



# Media over IP

Pavilion at AES NY 2022



AES67 WAN  
Transport  
Utilizing the  
Cloud

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**Media over IP**

Pavilion at AES NY 2022

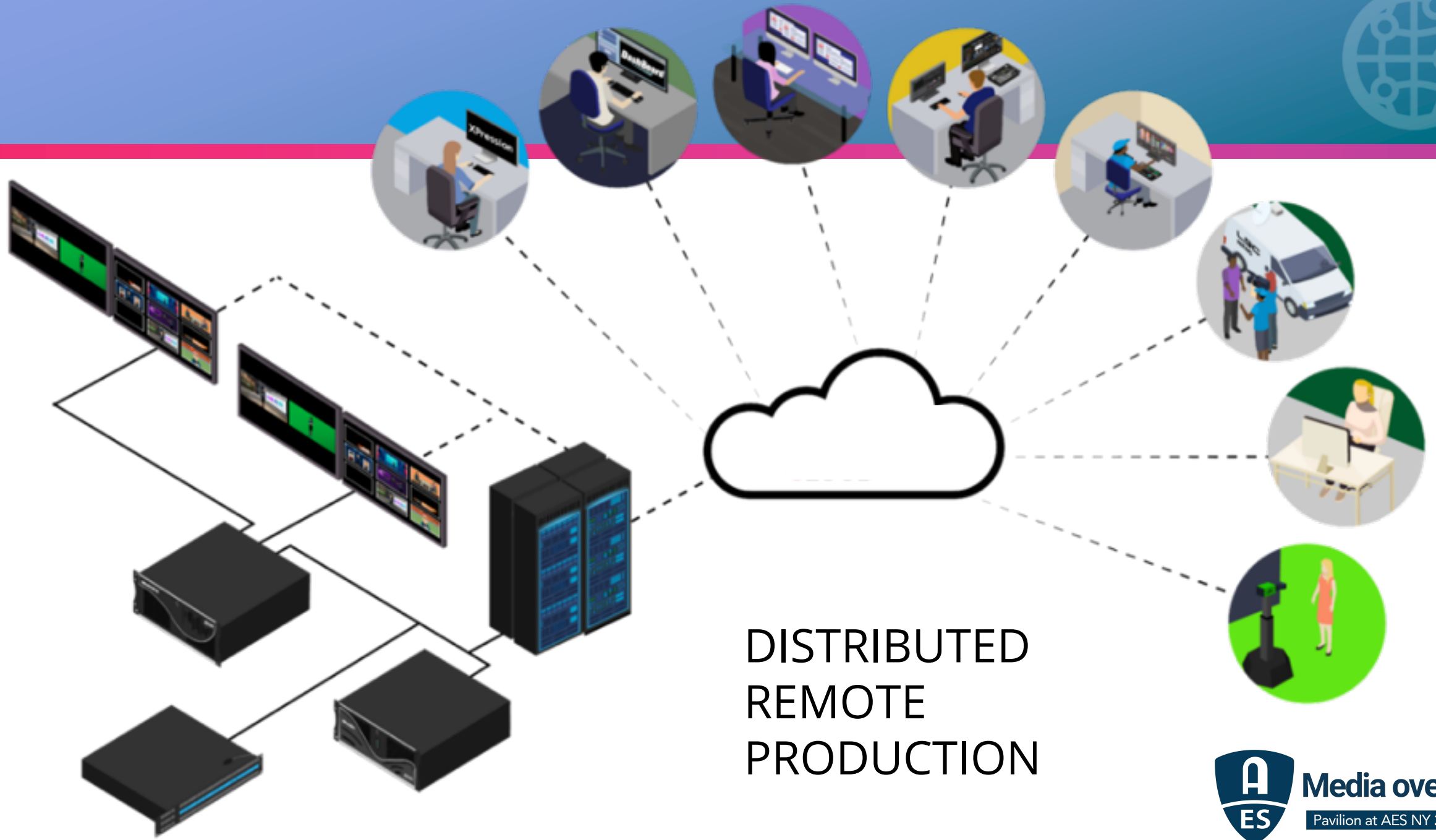
# TRADITIONAL REMOTE PRODUCTION



OUTSIDE BROADCAST VAN



REMI MODEL



# DISTRIBUTED REMOTE PRODUCTION



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**ROSS**<sup>®</sup>

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## Global AES67 over WAN Demo

- 2 continents
- 3 companies
- 4 sites

# PURPOSE



We embarked on this proof-of-concept demo to answer the following questions:

- Can RAVENNA / AES67 traffic be sent over the public cloud infrastructure?
- Across long distances?
- And maintain interoperability between companies?
- How?
- What challenges need to be overcome?



