

# Monitoring Audio-over-IP Audio

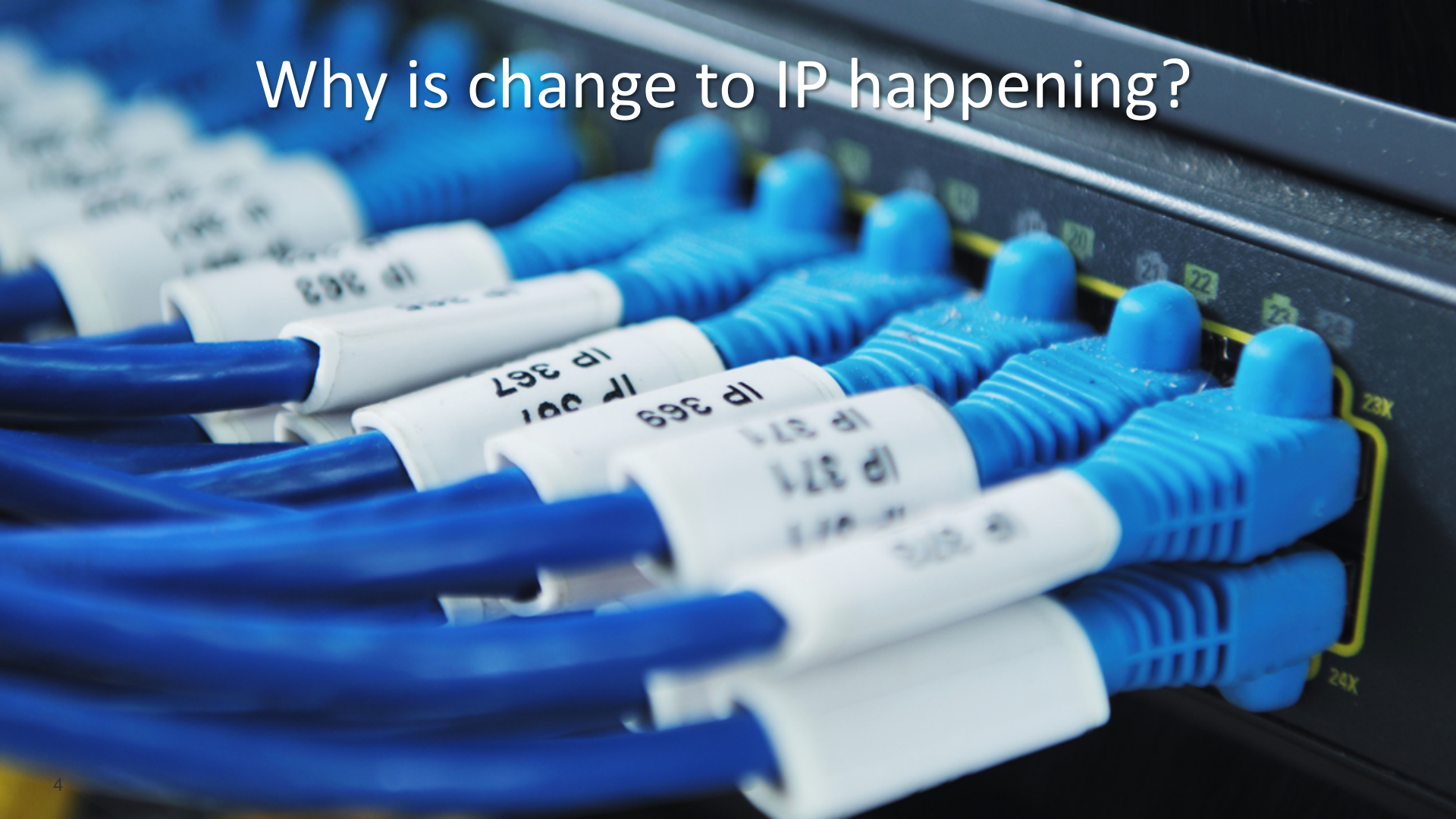
Aki Mäkivirta  
2019-10-16...19

**GENELEC®**

# Overview

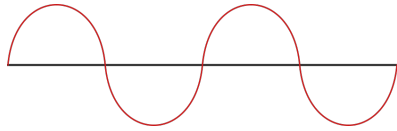
- Why is change to audio-over-IP happening?
- Audio-over-IP networking standards
- How does audio-over-IP work?
- PoE and single cable connectivity
- Setting up networked audio monitoring
- Case examples
  - Broadcast case examples
  - AV install case examples

Why is change to IP happening?



# Traditional connectivity of devices

# Traditional signal types, analogue

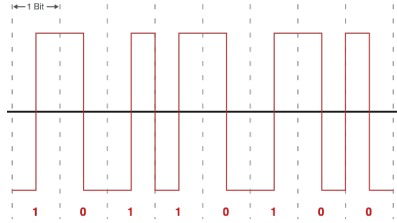


Analogue

Continuous  
voltage  
describes the  
audio signal

- one-to-one connectivity
- one-way signal
- one cable (two wires) per signal
- → N signals, N x 2 wires (balanced connections)
- limited dynamic range (max-noise)

# Traditional signal types, digital PCM

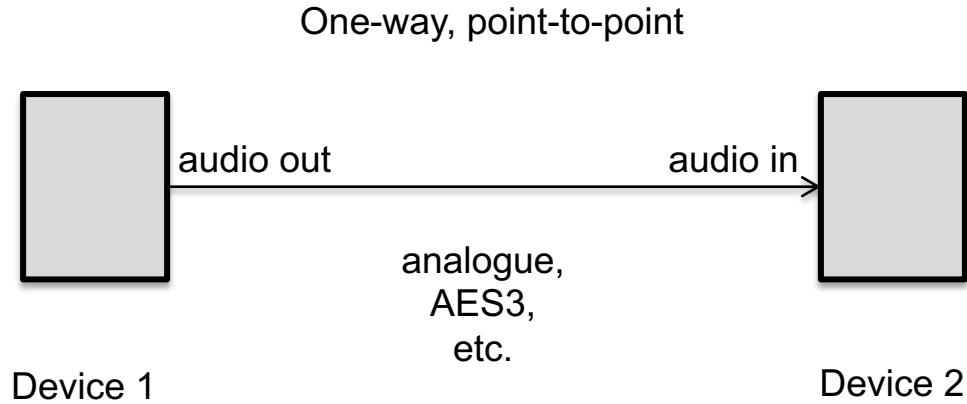


Digital  
PCM STREAM

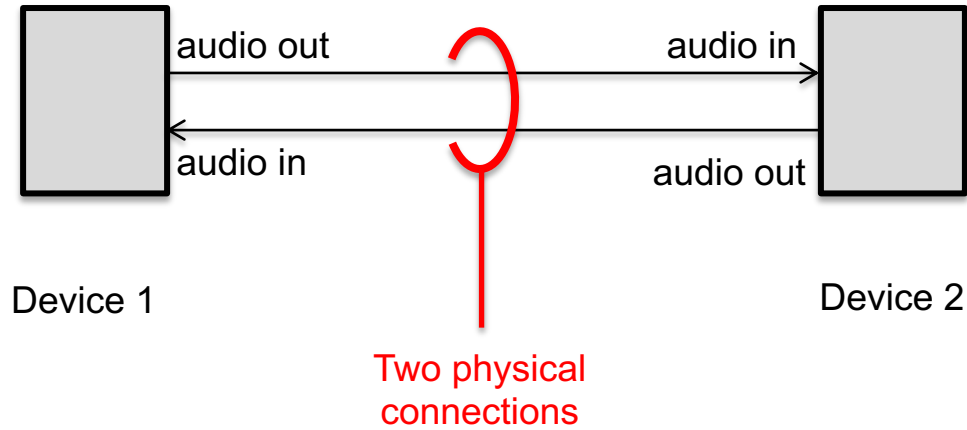
Sample values  
describes the  
audio signal

- one-to-one connectivity
- one-way signal
- one cable (two wires) per N signals (can be multiplexed)
  - max bit rate limits capacity of one cable
- flexible dynamic range (max-noise)
  - fixed point = fixed dynamic range
  - floating point

# Traditional interconnection

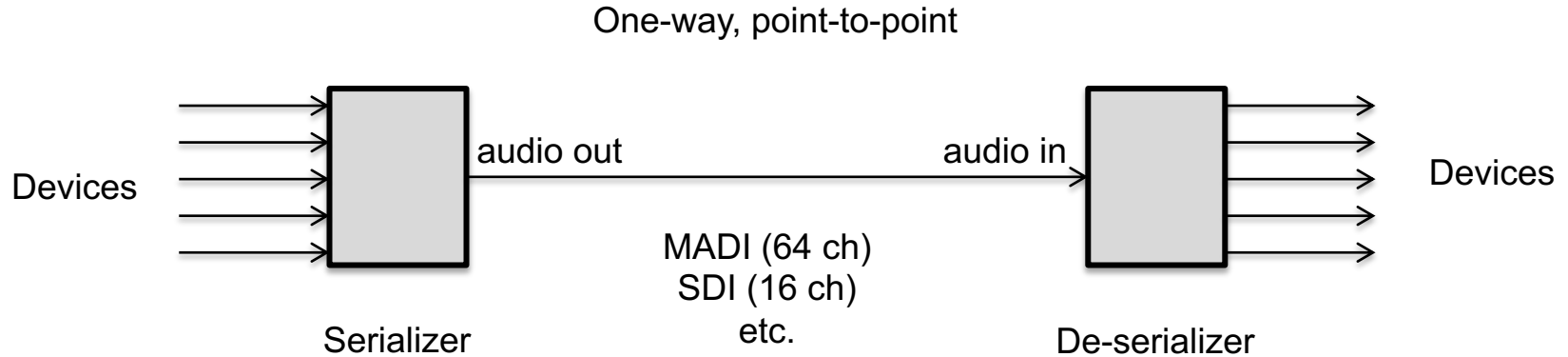


# Traditional two-way interconnection

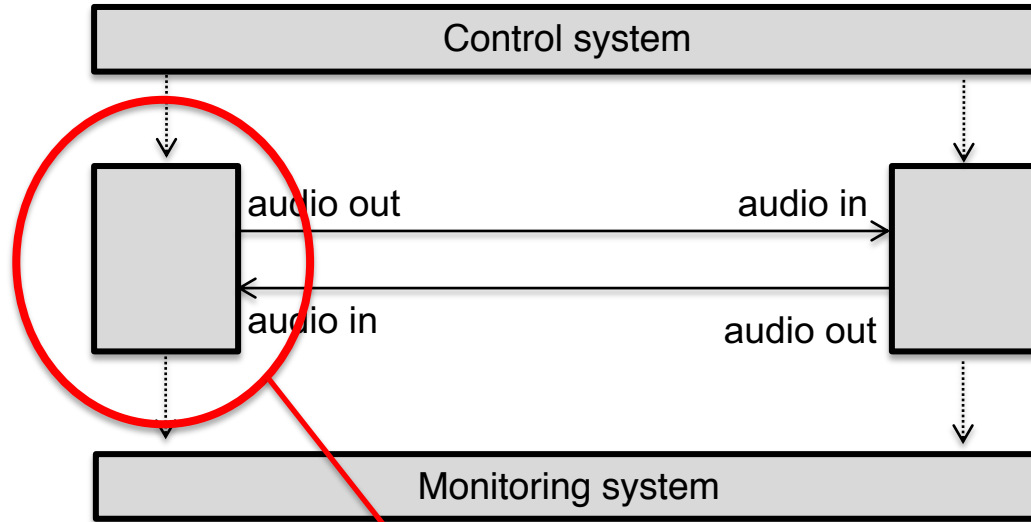




# Digital interconnection allows serialization



# Two-way interconnection with control and monitoring



Several, separate  
interfaces are needed to  
create a complete system

# Connectivity, good old days



Batch bay, one-to-one physical links between devices in studio

- Expensive
- Custom build
- Cabling requires experts to build and maintain/service

# Connectivity on IP networks

**OLD**



**NEW**



analog, AES3, MADI  
and SDI solutions



dedicated, special  
technology for  
audio only

IT technology



standard technology:  
IP switch infrastructure,  
standard cables, low cost

analog, AES3, MADI  
and SDI solutions



limited technical life time,  
new technology generations  
outdate previous systems

IT technology



IP technology has a long  
lifetime and good backwards  
compatibility, technology does  
not become obsolete suddenly

analog, AES3, MADI  
and SDI solutions



interface has hard limits:  
format, speed, channel count,  
etc.

