration of the possibilities and opportunities that are achievable with today's IP enabled broadcast technology for live production.

Based on experiences from the LiveIP Project, this paper attempts to address, in four separate chapters, the following questions: Why should a broadcaster change to IP and IT for live production? What is a possible technical realization? What operational changes will this bring? Will all this be possible at a lower cost?

The roadmap to a fully-fledged IP-based live production environment, including distributed workflows and service virtualisation, is an evolutionary one. The technology does not yet deliver all the anticipated possibilities, and many issues still need to be addressed, but it certainly is evolving in the right direction. And with the current state of technology, broadcasters can already initiate the transition and start building the know-how that will be critical for its success in the long run.

LiveIP: a practical explanation

1. Introduction

Over the last few years we have seen a change in the way media is being consumed. We have seen new ways to reach audiences as well as a multitude of devices coming into the market. The Internet and the increasing (mobile) broadband connectivity have changed the expectations of consumers who now want content to be available at any time, at any place on any device.

The phenomenal success of Internet technology in recent years is leading broadcasters to adopt new ways of working. Almost every part of a broadcaster's production chain has now evolved into an IT- and IP-based infrastructure. Today, only one part of the production chain still relies on dedicated SDI based networks: this is live production.

The LiveIP Project is a major proof of concept of just how far the industry has already moved towards a future that is based on open standards for live production.

This Technical Review will focus on the exploration that was made possible by building and operating a live TV production studio with state of the art IP-based equipment at VRT and using available interoperable open standards. LiveIP is the result of collaboration between Belgium’s national public broadcaster VRT, the European Broadcasting Union (EBU) and a group of innovative broadcast technology partners including Axon, D&MS, DWESAM, EVS, Genelec, Grass Valley, Lawo, Nevion, Tektronix and Trilogy.

The ongoing business transformation of media production studios would be difficult to achieve using the current and well known SDI-based technology as it is not flexible enough to adapt to the continuous and rapid changes the industry is facing. While IP is often publicised as the technology to support that transformation, there are still many questions in the industry regarding its readiness for live production e.g. can it, in its present state, bring to professional live media the transformation seen in other businesses such as telecoms or even media post production? Should broadcasters begin to move their facilities to IP, or wait?

The LiveIP Project attempts to provide answers to these questions, by examining the different challenges and evaluating possible solutions. In particular, this Technical Review reveals the current state of interoperability between available products from different vendors, as well as the benefits of using IP for live production applications.

2. Sandbox approach

2.1 Co-innovation

The LiveIP Project started in the VRT Sandbox[1], a technology accelerator programme. The main purpose of the VRT Sandbox is to set up partnerships for short-term pilot projects with relevant innovators. New media technology is built and tested in a realistic context – at the VRT, in their television, radio, and digital production and distribution facilities. Projects are built around short phases with clearly defined goals. This approach has proven to result in practical, hands-on findings rather than just theory.

2.2. Maturity of interoperability

The transition to IP video transport and IT architectures is a fundamental change for our industry. It brings more flexibility and modularity, but also adds complexity. The technology is also new to users as well as to many vendors.

This being said, we found that users have a baseline expectation: they want the same level of interoperability as with SDI, which gives them the liberty of building systems from best of breed components.

Therefore, there is a real advantage to bring together technology pioneers on the premise of a broadcaster to interconnect their hardware and software to address a real use-case – all while the technology is still evolving. Within this “safe haven”, engineers can cross company boundaries and work together on the interoperability between their products without behind burdened by commercial concerns. Moreover, the findings from this project are contributing to the international standardization of the technology e.g. through the EBU, VSF, SMPTE, AMWA, AIMS and JT-NM.

2.3 Iterative process

The LiveIP Project was set up with many phases, built around typical workflows.

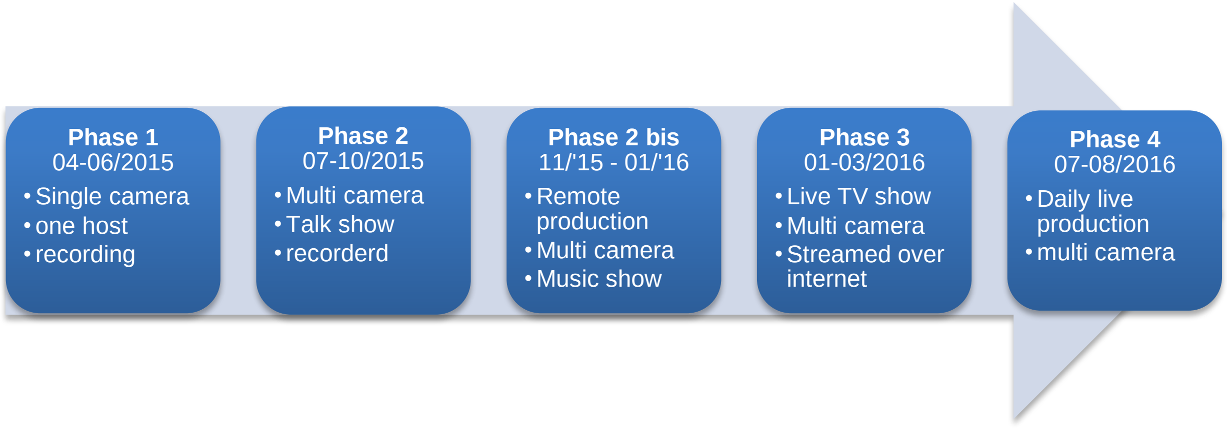

Figure 1 – Timeline of the LiveIP Project

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[1] http://sandbox.vrt.be/

2.3.1 Phase 1: Develop a basic, single camera studio[2] to establish the interoperability needed to produce content in an IP environment. The three locations (studio floor, data centre and control room [See 4.1, Architectural vision]) were and are still interconnected with only three fibre cables. The system was well received by VRT's operational team.

2.3.2 Phase 2: The system was expanded to four cameras with new components to reproduce full studio functionality. VRT simulated a multi-camera talk show production.

2.3.3 Phase 2 bis: The control room was connected to a remote studio floor, at the Bozar concert venue, situated in the centre of Brussels (5 km away from the VRT production centre). A live-to-tape recording of a piano concert was produced remotely.[3]

2.3.4 Phase 3: The network was scaled up to accommodate all of the sources and to drive additional screens on the stage set. In March 2016, an uninterrupted programme production of 90 minutes was streamed live over the Internet.[4]

2.3.5 Phase 4: At the time of writing this paper, the set-up is being prepared for a daily production to take place during the upcoming summer months.

2.3.6 Next phases: There is potential to further evolve the system to new standards as they become implemented in the products of technology partners.

2.4 Connect and share

By putting new technology in a real setting, the LiveIP Project provides an opportunity to evaluate the state of today's technology, but also to explore new workflow opportunities by collecting feedback from users. This valuable and unbiased information gained from this early experience is shared with the community of Public Service Media (PSM) and with the industry to accelerate the acquisition of know-how in this area.

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[2] The green key recording (https://youtu.be/MMLq3BSOZJo)

[3] The remote production use case (https://youtu.be/_A4nWM2dLsc)

[4] The LiveIP Live Debate (https://youtu.be/7cx4ZMcc3gY)

3. Business transformation

VRT has a live production facility that is as robust as possible with today's SDI-based technology; however, it needs to become a lot more flexible, scalable and shareable in the future.

