Audio over IP

Решения для телевидения

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Стенд A39

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AES67 Livewire+

In 2016 we spoke about the industry shift toward integrating IP into the production and distribution workflow. Now IP is an accepted method for program delivery and distribution and audio over IP is becoming an integral part of audio production for TV.

This year we can add the AES67 standardization of audio over IP (AoIP) and the SMPTE 2110 standard. SMPTE 2110-30 is particularly important because it standardizes distribution of AES67 AoIP streams for production.

Before SMPTE 2110, video over IP simply packetizes an complete SDI stream with embedded audio. To get to the audio the video IP stream must be unpacketized and brought back to SDI. Audio can then be de-embedded and mixed, replaced, processed, etc. The audio must then be re-embedded and then the SDI stream is packetized into IP an IP stream again.

SMPTE 2110 standardizes audio as an independent AES67 AoIP stream. The AoIP data can be accessed directly and independently from the video. PTP clock, part of the AES67 and SMPTE 2110 standards will allow accurate timing between audio and video to be maintained.

AoIP: Почему сейчас?

Экономия

Гибкость

Надежность

Инвестиции в будущее

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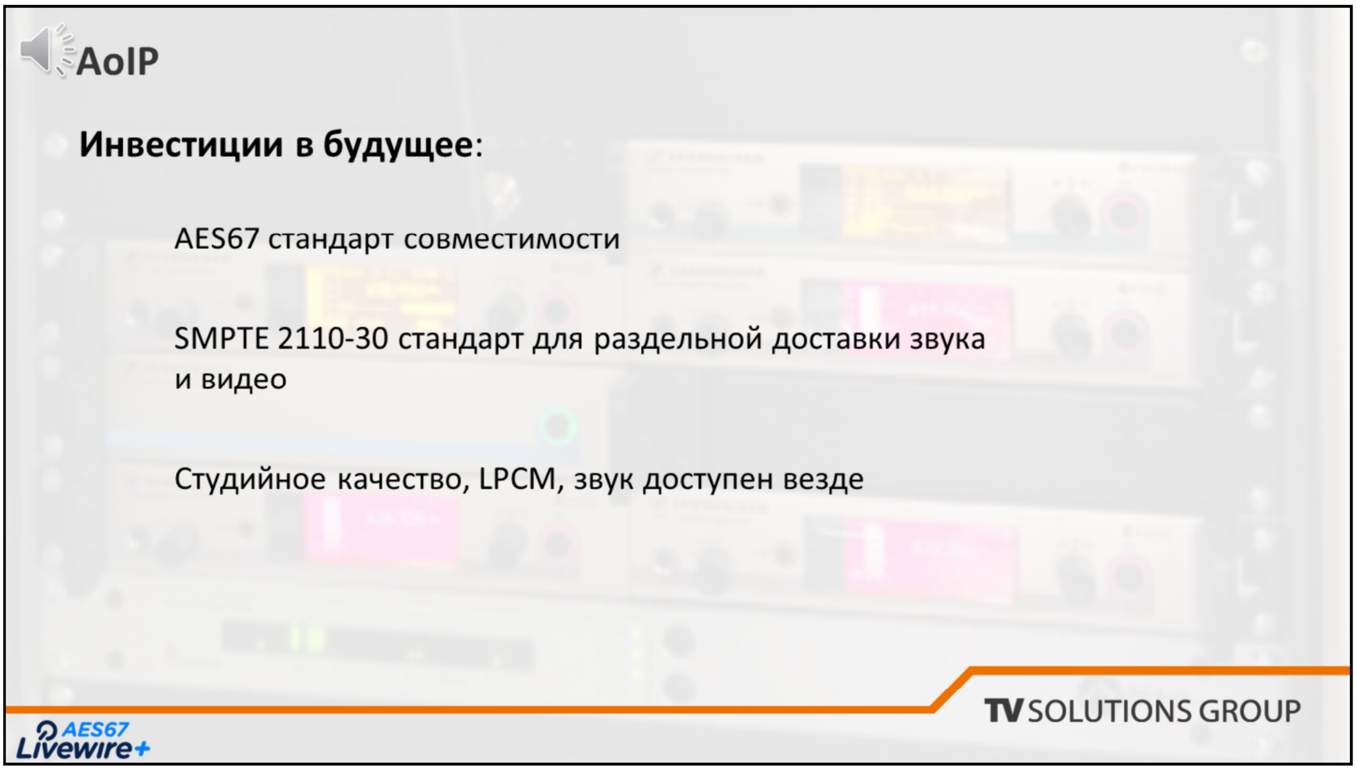
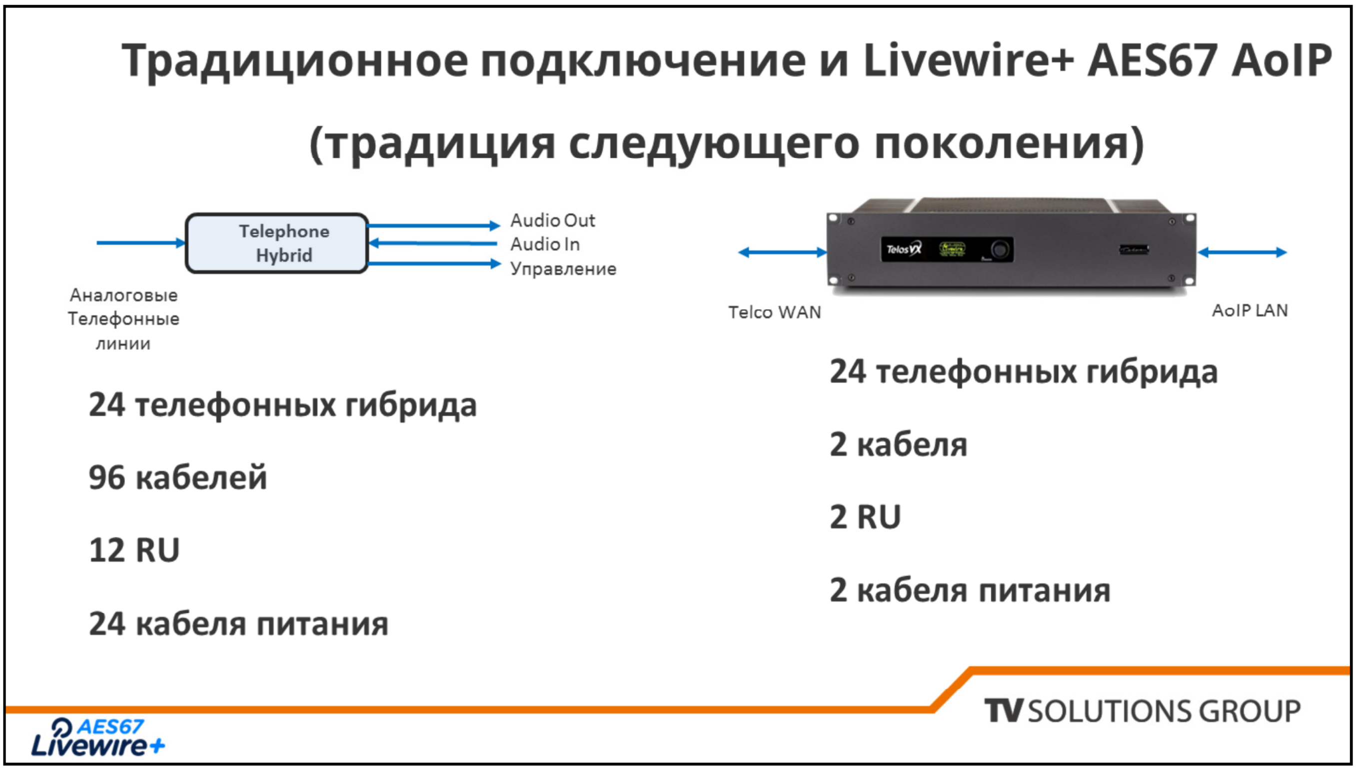
The reasons to use AES67 AoIP have not changed. We are going to look at range of completed projects that use AoIP, specifically Livewire + AES67 (LW+ AES67), to great advantage.