IT technology



- IP network speeds continuously increase;
- **network is agnostic**
- no fundamental limitations to formats, channel counts, precision, clock rates, etc.



analog, AES3, MADI  
and SDI solutions



basic balanced line connection  
is (fairly) standard

interface conventions can be  
limited to one vendor

IT technology



standard protocols are recognized  
by all devices, standardization  
improves cross-vendor compatibility

audio-over-IP is (being)  
standardized by AES (AES67) and  
SMPTE (ST2110)

analog, AES3, MADI  
and SDI solutions



expensive to design and build,  
needs specialists to maintain

IT technology



low-cost to build and maintain:  
systems use the same backbone  
technology

analog, AES3, MADI  
and SDI solutions



control, monitoring and  
management need a separate  
network/connectivity/system

IT technology



same IP network works for all  
functions, media (audio, video) +  
monitoring + management + all  
else

# Options for audio-over-IP connectivity

Multichannel HW  
interfaces and converters



Adapter w/  
standard  
loudspeakers



Audio-over-IP  
connectable  
loudspeakers  
(two-cable  
connectivity)



# Minimum audio monitoring system



Loudspeakers with audio-over-IP input



IP switch

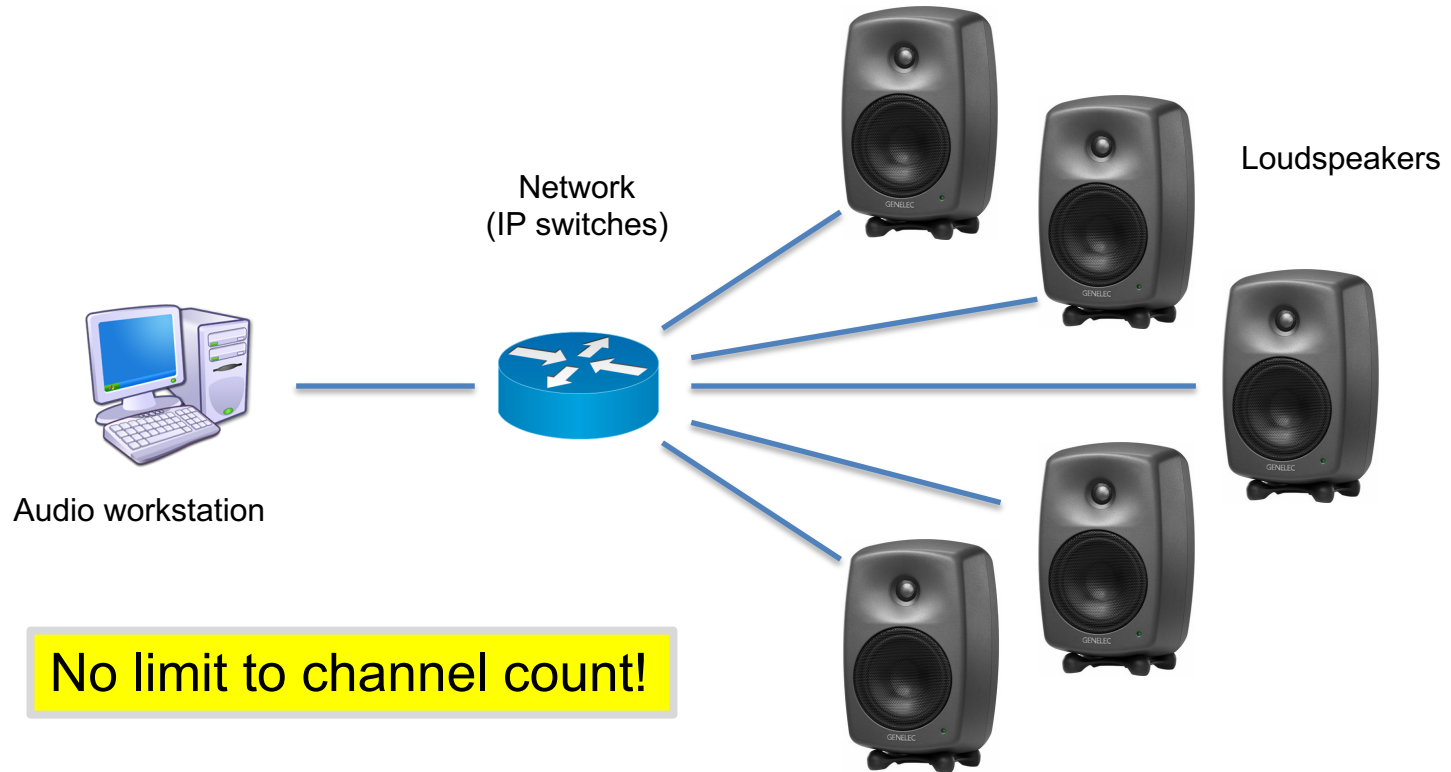


Computer for graphical user interface and possibly software soundcard



PTP clock source (HW audio inputs)

# Simple audio-over-IP monitoring network



**No limit to channel count!**

# Standards for Networking Audio

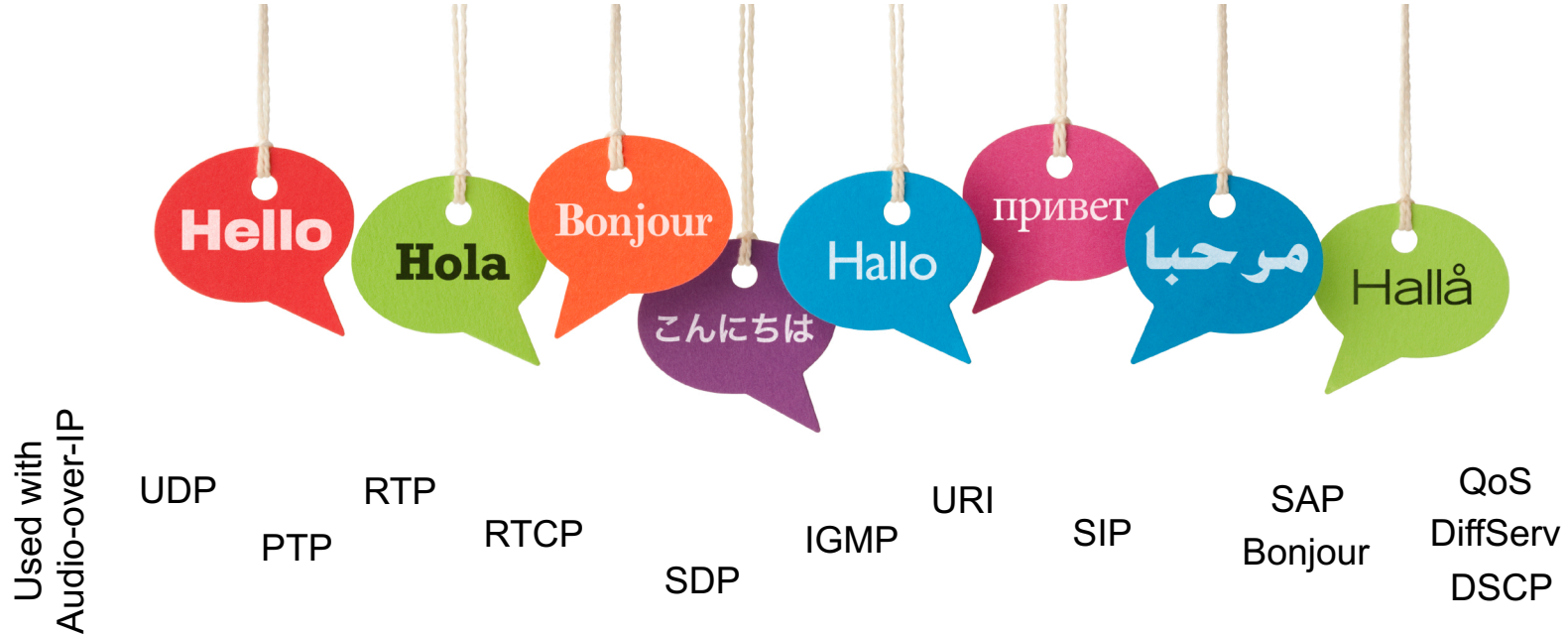


# IP protocols can create efficient communication





# IP protocols, the agreed methods of interacting




# Open System Interconnect (OSI) model

OSI Model				
Layer		Data unit	Function	Examples
<b>Host layers</b>	7. Application	Data		HTTP, FTP, SMTP, SSH
	6. Presentation		Translation of data between a networking service and an application;	HTML, CSS
	5. Session		Managing communication sessions, i.e. continuous exchange of information	RPC, PAP, SSL, SQL
	4. Transport	Segments/ Datagram	Reliable transmission of data segments between points on a network,	TCP, UDP
<b>Media layers</b>	3. Network	Packet	addressing, routing and traffic control	IPv4, IPv6, IPsec, ICMP
	2. Data link	Frame	Reliable transmission of data frames	MAC
	1. Physical	Bit	Transmission and reception of raw bit streams	Ethernet PHY (physical layer)

Audio-over-IP

# Legacy Audio-over-IP technologies

Technology	Purveyor	Date introduced	Synchronization	Transport
Livewire	Telos/Axia	2003	Proprietary	RTP
Wheatnet-IP	Wheatstone	2005	Proprietary	RTP
Dante	Audinate	2006	IEEE 1588-2002	UDP
N/ACIP	EBU	2007	Adaptive (per stream)	RTP
Q-LAN	QSC Audio Products	2009	IEEE 1588-2002	UDP
 RAVENNA	ALC NetworX	2010	IEEE 1588-2008	RTP
AVB	IEEE, AVnu	2011	IEEE 802.1AS	Ethernet, RTP

# Some RAVENNA Partners

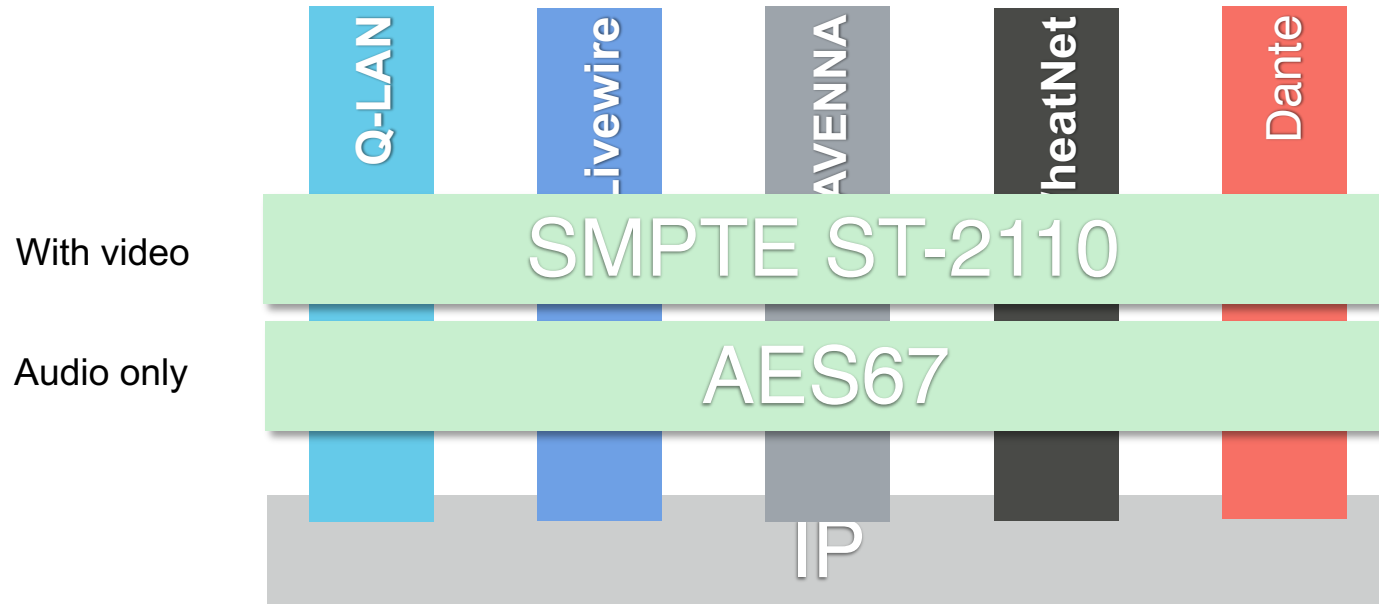


# Some Dante partners



Dante Domain Manager version 1.1. enables support for SMPTE 2110 and AES67 within Dante domains, allowing broadcasters to integrate SMTPE 2110/AES67 compliant devices in a managed Dante environment.

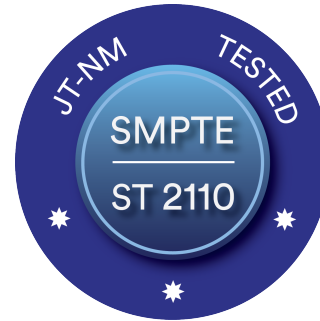
# AES67 – AES Standard for Interoperability in High-performance Audio-over-IP Streaming



# Standards: EBU and SMPTE



## ST 2110



The SMPTE ST 2110 Professional Media Over Managed IP Networks suite of standards is a major contributing factor in the movement toward one common internet protocol (IP)-based mechanism for the professional media industries.

Task Force on Networked Media (JT-NM) Open Interoperability

# What is AES67

**AES67** is a technical standard for audio over IP and audio over Ethernet interoperability. The standard was developed by the Audio Engineering Society.