Moreover, VRT plans to move to a new facility which should be operational around 2020. This is a real opportunity for VRT to embrace these new possibilities in a renewed infrastructure. The lessons learned during the LiveIP Project will provide support for making decisions about the types of technology that should be used in these new facilities.

3.1 Do more with less

The continuous challenge of having to produce more content with less (e.g. less resources) is not new for broadcasters. "Less" can mean a reduction in staff members but also in budget or time. However, there is a limit to how far resources can be reduced while the pressure to produce more content grows. New ways need to be found to increase efficiency and effectiveness in production. Most notably, new technological solutions are needed to automate the processes and enable new better workflows.

3.2 Evolving content production

Producers and directors always want to find better ways to tell their stories. They want to be the first using new innovative technology to bring their story in an original and attractive way. This includes using more live sources, recorded footage, stills, music, text and all sorts of enriched data. The capabilities now available in the "prosumer" market (e.g. action cameras, drones, etc.) are setting expectations for professional media facilities and the success of their show depends on it.

3.3 Digital shift

The digital shift has a clear impact on audiences. They are able to consume more media at anytime, anywhere and on any device. This pushes media organizations to adopt new formats and to repackage their content for different media outlets. This has to be highly automated and data-driven, both properties that characterize IT technology.

3.4 New business models

When the digital shift happened in post-production some years ago, the benefits turned out to be more than just the simple reduction of costs that were originally targeted. Non-linear editing offered a completely new way for images to be assembled together. It cleared the path for more creative possibilities using commercial, off-the-shelf products. New parties joined the media market and developed pure software tools for professional video production. The entry level into post-production has gone down to such an extent that a huge number of facilities were able to get involved, driving more competition and creativity.

If the digital shift in live production has an impact comparable to that seen in post-production, one can only imagine the changes it will cause in media organizations and how it will impact the competitive landscape.

Although hard to predict, we know that there will be business opportunities resulting from the following:

1) Possibility to use production infrastructure capabilities as pay-per-use or subscription-based services (OpEx model) rather than owning the equipment (CaPex model). This will result in the ability to limit investments to the minimal amount of resources needed on a daily basis and give the ability to scale out to the Cloud when peak usage occurs.
2) Potential improved transparency of costs by using “pay as you produce” approach.
3) Dedicated speciality hardware will make way for software-based applications running on standard IT infrastructure.
4) Capability to share the infrastructure and, consequently, make more cost effective use of investments.
5) Capability to get live feedback from the audience straight into the production chain using end-to-end IP connectivity natively.
6) Capability to accompany the live signal with any conceivable data, such as real-time telemetry or metadata automatically extracted from the scene. This will result in enhanced or completely new formats.
7) Object-based production enabling personalized and adaptive content will be made possible thanks to the data-rich nature of IT where metadata can be bundled with the media in the same flow.
8) Possibility to offer live production functionalities as a service from one’s own infrastructure.

4. Technical realization

4.1 Architectural vision

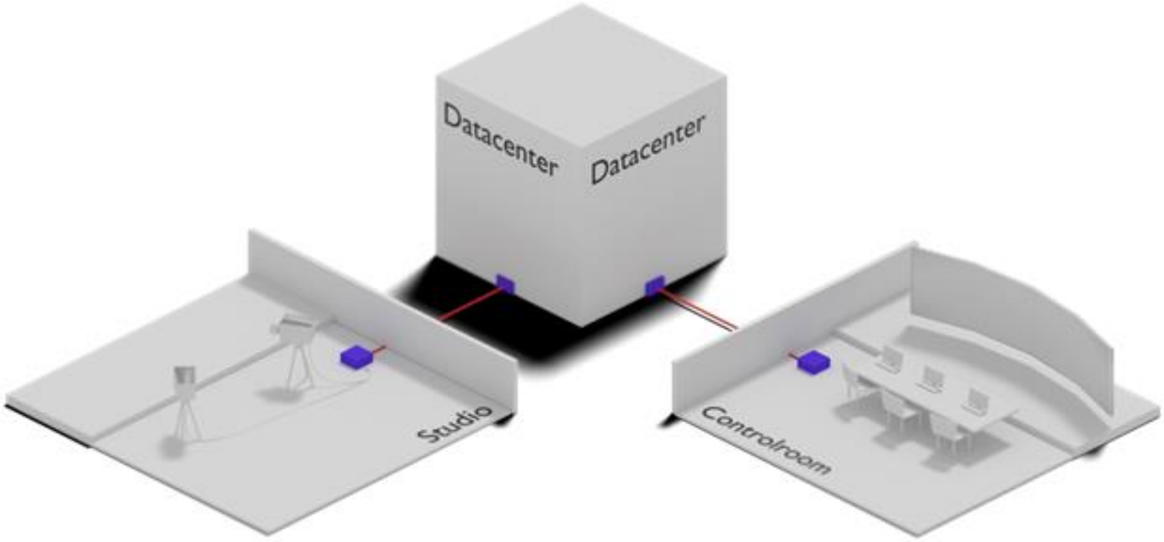

Figure 2 – Three rooms (architectural vision)

The original dream of the LiveIP Project was to end up with a system distributed in the three typical locations of the production studio: the control room, the data centre and the studio floor. These three locations would be interconnected with only a few fibre optic cables carrying managed network connectivity.

4.1.1 Open standards

One of the architectural prerequisites of this Live IP Project was the use of open standards. Products used in the project were to be market-ready or at least very close to being so. Thus, the choice was made to go with the most supported, open standards in the market at the beginning of the project in April 2015.

The reason for using open standards was to ensure that there would be no vendor lock in, therefore protecting the investments made during the project and having the choice to select the product with the best suitable features for the specific use case.

4.1.2 Remote, shared and automated

With this streamlined connectivity, the distance between a studio, its data centre and the control room are no longer limited by the length and the multitude of SDI cables.

Therefore, in a single production centre, it is possible to link different studios and control rooms with the data centre. Furthermore, with this set-up remote production can be achieved by simply extending the network connection to the off-campus location.

Since the back-end equipment is located in the data centre, the resources are centralized and pooled. This means they can be shared amongst different studios and control rooms and can be allocated according to the immediate needs of a production. This way, the investment can be optimized in a much better way than with the current SDI-based studio that has to be designed for the worst-case scenario.

Ultimately, in order to keep the operations simple and to achieve the desired flexibility without the need for network expert support every time resources are needed by the production, a high-level of automation of the network and devices management is required.

In practice, even though not all of the technologies were available to fully implement this architectural vision, the *remote, shared and automated principles*[5] helped guide the system design and the technology providers at every stage of the project.