Now that AES67 is finding increasing acceptance with manufacturers and SMPTE 2110 is nearing completion AoIP has taken on a new meaning for the future. Now lets look at some projects that use AoIP to great advantage.

Premiere League football stadiums are linked to the broadcast center, in Manchester, via fiber. Since the bandwidth is available for high speed Ethernet AES xNodes were placed in every stadium and LW+ AES67 was run on the network. Multiple connections of production quality audio (48 KHz sampled, 24 bit, LPCM) were now available between all of the stadiums. By interfacing the audio with the intercom and master control, in Manchester, communications and program can share high quality audio. Commentators in any of the stadiums can talk with one another and this can also be taken to air.

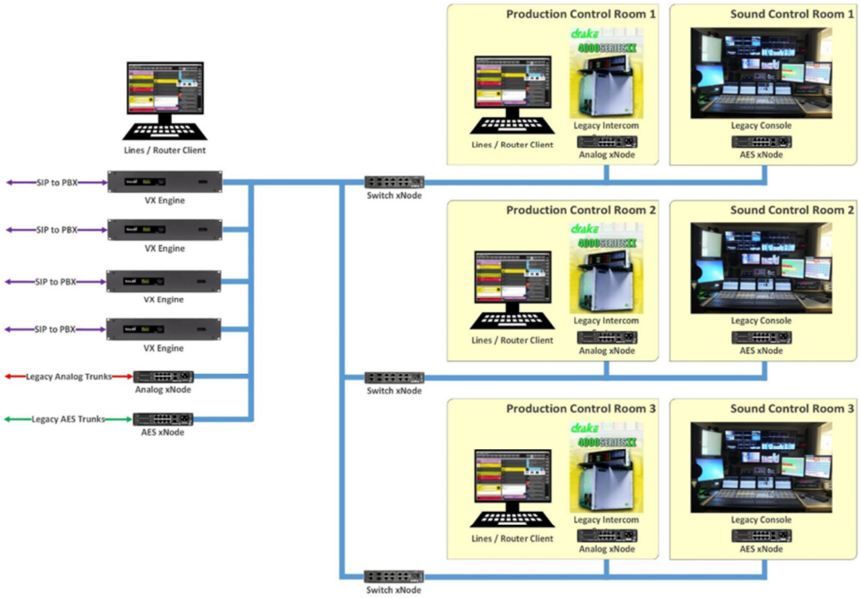


Interfacing audio consoles and intercom systems to telephones has not changed since Telos invented the digital telephone hybrid and began selling them in 1985.

Telos began the AoIP revolution with Livewire in 2002 and royalty free licensing of the technology in 2003.

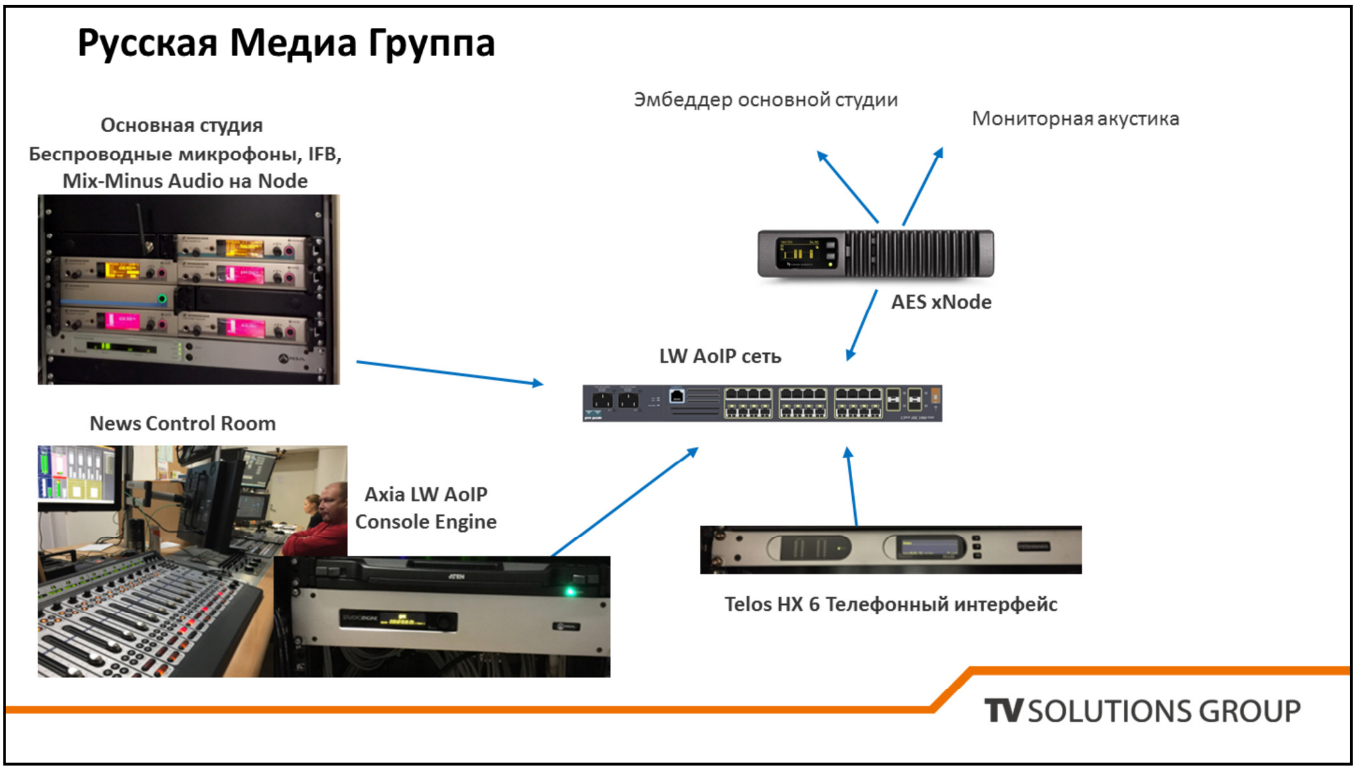
The VX combines the latest digital hybrid technology with state of the art telephony and Livewire+ AES67 AoIP.

