

# AES67-101: The Basics of AES67



**Anthony P. Kuzub**

IP Audio Product Manager

[Anthony@Ward-Beck.Systems](mailto:Anthony@Ward-Beck.Systems)

[www.Ward-Beck.Systems](http://www.Ward-Beck.Systems)



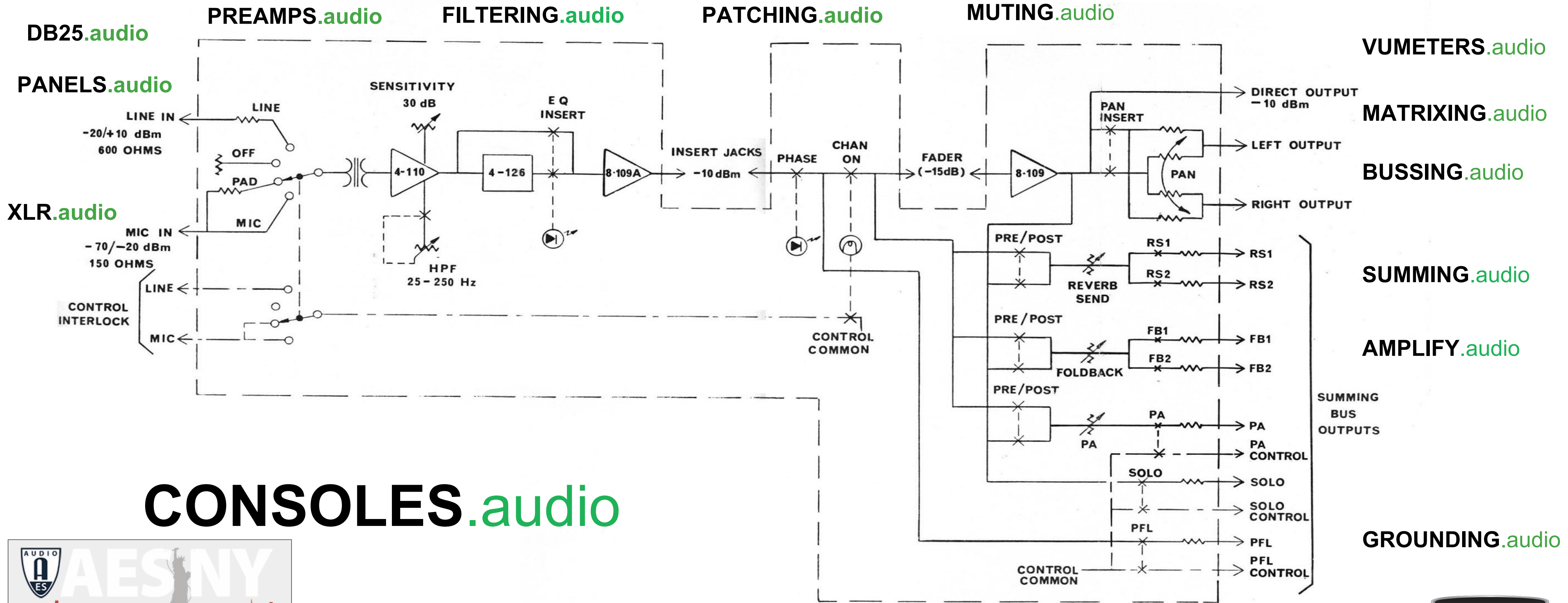
**WARD-BECK SYSTEMS**

[TorontoAES.org](http://TorontoAES.org)

Vice-Chair



# Ward-Beck Systems - Audio Domains



## CONSOLES.audio



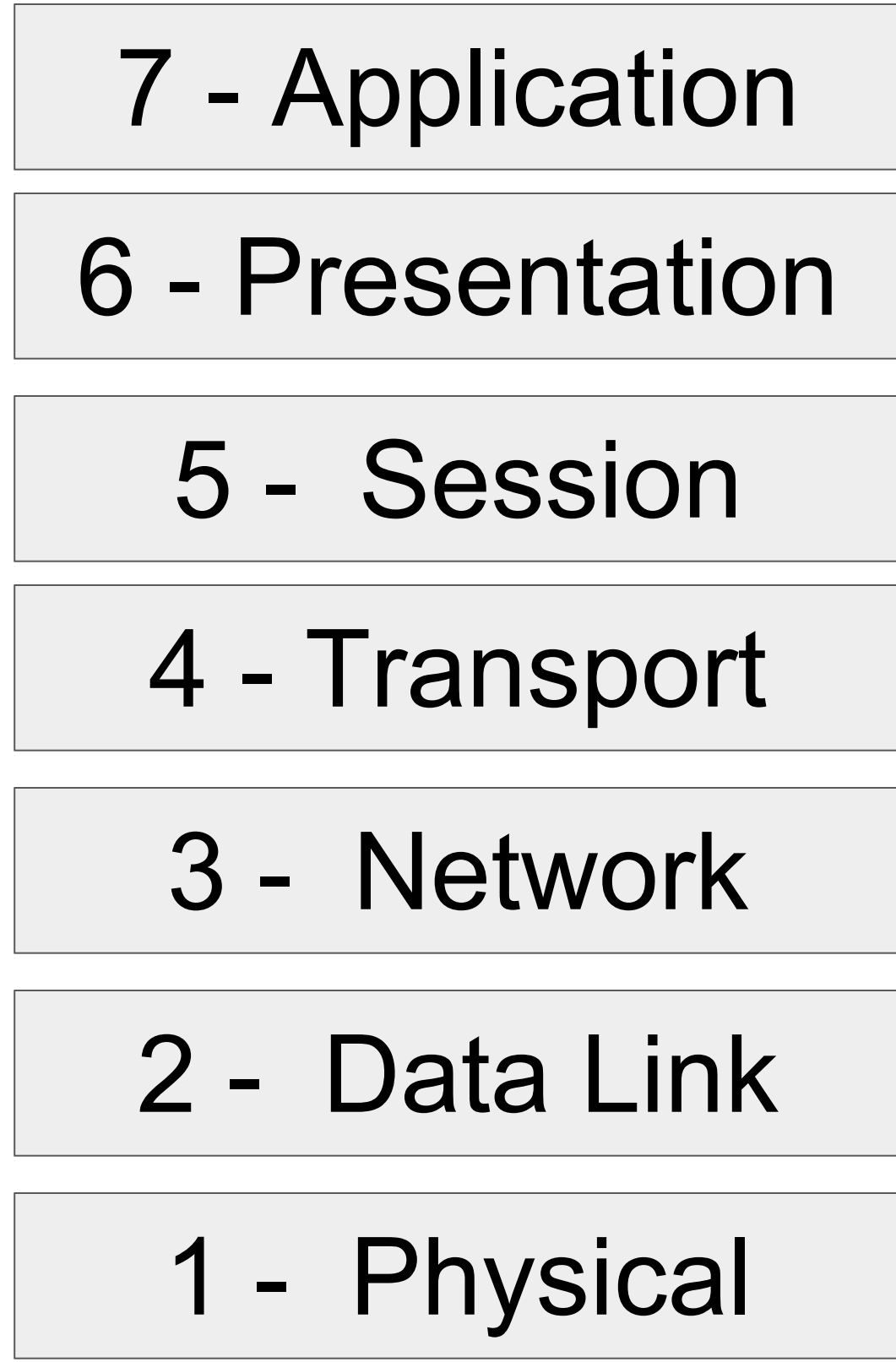
The least you **SHOULD**  
know about networking:

**The physical datalink networks  
transported sessions  
presented by the application**

**AES67.audio**  
**2110-30.AES67.audio**

**IGMP.audio**  
**CLOCKING.audio**

**TX  
DATA**



**RX  
DATA**

**PATCHING.audio**  
**MATRIXING.audio**  
**BUSSING.audio**  
  
**ROUTING.audio**  
  
**SWITCHING.audio**  
**MULTICASTING.audio**

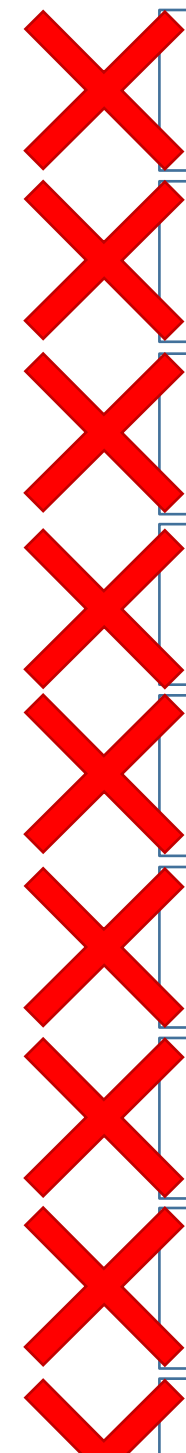


**RJ45.audio**



# The Road to Incompatibility...

	Dante	RAVENNA	QLAN	Livewire
Control & Monitoring	Proprietary	HTTP, Ember+	TCP, HTTP	HTTP, Proprietary
Discovery	Proprietary	Bonjour	Proprietary	Proprietary
Connection Management	Proprietary	RTSP, SIP, IGMP	Proprietary	Proprietary, HTTP, IGMP
Session Description	Proprietary	SDP	Proprietary	Channel #
Transport	Proprietary, IPv4	RTP, IPv4	RTP, IPv4	RTP, IPv4
Quality of Service	DiffServ	DiffServ	DiffServ	DiffServ/802.1pq
Encoding & Streaming	L16-32, ≤4 ch/flow	L16-32, ≤64 cha/str	32B-FP, ≤16 ch/str	L24, st, surr
Synchronization	PTP1588-2002	PTP1588-2008	PTP1588-2008	Proprietary
Media Clock	44.1kHz, 192kHz	44.1kHz - 384kHz	48kHz	48kHz



# AES67



***AES67-2013 Standard for audio applications of  
networks:  
High-performance streaming audio-over-IP  
interoperability***



# AES67

***AES67-2018 Standard for  
audio applications of networks:  
High-performance streaming a  
udio-over-IP interoperability***

## References

### **Audio Engineering Society, New York, NY., US.**

AES11 - AES recommended practice for digital audio engineering  
Synchronization of digital audio equipment in studio operations

### **Institute of Electrical and Electronics Engineers (IEEE)**

IEEE 1588-2008 - IEEE Standard for a Precision Clock Synchronization Protocol  
for Networked Measurement and Control Systems, July 2008

### **Internet Engineering Task Force**

RFC 768 – User Datagram Protocol”  
RFC 791 – Internet Protocol  
RFC 1112 – Host Extensions for IP Multicasting  
RFC 2236 - Internet Group Management Protocol, Version 2  
RFC 2474 – Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers  
RFC 2616 - Hypertext Transfer Protocol - HTTP/1.1  
RFC 2974 – Session Announcement Protocol  
RFC 3190 – RTP Payload Format for 12-bit DAT Audio and 20- and 24-bit Linear Sampled Audio  
RFC 3261 - SIP: Session Initiation Protocol  
RFC 3264 - An Offer/Answer Model with the Session Description Protocol (SDP)  
RFC 3376 - Internet Group Management Protocol, Version 3  
RFC 3550 – RTP: A Transport Protocol for Real-Time Applications  
RFC 3551 - RTP Profile for Audio and Video Conferences with Minimal Control  
RFC 4566 – Session Description Protocol  
RFC 5939 – Session Description Protocol (SDP) Capability Negotiation  
RFC 7273 – RTP Clock Source Signalling



# AES67.audio

Device Control

Connection Management

Transport

Timing

# DOMAINS:

Control & Monitoring

Discovery

Connection Management

Session Description

Transport

Quality of Service

Encoding & Streaming

Synchronization

Media Clock

**AES70.audio**

**PATCHING.audio**

**MATRIXING.audio**

**SWITCHING.audio**

**IGMP.audio**

**MULTICASTING.audio**

**ROUTING.audio**

**CLOCKING.audio**



# AES Standards

AES 3 - Serial Digital Audio

AES 10 - Serial Multichannel Audio Digital Interface (sMADI)

AES 14 - XLR-type polarity and gender

AES 59 - 25 pin D-Sub connector

AES67 - High-performance streaming audio-over-IP

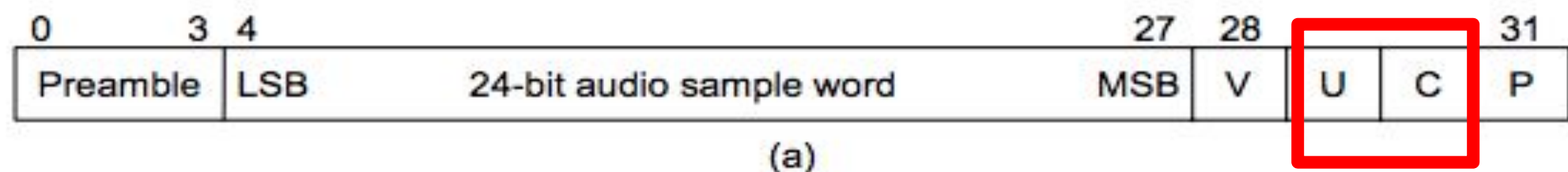
AES70 - Audio applications of networks - Open Control Architecture



# AES3-1992

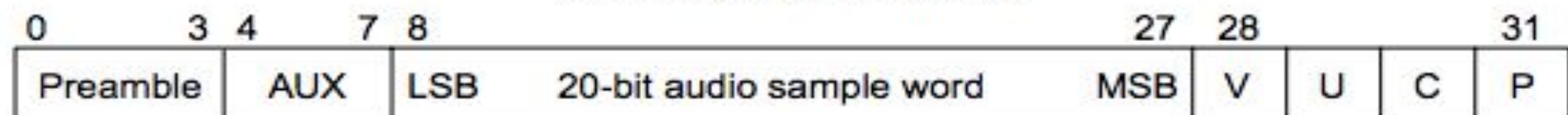
REVISED

**AES standard for digital audio —  
Digital input-output interfacing —  
Serial transmission format for two-  
channel linearly represented digital  
audio data**

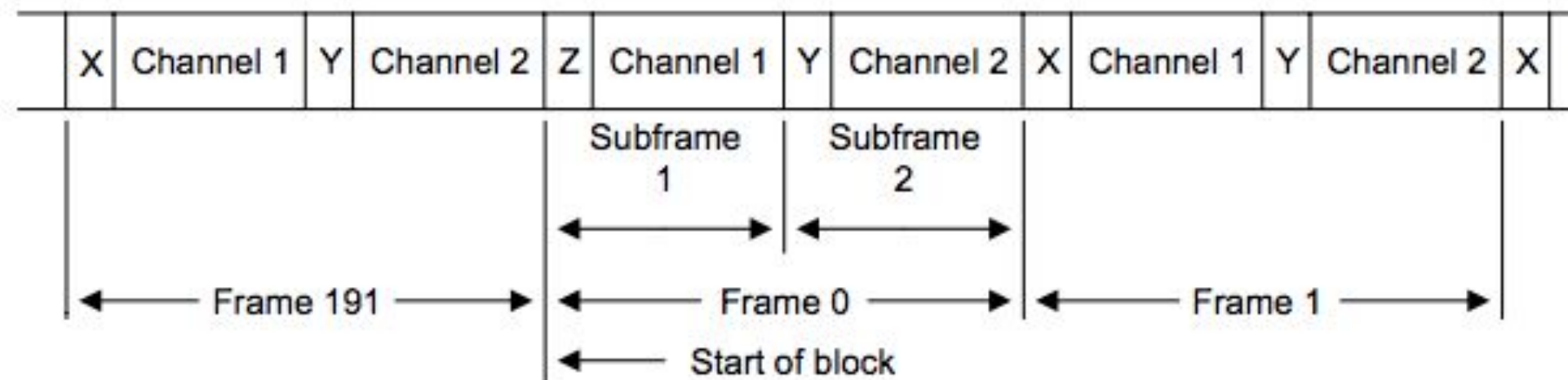


(a)