<http://www.aes.org/publications/standards/search.cfm?docID=96>





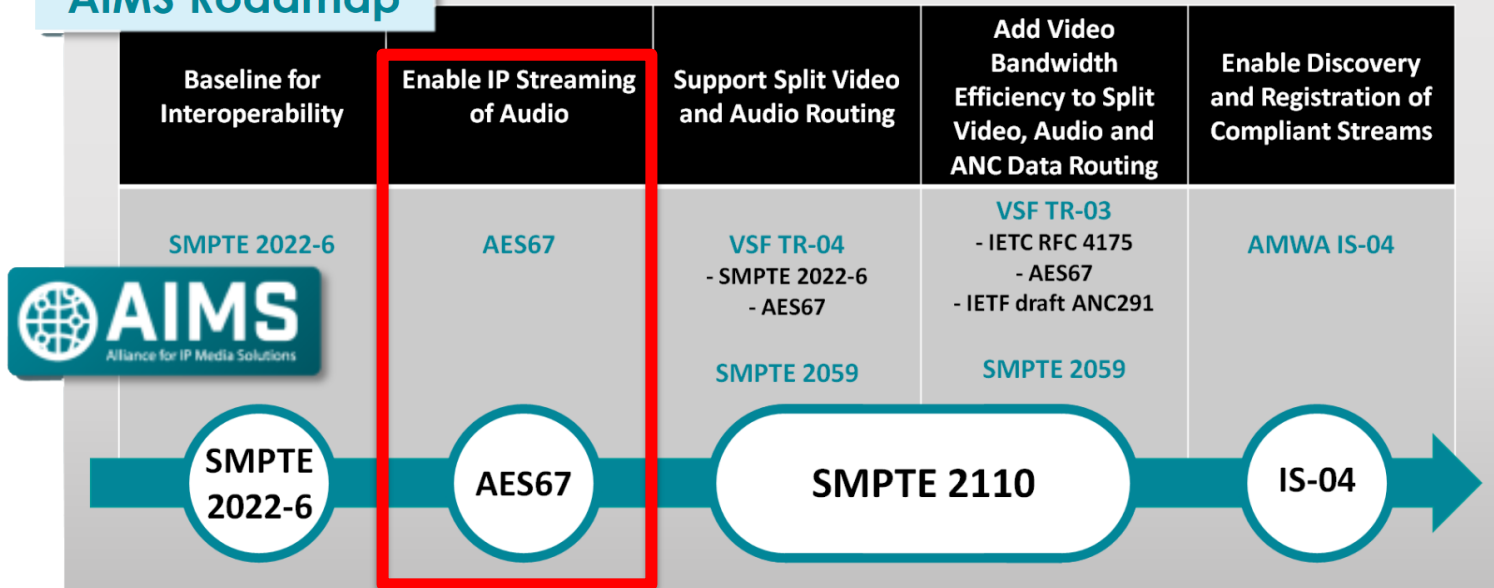
# What is SMPTE ST-2110

The SMPTE ST 2110 routes video, audio, and ancillary data streams separately to gain efficiency.

The standards suite is multipart. The following is the current status of documents included in the suite as of 10 September 2018:

- [SMPTE ST 2110-10/-20/-30](#) — addressing system concerns and uncompressed video and audio streams
- [SMPTE ST 2110-21](#) — specifying traffic shaping and delivery timing of the uncompressed video
- [SMPTE ST 2110-31](#) - specifies the real-time, RTP-based transport of AES3 signals over IP networks, referenced to a network reference clock
- [SMPTE ST 2110-40](#) — maps ancillary data packets (as defined in SMPTE ST 291-1) into Real-Time Transport Protocol (RTP) packets that are transported via User Data Protocol/Internet Protocol (UDP/IP) and enables those packets to be moved synchronously with associated video and audio essence streams

## AIMS Roadmap



“The Alliance for IP Media Solutions (AIMS), is a non-profit trade alliance that promotes the open standards that broadcast and media companies use to move from legacy SDI systems to a virtualized, IP-based future”

# AIMS alliance

<https://aimsalliance.org/>



The Alliance for IP Media Solutions (AIMS), is a non-profit trade alliance that fosters the adoption of one set of common, ubiquitous, standards-based protocols for interoperability over IP in the media and entertainment, and professional audio/video industries

# EBU position: Technology Pyramid

Minimum User Requirements to Build and  
Manage an IP-Based Media Facility

<https://tech.ebu.ch/pyramid>

# The Technology Pyramid for Media Nodes

Minimum User Requirements to Build and Manage an IP-Based Media Facility.



## Media Transport

- Single link video SMPTE ST 2110-20
- Software-friendly SMPTE ST 2110-21 Wide video receivers
- Universal, multichannel and low latency audio SMPTE ST 2110-30 Level C
- Stream protection with SMPTE ST 2022-7



EBU TECH 3371 - December 2018

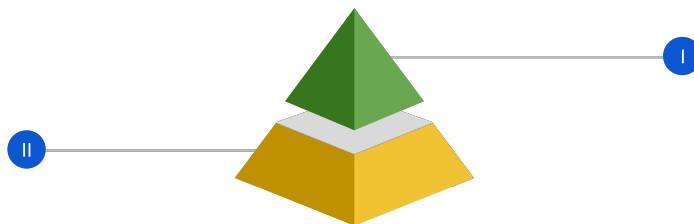


# The Technology Pyramid for Media Nodes

Minimum User Requirements to Build and Manage an IP-Based Media Facility.

## Time and Sync

- PTPv2 configurable within SMPTE and AES profiles
- Multi-interface PTP redundancy
- Synchronisation of audio, video and data essences



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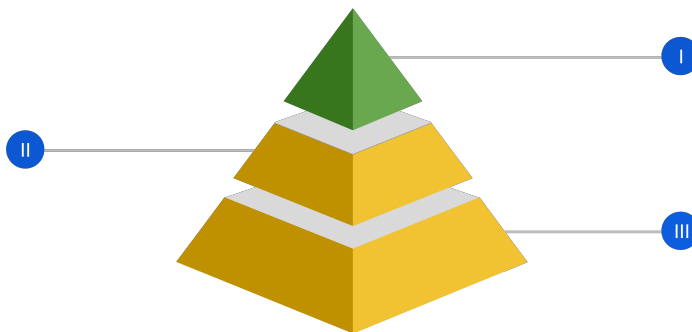


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## Discovery and Connection

- Discovery and Registration: AMWA IS-04
- Connection Management: AMWA IS-05
- Audio channel mapping: AMWA IS-08 (in dev.)
- Topology discovery: LLDP



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# The Technology Pyramid for Media Nodes

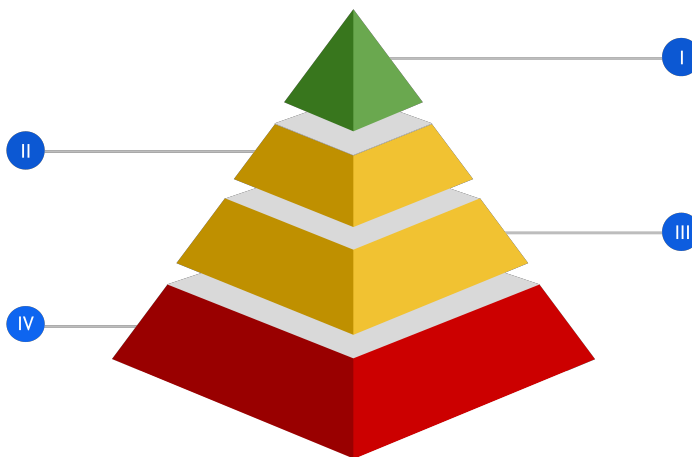
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## Configuration and Monitoring

- IP assignment: DHCP
- Open configuration management - e.g., API, config file, SSH CLI, etc.
- Open monitoring protocol - e.g., syslog, agent, SNMPv3, etc.



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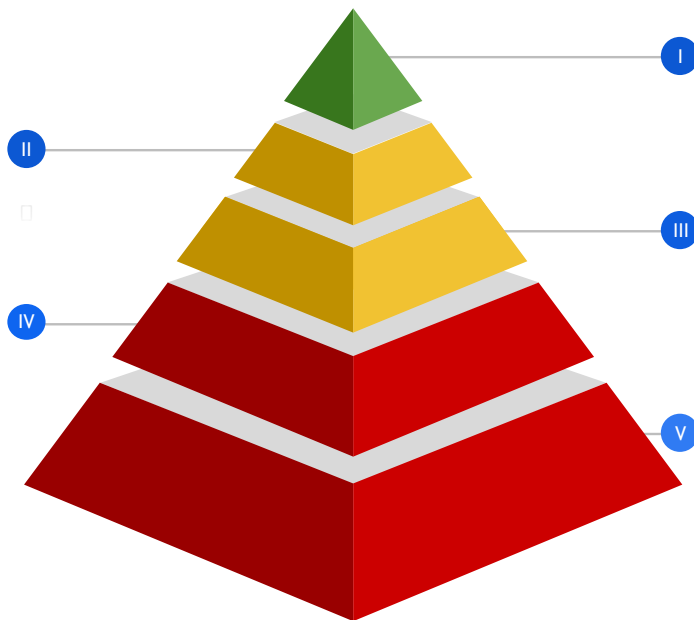
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## Security

- EBU R 148 Security Tests
- EBU R 143 Security Safeguards
- Secure HTTPS API calls



# AES 67 Covers a part of the EBU Full Stack

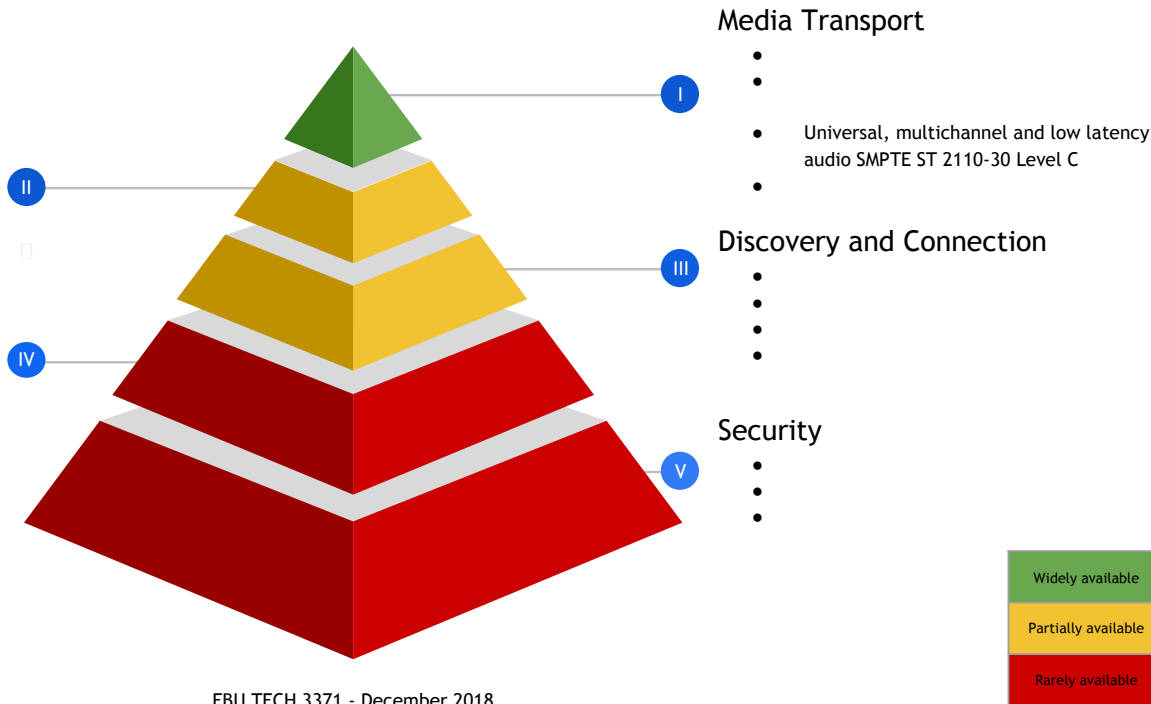


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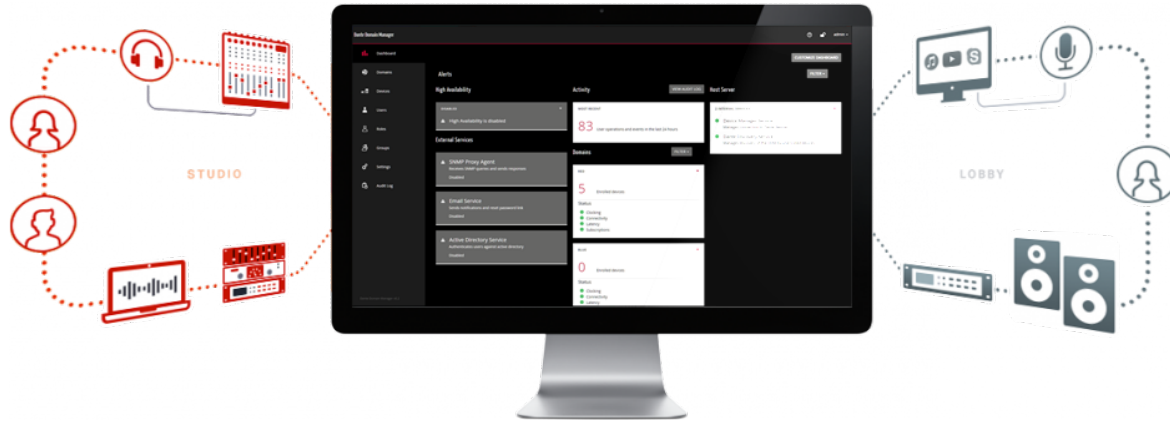
## Configuration and Monitoring

- IP assignment: DHCP
- 
- 



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# Examples of implementing ST2110 compliance