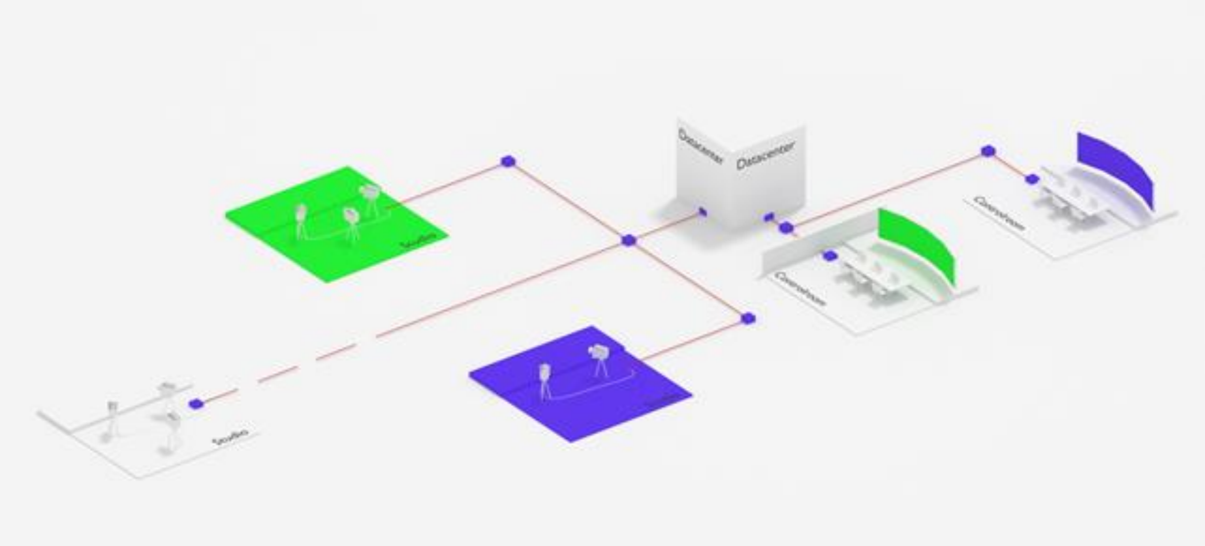

Figure 3 – Remote, shared and automated

4.2 System diagram

The block diagram (Figure 4) provides a high-level view of the main components of the system as used in the LiveIP Project, Phase 3.

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[5] LiveIP RSA concept: Remote, Shared & Automated (https://youtu.be/U0-NLeYjmSE)

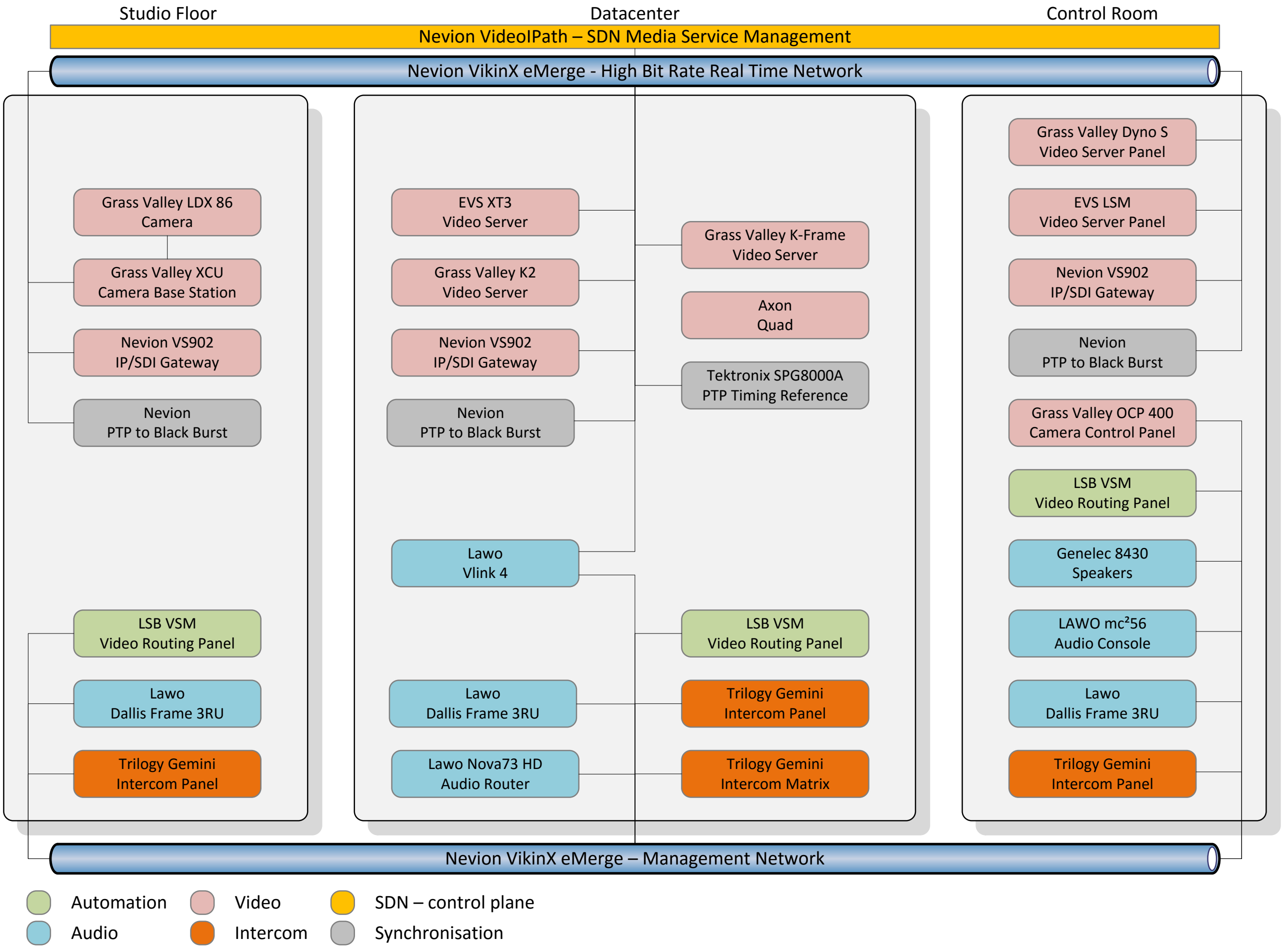

Figure 4 – High level system diagram

4.3 Network architecture

The LiveIP network was constructed as two separated networks. One IP network was assigned for the high bitrate real-time signal e.g. the video, the audio and the timing. The other IP network was used for the management of the devices.

The real-time network is based on a Software Defined Network (SDN), whereas the management network was built as a standard IP network.

4.4 High bit rate real-time network

4.4.1 Software Defined Network: The network solution in this project uses a Software Defined Network (SDN) approach. Per definition, a SDN separates the control plane from the data plane. The data plane remains on the individual switches, the control plane (routing decisions) is moved to a separate central controller.

There are several flavours of SDN emerging on the market of real-time media networks. Following the requirement to use open standards, the choice was made to deploy technology that uses OpenFlow 1.3.

However, not every switch vendor that supports OpenFlow 1.3 has implemented the same set of features that are needed by this implementation. Therefore, it might be safer to use switches tested and approved by the SDN supplier, however this might limit the choice of suitable switches.