Система телефонии – ITN UK



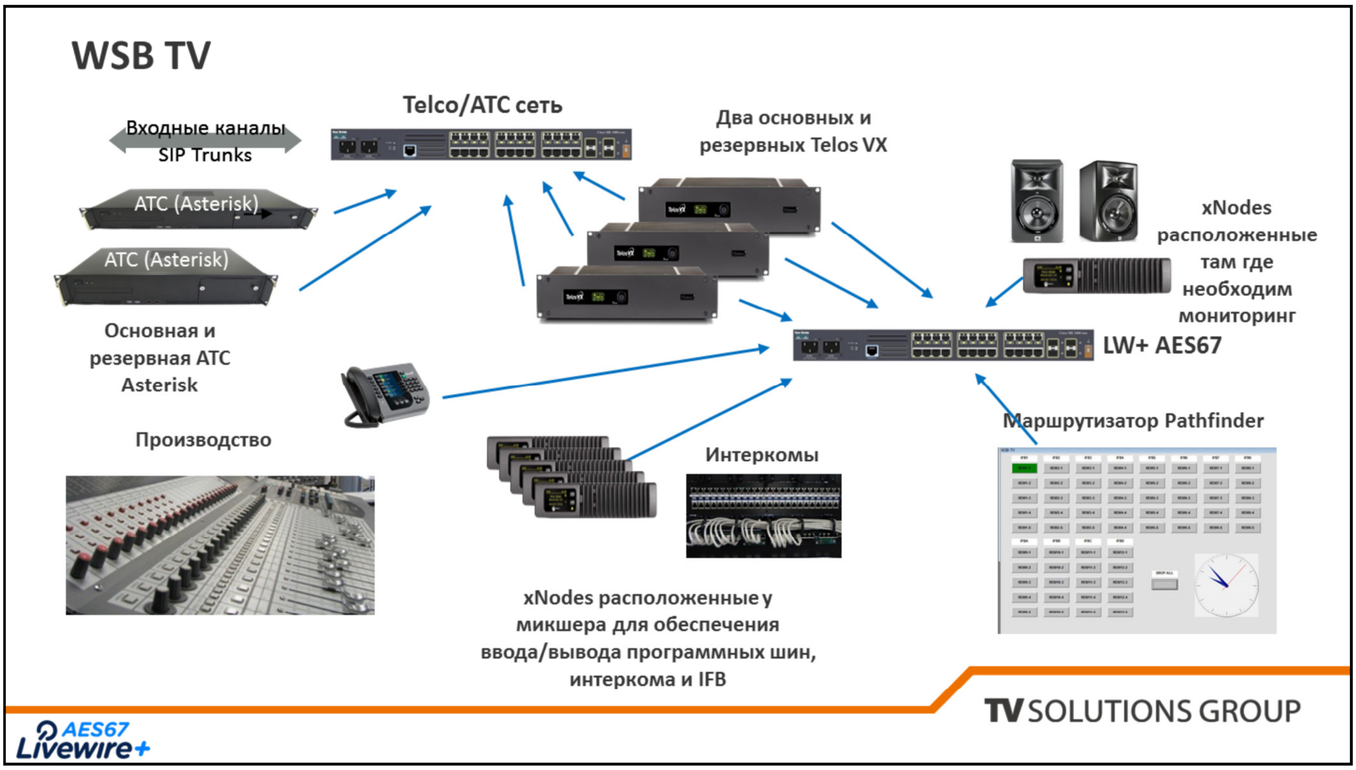
At ITN in London there are 5 control rooms and associated studio spaces. These are leased out to a variety of users, some long term and some for a single show or production. A way to share facility resources among the 5 control rooms was needed. Allocation of resources like production quality telephone services, intercom and audio devices had to be easy to assign and change at a moments notice.

3 of the 5 studios are shown here. LW+ AES67 was used to link all the control rooms to the various legacy intercoms, analog and AES digital audio devices and to Telos VX telephone systems. Telos angineers custom configured a LW+ AES67 AoIP control system and the 5 control rooms now share resources. It is quick and easy to make changes to accommodate different client and show requirements.