- V Validity bit
- U User data bit
- C Channel status bit
- P Parity bit
- AUX Auxiliary sample bits



(b)



# AES 10 - MADI

## AES Recommended Practice for Digital Audio Engineering — Serial Multichannel Audio Digital Interface (MADI)

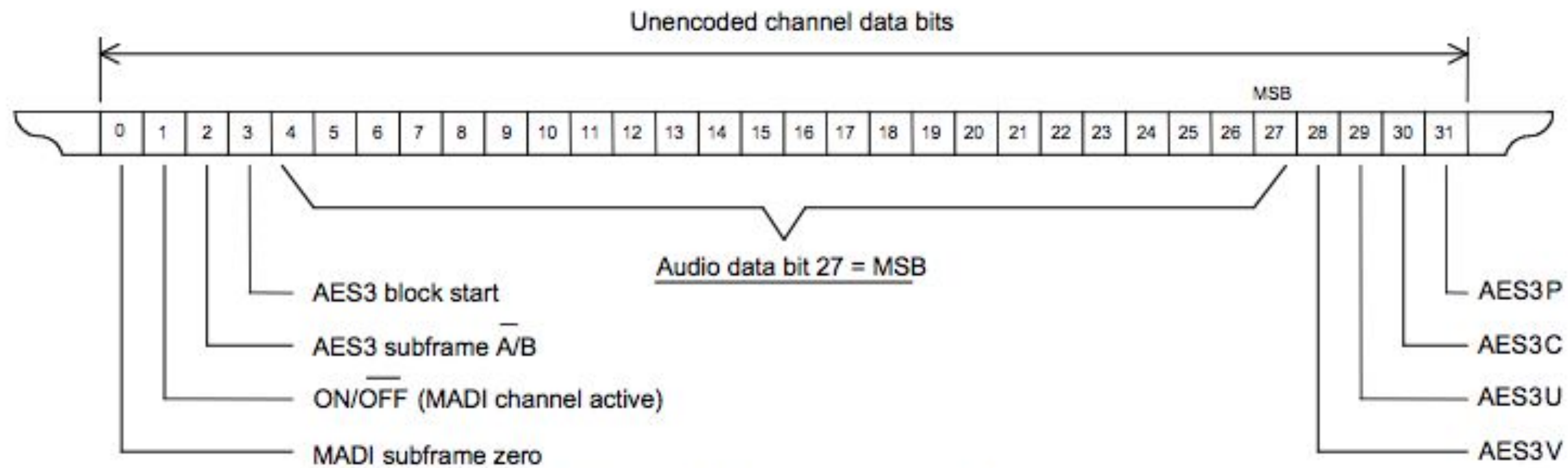


Figure 5 - Channel data format

MADI subframe  
Audio channel  
Sample number  
AES3 subframe

0	1	2	3	4					54	55	0
Ch 0	Ch 1	Ch 2	Ch 3						Ch 54	Ch 55	Ch 0
n	n	n	n						n	n	n + 1
A	B	A	B						A	B	A

← 20,8 μs →

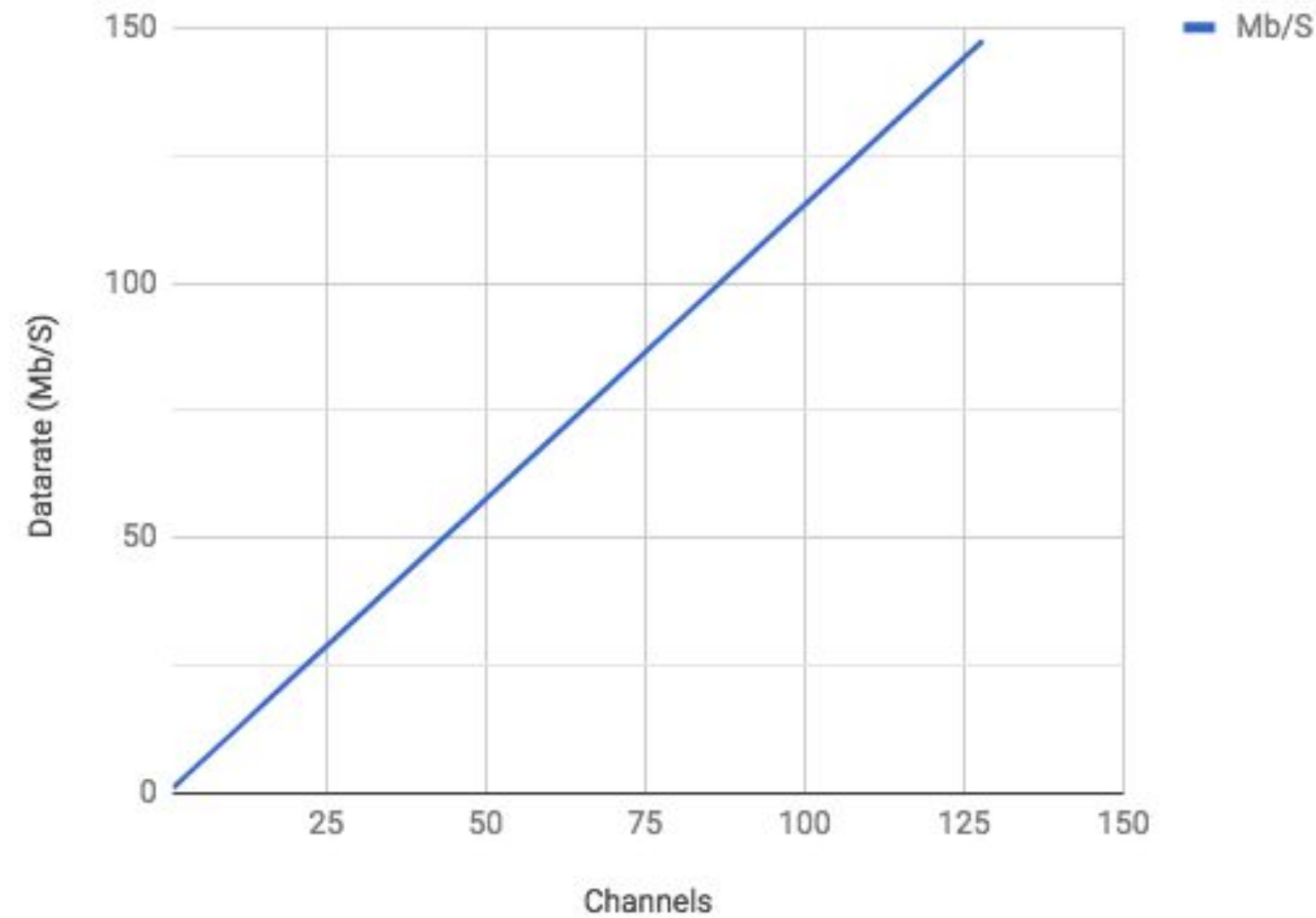
Figure 2 - 48 kHz with 56 channels working



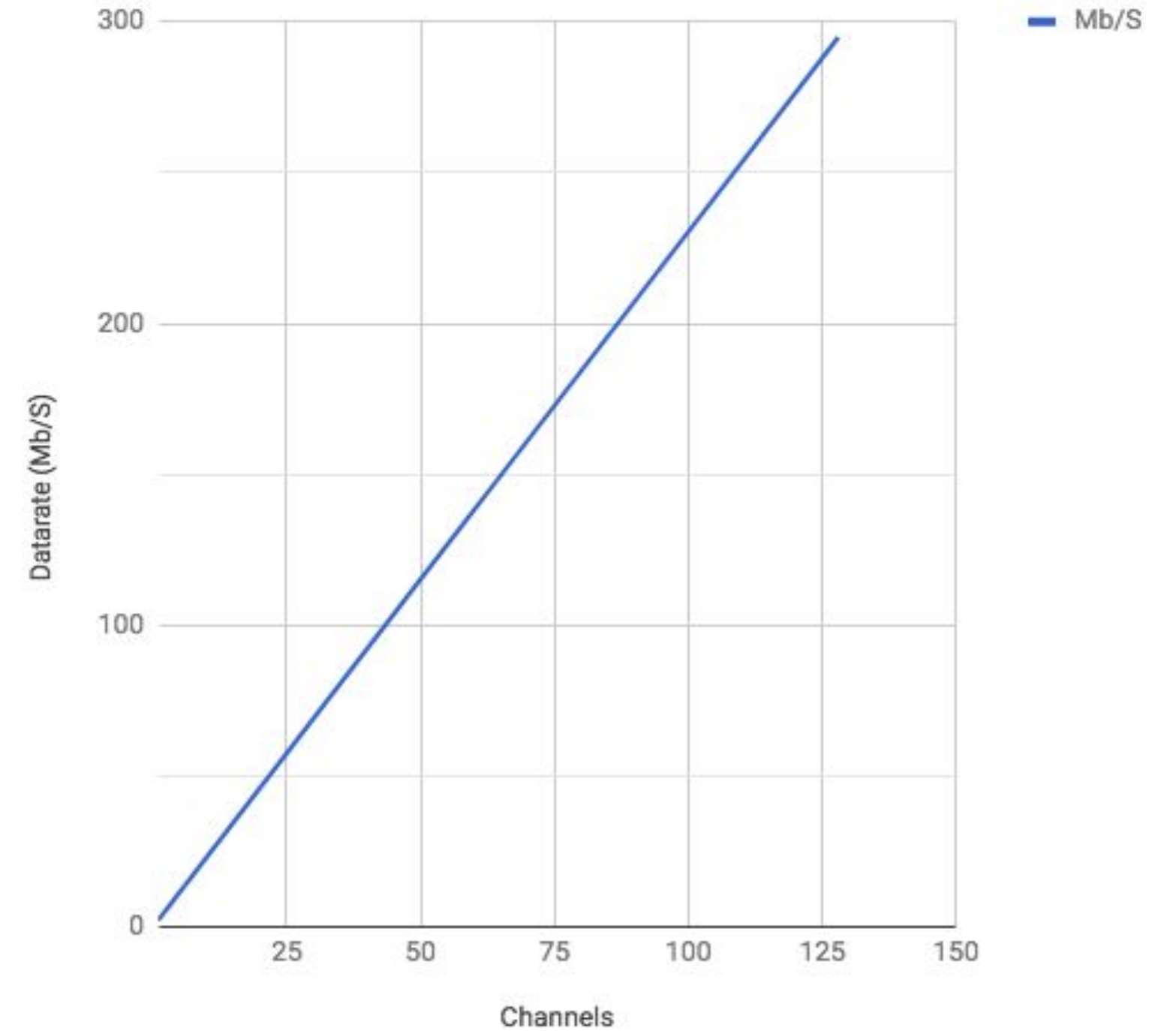
# DATA RATES

- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming**
- Synchronization
- Media Clock

Channels / Mb/S a=rtpmap:98 L24/48000/X  
a=framecount:48



Channels / Mb/S a=rtpmap:98 L24/96000/X  
a=framecount:48



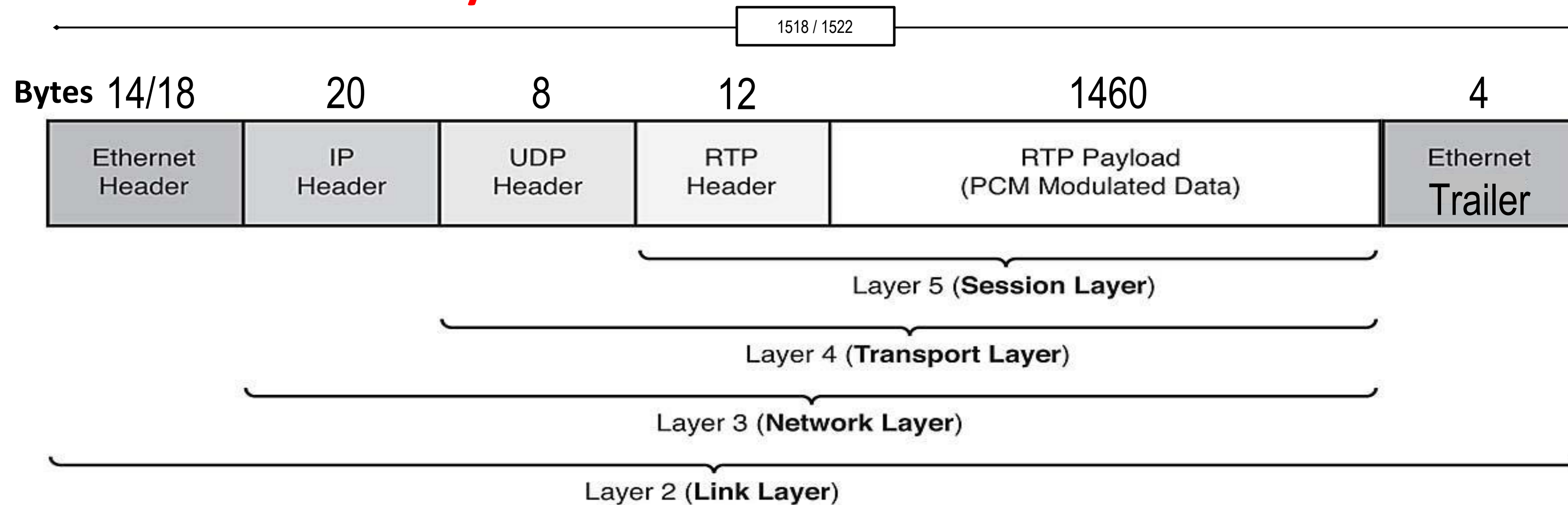
# RTP Packets - Required and recommended Packet Times

	<b>EXACT Packet time</b>	Packet samples (48 kHz)	Packet samples (96 kHz)	Packet samples (44,1 kHz)
Control & Monitoring				
Discovery				
Connection Management	1/8 millisecond "125 microseconds"	6	12	6
Session Description	1/4 millisecond "250 microseconds"	12	24	12
Transport				
Quality of Service	"333 microseconds"	16	32	16
<b>Encoding &amp; Streaming</b>	<b>"1 millisecond"</b>	<b>48</b>	<b>96</b>	<b>48</b>
Synchronization	"4 milliseconds"	192	n.a.	192
Media Clock				