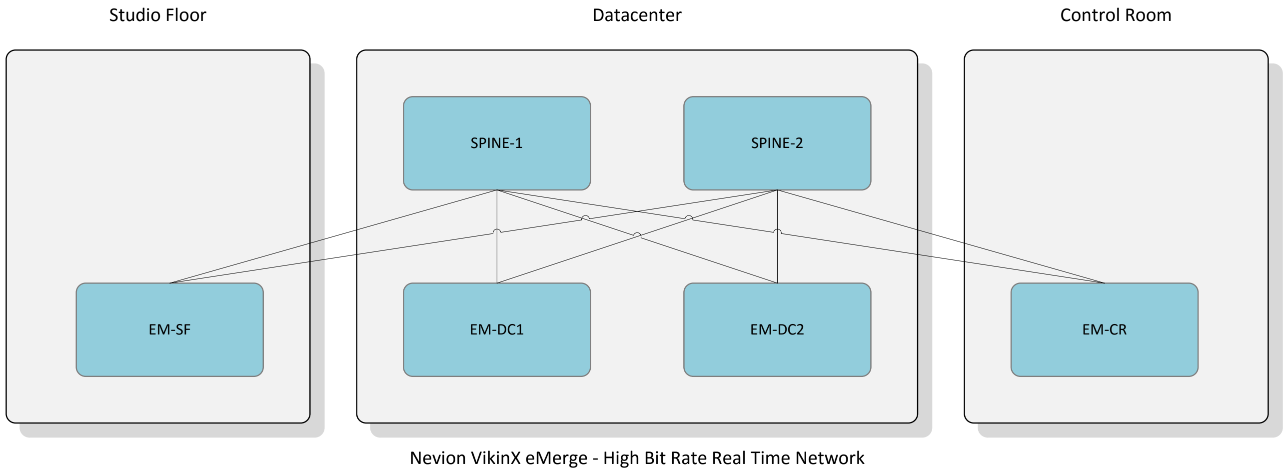

Figure 5 – High bit rate real-time network

4.4.2 Flow routing: In the LiveIP set-up, both sources and destinations used fixed multicasts addresses, in other words, a source (e.g. a camera) would always “broadcast” its stream on a specific multicast address, and a destination would always “listen” in on a particular multicast address.

4.4.3 Fast switching: The connection between a source and a destination is done by making sure the network fabric’s Network Address Translation (NAT) is configured in such a way that the source multicast address is translated to the destination multicast address. At the network level, switching between sources is achieved by rapidly reconfiguring the NAT.

A major advantage of this approach (“break-before-make”, as Nevion calls it) - compared to the “destination switching” technique - is that there is no need for both sources to be forwarded to the destination, so that it doesn’t use twice the bandwidth. And most importantly, the switching in the fabric means that there is no need to have drivers for every destination device like in “destination switching”.

4.4.4 Clean switching: Since the IP network is not content aware, the switching may take place at any point in the video frame. Therefore, in order to get clean switching, there is a need to “clean-up” the stream so that the first and seconds sources are switched at the right points in their respective streams (e.g. top of frame) before used by the destination device(s). In the LiveIP Project, this is performed by the edge media gateways.

This technique was chosen because the speed of operation provides a switching impression that is “clean enough” for most cases, as experienced by the operation crew (e.g. for preview operations). However, some operations like those going "on air" could require the option for a "clean perfect" switching like it would be produced in "destination switching" - which the equipment also supports, along with IGMP switching. Nevertheless, this was not an issue with the LiveIP Project set-up since the "on air" signal was realized only by the vision mixer.

Finally, to support multi-vendor interoperability, this "break-before-make" switching technique would benefit from being standardized.

4.4.5 Spine-Leaf: For the third phase of the project when more bandwidth was required, the real-time network was scaled up using a spine-leaf architecture.

In such topology, a series of leaf switches form the access layer. These switches are fully meshed to a series of spine switches. This topology, when built with non-blocking and low-latency switches, minimizes the amount of buffers to pass through, the latency and the likelihood of bottlenecks between access-layer switches. This architecture is used in data centres where it has proven scalability.

4.5 Management network

A separate network was built in parallel to the high bit rate real-time network. The management of all the connected devices, including the SDN control of the high bit rate real-time network was done via this network.

This simple layer 2 switched network was used to manage and control the SDN network and most of the audio essence runs on this network as well.

In the first three phases of the project, the RAVENNA audio streams were carried on the management network, since they are not supported on the SDN network. For the following steps, AES67 [See Reference 1] audio streams were added and carried over the high bitrate real-time network.

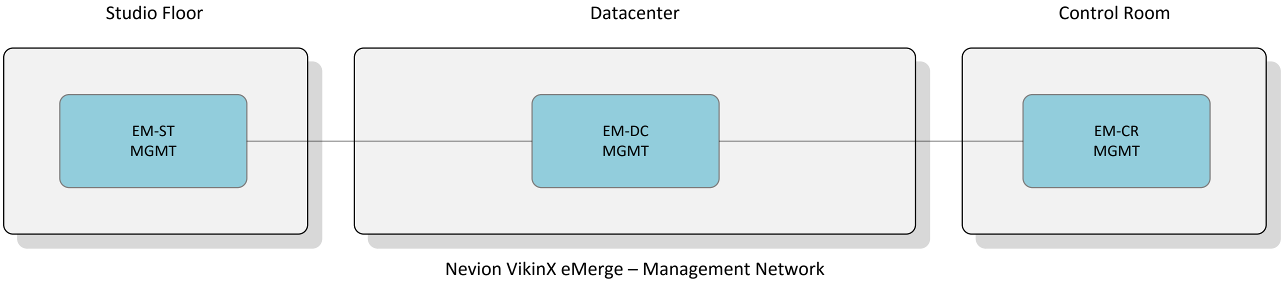

Figure 6 – Management network

4.6 Protection mechanisms

4.6.1 Forward error correction

Forward error correction (FEC) is a technique used for correcting errors in data transmission over unreliable or noisy communication channels.

SMPTE ST 2022-5 [See Reference 3] FEC was not used in this set-up. This was decided because the network used was designed for non-blocking operations and is fully managed, so no packet loss was expected. Another reason for not using FEC is to avoid additional latency that is not wanted for live operations.

To confirm the hypothesis, no artefacts were ever noticed during the operation of the system. More in-depth tests would be required to verify the exact amount of packet errors present in this configuration. In the future, it could be interesting to find out what minimal amount of FEC could make the system robust to random bit errors at a level of acceptable latency and bandwidth overhead.

4.6.2 Seamless protection switching

SMPTE ST 2022-7 Seamless Protection Switching of SMPTE ST 2022 IP Datagrams [See Reference 5] describes a way of sending two matching streams of packets from a source to a destination over different paths, and having the receiver switch automatically to the secondary stream when the primary stream gets degraded.

In the project, a very limited testing of redundant inter-switch connection was done by removing the primary link without visual disruption of the signal. More tests would be required to confirm the applicability of this technology as a generalized protection mechanism.

4.7 Synchronisation

Just like in a traditional SDI-based studio, an SDI over IP studio requires the end devices to be precisely synchronized, in frequency and phase, in order to enable clean switching and avoid building in additional delay through the use of time buffers to align signals.

IEEE 1588-2008 [See Reference 8] or Precision Time Protocol (PTP) is the recognized way to achieve precise-enough time distribution over a network.

During the project, the PTP reference clock signal was used by the audio and intercom equipment. However the video devices did not yet support PTP and had to be synchronized with conventional Black Burst (BB). During Phase 3 of the Project, a card was put into place to derive from the PTP the BB and word-clock distributed in the three rooms, as shown in Figure 7. In Phase 4, the target is to have all video IP gateways synchronized directly via PTP.