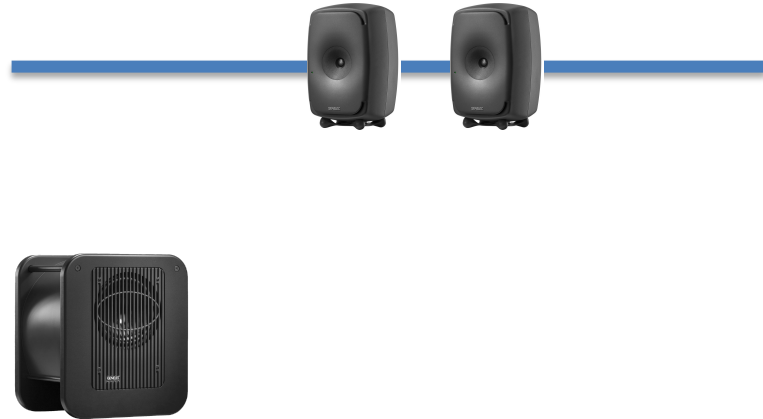


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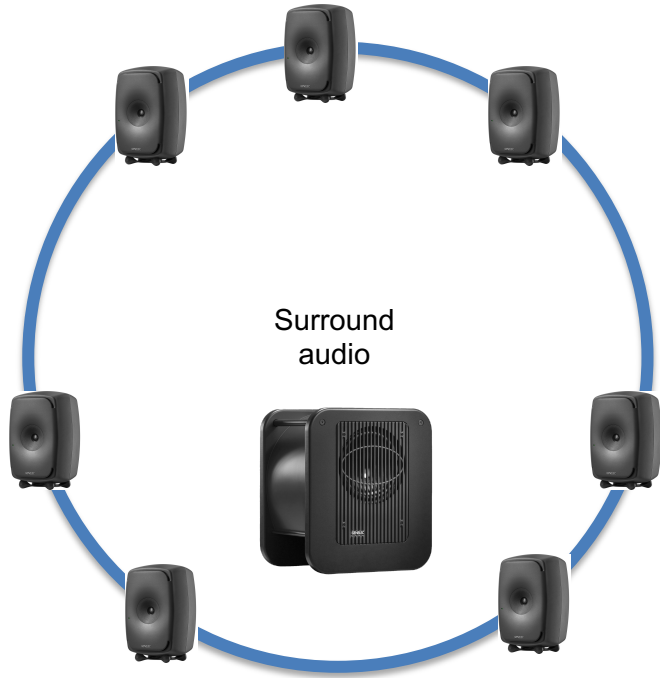
# Monitoring Networked Audio



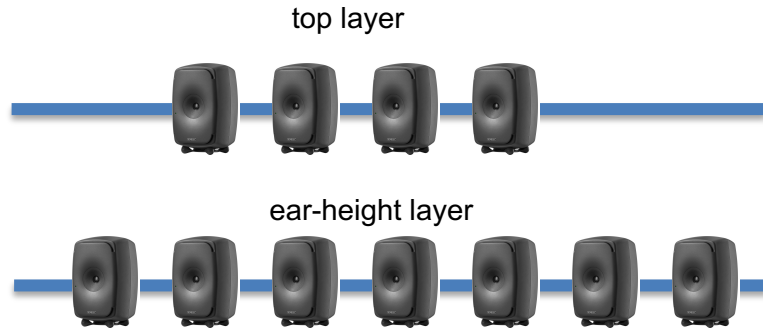
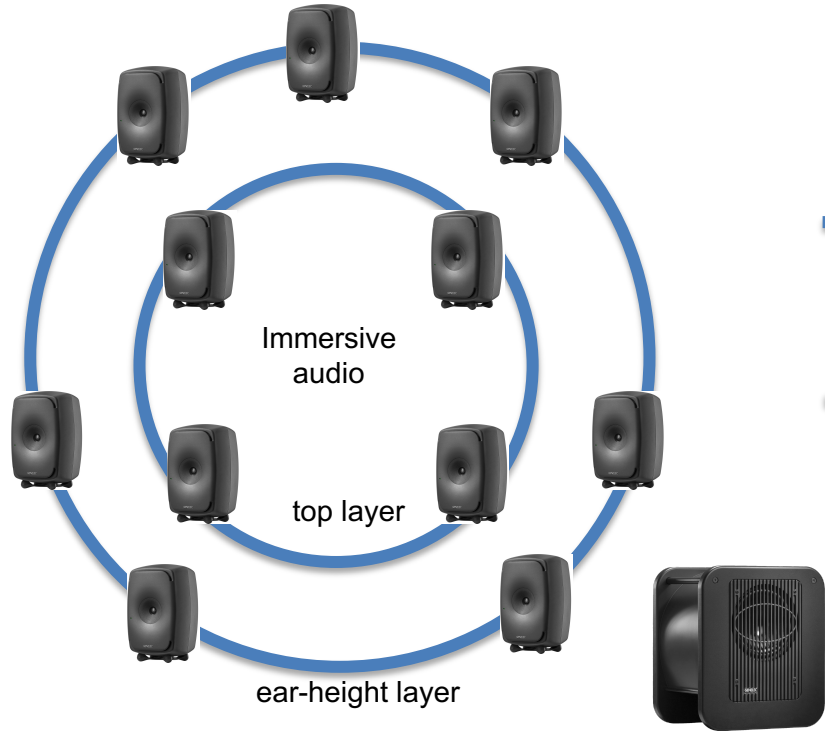
# Single-layer systems: stereo



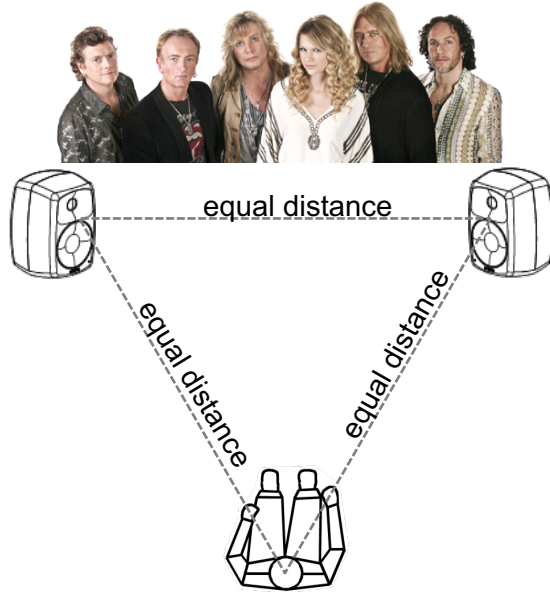
# Single-layer systems: surround (typically 5...7 ch)



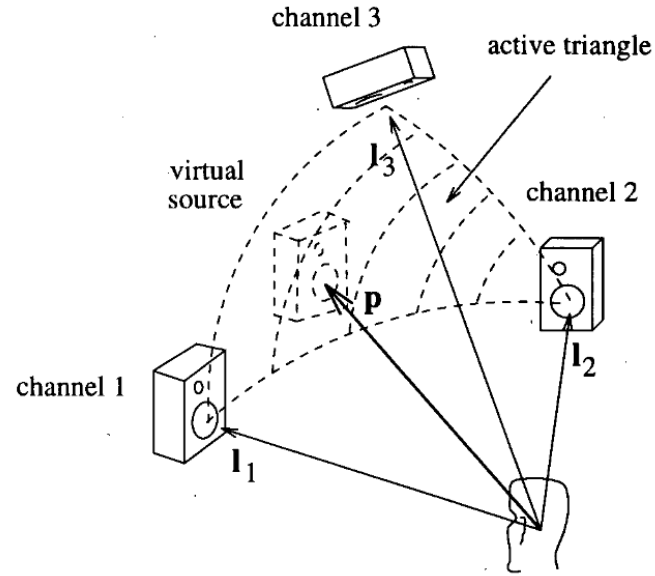
# Two-layer immersive systems



# Creating immersive sound stage



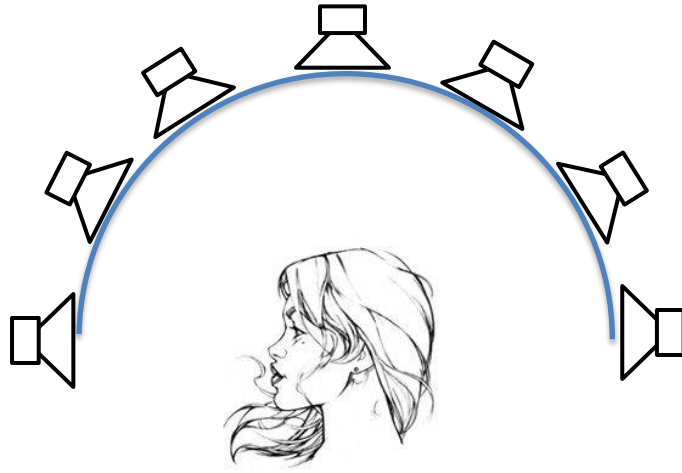
Single-layer virtual images can be created using level panning



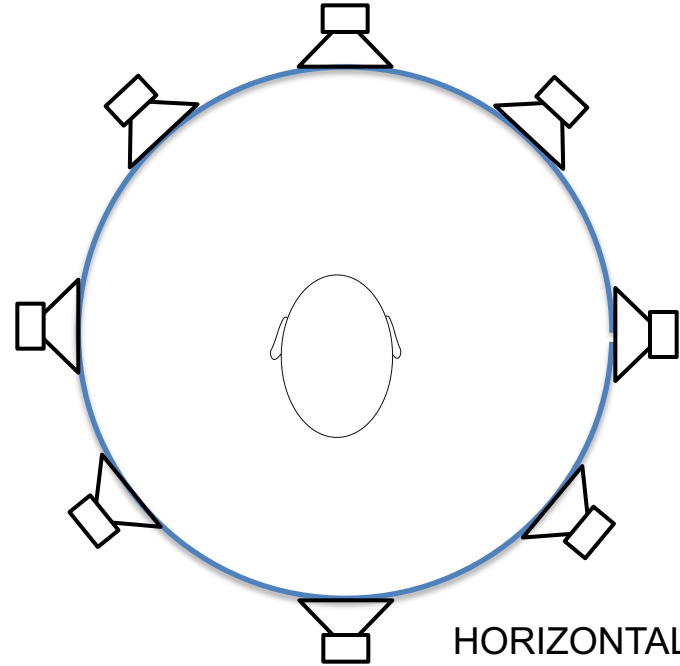
Interlayer virtual images created using Vector Base Amplitude Panning (VBAP)



# Equidelay, equidistance, and equal level at listening location

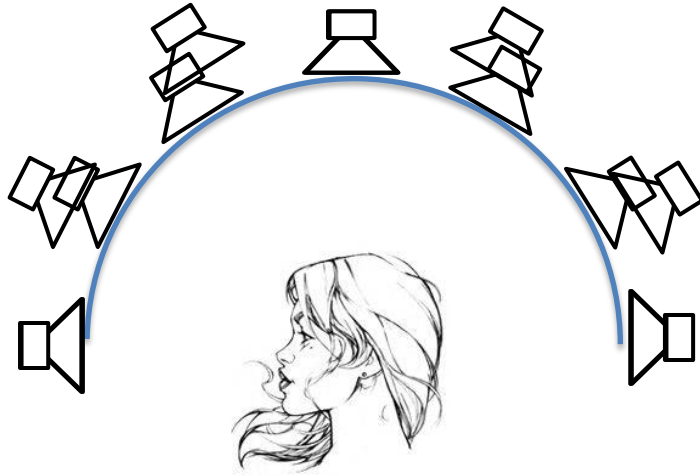


VERTICAL  
ORIENTATION

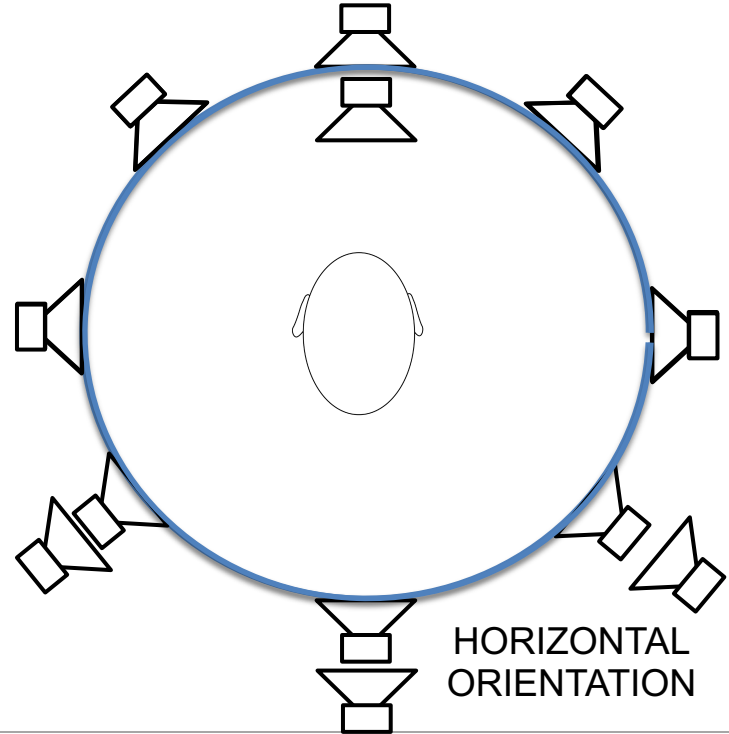


HORIZONTAL  
ORIENTATION

# Compensating for delay and level differences

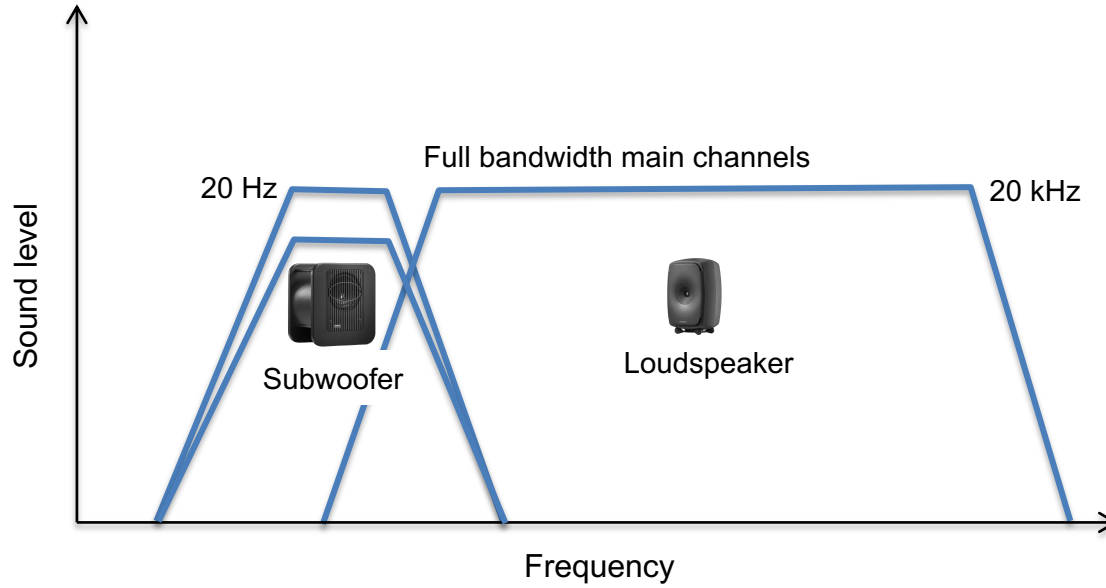


VERTICAL  
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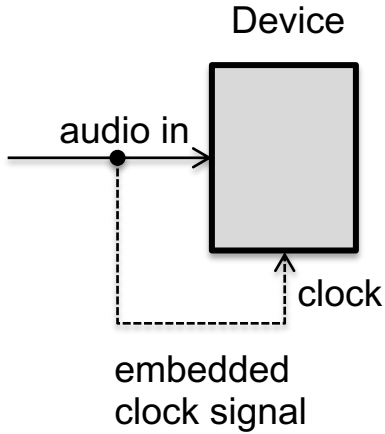
HORIZONTAL  
ORIENTATION