# Layered Packet Encapsulation

IP + UDP + RTP overhead  
 $20 + 8 + 12 = 40$  Bytes

RTP overhead  
**12 Bytes**

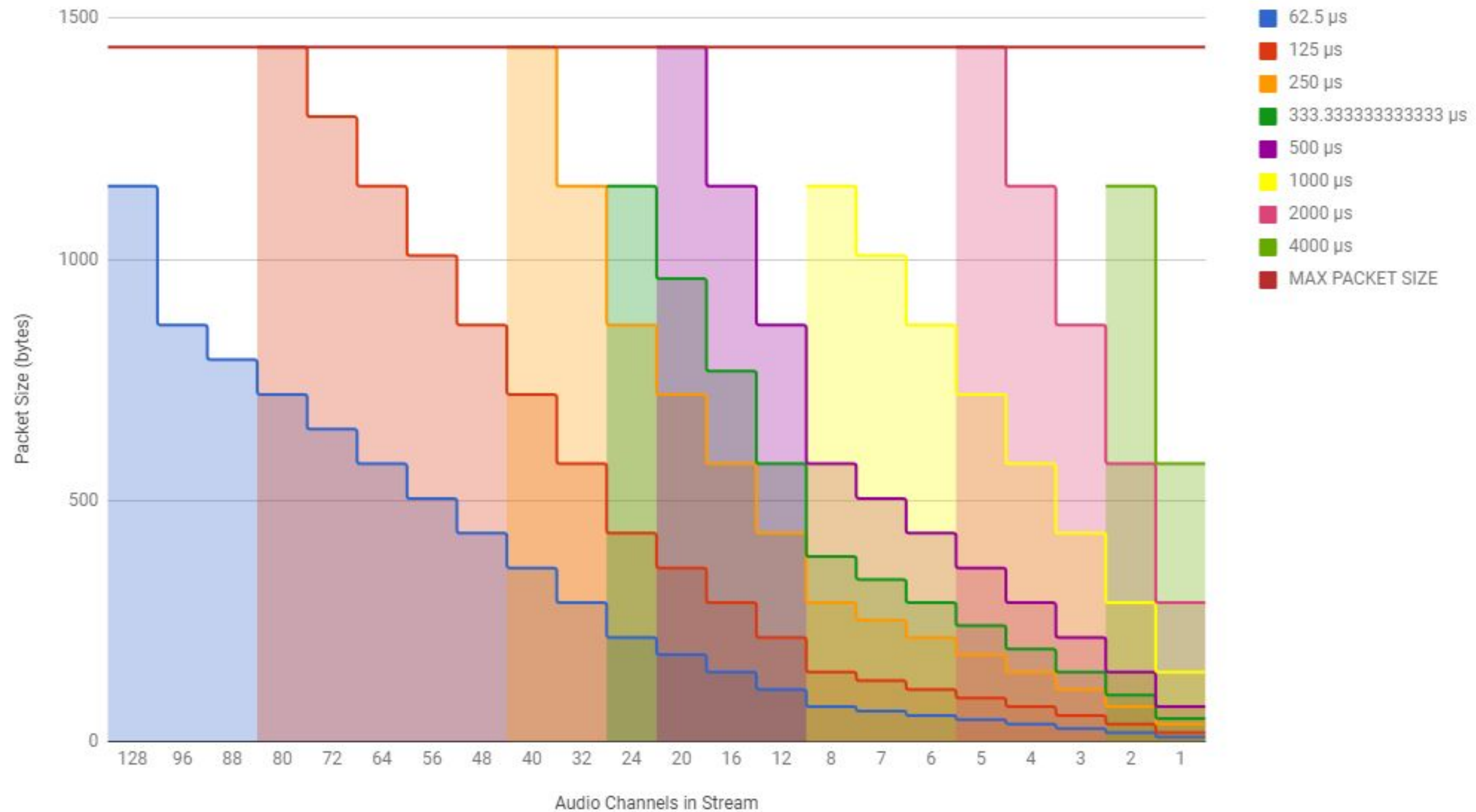


**MTU** (maximum transmission unit, largest size of a packet that can be transmitted without being split) 1500 Bytes in an IP/Ethernet LAN: in principle **0 to 1460 bytes** available for RTP **payload** data per packet

# RTP PACKETS - Required and recommended packet times

- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming
- Synchronization
- Media Clock

Packet Timing in Microseconds - Bytes Per Packet VS Channels in stream - 48KHz - 24-Bit



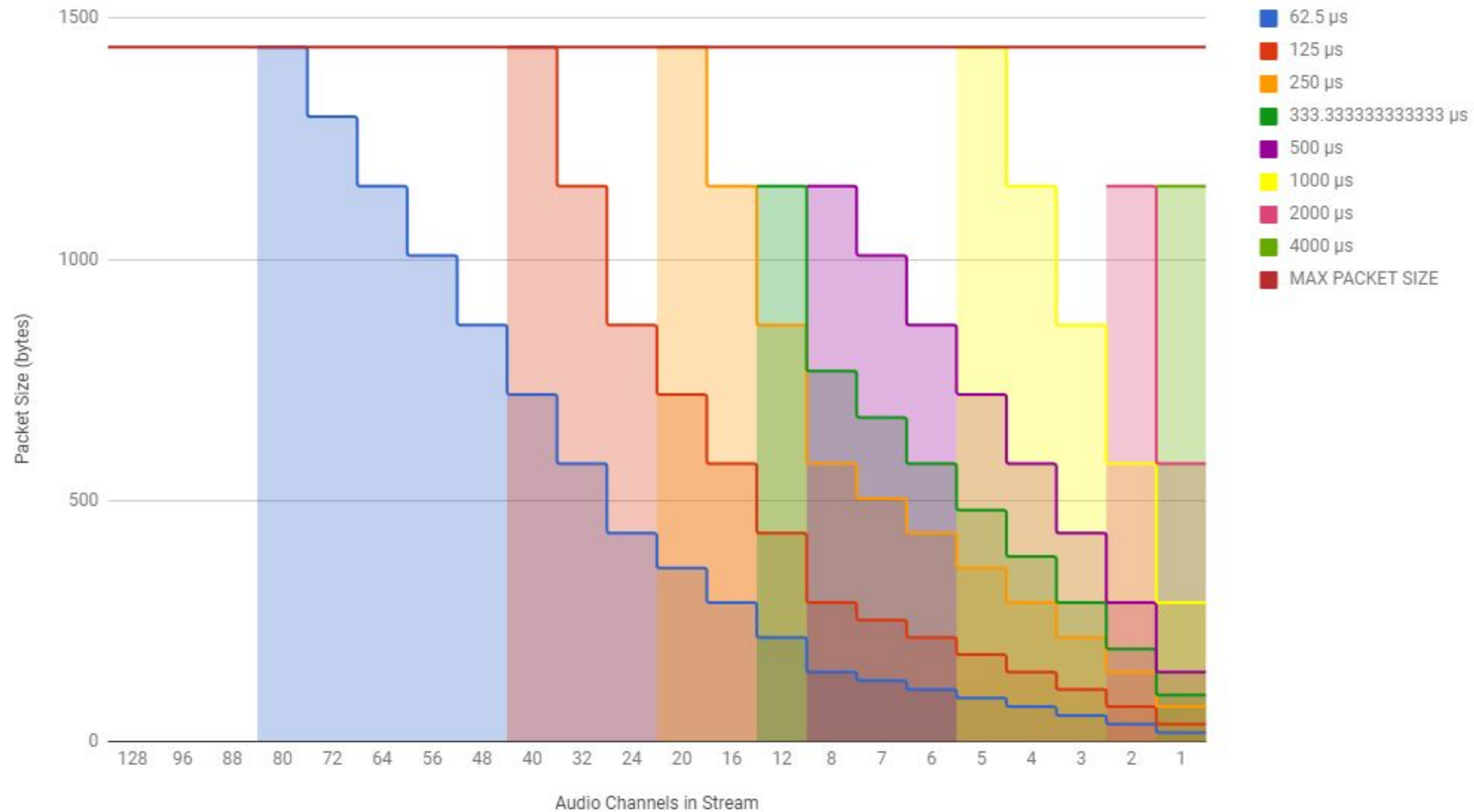
# RTP Packets - Required and recommended Packet Times



TTC2018

- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming
- Synchronization
- Media Clock

Packet Timing in Microseconds - Bytes Per Packet VS Channels in stream - 96KHz - 24-Bit

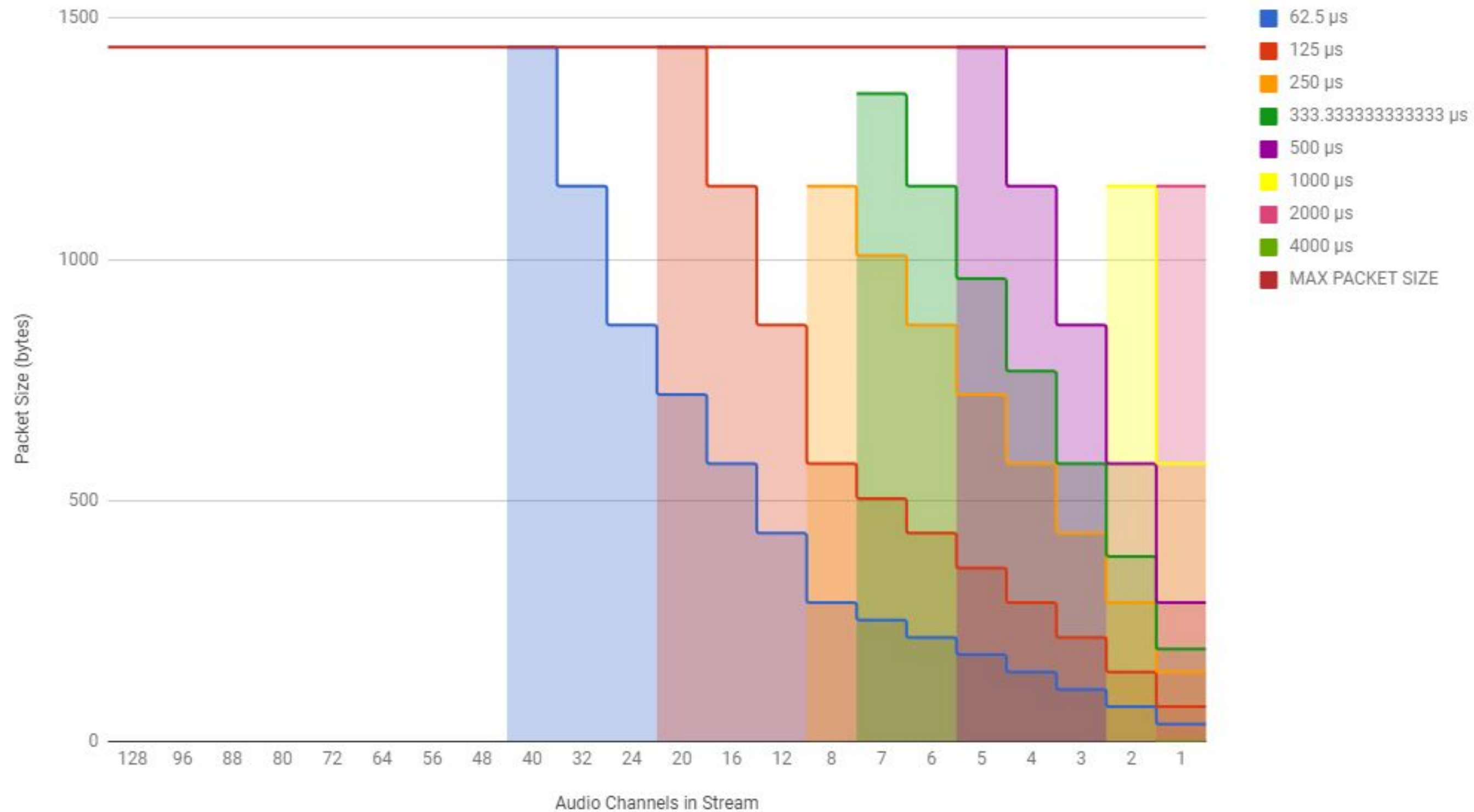




# RTP PACKETS - Required and recommended packet times

- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming
- Synchronization
- Media Clock

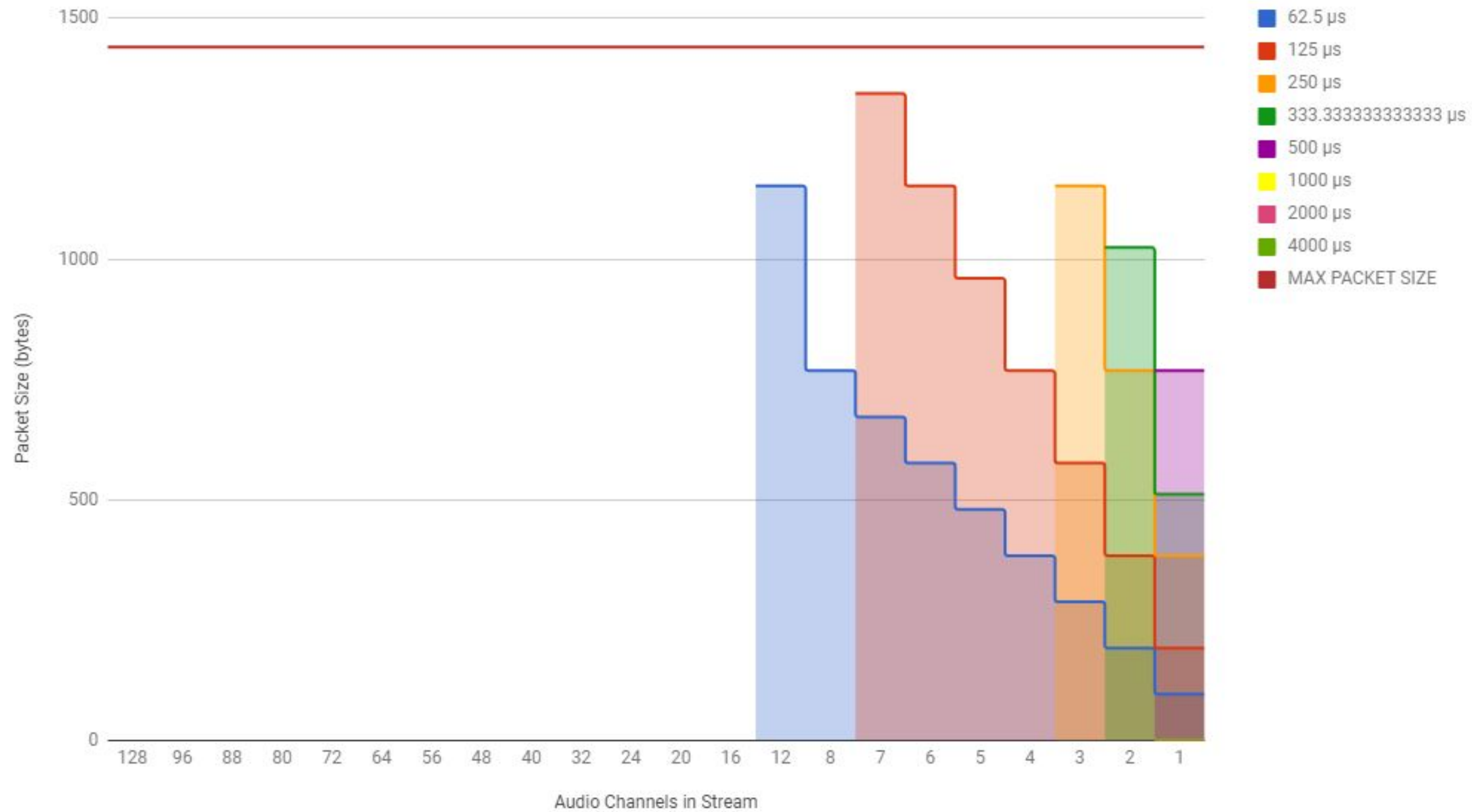
Packet Timing in Microseconds - Bytes Per Packet VS Channels in stream - 192KHz - 24-Bit