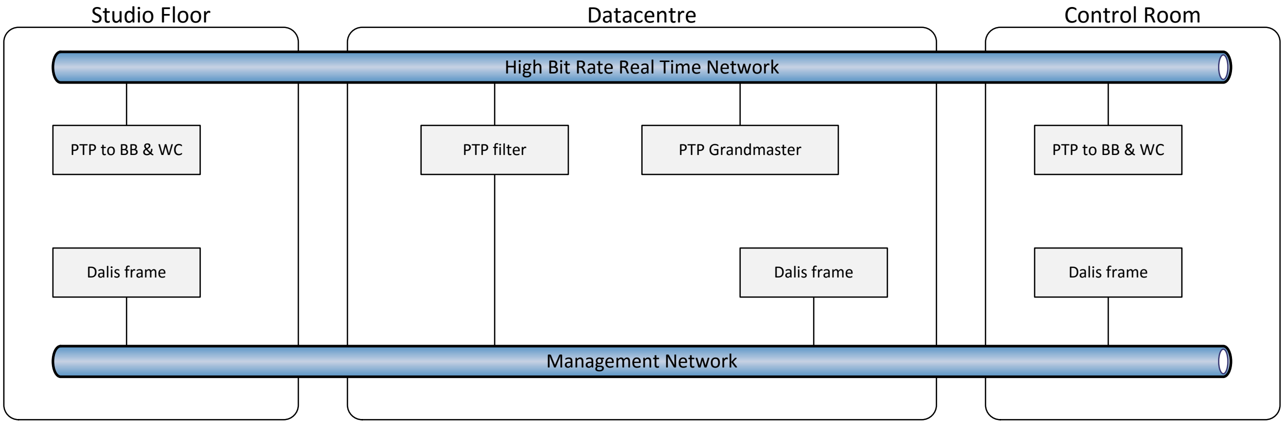

Figure 7 – Synchronisation distribution

The switches in the real-time network are PTP enabled. They run in PTP E2E transparent operation and will compensate for their own delay when forwarding PTP synchronization packets.

The switches used in the management network support PTP transparency, without compensating for their own delay when forwarding PTP packets.

PTP includes several parameters and those are constrained in profiles that are specific to certain applications. In this project, two PTP profiles are used by the different devices. The *SMPTE Profile for Use of IEEE-1588 Precision Time Protocol in Professional Broadcast Applications* as described in SMPTE ST 2059-2:2015 [See Reference 7] and the profile described in AES67-2015 [See Reference 1] for professional audio applications.

As shown in Table 1, some attributes have different default values and different ranges are allowed, depending on the profile.

Table 1 – PTP profile comparison

Attributes	AES67-2015 profile	SMPTE ST 2059-2:2015 profile
defaultDS.domainNumber	default value: 0 configurable range: 0 to 255	default value: 127 configurable range: 0 to 127
portDS.logSyncInterval	default value: -3. configurable range: -4 to +1.	default value: -3 configurable range: -7 to -1
portDS.logAnnounceInterval	default value: 1 configurable range: 0 to 4.	default value: -2 configurable range: -3 to +1
portDS.logMinDelayReqInterval	default value: 0. configurable range: -3 to 5 or portDS.logSyncInterval to portDS.logSyncInterval +5, whichever is more restrictive.	default value: portDS.logSyncInterval configurable range: portDS.logSyncInterval to portDS.logSyncInterval+5
portDS.announceReceptTimeout	default value: 3 configurable range: 2 to 10	default value: 3 configurable range: 2 to 10

To be able to use one clock for both audio and video devices a new set of default values should be used within the common configurable range.

Table 2 – PTP profile overlap

Attributes	Value	Common Range
defaultDS.domainNumber	default value: 0	configurable range: 0 to 127
portDS.logSyncInterval	default value: -3	configurable range:−4 to -1.
portDS.logAnnounceInterval	default value: 1	configurable range: 0 to 1.
portDS.logMinDelayReqInterval	default value: portDS.logSyncInterval	configurable range: -4 to +4 portDS.logSyncInterval to portDS.logSyncInterval+5
portDS.announceReceptTimeout	default value: 3	configurable range: 2 to 10

The issue of multiple PTP profiles is under investigation jointly by SMPTE and AES. A report from AES has been published [See Reference 2] and more interoperability tests are planned to confirm those recommendations.

4.8 Tally

Virtual Studio Manager (VSM) is used in the LiveIP set-up as automation systems. The tally lights are driven by VSM.

4.9 Video

The system uses SMPTE ST 2022-6 [See Reference 4] to transport uncompressed video (SDI over IP). This standard was originally intended for real-time audio/video applications such as contribution, primary distribution, and digital cinema. It specifies that the entire payload of the Serial Digital Interface (SDI) signal, including VANC and HANC ancillary data spaces carrying audio and data such as timecode and closed-captions, is encapsulated as one stream for a limited number of formats up to full HD (3G-SDI). In this set-up the video spatial and temporal resolution are set to 1080i/25 (1.5 Gbit/s).

4.10 Audio

As in many SDI-based television studios, audio from the LiveIP is carried separately from the video. For the audio transport a combination of AES67 [See Reference 1] and RAVENNA is used. The audio equipment is fixed to use 48 kHz as a sampling frequency and a bit depth of 24 bit.

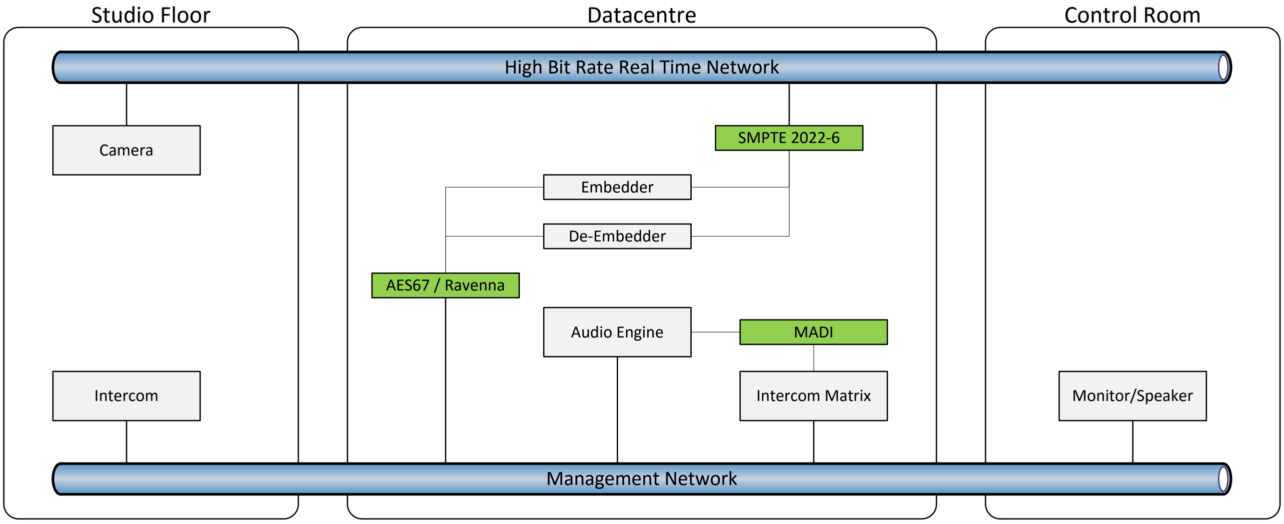

Figure 8 – Audio between two networks

The audio is deployed over a separate VLAN on the management network. Figure 8 shows how the intercom is interconnected with a camera.