# Subwoofer alignment

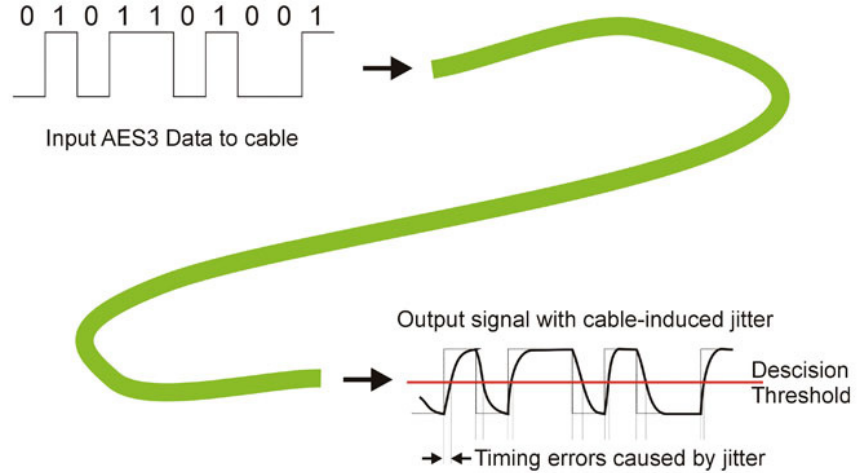


# Clock and time synchronization

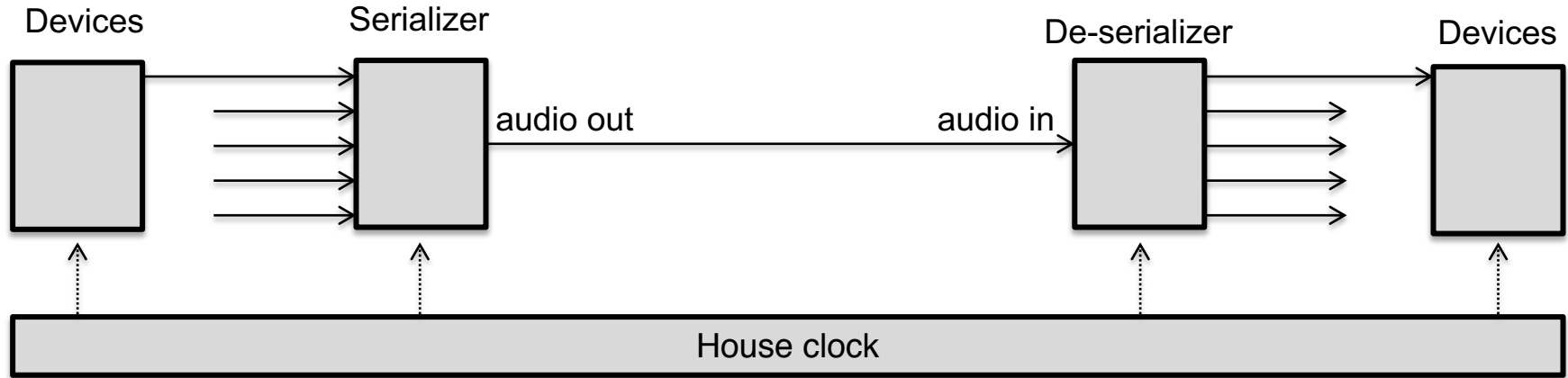
# Self-clocking digital interconnects can have jitter



AES3

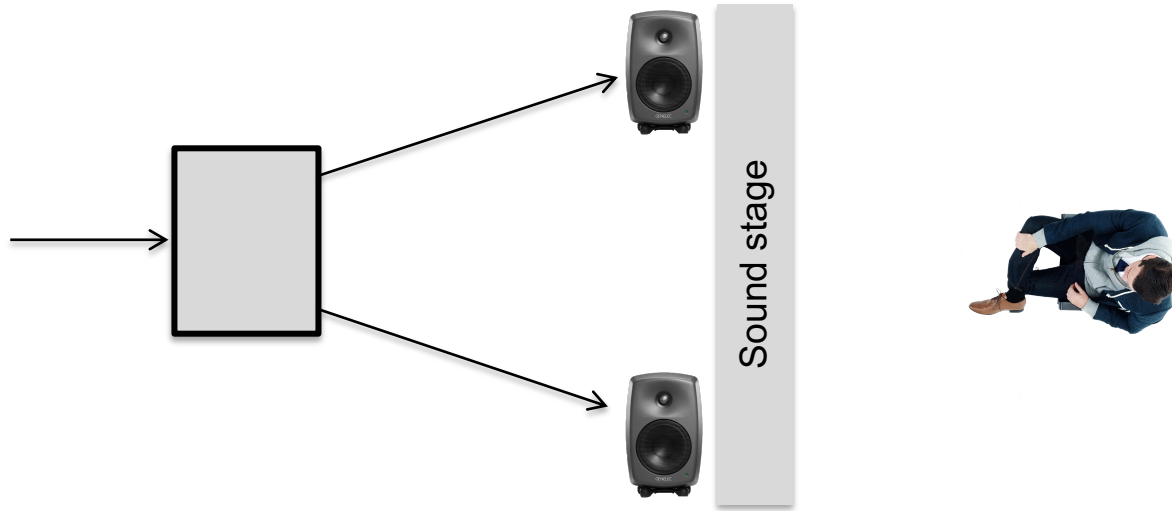


# Traditional digital interconnection uses a house clock



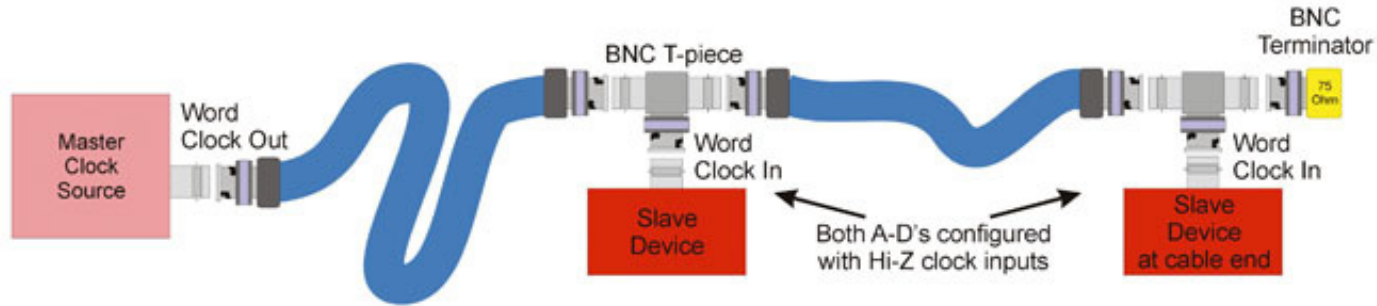
The House Clock ensures that all devices work at the same speed and that clock jitter is at minimum.

# Traditional loudspeaker interconnection



Time synchrony to form the stereo stage is guaranteed by the fact that all channels originate in one piece of equipment

# All devices see the same house clock signal

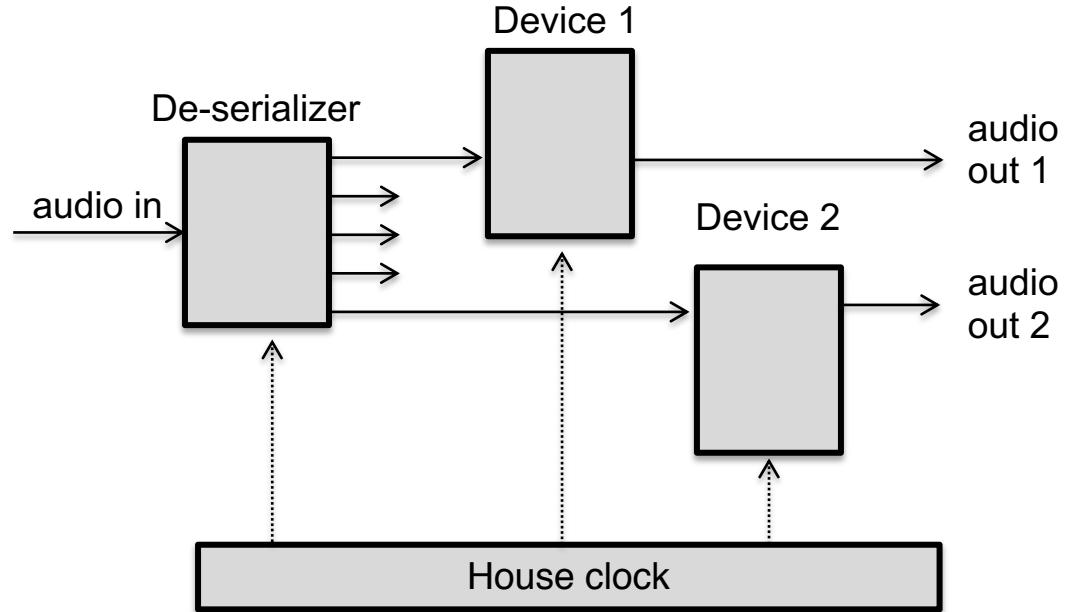


A typical way of distributing the House Clock signal.

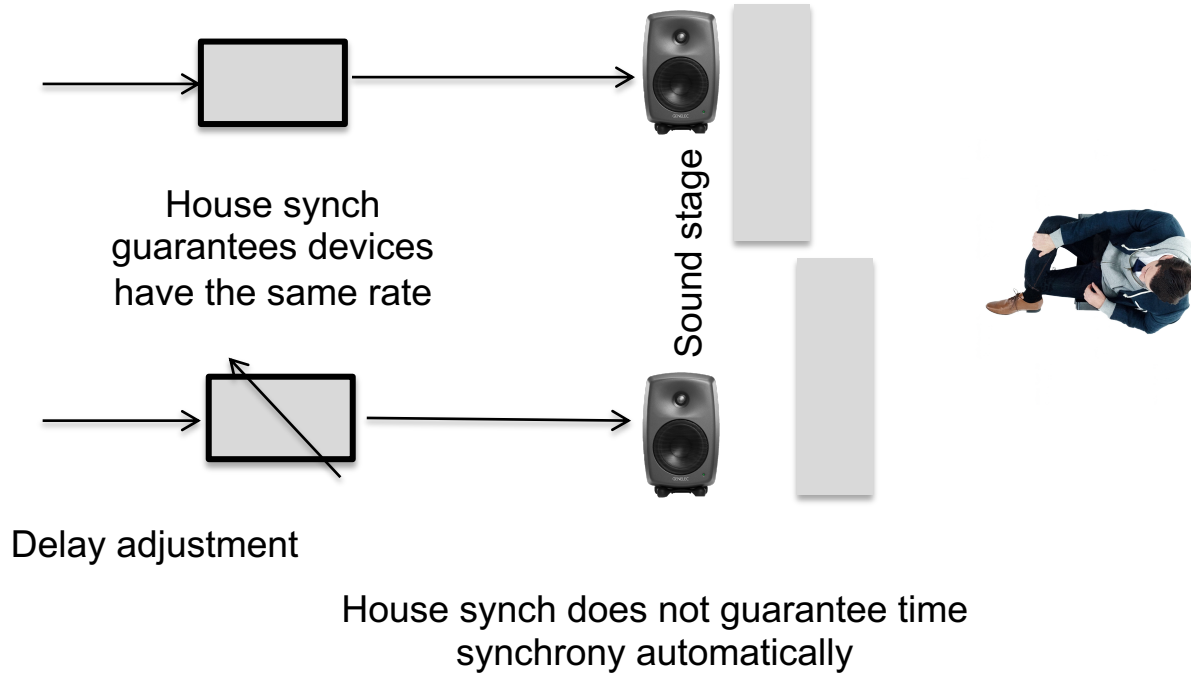


# House clock does not create absolute time reference

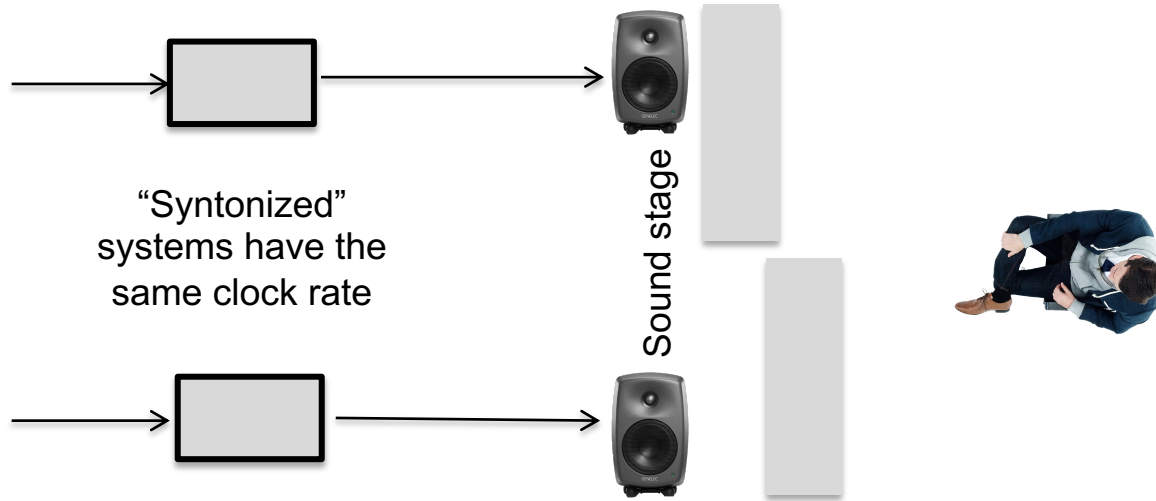
- Devices do not know about absolute time
- Device 1 and Device 2 can have different latency for audio output
- This is typically a fixed value, latency can be manually trimmed off in most cases
- House Clock does not guarantee time synchrony for outputs (but helps in achieving it)
- use of House Clock **assumes fixed timing in the signal transport**