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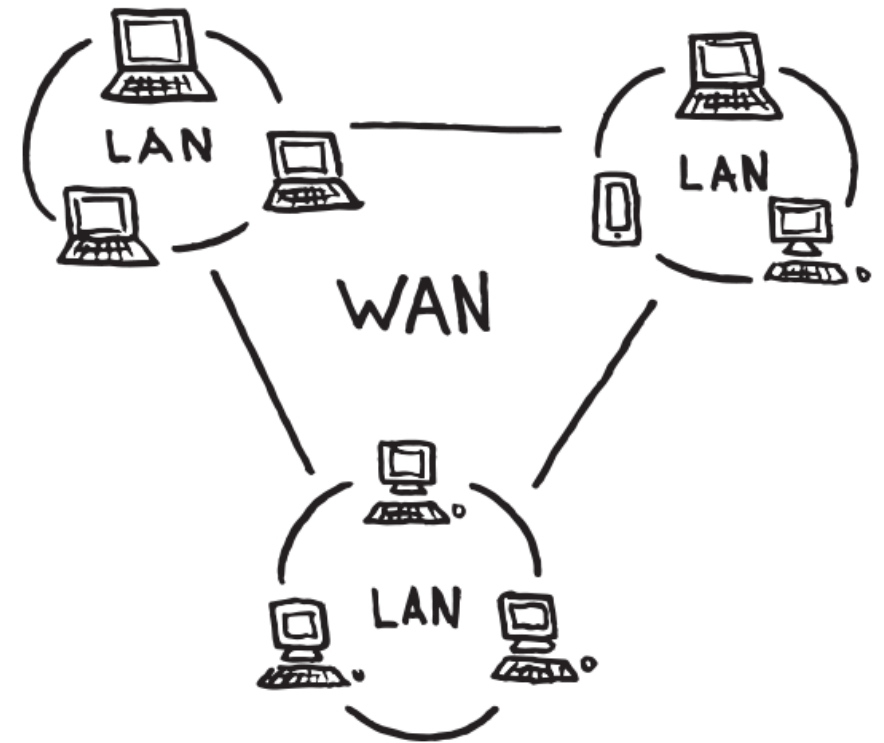
Pavilion at AES NY 2022



# SOME BACKGROUND ON AES67



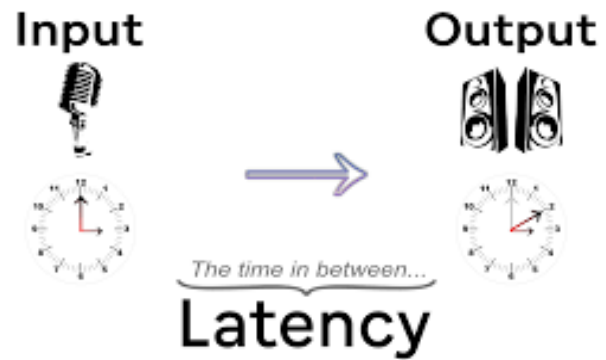
- Designed for local area networks synchronized to PTP that don't drop packets
- Now being used over long distances in WAN applications across private dedicated infrastructures using fiber (even though it was not contemplated by the standard)
- Public or "best-effort" networks tend to be congested, suffer from packet loss, and have increased latency due to re-transmissions