If both the camera and intercom matrix would support AES67, the system would be simplified as shown on Figure 9.

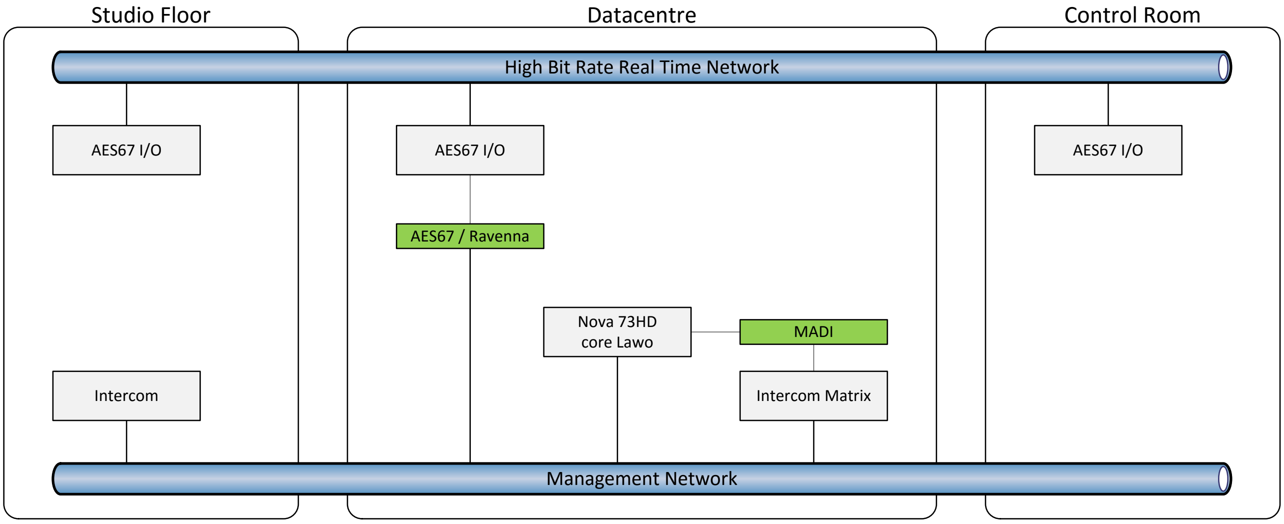

Figure 9 – Audio schema can be simplified

4.11 Progress towards all IP

At the beginning of the LiveIP Project, not all of the components were available to support the chosen standards. Therefore, a number of gateways to convert between standards were necessary. Moreover, a baseband bridge was originally necessary for embedding and de-embedding audio in and out of the 2022-6 streams. By Phase 3, gateway equipment could handle the embedding operations internally thus reducing the need for extra SDI interconnections. As we can see in the chart below, throughout the project, vendors made progress to bring their equipment closer to the target of 100% of flows using IP.

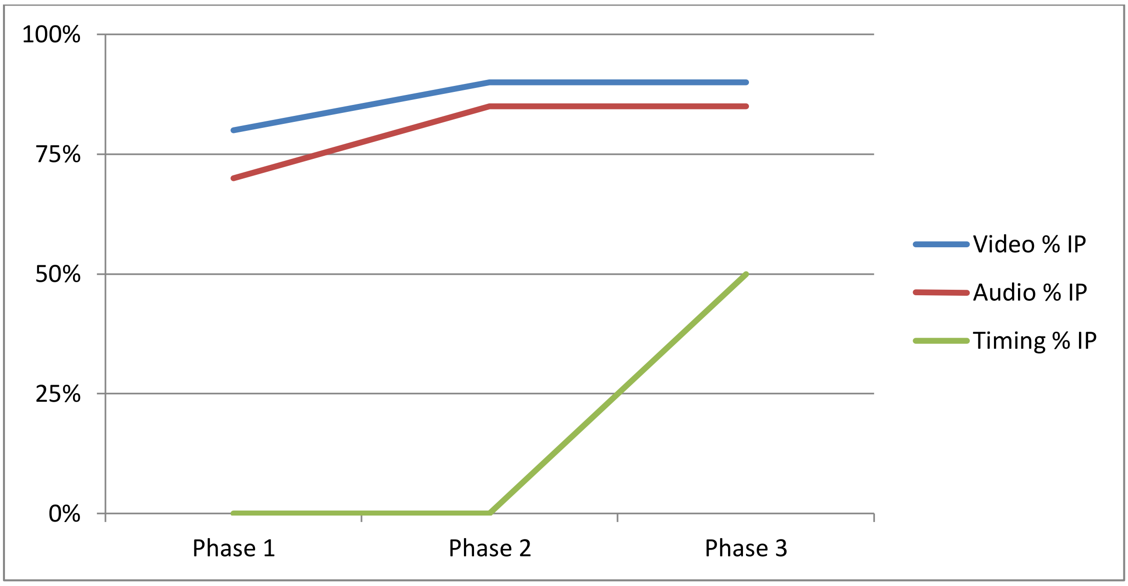

Figure 10 – Progress towards all IP

4.12 Technical learnings

By building, configuring and operating this LiveIP studio, a lot of insights were gained. The reduction of cabling between the different locations is an obvious benefit.

The main technical findings:

1) PTP: a common profile for audio and video devices is needed for using one common clock.
2) PTP: PTP support for video equipment is not widely available yet.
3) FEC: was not enabled on the set-up but this did not have any noticeable issues.
4) A small number of gateways are still needed today. However this will fade away when all equipment will support IP.
5) During the set up and configuration of equipment, it is not so easy to identify or to detect problems or configuration mistakes. One has to rely on software tools to measure what happens on the network. This needs an in depth knowledge of the system.
6) Controlled automatic discovery of newly inserted equipment will be needed to create the flexibility and ease of set-up in dynamic environments such as for outside broadcast productions and at larger scale facilities.
7) If AES67 were supported by more devices (e.g. camera, intercom, etc.), the set-up would be simplified, removing extra glueware.

5. Operational changes

5.1 Simplification of remote production

Many broadcasters are already comfortable and have been using remote production workflows for quite some time. Remote production workflows have many benefits, such as:

1) Reduced size and cost of the set-up (e.g. no OB van).
2) Reduced time to install on-site.
3) Reduced travel time, inconvenience and expenses due to having less staff on the road.
4) Easier access by the core of the production crew (which remains in the production centre) to regular resources like the archives, graphic artists and even the company restaurant.
5) Ability of the crew to work on several productions by staying in the same production centre.

Now, with the simplification of the connectivity using Ethernet and IP networking running over fibre optic, remote production has never been so simple.