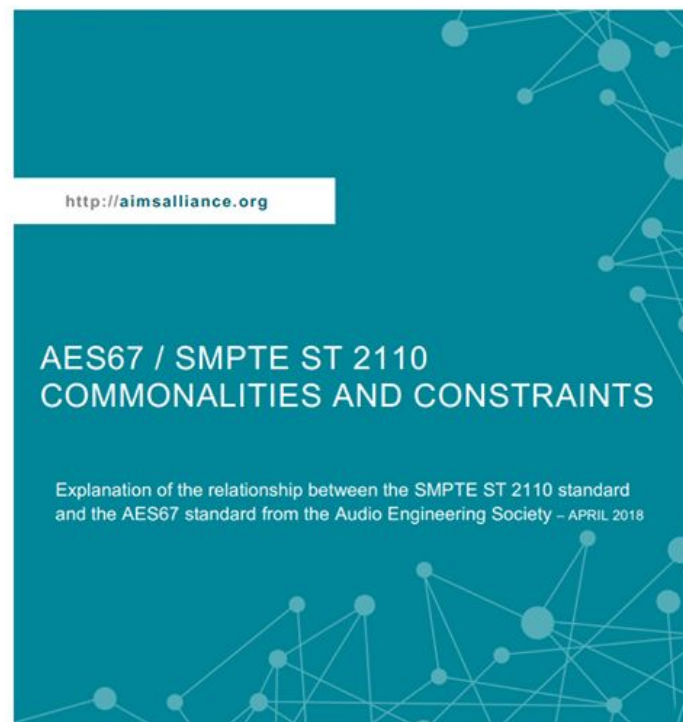
# RTP Packets - Required and recommended packet times

- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming
- Synchronization
- Media Clock

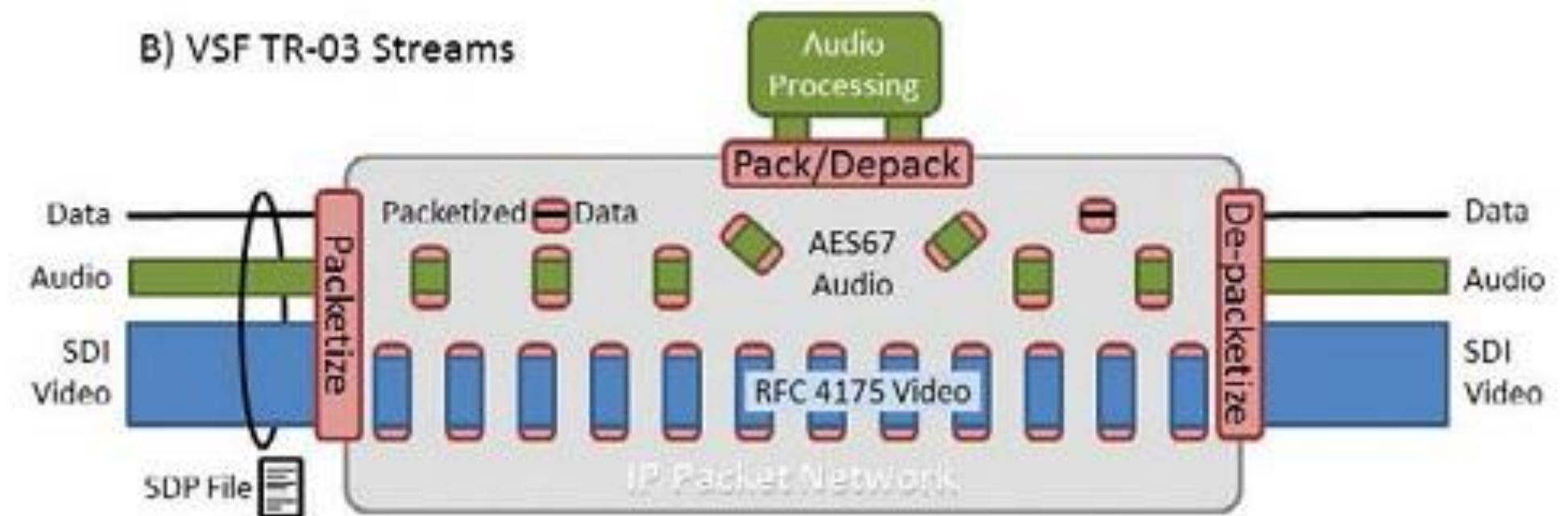
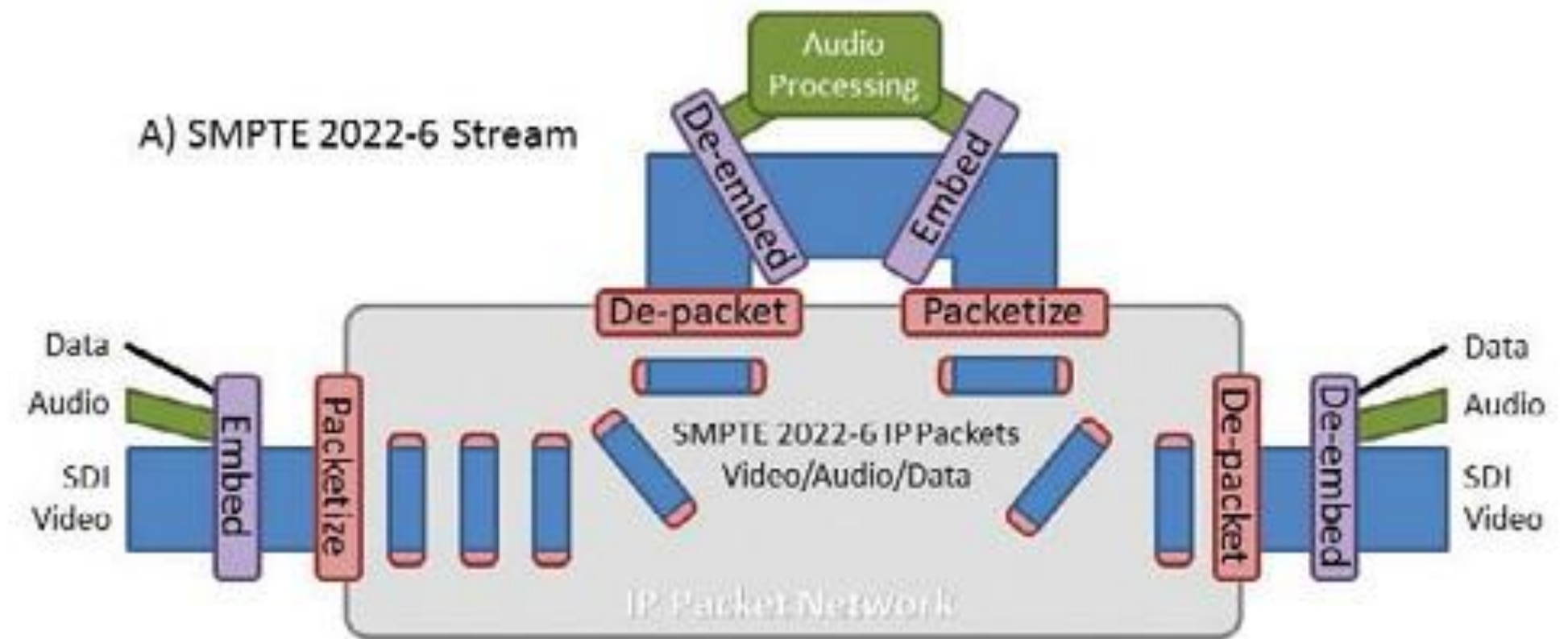
Packet Timing in Microseconds - Bytes Per Packet VS Channels in stream - 384KHz - 24-Bit



# 2110-30



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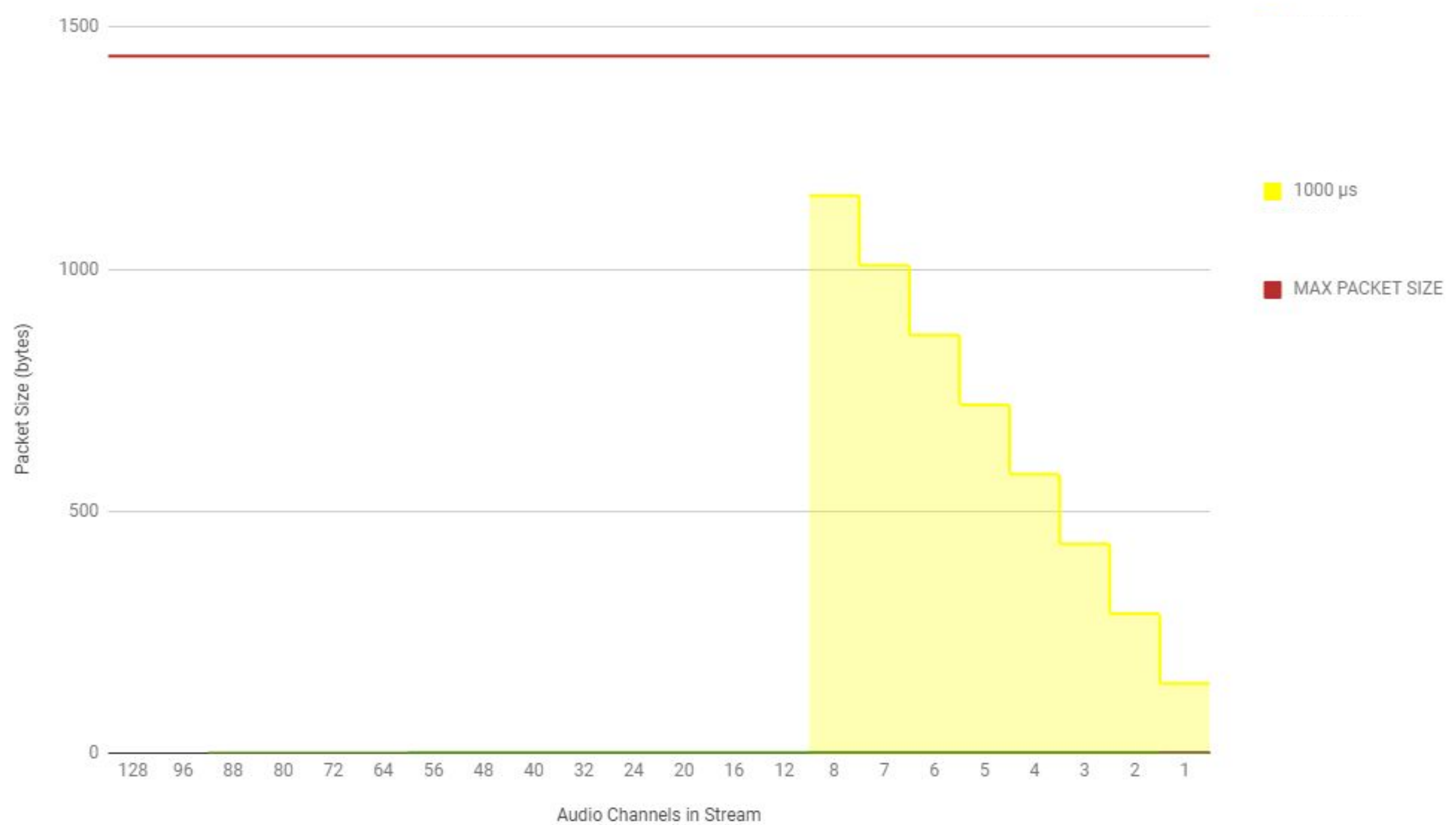


# AES67 - RX SHALL SUPPORT

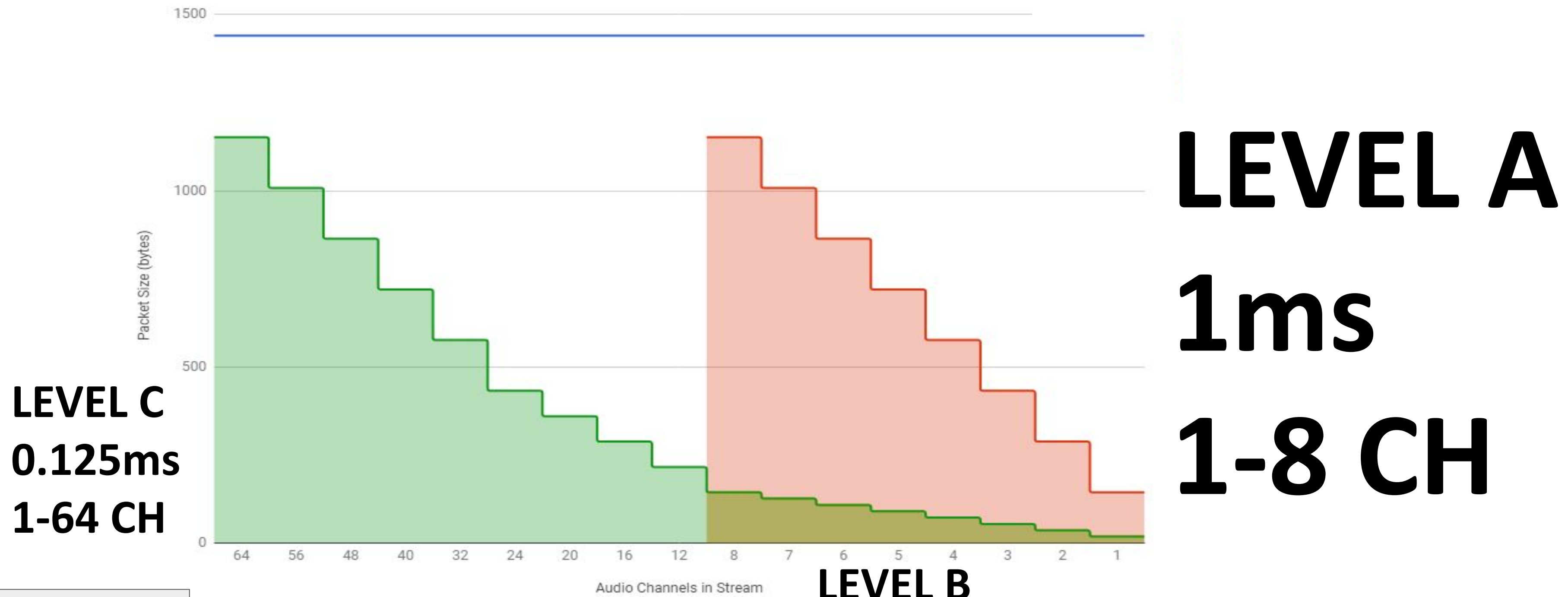
## 1ms - 1-8 Channels 48KHZ - 24BIT

- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming**
- Synchronization
- Media Clock

Packet Timing in Microseconds - Bytes Per Packet VS Channels in stream - 48KHz - 24-Bit - SHALL SUPPORT