# Achieving synch with house clock

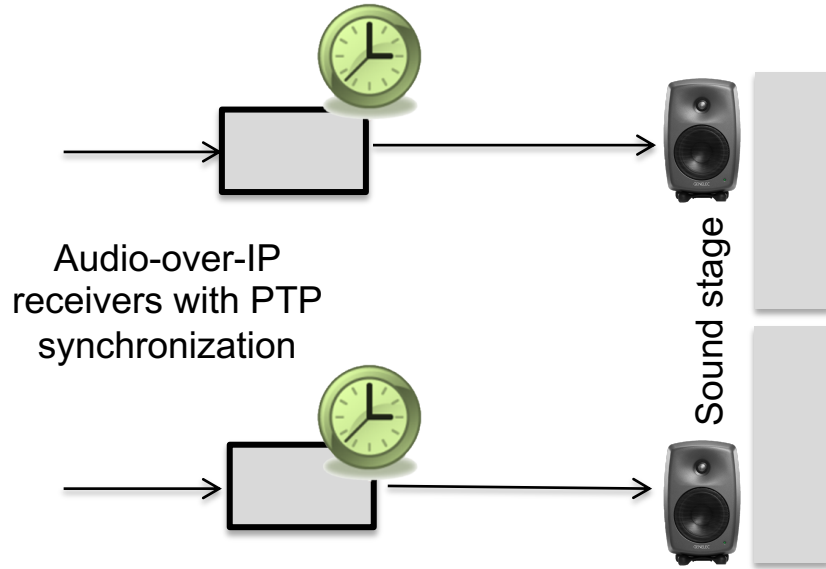


# Audio-over-IP loudspeaker interconnection



Syntonized clocks guarantee the same clock rate but no absolute time synchrony

# Audio-over-IP loudspeaker interconnection

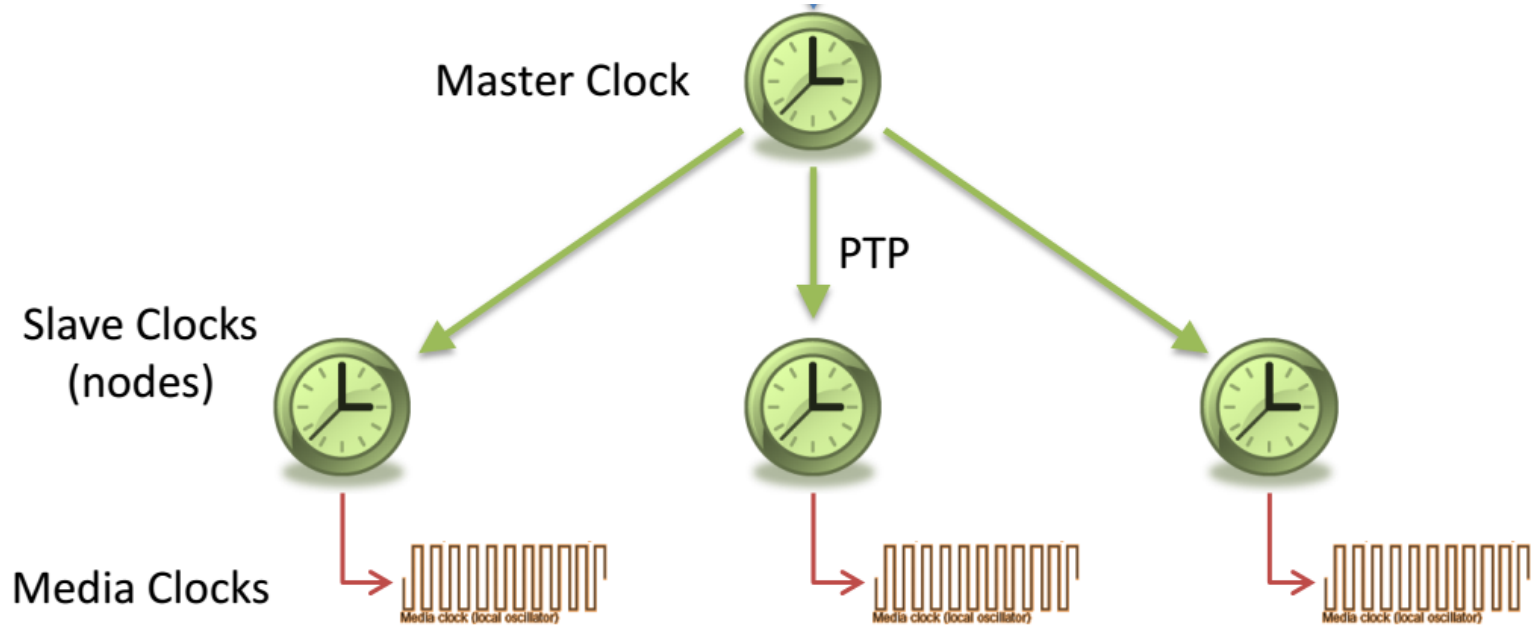


Time synchrony can be guaranteed using PTP  
clock synchronization protocol

# Time

- House clock sets the correct speed for clocks but does not represent absolute time
- Time concepts for audio-over-IP
  - Presentation time (link offset: ingress time and egress time, link offset)
  - Packet time
  - Jitter absorption buffer
- “Buffer too late” or “wrong PTS value”
- relativity of time
  - synchronized vs. syntonised (PTP standard, IEEE 1588)

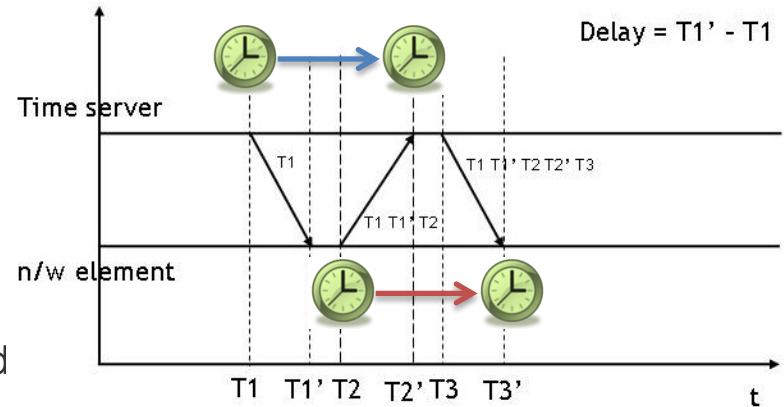
# Clock synchronization using IEEE 1588-2008 (PTP v2)



# AES67 – synchronizing devices

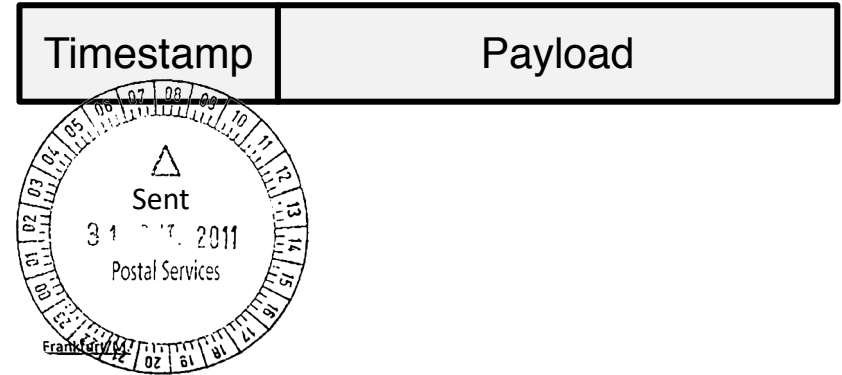
- The Precision Time Protocol (PTP) is a protocol used to synchronize clocks throughout a computer network
- PTP version 2 achieves clock accuracy in the sub-microsecond range
  - all devices synch to one device (called grandmaster)
  - timestamps in packets are corrected with the packet transport delay in network
- PTP was originally defined in the IEEE 1588 standard "Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control Systems"

Method of synchronizing clocks



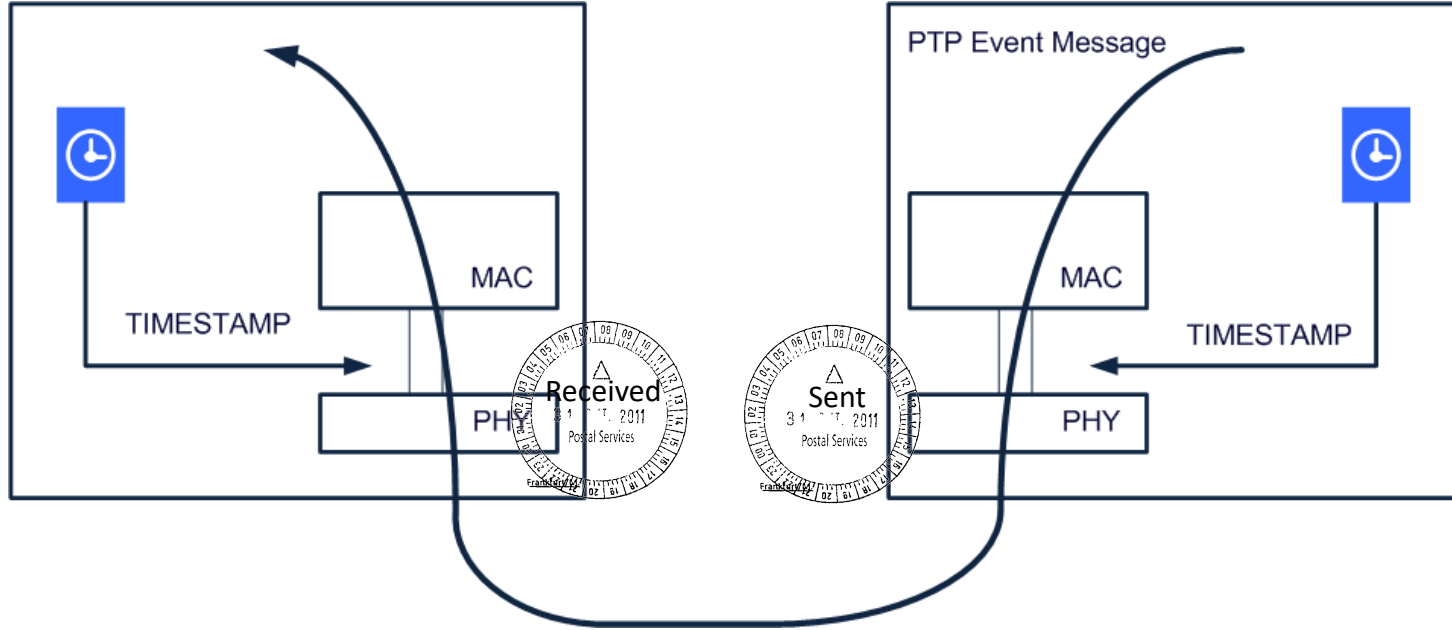
# Timestamp

- Each audio packet is timestamped
- The timestamp reflects the sampling instant of the first octet (i.e. start of data) in the RTP data packet (RFC 3550)
- *Truncated IEEE 1588v2 PTP timestamp format* uses 64 bits, including a 32-bit seconds field and a 32-bit nanoseconds field
- Timestamp is the **origination time** of the packet on the network
  - → timestamp is set by the sender
- Presentation time = origination time + link offset