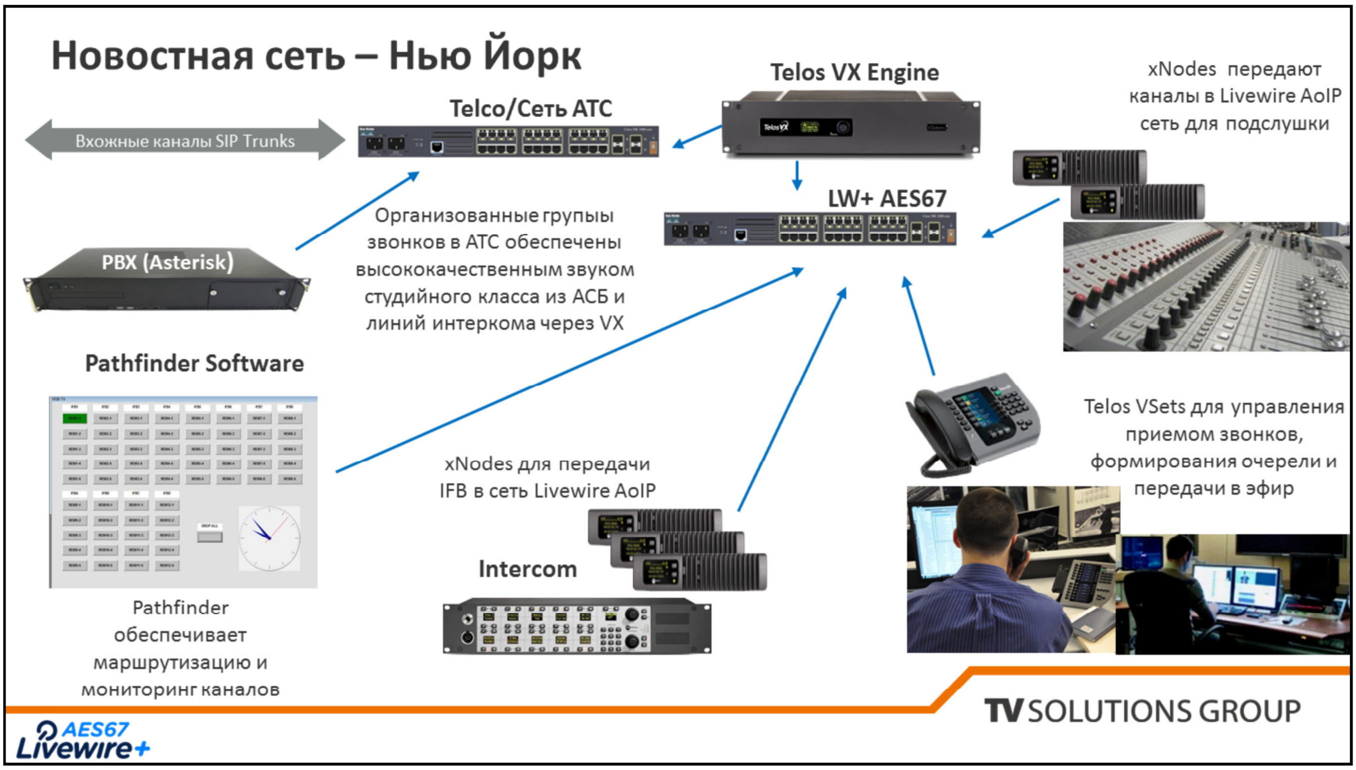
Here in Moscow Russian Media Group has several radio stations and a TV station co-located. Seeing the success the radio stations had moving to a Livewire AoIP infrastructure the TV station decided to use this technology to its advantage. This was done several years ago before AES67 or SMPTE 2110. The selection was based on the system's potential for savings, ease of use, and features.

An Axia AoIP console engine and control surface were installed in the news control room. In the news studio wireless mics, wireless IFB, and studio monitor speakers were all connected to the AoIP netowrk by an Axia xNode. An Axia Intercom, natively using LW AoIP was installed. A Telos multiline studio phone system, with digital telephone hybrids and LW AoIP I/O was used. Another xNode delivered new studio audio to the SDII audio embedder and also provides audio outputs for audio monitoring. All of this audio I/O is interconnected with just a few Cat5/6 Ethernet cables.