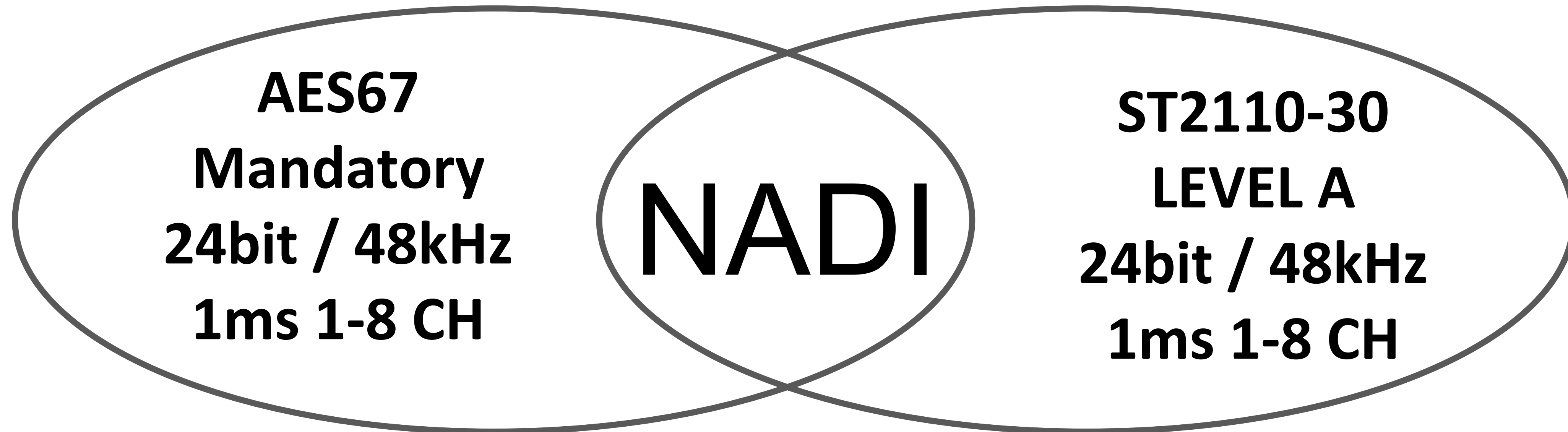
# SMPTE 2110-30



**LEVEL C**  
**0.125ms**  
**1-64 CH**

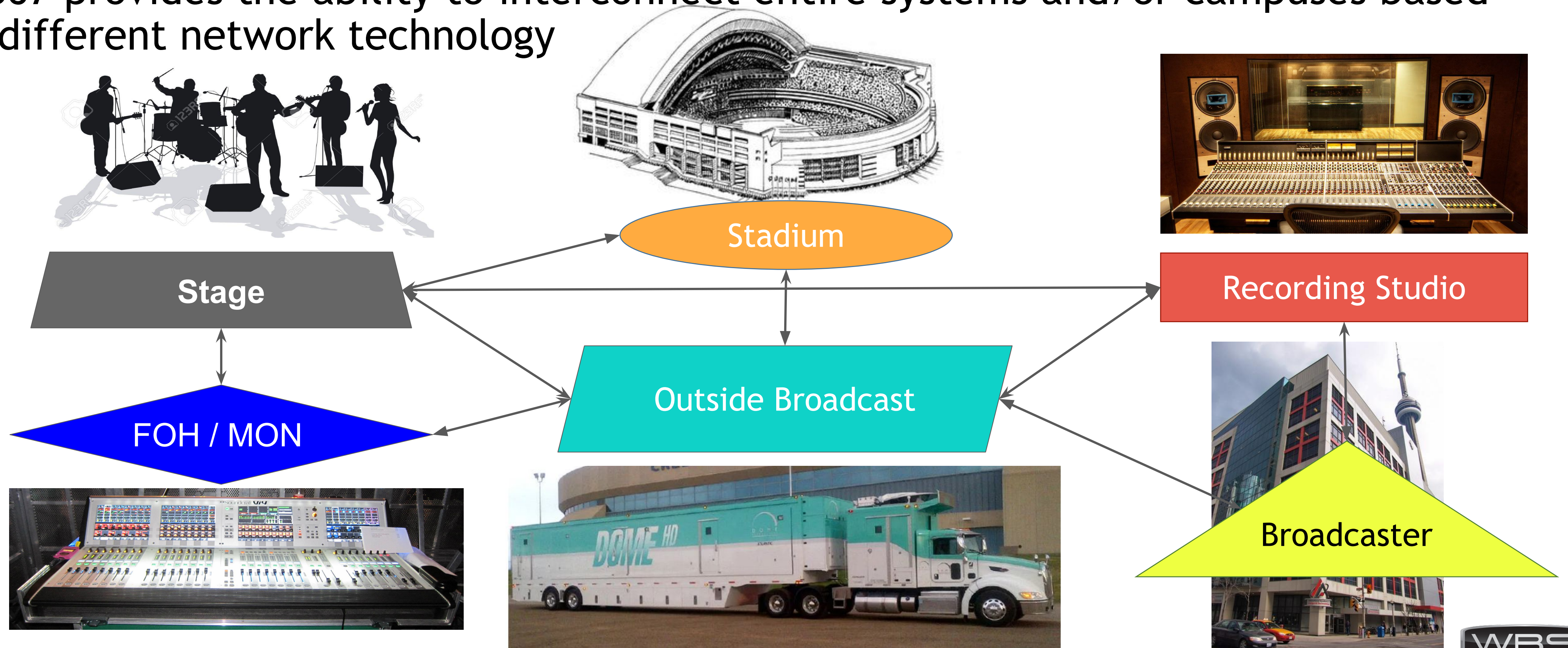
**LEVEL A**  
**1ms**  
**1-8 CH**

**LEVEL B**  
**0.125ms**  
**1-8 CH**



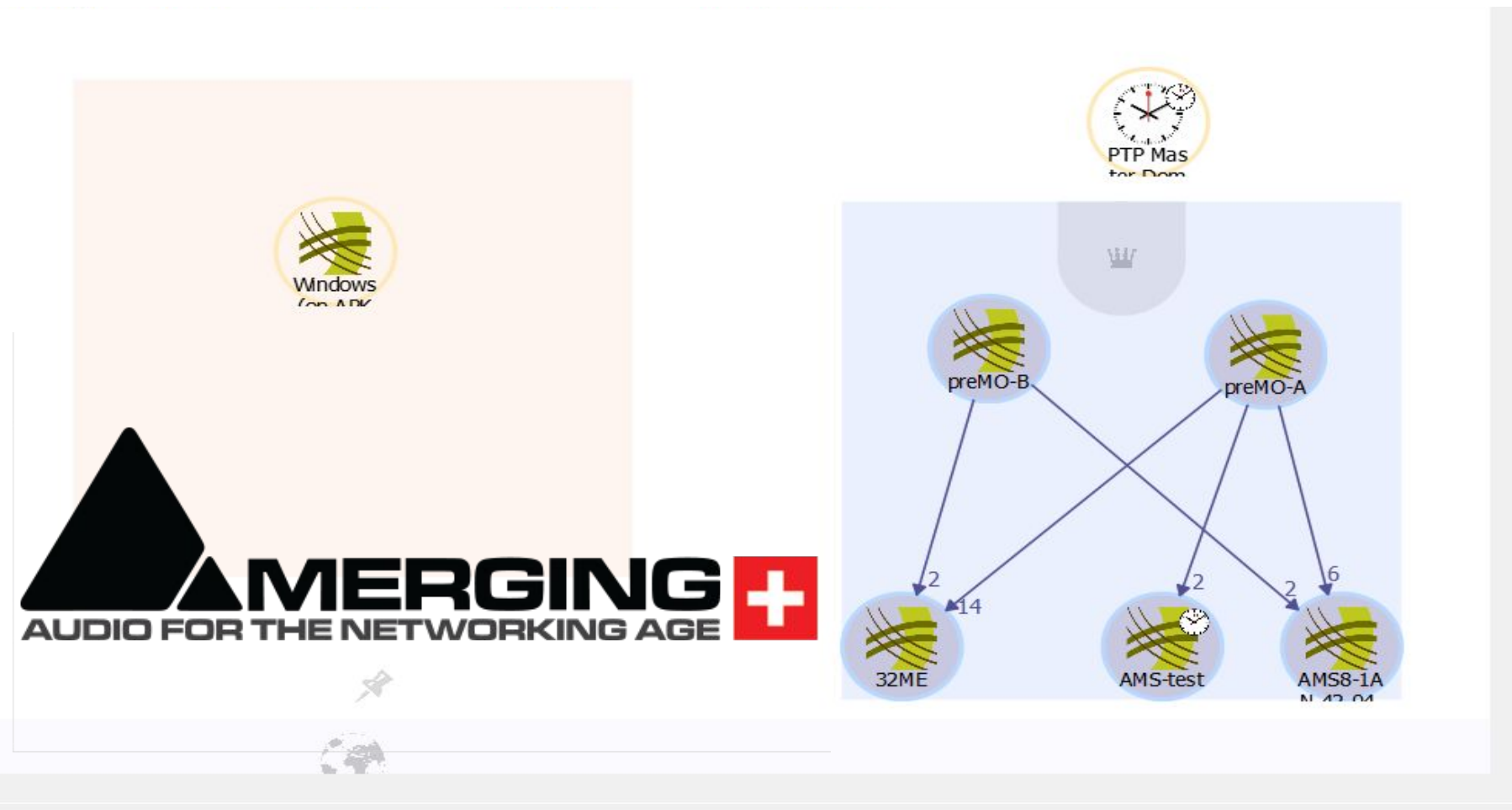
# AES67 SOLUTION:

AES67 provides the ability to interconnect entire systems and/or campuses based on different network technology

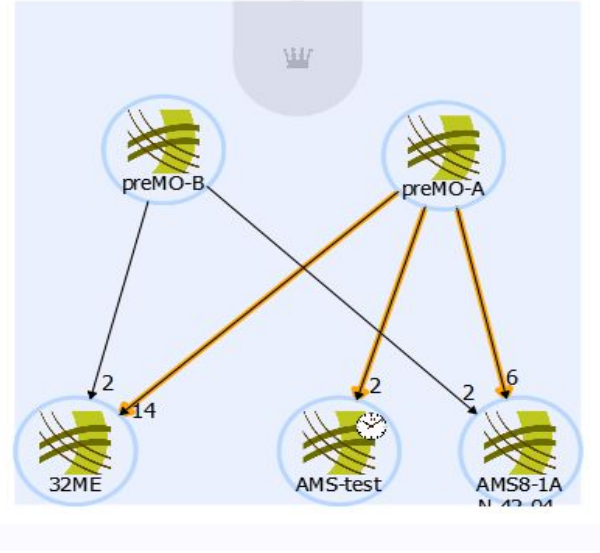


# ANEMAN.AES67.aUDIO

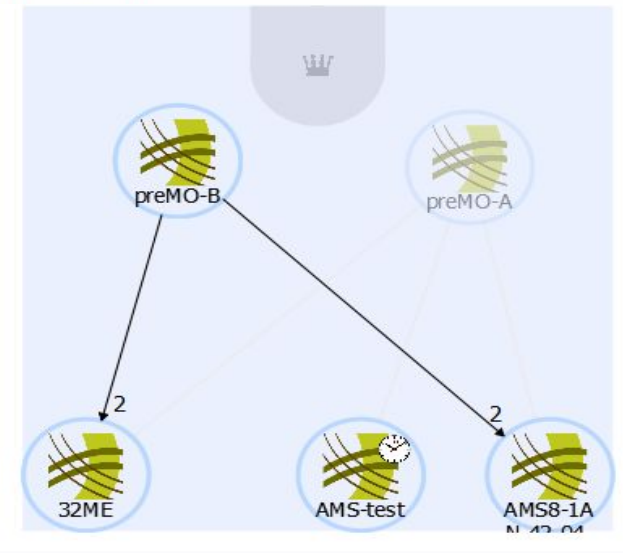
- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming
- Synchronization
- Media Clock



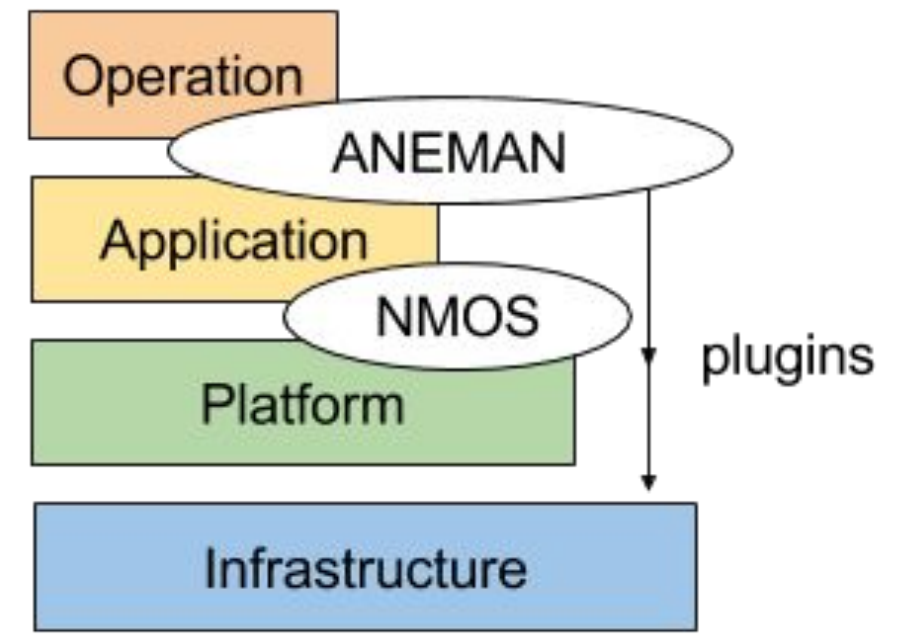
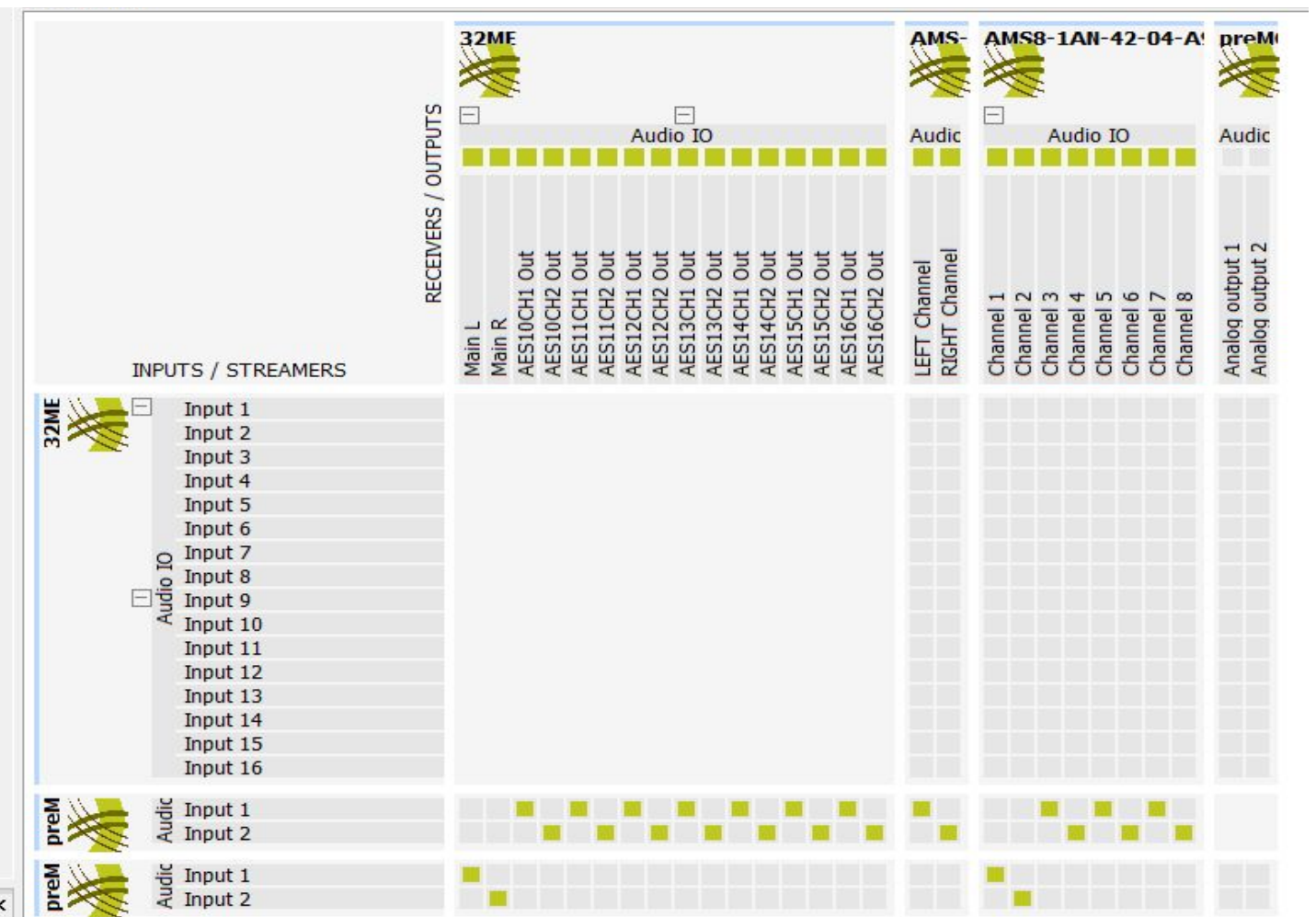
Name	Type	Version	Company	Product	Serial
preMO-B		0	Ward-Beck Syst...	PreMo	00:07:F5:00:52:B6
preMO-A		0	Ward-Beck Syst...	PreMo	00:07:F5:00:52:DE
Windows (o...		0	Unknown	Unknown	Windows (on APK_Bench_PC)
PTP Master ...			Unknown	PTP Master	00-07-F5-FF-FE-42-02-E8
AMS8-1AN-...		0	Ward-Beck Syst...	AMS8-1AN	00:07:F5:42:04:A9
AMS-test		0	Ward-Beck Syst...	AMS2-N	00:07:F5:42:02:E8
32ME		0	Ward-Beck Syst...	32ME-N	00:07:F5:42:05:87