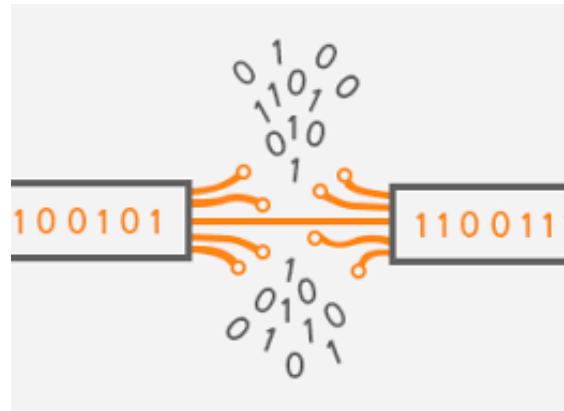
# PUBLIC INFRASTRUCTURE CHALLENGES



## Latency and Packet Jitter



## Packet Loss



## Timing and Synchronization





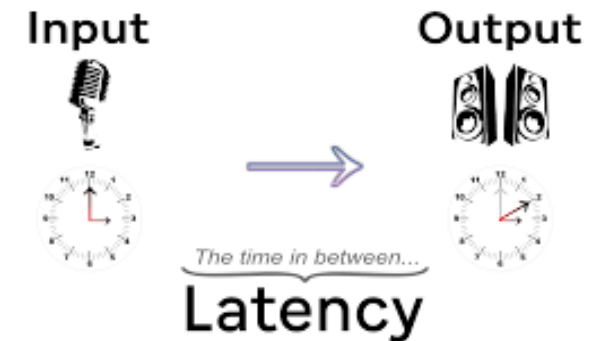
# CHALLENGE: LATENCY AND PACKET JITTER



- RAVENNA receivers are designed to handle increased packet jitter & latency
- The large receiver buffers can compensate for added delay
- RAVENNA specifies receiver buffers must handle a minimum delay of 20 msec; AES67 only requires 3 msec, but also recommends 20 msec
- Most well-designed AES67 and RAVENNA solutions have even bigger buffers
- The AES Standard Committee working group SC-02-12-M is focused on AES67 over WAN; a key recommendation is to increase the buffer size within devices
- Solutions can also be manually tuned to the network delay



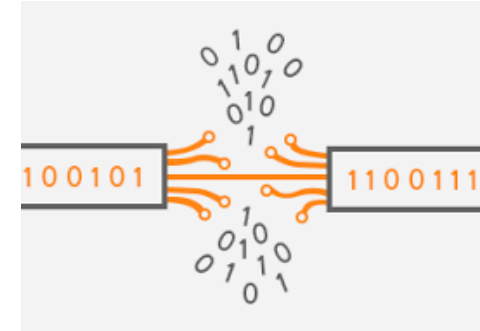
*Increased latency of the public network can be handled by the larger buffers used in well-architected AES67 and RAVENNA solutions*



# CHALLENGE: PACKET LOSS



Leverage transport protocols designed for reliable transmission of media over lossy networks with low latency and high quality



- Secure Reliable Transport (SRT)
  - Open source protocol developed by Haivision and backed by the SRT Alliance
- Zixi
  - Widely used proprietary solution developed by company of the same name
- Reliable Internet Stream Transport (RIST)
  - An open source, open specification protocol intended to be more reliable than SRT and an alternative to proprietary solutions like Zixi, VideoFlow, Qvidium etc.



*We are using SRT for the proof-of-concept demo but any of these will work*



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# CHALLENGE: TIMING AND SYNCHRONIZATION



- At each location, a PTP GM running SMPTE 2059 profile is synchronized to GPS
- The equipment at each site is locked to PTP locally
- PTP packets are not sent across the WAN as this is not currently practical (packet jitter is too high)



*Since the PTP GM at each location is locked to GPS, synchronization is maintained across the WAN*