# Timestamp

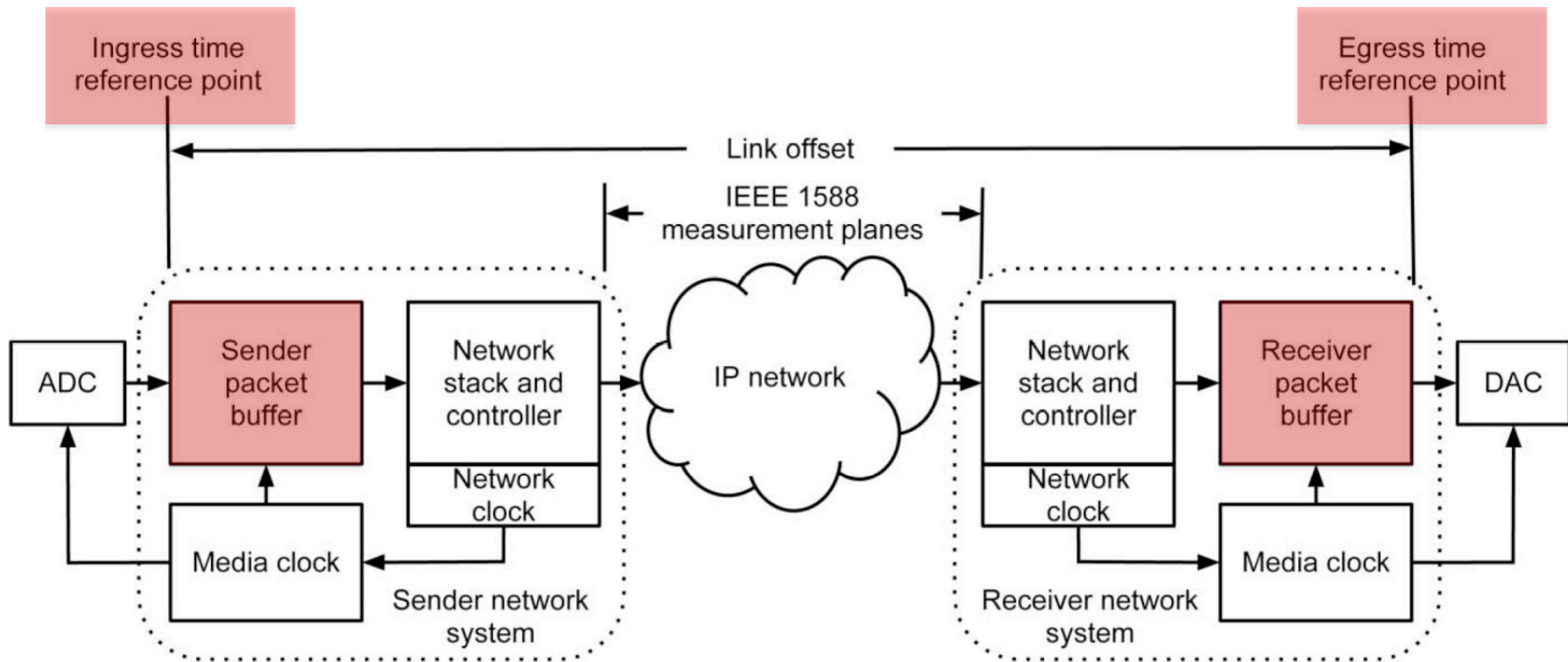


# Quality of Service (QoS) using DiffServ DSCP

Differentiated Services Code Points (DSCP) is a 6 bit value in the IP header DS field, guiding the switch device priority of handling packets.

↑  
increasing importance

<b>Class name</b>	<b>Traffic type</b>	<b>Default DiffServ class (DSCP decimal value)</b>
Clock	IEEE 1588-2008 <i>Announce, Sync, Follow_Up, Delay_Req, Delay_Resp, Pdelay_Req, Pdelay_Resp</i> and <i>Pdelay_Resp_Follow_Up</i> packets	EF (46)
Media	RTP and RTCP media stream data	AF41 (34)
Best effort	IEEE 1588-2008 signaling and management messages. Discovery and connection management messages.	DF (0)

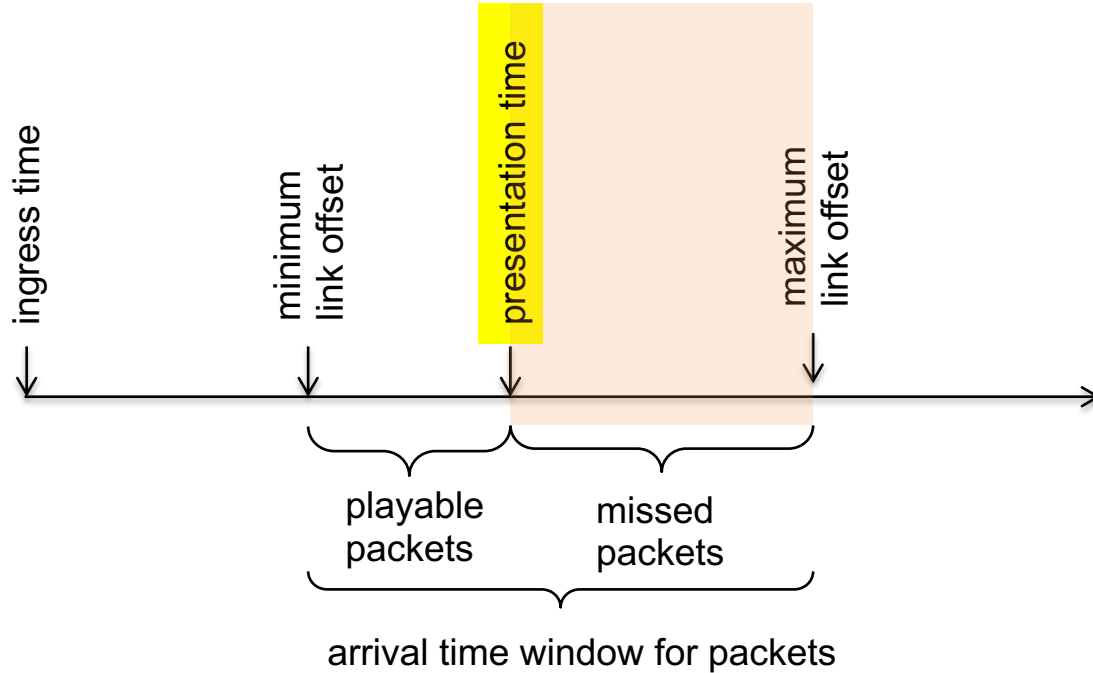




# Link offset and packet jitter allowance

Packet time (ms)	Minimum link offset (2 x packet time) (ms)	Maximum link offset (ms) (20+1) x packet time (max 20 ms)	Sender maximum jitter (ms) (17 x packet time) (max 17 ms)
0,125	0,25	2,625	2,125
0,25	0,5	5,25	4,25
0,333	0,666	6,993	5,661
1	2	20	17
4	8	20	17

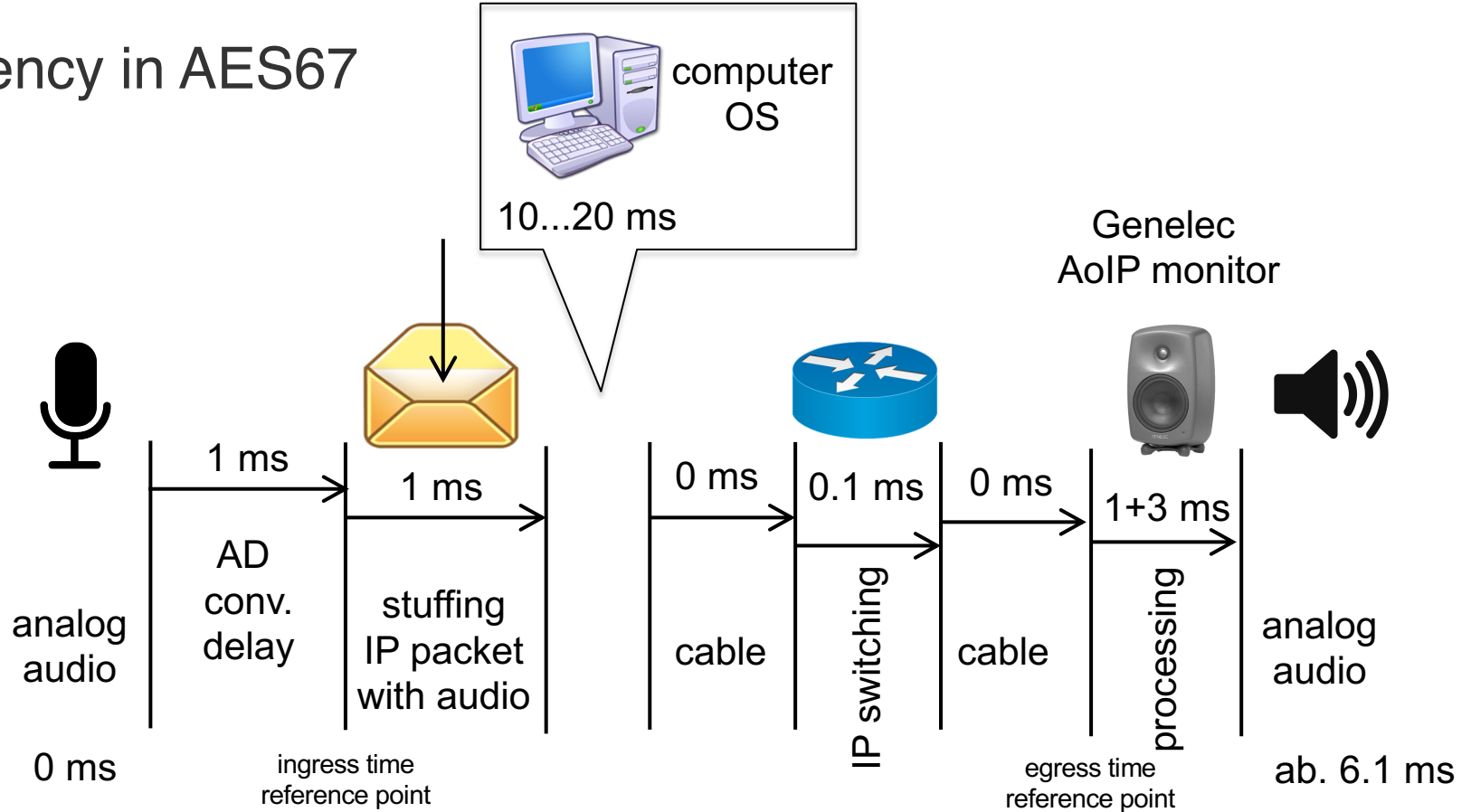
# Missing the presentation time window



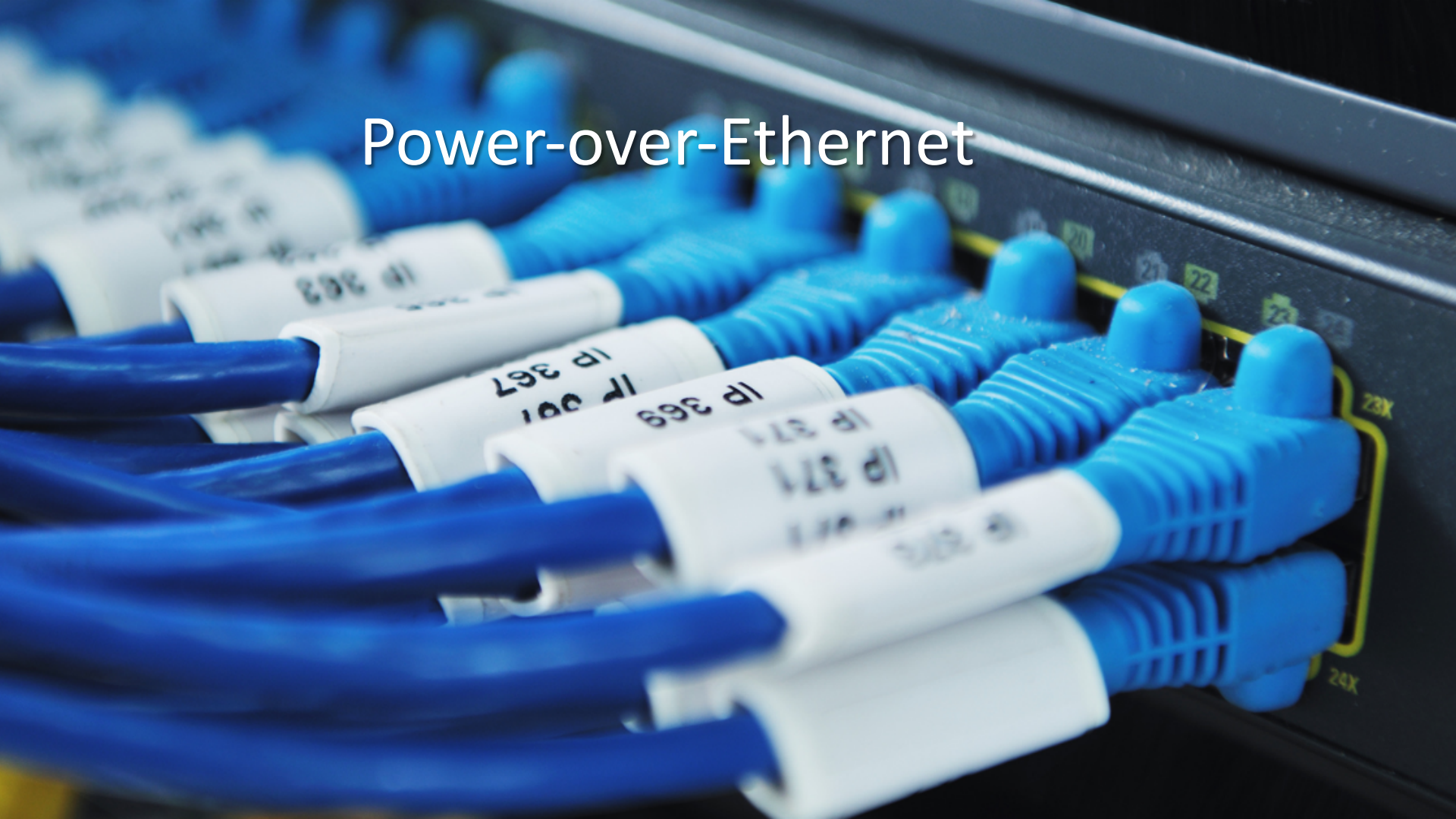
# Synchronized vs. syntonised (PTP standard, IEEE 1588)