Connection Change



Broken Connection





# MIX and MATCH

- Control & Monitoring
- Discovery
- Connection Management
- Session Description
- Transport
- Quality of Service
- Encoding & Streaming
- Synchronization
- Media Clock

The screenshot displays the AUDIOLAN web interface for AES67 Networking. At the top, it shows '19 devices found in network'. Below this is a grid of device cards, including 'MOBILE-32ME', 'APK-AMS2-42-0...', 'APK-BENCH-W...', 'APK-WBS-AMS8', 'FRONT-ROOM...', 'FRONTTESTuN...', 'MOBILE-AMS-2X2', 'MOBILE-AMS81', and 'MOBILE-GUEST...'. The 'MOBILE-32ME' device is selected, and its configuration page is shown. This page has tabs for 'Stream consumer', 'Stream provider', 'General settings', and 'Advanced'. The 'Advanced' tab is active, showing three channel configurations:

- Channel 1:** Name: RX via RAVENNA, Source stream: Front Room - DARS, Channel number: Channel 1. Includes 'Manual SDP' and 'Show current SDP' buttons.
- Channel 2:** Name: RX via SAP, Source stream: sap:Front Room - DARS, Channel number: Channel 1. Includes 'Manual SDP' and 'Show current SDP' buttons.
- Channel 3:** Name: RX via Manual SDP. Shows a text area with SDP details:
 

```
v=0
o=- 4 0 IN IP4 192.168.11.109
s=Front Room - DARS
c=IN IP4 239.1.11.109/1
t=0 0
a=clock-domain:PTPv2 0
m=audio 5004 RTP/AVP 98
a=rtpmap:98 L24/48000/8
a=recvonly
a=framecount:48
a=sync-time:0
```

 Includes a 'Normal Mode' button.

At the bottom of the interface, logos for RAVENNA and Dante (with 'Now with AES67' text) are visible.



DEVICES IN SUBNET

STREAM NAME OF DEVICE

API / MANUAL SDP

CH IN STREAM

**AUDIOLAN** AES67 Networking

11 devices found in network

PIANO-R	32ME-N	AMS2-N-LIVEFL...	AMS8-1AN	MLC8-N	Merging Technol...	Merging Technol...	PreMo2	Talkback	Vocal
Ward-Beck Syst... PreMo	Ward-Beck Syst... 32ME-N	Ward-Beck Syst... AMS2-N	Ward-Beck Syst... AES8-1AN	Ward-Beck Syst... MLC8-N	111.13.5.5	111.13.5.103	Ward-Beck Syst... PreMo	Ward-Beck Syst... PreMo	Ward-Beck Syst... PreMo

**PIANO-R**

Stream provider | General settings | Advanced | Discard changes | Apply all

Stream name: preMO - PIANO R | Activate:  | SDP: Show

Use automatic configuration

STREAM SDP INFORMATION

**WBS-32ME**

Stream consumer | Stream provider | General settings | Advanced | Discard changes | Clear all | Apply all

Channel 1	Channel 2	Channel 3
Name: AES9CH1 Out	Name: AES9CH2 Out	Name: AES10CH1 Out
SDP: v=0, o=- 1 0 IN IP4 192.168.111.208, s=Vizrt - VizEngine, t=0 0, a=clock-domain:PTPv2 0, m=audio 50000 RTP/AVP 98, c=IN IP4 239.31.208.1/1, a=rtptime:98 L24/48000/2, a=sync-time:0, a=framecount:48, a=ptime:1	SDP: v=0, o=- 1 0 IN IP4 192.168.111.208, s=Vizrt - VizEngine, t=0 0, a=clock-domain:PTPv2 0, m=audio 50000 RTP/AVP 98, c=IN IP4 239.31.208.1/1, a=rtptime:98 L24/48000/2, a=sync-time:0, a=framecount:48, a=ptime:1	Source stream: -- Select Stream --
Channel: 1	Channel: 1	Channel: 1

Manual SDP | Show current SDP > >>

RAVENNA SOURCES



NAME +  
IP ADDRESS  
CONFIG

PIANO-R

Stream provider | General settings | Advanced | Discard changes | Apply all

**Network configuration**  
The device has the IP 111.13.5.93, mask 255.255.0.0 and gateway 111.13.5.1.

Device name: PIANO-R

DHCP (recommended)  
 Static

IP address: 111.13.5.93  
IP mask: 255.255.0.0  
IP gateway: 111.13.5.1

**Audio configuration**  
Sampling Rate: 48 kHz  
Resolution: 24 bits

**Firmware update**  
No BSL.  
Main Firmware revision 67  
Start

PCM AUDIO  
RATE

PTP  
CONFIG

WBS-preMO-217

Stream consumer | Stream provider | General settings | Advanced | Discard changes | Apply all

**PTP**  
State: MASTER  
Grandmaster ID: 00-07-F5-FF-FE-00-53-E5  
Current offset: 0 ns  
Domain: 0  
 Preferred master

**SIP configuration**  
Changes in these parameters need a reboot to take effect.

Stream 1	Username	WBS-PREMO-217-TX
Receivers	Username	WBS-PREMO-217-RX

**SAP configuration**  
 SAP browsing enabled

Multicast address: 239.255.255.255  
Port: 9875  
Session timeout (seconds): 3600

SAP  
CONFIG

SIP  
CONFIG



# SYSTEM CONFIG

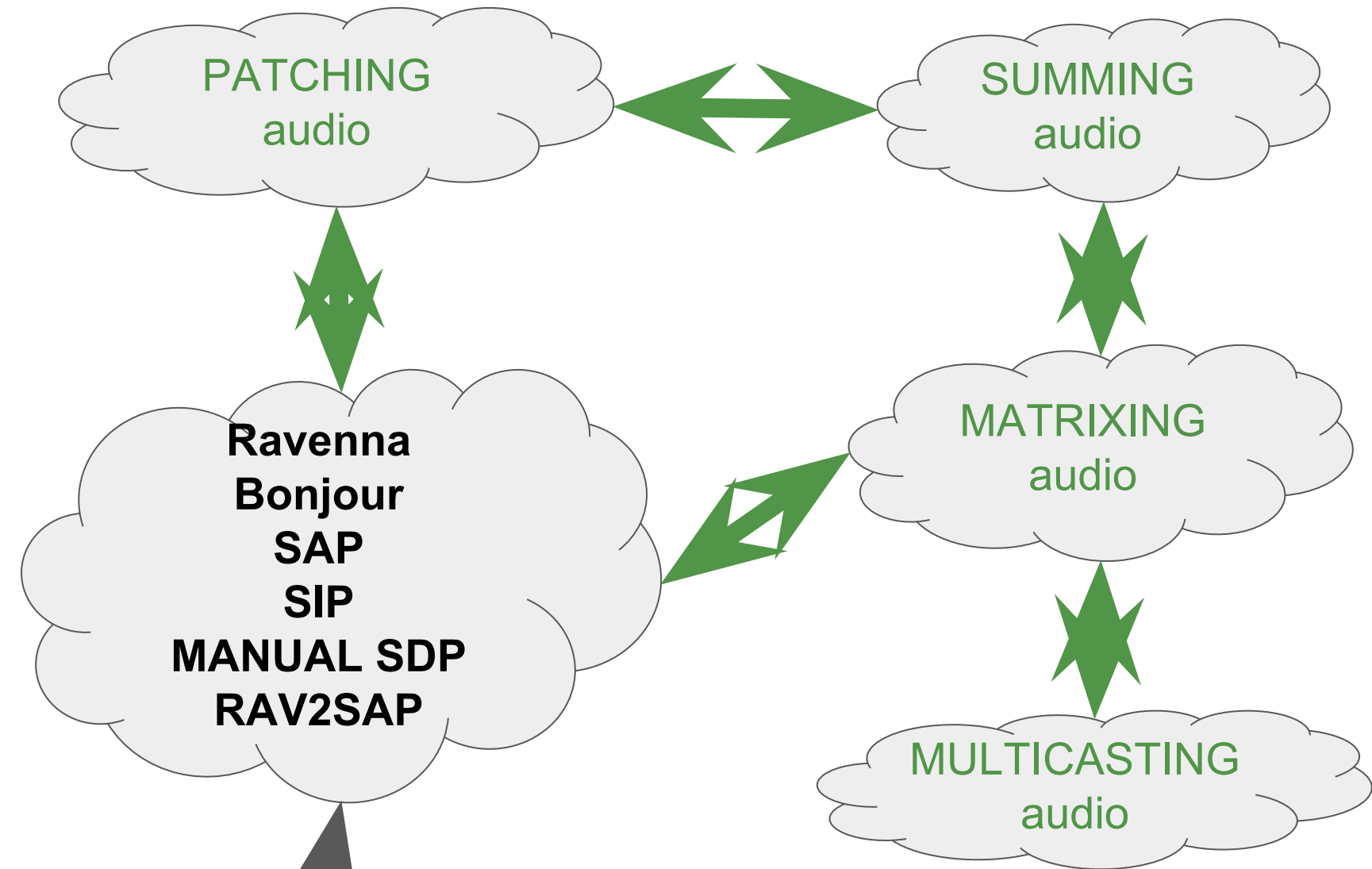
Automation Controller

# ROUTING REQUEST

**API:**  
**STREAM NAME**

- TALENT
- MIC
- LINE
- AES
- CITY:BUILDING:FLOOR:STUDIO

**TX AES67**  
SDP:JSON



**RX AES67**  
API:CH NAME SDP:JSON

**CONSUMER NAME**

- POOL
- TALENT
- ANALOG
- AES

**Channel 1** Receiving (no sync)

Name: Headphone Left

Source stream: sap:TORONTO preMO-221 (00-52-...)

Channel number: Channel 1

Show advanced settings

**Manual SDP** **Show current SDP** > >>

Stream name	Activate	SDP	Status	Multicast address	port
Stream 1	<input checked="" type="checkbox"/>	<a href="#">Show</a>	Transmitting	239.1.11.135	5004

Use automatic configuration

# REMOTE CONTROL OF DEVICE APPLICATION FUNCTIONS

Control & Monitoring
Discovery
Connection Management
Session Description
Transport
Quality of Service
Encoding & Streaming
Synchronization
Media Clock

- device and channel names
- network settings
- sample rate, bit rate
- Status and health monitoring
- Status reporting

AES70  
Ember+  
Ember  
Midi  
SNMP  
RS232  
RS485 bus

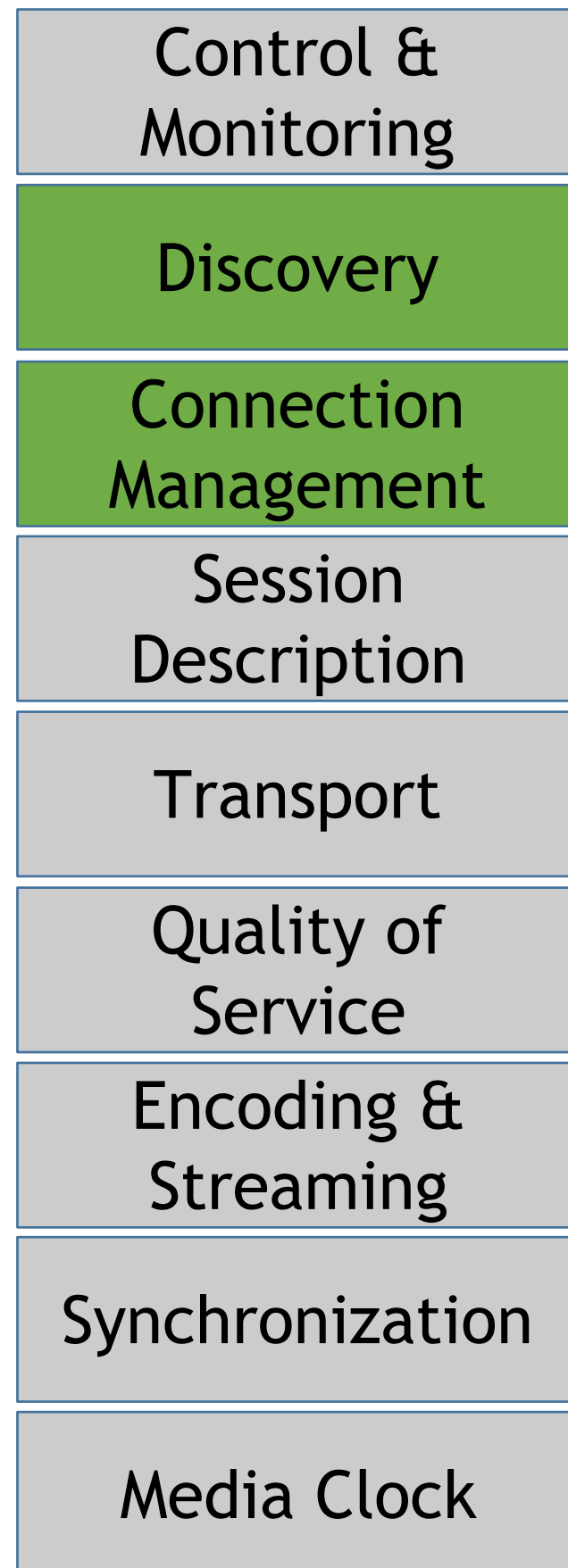
Copperlan  
VISA  
Eucon  
Mackie HUI  
JSON  
Service Directory  
Made up - UDP / TCP