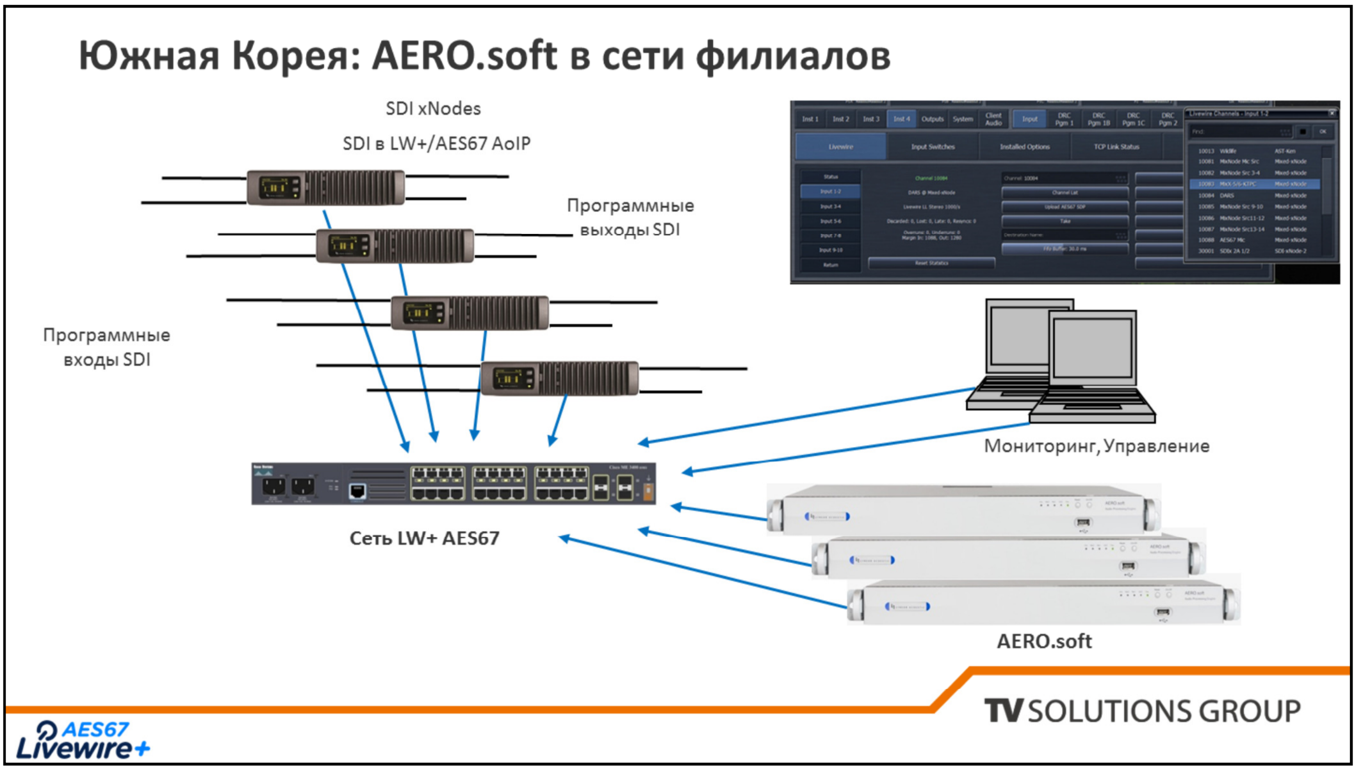


In Atlanta, in the US, WSB television was looking for a better way make studio quality telephone calls available to their various news studios, edit rooms and reporters and IFB for crews that were on location.

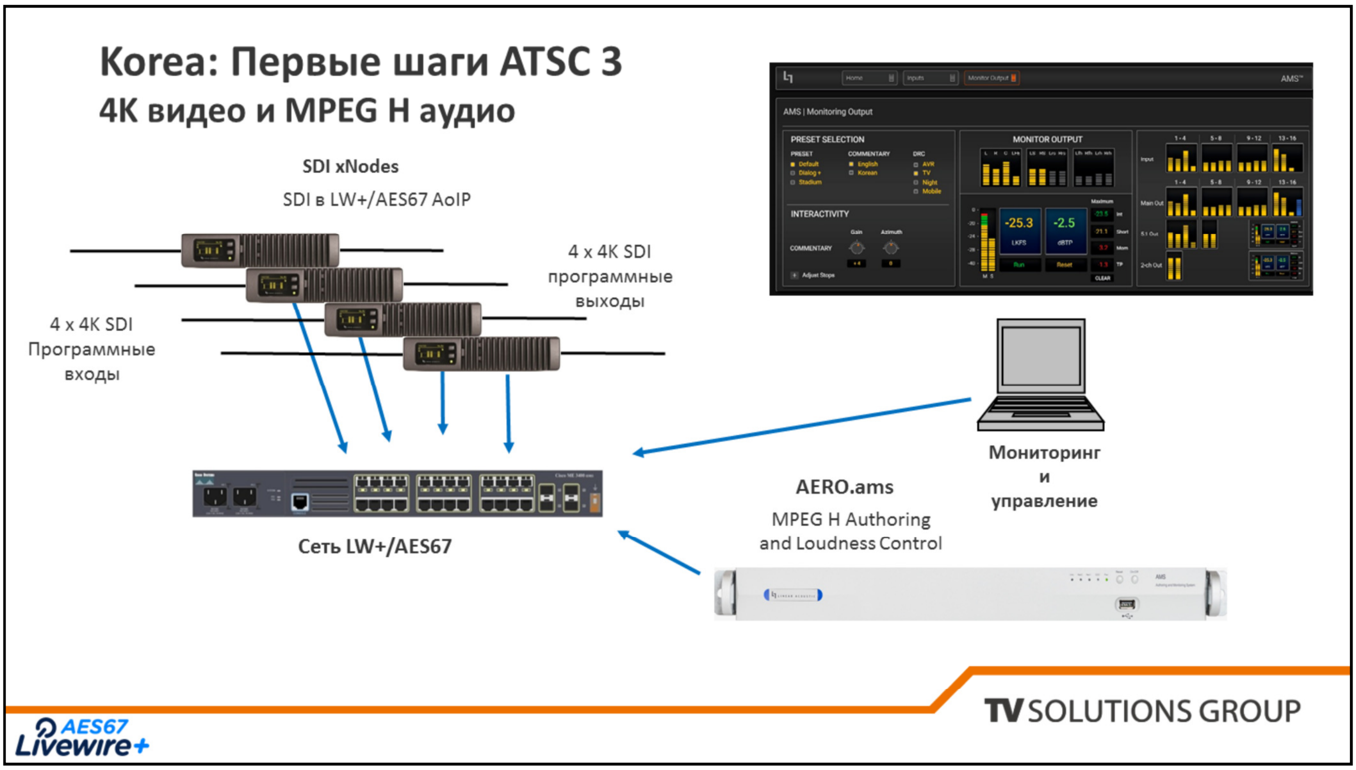
They replaced racks of telephone couplers, telephone hybrids and patch panels with Telos VX and a Livewire+ AES67 infrastructure. Control rooms, edit bays and desktops share the digital hybrids and phone coupler capabilities of the VX and also gained control over IFB using Axia Pathfinder software.