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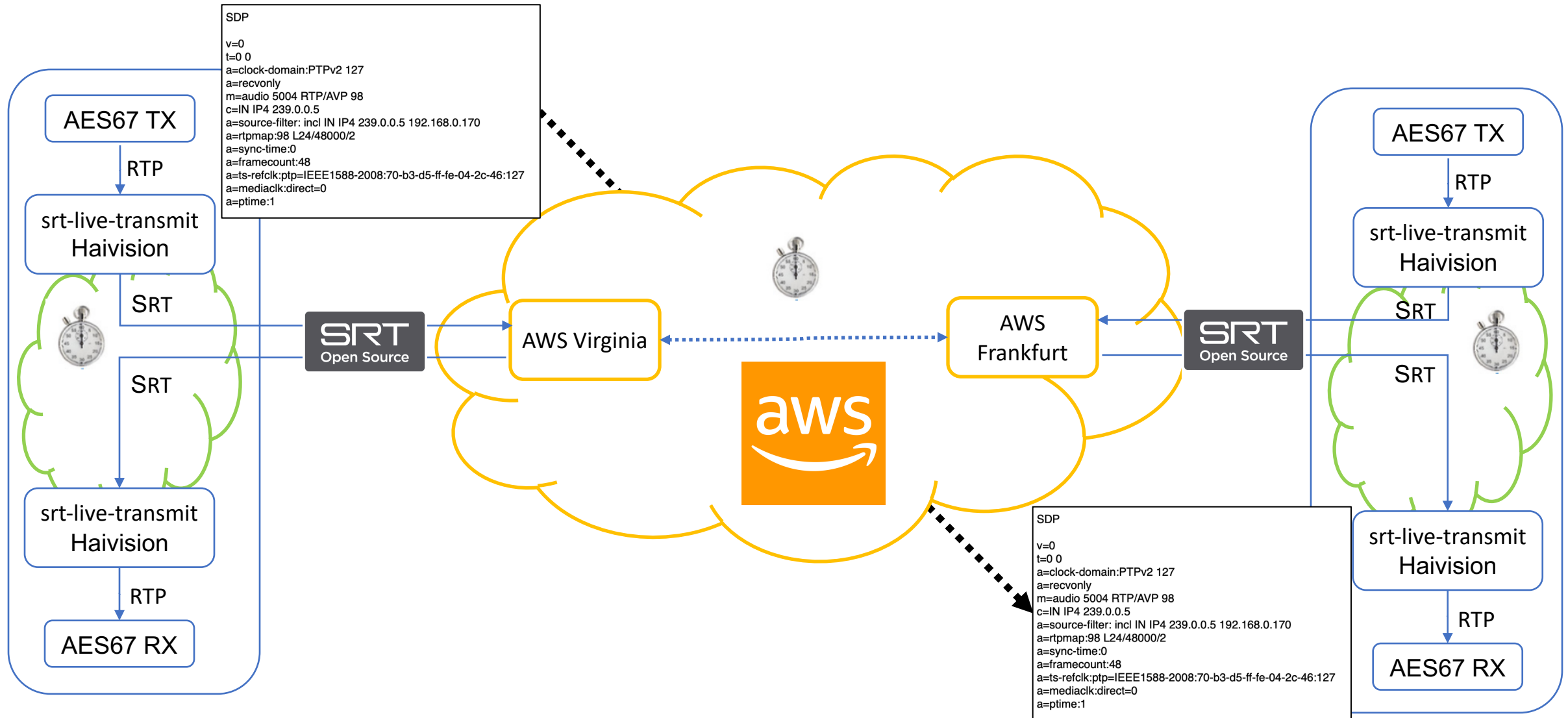
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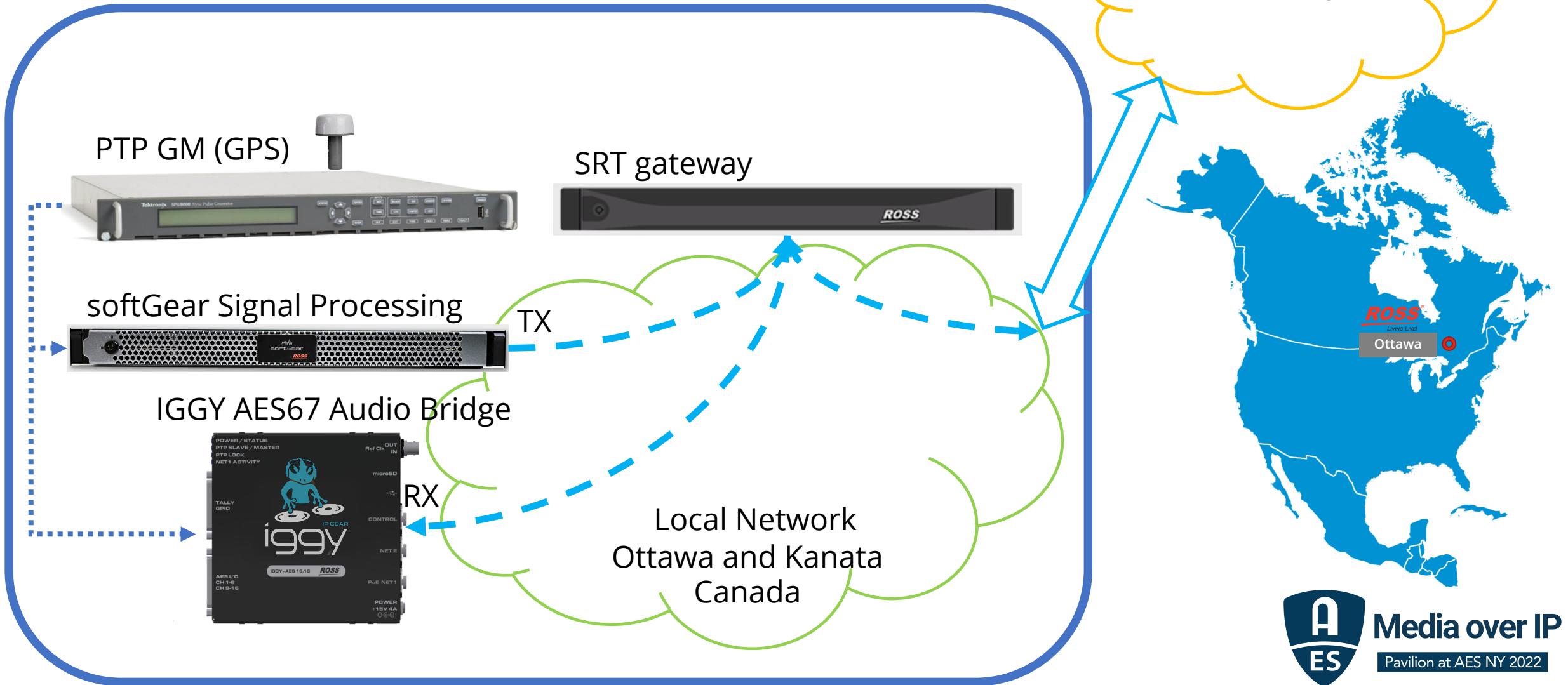
## Global AES67 over WAN Demo