# DISCOVERY.AES67.audio

Control & Monitoring
<b>Discovery</b>
Connection Management
Session Description
Transport
Quality of Service
Encoding & Streaming
Synchronization
Media Clock

- SIP - Session Initiation Protocol
- SAP - Session Announcement Protocol
- BONJOUR - apple zeroconf
- AES70 - Connection Management MDNS
- NMOS -
- SNMP
- KLV within stream
- FAST METADATA (SDP extension)

# NMOS



- Stream identification
- Timing
- Relationships
- Discovery and registration
- Connection management
- JSON data structures



# SESSION DESCRIPTION PROTOCOL

Control & Monitoring
Discovery
Connection Management
Session Description
Transport
Quality of Service
Encoding & Streaming
Synchronization
Media Clock

v=0 v= (protocol version number, currently only 0)

o=- 2 0 IN IP4 192.168.110.178 (originator and session identifier : username, id, version number, network address)

s=WardBeck 32ME s= (session name : mandatory with at least one UTF-8-encoded character)

t=0 0 t= (time the session is active)

a=clock-domain:PTPv2 0 a=\* (attribute Clock version and domain)

m=audio 50000 RTP/AVP 98 m= (media name and transport port and IP address / RTP profile 98)

c=IN IP4 239.30.178.1/100 c=\* (connection information/TTL)

a=rtpmap:98 L24/48000/8 a=\* (attribute RTP profile / Bit Rate / Sample Rate / Channels)

a=framecount:48 a=\* (attribute )

a=sync-time:0 a=\* (attribute required by ST2110-30)

a=ptime:1 a=\* (attribute: Packet Time)

a=ts-refclk:ptp=IEEE1588-2008:EC-46-70-FF-FE-00-8F-C8:0 a=\* (attribute)

a=mediack:direct=0 a=\* (attribute)

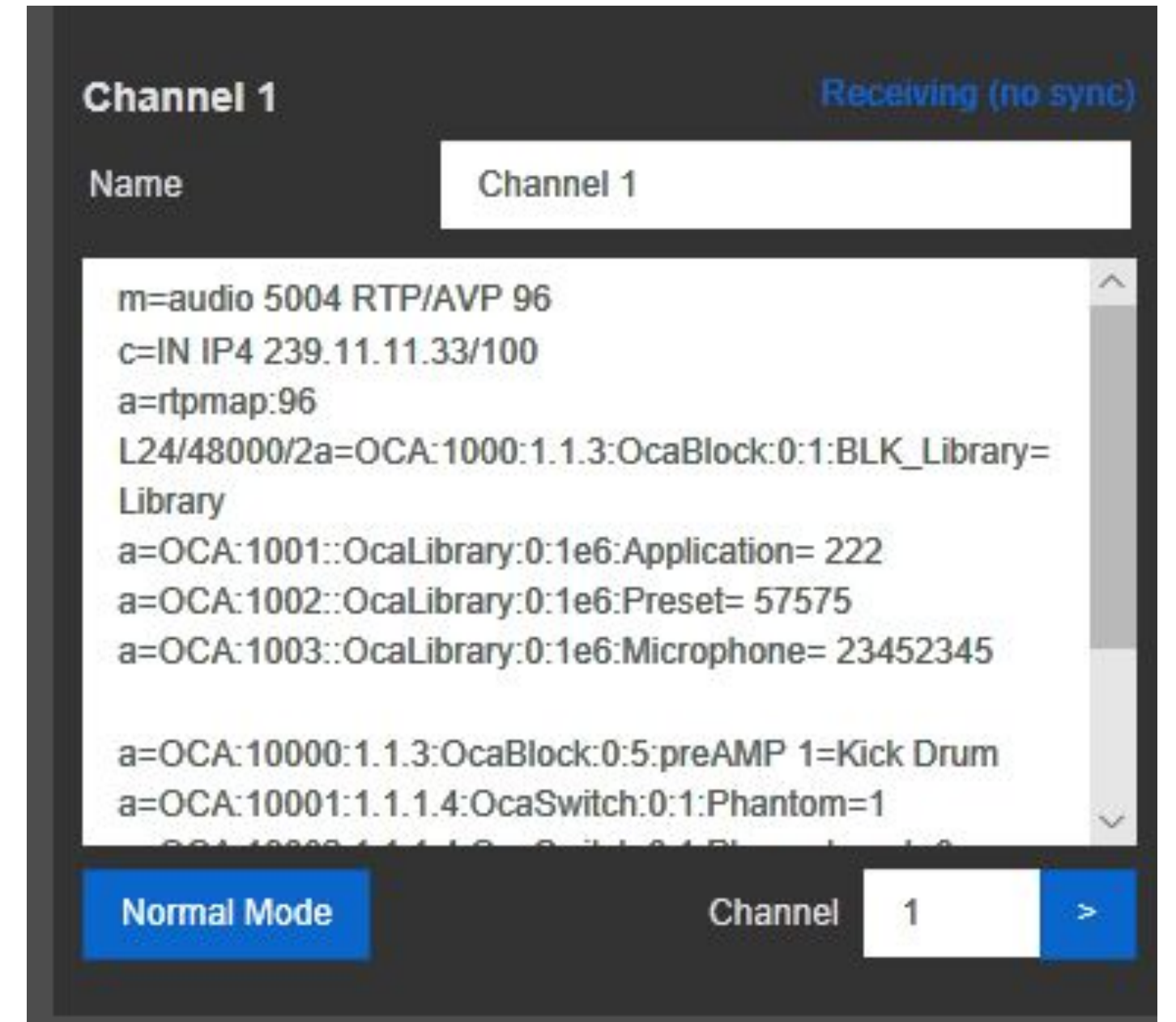


# OCA EXTENDED SDP TLV - KLV

Control & Monitoring
Discovery
Connection Management
<b>Session Description</b>
Transport
Quality of Service
Encoding & Streaming
Synchronization
Media Clock

```
a=OCA:1000:1.1.3:OcaBlock:0:1:BLK_Library=Library
a=OCA:1001::OcaLibrary:0:1e6:Application= 222
a=OCA:1002::OcaLibrary:0:1e6:Preset= 57575
a=OCA:1003::OcaLibrary:0:1e6:Microphone=23452345
a=OCA:10000:1.1.3:OcaBlock:0:5:preAMP 1=Kick Drum
a=OCA:10001:1.1.1.4:OcaSwitch:0:1:Phantom=1
a=OCA:10002:1.1.1.4:OcaSwitch:0:1:Phase_Invert=0
a=OCA:10003:1.1.1.4:OcaSwitch:0:1:Input(MIC/LINE)=0
a=OCA:10005:1.1.1.5:OcaGain:0:63:Pre Amp Gain=45
a=OCA:10100:1.1.3:OcaBlock:0:3:DSP Input 1=2
a=OCA:10101:1.1.1.4:OcaSwitch:0:1:HPF Enable=1
```

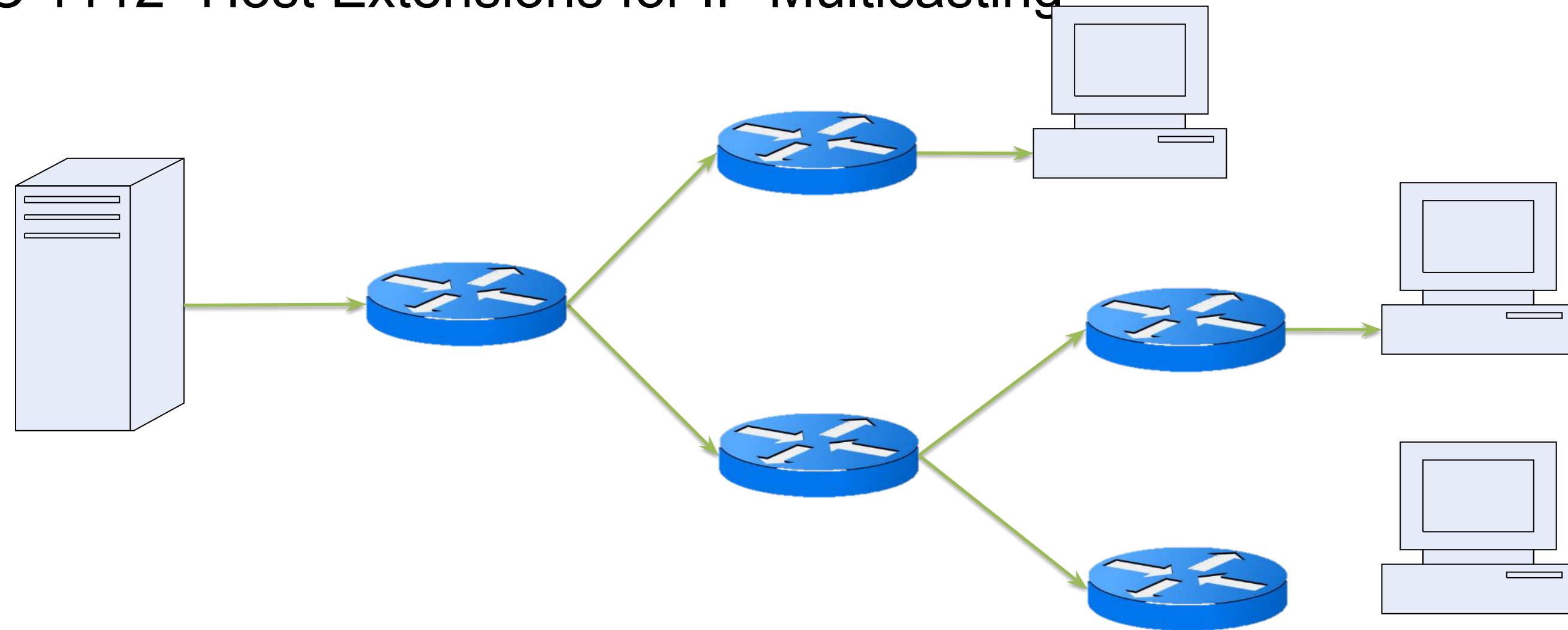
```
a=OCA:1000=Library
a=OCA:1001= 222
a=OCA:1002=57575
a=OCA:1003= 23452345
a=OCA:10000=Kick Drum
a=OCA:10001=1
a=OCA:10002=0
a=OCA:10003=0
a=OCA:10005=45
a=OCA:10100=2
a=OCA:10101=1
```



# http://MULTICASTING.audio

## Network Layer(3) : (one-to-many)

- Only one connection per stream on transmitter side
- Switches copy packets as required
- Network traffic increases on last (closest to receiver) segment(s) of network path only
- IETF RFC 1112 “Host Extensions for IP Multicasting”



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<b>Transport</b>
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Encoding & Streaming
Synchronization
Media Clock

# QOS - DIFFERENTIATED SERVICES (DIFFSERV)

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Class name	Traffic type	Default DiffServ class (DSCP decimal value)
<b>Clock</b>	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	<b>EF (46)</b>
<b>Media</b>	RTP and RTCP media stream data	<b>AF41 (34)</b>
<b>Best effort</b>	IEEE 1588-2008 signaling and management messages. Discovery and connection management messages.	<b>DF (0)</b>

# IEEE1588 Precision Time Protocol (PTP)

## PTP Domain

Logical grouping of clocks that are synchronised to each other using PTP, but may not be synchronised to other clocks in another domain

### Grandmaster clock (GM)

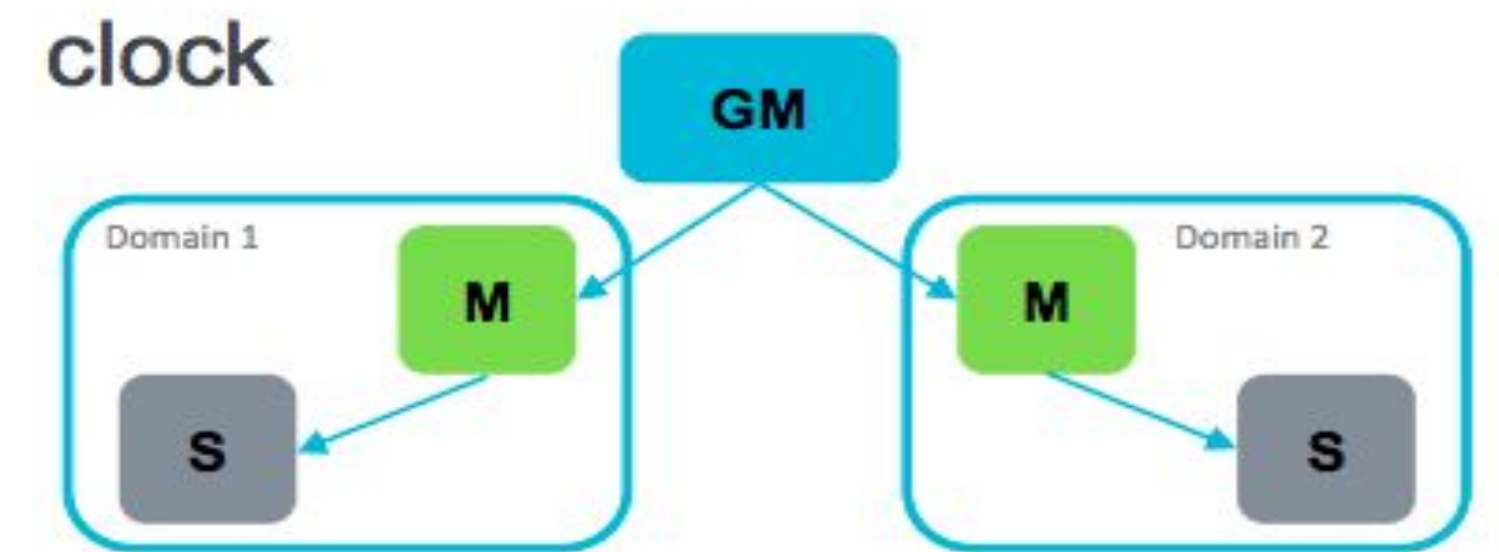
A grandmaster clock is the highest-ranking clock within its PTP domain and is the primary reference source for all other PTP elements.

### Master Clock (Mode)

A clock that is the source of time to which all other clocks in that domain are synchronised

### Slave clock

A slave clock receives the time information from a master clock by synchronizing itself with the master clock. It does not redistribute the time to another clock. In the data center, servers are typically PTP slave clocks.