- synchronized clocks:
  - Two clocks are synchronized to a specified uncertainty if they have the same epoch and their measurements of the time of a single event at an arbitrary time differ by no more than that uncertainty.
  - → Clock shows the same time
- syntonized clocks:
  - Two clocks are syntonized if the duration of the second is the same on both, which means the time as measured by each advances at the same rate. They may or may not share the same epoch.
  - → Two clocks may show different time, but advance at the same rate

# Latency in AES67

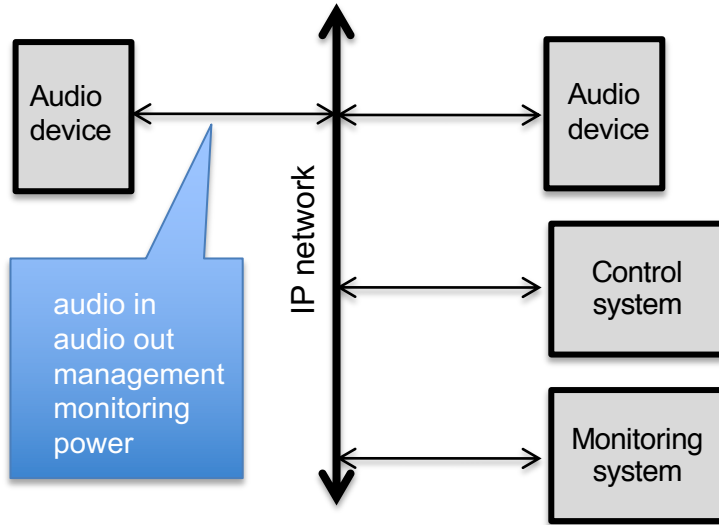


# Power-over-Ethernet





# One-cable-connectivity to a device is possible



- Traffic can co-existing in the same cable
  - audio
  - (video)
  - system management
  - device monitoring
  - system management
  - other data (for example, office)
- Also power to device is available: PoE

# Power-over-Ethernet PoE

Name	Standard	Power to Device	Voltage, current at Device	Power management
PoE	IEEE 802.3af-2003	<13 W	37-57 VDC, < 350 mA	signature
PoE+	IEEE 802.3at-2009	< 25.5 W	42.5-57 VDC	signature + LLDP
4PPoE	IEEE 802.3bt-2018	< 51 W (Type 3) < 71 W (Type 4)	42.5-57 VDC, < 600mA 41.1-57 VDC, < 960 mA	signature + LLDP

LLDP, Link-Layer  
Discovery Protocol

# Single cable connectivity solution

Audio-over-IP stream  
(all standards available)



Management and monitoring  
(GLM-for-Install)



Power-over-Ethernet (PoE+)



Analogue balanced input  
on a Phoenix screw  
mount connector



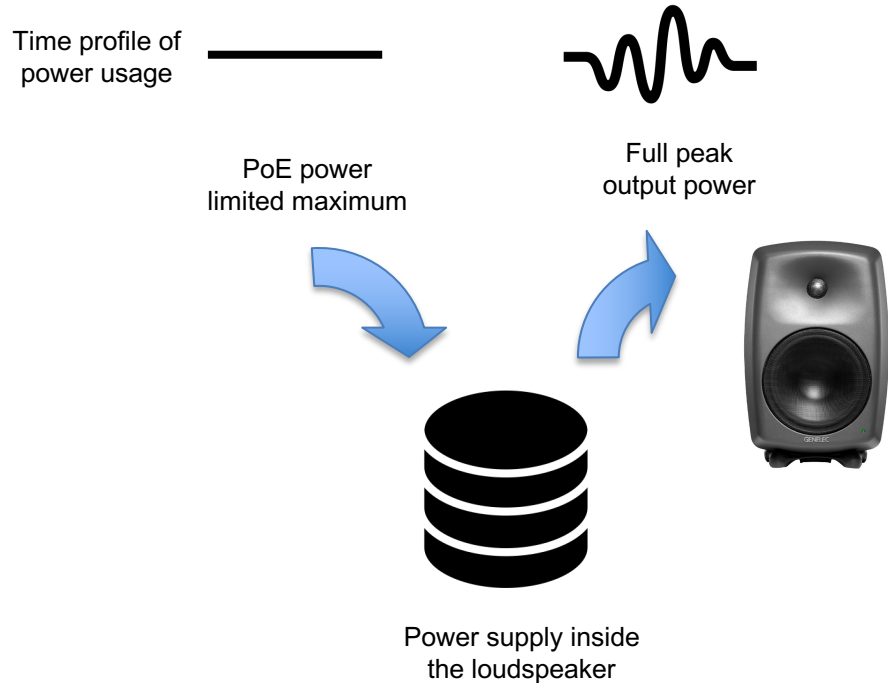
Single standard  
CAT cable



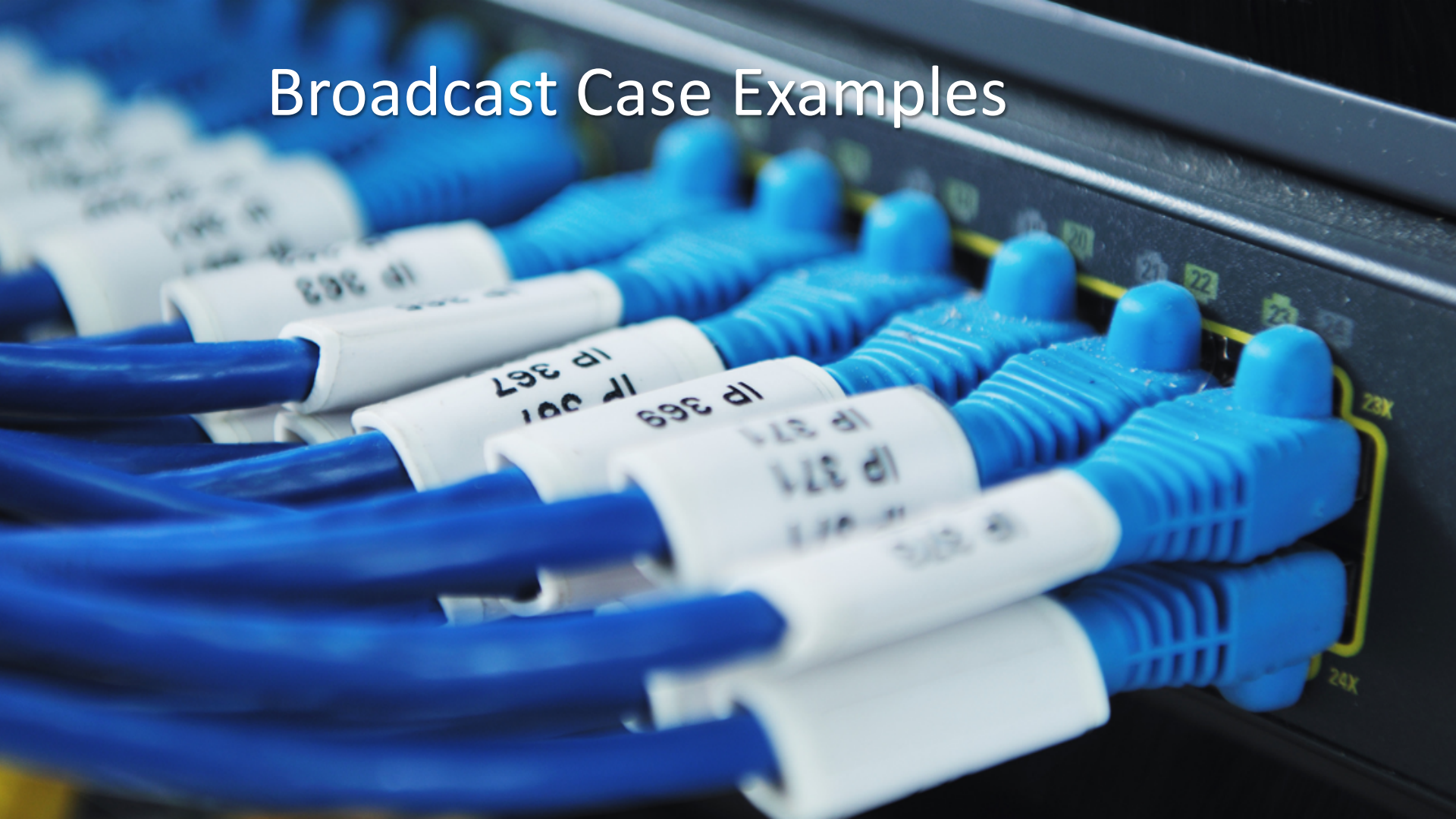
High performance active  
loudspeaker on IP network with  
high acoustic flexibility,  
versatile management and full  
status monitoring over IP

# PoE benefits and use

- Low cost standard cabling
- PoE is intrinsically safe voltage
  - Cost savings: anyone can install cabling and systems
- Standard off-the-shelve technology
- PoE+ supplies max 24.5 W of (constant) power
- Loudspeaker power supply enables high output power at music peak levels



# Broadcast Case Examples





BBC IP Studio Project (2017)

sverigesradio  
2017



# sverigesradio

2017







# Virtual Radio Studio (2017)



Genelec 8430A

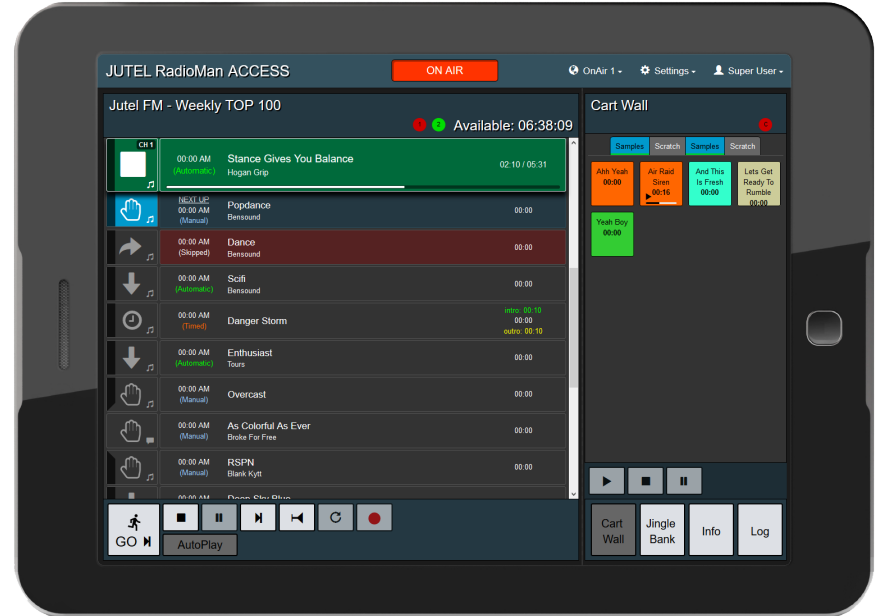
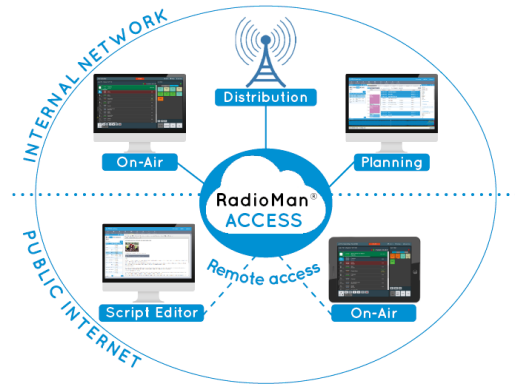


Genelec 8430A

Axia Audio IP-Tablet Virtual Radio software, designed by IP-Studio

<https://www.telosalliance.com/Axia/ip-tablet-virtual-radio-software>

# Jutel RadioMan Access (2018)



RadioMan® ACCESS On-Air tablet interface

Hogeschool, Utrecht

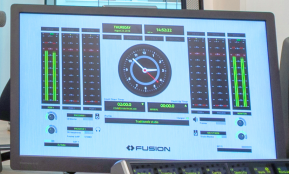
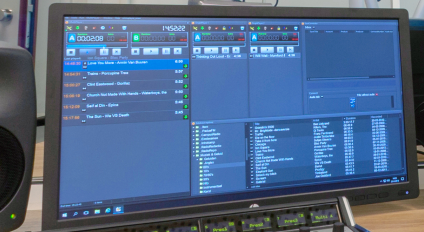


Hogeschool, Utrecht



donderdag 23 augustus 2016  
14:52:22  
Totale programmaduur 00:00:00  
Tijd tot volgend item 00:00:00  
Overname 00:00:00  
00:02:09  
ON AIR MIC LIVE STUDIO LIVE TELEFOON  
CHROMAFLUX

TRIPLE AUDIO  
Engineering & Broadcast



# CCTV EFP system in Djakarta, control room 1



开播 09:00:00  
14:13:34  
倒计时 18:46:27

2018雅加达亚运会  
前方演播室  
主路



CCTV EFP system in Djakarta, control room 2



开播 01:00:00  
15:11:02  
倒计时 09:48:58

CCTV EFP system in Djakarta, control room 3



# AV Install Case Examples





Maaninka Church, Finland



Restaurant Nallikari, Finland

**the sonic reference**