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# BIG PICTURE SUMMARY OF AES67 OVER WAN DEMO

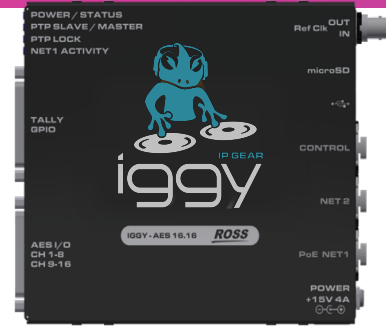


# NORTH AMERICA- OTTAWA





# OTTAWA SETUP



**Dashboard by Ross Video**

File Edit Layouts Views Window Help

PanelBuilder Edit Mode

Switchboard Global Labels

BAP - Dolby RTLL BAP - Nielsen Wate... BAP - DD DDP Encoder NWE-IP - Slot 6 - ...

Mem: Current User: (none)

Basic Tree View

Filter:

- BAP
- Frame
- Loudness-Control
- NWE-IP
- NWE-IP
- NWE-IP-101
- Dashboard Services

Slot 6: AES67 Player

Card state: OK

Connection: ONLINE

Product: AES67 Player

Orchestration Performance Status

Supplier: Ross Video, Limited

Product Name: AES67 Player

Version: 1.1.0

Build Timestamp: Mon Mar 23 15:00:03 UTC 2020

SOP SDK Version: 1.1.2

Signal Source

Ethernet Interface: NET1

IP Address: 239.111.48.106

Port: 5004

Ethernet Interface 2: NET2

IP Address 2: 239.111.48.106

Port 2: 5000

Audio Channels: 2

Codec: L24

Source File: Ross\_demo\_BACH.raw

Total Samples: 103032371

OK

Apply Changes

Packet Time: 1000us

Timestamp Option: Adjust from current

Timestamp Adjustment: 0

Source: Input

Tone Gen Freq: 1000

Stream: Disabled Enabled

Refresh Upload Reboot Close

Welcome Initial Setup Connections Presets Audio Gain Advanced

## Advanced Settings and Status

ROSS

Status Device Setup Ethernet I/O Receivers Destinations Senders Discovery Timing TSL/Rosstalk Alarms Logs