In an installation very similar to WSB one of the major news networks and their Washington, DC bureau, installed VX interfaces, xNodes and Axia intercom connected by a Livewire+ AES67 AoIP network.

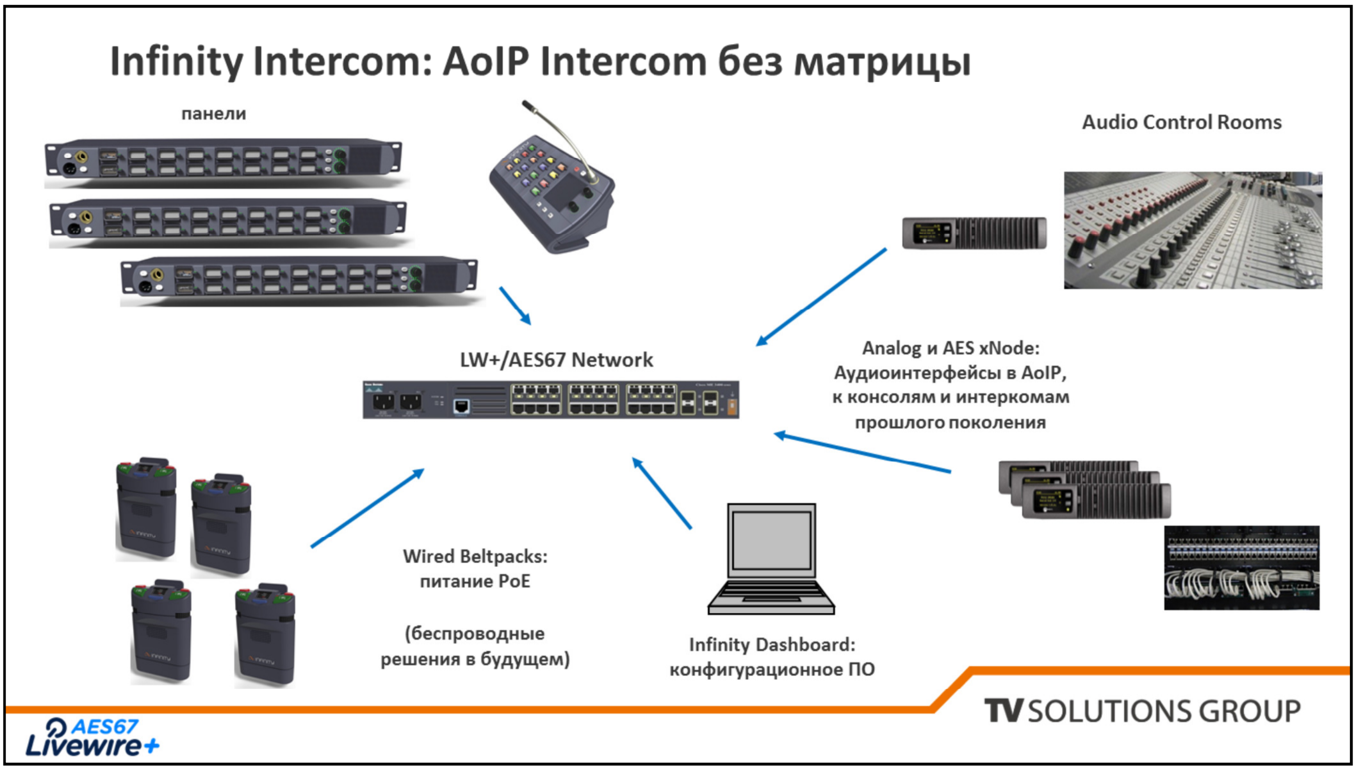


In the past few years several broadcasters in Korea have installed AERO.soft loudness control processors in broadcast television distribution facilities. Using a Livewire+ AES67 AoIP infrastructure, SDI xNodes de-embed and re-embed audio from and to SDI streams. Audio loudness control and processing is done in AERO.soft which uses LW+ AES67 I/O.