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# IEEE1588 Precision Time Protocol (PTP)

## Ordinary clock

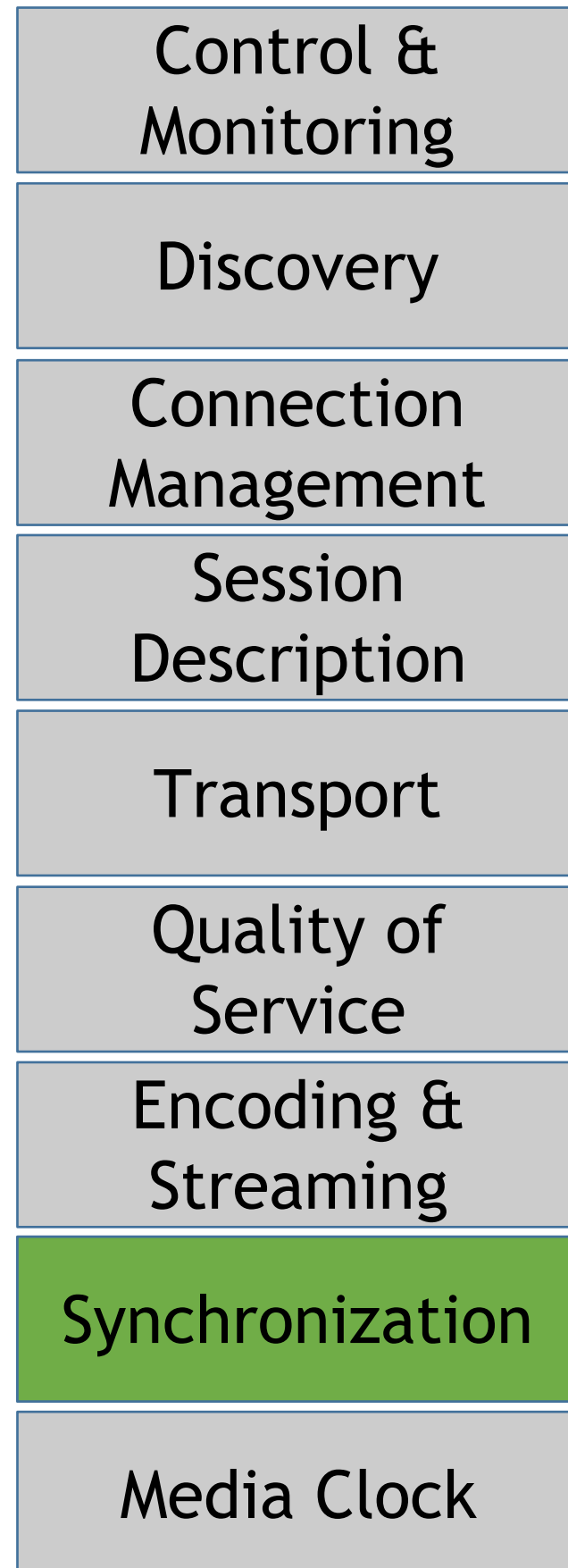
An ordinary clock is a PTP clock with a single PTP port. It could be a master clock (grandmaster) or a slave clock.

## Boundary clock (BC)

A boundary clock is the intermediary device between a PTP grandmaster and its PTP slave clients. It has multiple PTP ports in a domain and maintains the time scale used in the domain. Different ports on the boundary clock can be master ports or slave ports. The boundary clock terminates the PTP flow, recovers the clock and time stamp, and regenerates the PTP flow. Effectively, there is a slave port to recover the clock and master ports to regenerate the PTP packets.

## Transparent clock (TC)

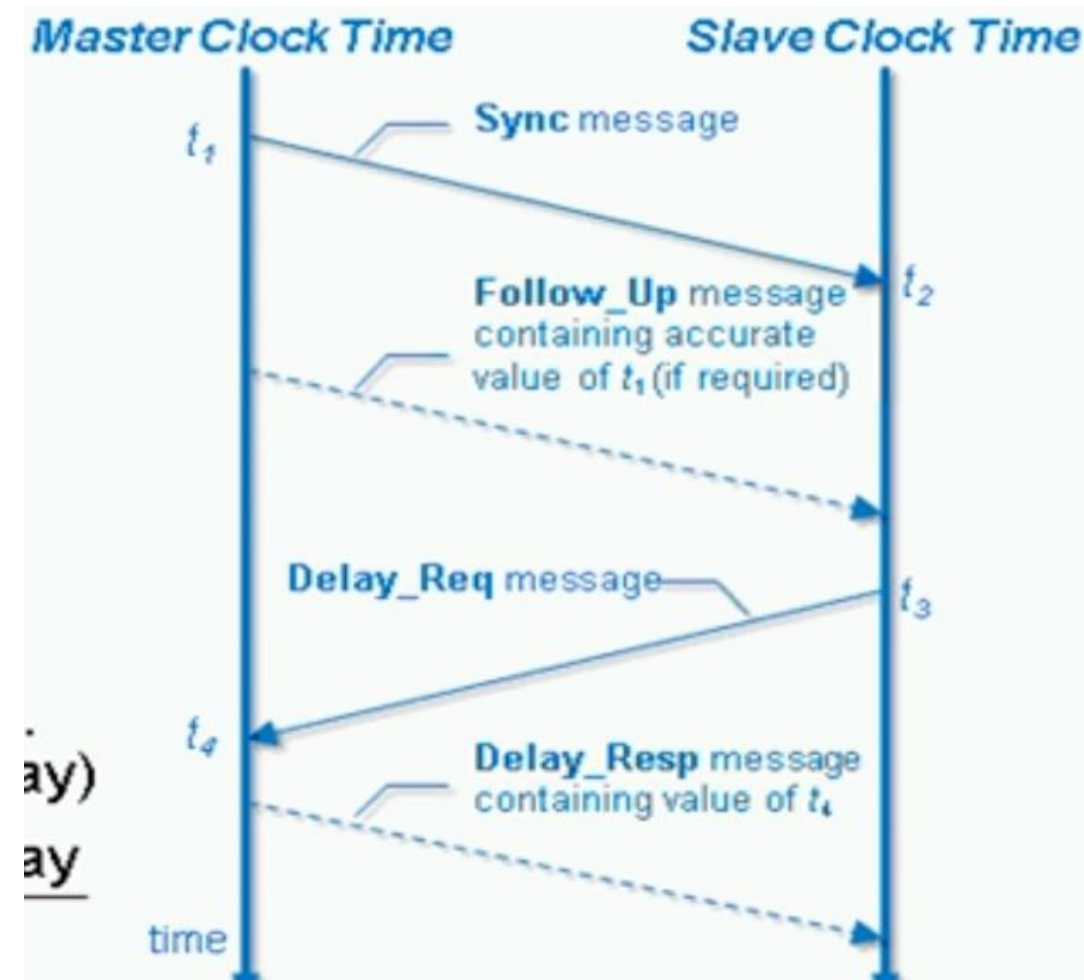
A transparent clock measures the time needed for a PTP event message to transit the device and then compensates for the packet delay.



# AES67 PTP MEDIA PROFILE:

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Delay Mechanism: **E2E**  
Announce Interval: **2 secs**  
**Announce Timeout: 3 secs**  
**Sync Interval: 1/8 sec**  
**Delay Req Interval: 1 sec**  
Peer Delay Req Interval: **1 sec**  
Priority 1: **128**  
Priority 2: **128**  
Domain: **0**  
Slave Only: **Disabled**



**Announce Timeout** – time to wait for an announce message and is a multiple of announce intervals. No announce messages are received within the timeout interval, the unit will assume the role of grandmaster.

**Sync Interval** – Rate that one sync message is sent. The sync and follow up messages are sent from the master to the slave to determine the difference in clock frequency. This information is used in conjunction with the network delay to synchronise the clocks.

**Delay Req Interval** – rate at which a slave clock sends delay request messages to the master. The delay request message allows a slave device to calculate the network delay from the slave to the master. This option is only valid when using the E2E delay mechanism.

# SMPTE ST 2059-2:2015 "SMPTE PROFILE FOR USE OF IEEE-1588 PRECISION TIME PROTOCOL IN PROFESSIONAL BROADCAST APPLICATIONS"

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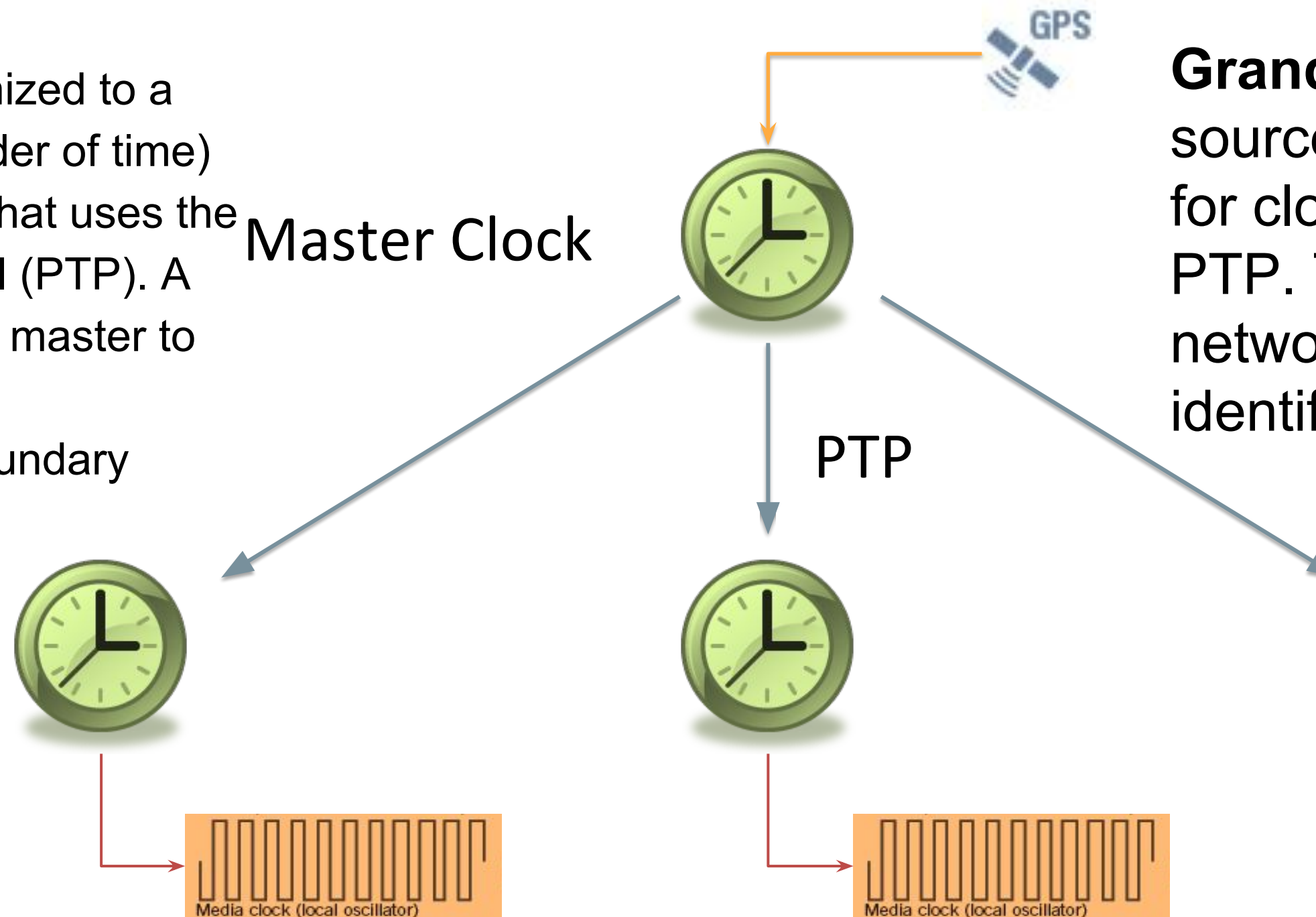
## Slave Clock

A clock that is synchronized to a master clock (the provider of time) within an environment that uses the Precision Time Protocol (PTP). A slave may, in turn, be a master to another clock and may simultaneously be a boundary clock.

Slave Clocks (nodes)

Media Clocks

**Media clock:** The clock used by senders to sample and receivers to play digital media streams. The media clock for audio streams reads in units of samples.

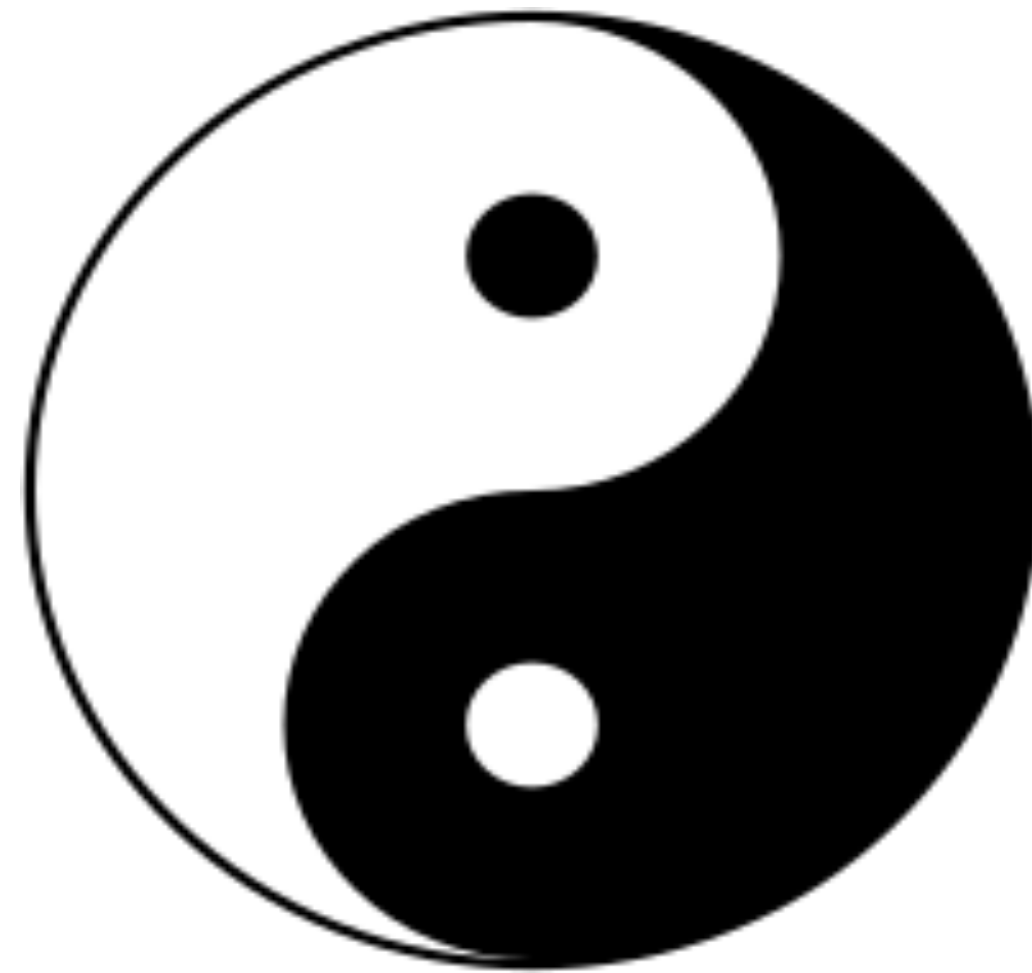


**Grandmaster:** The master source of synchronization for clock distribution via PTP. The grandmaster is a network device and is identified by an EUI-64.

# NETWORKING . audio



AES67.audio



OPEN CONTROL ARCHITECTURE

AES70.audio