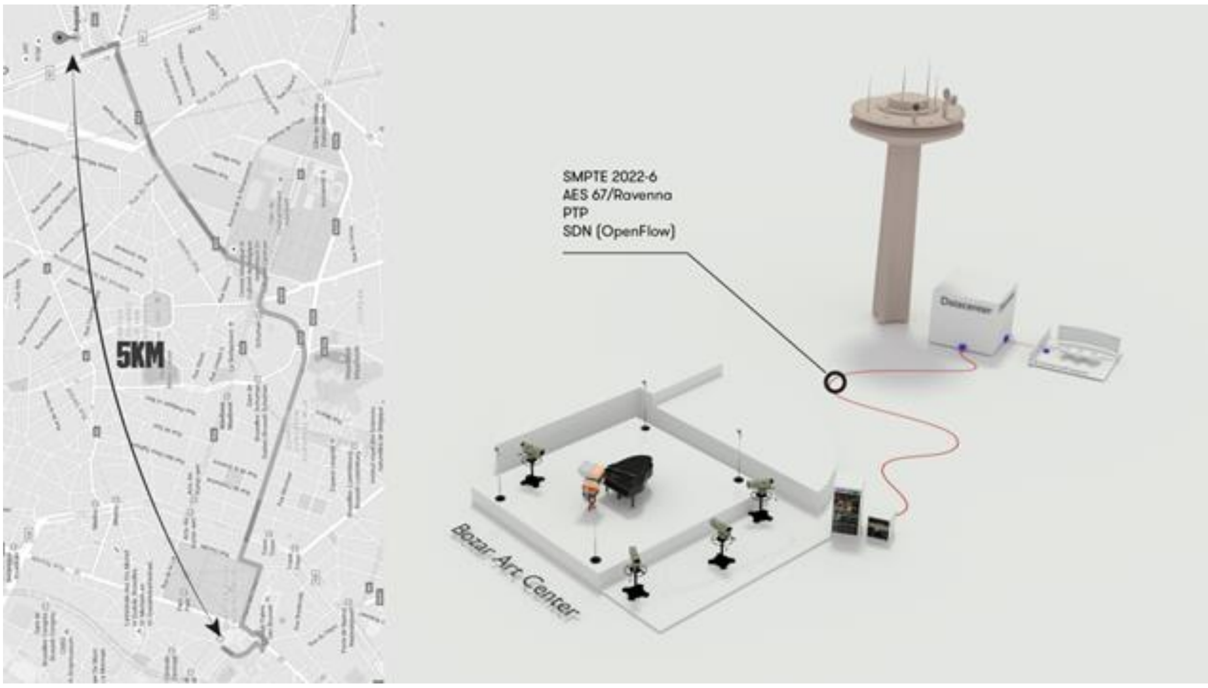

Figure 11 – Remote production

Demonstrating this was the focus of Phase 2 of the project, when the LiveIP set-up was used to produce a piano concert taking place in a concert hall 5 km away from the VRT production centre where the control room is located. By using a single dark fibre link that was already installed between these two locations, and multiplexing the Ethernet networks using CWDM, the connection between the studio's equipment rack and the data centre could be extended transparently and without any noticeable extra latency for the operators, giving the impression that both locations were next door to each other. As soon as the rack was connected to the fibre, the remote production facility was up and running. Tests and configuration was done upfront while still at the production centre. As a result, the whole set-up on location took only 30 minutes!

5.2 Shift in skill sets

These changes in technology and workflows require a different set of skills and a change in the organization of the work across all domains of production: operations, technical support, and project engineering teams.

5.2.1 Operational teams

The equipment used in the project provided a smooth transition for the operational crew. The LiveIP studio can be operated like a classic studio, with identical panels and controls. Therefore the crew didn’t need special training before starting to use the LiveIP system.

However, when the technology will evolve to more software-based user interfaces, this will enable changes in the way the functions are distributed. This might lead to a more flexible organization of the work, where people might have to adapt to changing and mixed roles according to the needs of the specific production.

In the “remote production” trial, it was found that, as the crew was split between two locations, special attention needed to be paid to communication, to compensate for the limited visibility crew members had of each other – even with telepresence cameras that were specifically installed to help overcome this lack of visibility. Related to this, extra care needs to be given to building and maintaining trust between team members in order to get good results when distributing the working positions geographically.

5.2.2 Technical support

The current generation of IP-enabled equipment used in the project still featured plenty of legacy technology (such as Black Burst and some SDI links), however, the real novelty was that the backbone of the system was a standards-based IP network. This means that the production crew needs people with advanced network management skills as well as the more traditional broadcast engineering skills (including a good understanding of the business requirements). Since it is still rare to find this combination of specialist knowledge in the same person, multidisciplinary teams are needed. However, all of the roles in the support team need to understand the layers of the architecture in order to communicate in the same language.

The current technology requires significant manual configuration of the network and devices. As the technology evolves, we expect more automated network management features that will help the team to support more and bigger IP-based systems.

Another observation from the project is the need for proper tools and methods to quickly analyse and solve incidents and problems.

However, the use of SDN technology does provide more visibility of individual signal paths than a traditional automatically routed IP network would do. It is possible to get detailed information from the system on the exact network path that each single stream uses.

Another fundamental architectural difference in IP is the routing, which is distributed in many nodes (switches) rather than at a central point (cross point router). At the moment, a solid understanding of the architecture and stack of technology is required to pinpoint the issues.

5.2.3 System design and implementation teams

Again, the current state of the technology used in LiveIP requires a hybrid skill set of broadcast system design with network architecture for the interconnections.

At this stage, the system is mainly composed of specialised devices interconnected using IP transport. As we expect in the future to use more and more software applications running on standard IT hardware, software integrators and developers, both for front and back-end layers, will need to join forces with the design teams until, eventually, software knowhow will become more widespread.

Project management will also need to become familiar with the best practices of IT, such as the formalized process of ITIL and agile practices.

6. Financial viewpoint

A first effort to clarify the financial impact of this digital shift is to look at the investment of all of the necessary equipment. Is all of this more or less expensive? Will IP enable us to do more with less?

WARNING: The reader should be very careful interpreting these values. The equipment used in this set-up might be different from any other one. In this case, top range equipment is used, even if this was not needed due to the smaller scale or required features. Therefore, the analysis presented here cannot be generalized and should be redone for each individual project.

6.1 Methodology

The system integrator partner in the project, D&MS, was asked to make a Bill Of Material (BOM) that would be required to build the LiveIP studio in its Phase 2 version. For making the comparison, they also designed and prepared a BOM for a classic studio based on SDI technology and with equivalent components to those used in the Phase 2.

Each partner vendor was asked to fill in the list with prices for their products in the Benelux region. These results were compiled into one spreadsheet in order to make the final analysis.

6.2 Comparison

To get meaningful data out of more than 500 items in a list, the items were classified into seven categories:

1) **Build and install:** includes the cost to engineer the solution, the time to build it, the cables and cabling of the system.
2) **Network:** includes the IP switches, SDN or the SDI matrix.
3) **Audio:** includes all of the audio equipment. In both cases, IP-based equipment was used for the audio part.
4) **Common equipment:** includes camera head, lenses, intercom, monitoring, desks, etc.
5) **Hybrid equipment:** includes the camera system, the multiviewer, recording and replay units, synchronisation unit, and vision mixer mainframe.
6) **Gateways:** includes all of the equipment needed to transform the signal from IP to SDI or vice versa, because not everything is equipped with an IP connection (e.g. video monitors, mutliviewer, etc.).
7) **Glueware:** includes all of the bits and pieces needed to convert from one to the other format. In an all-IP system, they should disappear.

Table 3 shows a comparison to the lowest cost solution. The BOM of the IP-based system is 6.8% more expensive than the classical SDI based set-up.