In our final example broadcasters in Korea have agreed to be early adopters of ATSC 3 even though the standard has not yet been completed. Several broadcasters have already begun broadcasting ATSC 3 4K UHD signals.

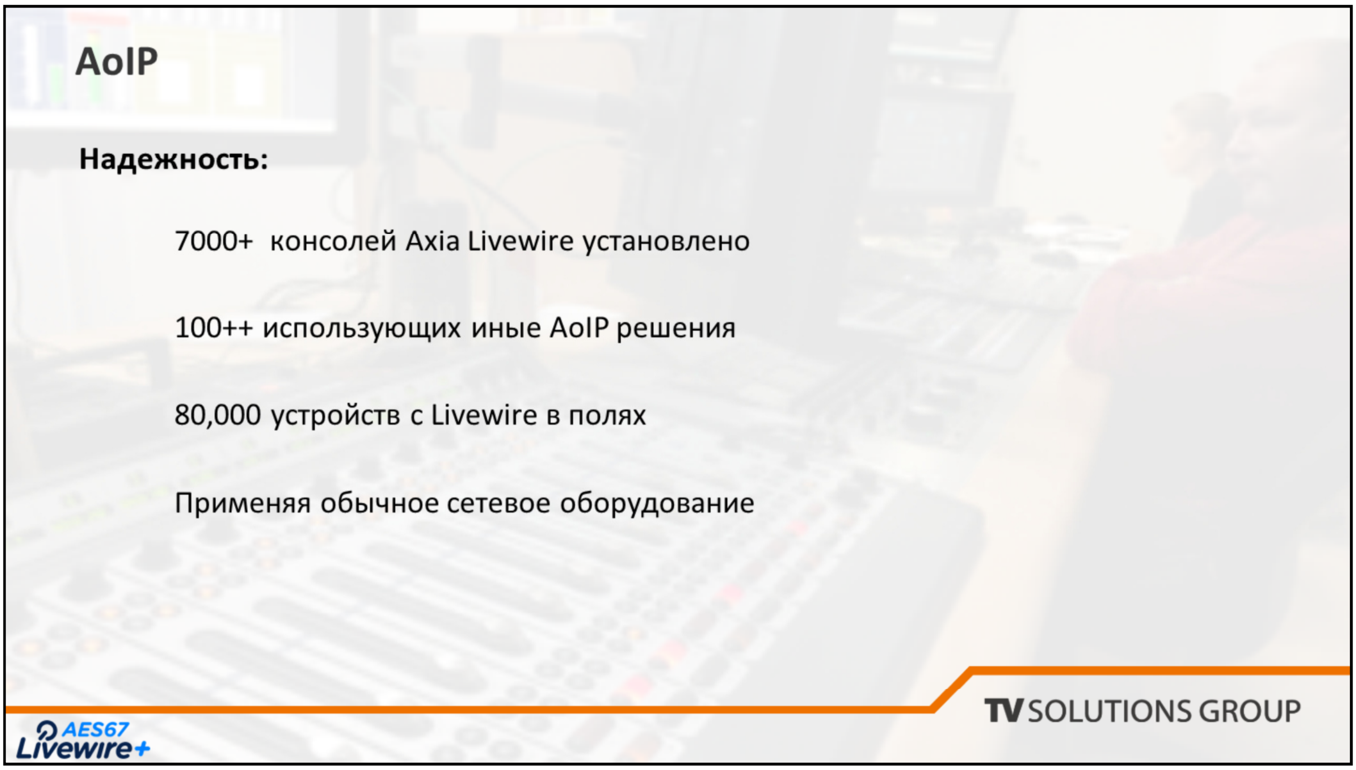
Under test by some Korean broadcasters is the Linear Acoustic AMS MPEG H authoring system. Prototype SDI xNodes de-embed and re-embed audio and metadata in the primary 4K quadrant's SDI stream. The AMS provides authoring and outputs the metadata required for receivers to correctly interpret the audio channels and provide viewers with the selections and mixes desired or needed.



The Infinity Intercom has all of the intelligence of the intercom system within each panel and beltpack. The system can expand to the limits of IP multicast addresses. (32,000 endpoints)

Every time a panel, beltback or port interface is added to Matrix based intercom the resources of the intercom are reduced. At a point the matrix' resources are totally used up. Using Infinity, each time an Infinity panel or Infinity beltpack is added to the system the resources expand.

The audio is 48KHz sampled and has a 24 bit depth. The Infinity intercom system is program quality audio.



The use of AoIP connectivity continues to grow. Not only The Telos Alliance but other manufacturers are also investing in and installing broadcast systems that use AoIP. The AES67 standard now provides interoperability between manufacturers whose AoIP interfaces are AES67 compliant or compatible.

Спасибо за внимание

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