Table 3 – IP-based compared to SDI-based set-ups (budget spent on each category)

Category	IP-based	SDI-based
Build and install	3.0%	5.5%
Network	7.1%	4.9%
Audio	9.5%	9.5%
Common equipment	43.8%	44.3%
Hybrid equipment	39.7%	33.8%
Gateways	3.7%	0.0%
Glueware	0.00%	2.0%
Total	**106.8%**	**100.00%**

Table 4 demonstrates the shift of expenditure between the different categories for each approach.

Table 4 – Expenditure shift in categories (budget spent on categories during set up)

Category	IP-based	SDI-based
Build and install	2.8%	5.5%
Network	6.6%	4.9%
Audio	8.9%	9.5%
Common equipment	41.0%	44.3%
Hybrid equipment	37.2%	33.8%
Gateways	3.4%	0.0%
Glueware	0.0%	2.0%
Total	**100.0%**	**100.00%**

For example:

1) Build and install:
    - A significant decrease in price by 45% for building and installing an IP-based system compared to an SDI-based system. Zoomed in on the detailed level of cabling, connectors and hours spent on installing them, there is a reduction of 66% compared to an SDI-based installation.
    - Project engineering and implementation are levelling out. Due to the fact that IP is new, the integrator estimates to spend extra time on project implementation.
    - These make 2.8% of the IP-based BOM
    - These make 5.4% of the SDI-based BOM
2) Network:
    - The high bandwidth real-time media network including the software defined network consumes 6.6% of the IP-based BOM. It brings the possibility to extend the “matrix” and to have a part of the matrix on different locations.
    - The SDI matrix takes 4.9%.
3) Audio:
    - As the audio is in both configurations IP-based, there is no difference.
4) Common equipment:
    - There are components which remain the same: e.g. camera head, lenses, production desk, etc.
5) Hybrid equipment versus SDI-based equipment:
    - The camera system, recording and replay units and vision mixer are hybrid in the IP-based set-up. This means they have as well SDI as IP

interfaces. In the future this cost will be shareable over different productions. These systems make 37.2% of the IP-based BOM.
    - The SDI based BOM has a share of 33.8% of these systems.
6) Gateways:
    - There are a few gateways needed to convert the IP signal into an SDI signals because the multiviewer and video monitors are not IP enabled.
    - There is a gateway needed to infuse to audio into the SMPTE ST 2022-6 streams (Lawo V-Link4).
    - These components are expensive and account for 3.4% of the IP-based investment, but will disappear as more native IP-based products become available.
    - This comparison shows no need for IP-SDI gateways in the SDI-based setup. However, as edge equipment will soon be delivered with IP interfaces as a standard, SDI-IP gateways will be needed.
7) Glueware
    - Glueware disappears in the IP-based environment.

7. Future work

7.1 New workflows

Since this system is basically about interconnecting existing studio devices using IP for the transport of media, timing and control signals, the workflow (as experienced by the crew) reproduced a typical SDI-based live production studio. This has the great benefit of familiarity for users.

In the future, since the IP network is at the core of the system, it will be interesting to try out new possibilities, such as more geographic distribution of the crew, object-based production and mixed live, near-live and non-live workflows.

7.2 The next evolution of standards

This project started in the spring of 2015 and the design of the system was based on technology that was available at the time and that vendors had committed to provide working implementations during the course of the project.

In the meantime, the industry has been actively developing further standards and open specifications that should enable the realisation of the RSA architectural vision in the near to medium term future.

The EBU expert group on Future Network Systems (FNS) is maintaining a roadmap of the foreseeable evolution of technology available[6], thanks to the LiveIP experience and the follow up of international standardization.

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[6] EBU Technology & Innovation website: (https://tech.ebu.ch/live-ip)

The next major evolutions will:

1) enable the transport of separate essence flows and support UHD formats (e.g., VSF TR-03 being standardized as SMPTE ST 2110).
2) provide automatic discovery of devices and flows (e.g., AMWA NMOS).

This will enable new workflows and larger scale systems by reducing the need for manual configuration steps.

The LiveIP Project is investigating the possibility of upgrading the components of the system to take advantage of new standards as they become available.

7.3 Measurements

The LiveIP Project approach was to “make it and try” and this resulted in a very positive and rapid learning for the VRT, the technology partners and the EBU community.

So far, a minimal amount of formal measurement was done to quantify the assessment that was received from the users. Now that the set-up is up and running, there is an opportunity to quantify parameters such as latency, video and audio transparency, bit-error-rate, degradation behaviour in case of packet loss, etc. To do so, new procedures need to be investigated and new measurement tools are required.

7.4 Cybersecurity

As the LiveIP Project operates in an isolated, controlled network that requires that the SDN controller to open the route for any flow of data, security risks have been limited to something similar to those of an SDI-based system. It is clear that as soon as the system is linked to the Internet to benefit from a greater interactivity with the audience or to get access to some distant content, security will need to be tightened.

The EBU has published a Recommendation for the manufacturers of video equipment to follow a minimal set of best practices regarding security[7].

8 Acknowledgements

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[8] IEEE 1588-2008 Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control Systems

10 Author biographies

Willem Vermost (EBU)
Willem joined EBU Technology & Innovation as Network IP Media Technology Architect in 2016. He obtained a Master's degree in electronic engineering and a Master's degree in applied computer science. Before this, Willem gained 16 years of experience at the Belgian public broadcaster VRT in different roles. He has always sought to combine broadcast and IT technology in the best possible ways and in many different projects. Willem is a member of SMPTE and the AES.

Felix Poulin (EBU)
Felix is a topic leader for Networked Media Production at EBU Technology & Innovation. He completed his diploma in electrical engineering at Montreal's Polytechnic with his final thesis done at MIT. He began working as an audio engineer at Cirque du Soleil and other international productions. Prior to the EBU, he also worked for CBC/Radio-Canada as an adviser in new broadcast technologies. Felix co-chairs the Joint EBU/SMPTE/VSF/AMWA Task Force on Networked Media and coordinates the annual Network Technology Seminar as well as the Strategic Programme on Future Networks. He also collaborates with the VRT Sandbox, the VRT LiveIP project, the AMWA Network Media Incubator and some SMPTE standards committees.

Wouter De Cuyper (VRT)
Wouter spent the last 14 years at the Belgian public broadcaster VRT where he took on many different technology-related roles and gained a high level of experience in a variety of media topics including television, radio and digital products. He built a substantial part of the file-based video production system and was part of starting the digital shift in distribution and production at the VRT. Currently, Wouter is working as a Technology Architect responsible for implementing state of the art media facilities in the VRT's new building, planned for 2020.

Karel De Bondt (VRT)
Currently working as a Project Manager at VRT's Technology & Operations department, Karel De Bondt is involved in VRT's Sandbox technology accelerator program. On top of this, he's managing projects where technology, operations and content production meet. Typically this involves introducing new workflows and implementing new technology infrastructures. Karel has also worked as a business analyst and business relationships manager at VRT.

Michel De Wolf (DWESAB)
Michel De Wolf graduated in 1982 as a civil electronics engineer from the University of Leuven. He founded DWESAB engineering in 1987. It became a small profitable company active in the Benelux in the field of automotive, industrial control systems and broadcast solutions. In 2008 EVS Broadcast Equipment acquired this company, including the staff. It became the EVS Brussels Development Center, where mainly studio and archive applications are being developed. At the end of 2012, he was nominated CTO of EVS. In mid-2014 he left EVS, in order to focus on new activities in DWESAM.

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