Status

X-Connect

NET Bandwidth Allocation\*

NET 1: 3.3 Mb/s

NET 2: 0.0 b/s

\*Bandwidth allocation bars represent allocated bandwidth only. They are not a reflection of actual traffic on the link.

Status	Name	Disconnect			
AES A: OK	Destination 1	Disconnect			
Audio: OK	From_WAN.audioA	239.1.1.135	5004	NET 1	20000
AES B: Not In Use	Destination 2	Disconnect			

Refresh Upload Reboot Close

# EUROPE- LAUSANNE AND GRENOBLE



Anubis AD/DA + Mixer

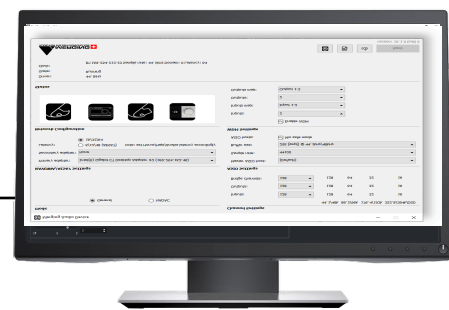


PTP GM



Local Area Network  
Setup In Lausanne, Switzerland  
and Grenoble, France

SRT  
Gateway



MAD Virtual Sound Card



# LAUSANNE AND GRENOBLE SETUP



**Merging Audio Device**

**Mode**  
 General  NADAC

**Channel Settings**  
44.1/48k 88.2/96k 176.4/192k 352.8/384k/DSD

Inputs:	128	128	64	32	16
Outputs:	128	128	64	32	16
Bridge channels:	128	128	64	32	16

**RAVENNA/AES67 Settings**  
Primary adapter: Intel(R) Gigabit CT Desktop Adapter #2 (169.254.162.46)  
Secondary adapter: None  
Latency:  6/12/48 (AES67) note: set Horus/Hapi/Anubis latency accordingly  
 16/32/64

**ASIO Settings**  
Master ASIO host: (Default)  
Sample rate: 44100  
Buffer size: 256 [smp] @ 44.1kHz/48kHz  
ASIO hosts:  Mix safe mode

**WDM Settings**  
 Enable WDM

Inputs:	2
Inputs map:	Input 1-2
Outputs:	2
Outputs map:	Output 1-2

**Network Configuration**

**Status**  
Driver: 44.1kHz  
State: Running  
Clock: IP:169.254.213.23 Sample rate: 44.1kHz Domain: 0 Latency: 64

version: 10.1.0 build 0

**Configuration**

IO: Stream  
Label:  
Description:  
Source: manual://RavennaWanTest  Manual

```
v=0
o=- 1 0 IN IP4 192.168.1.135
s=RavennaWanTest
c=IN IP4 239.9.9.1/15
t=0 0
a=clock-domain:PTPv2 0
a=ts-refclk:ptp=IEEE1588-2008:D0-A4-B1-FF-FE-00-05-9A:0
a=mediaclock:direct=0
m=audio 5004 RTP/AVP 98
c=IN IP4 239.9.9.1/15
a=rtptime:98 L24/48000/2
a=source-filter:incl IN IP4 239.1.1.135
192.168.1.135
```

Delay (samples): 16000 (~333.3 ms)  
 accept source locked to any PTP Master  
 accept source with lower channel count

Channels: Channel count 2  
Count adapted

**Session Info**

**Session status** Connected  
**RTP status** Receiving

Session name: RavennaWanTest  
Playout delay: 16000 (~333.3 ms)  
RTSP Host:

**Interface 1**

**RTP status** 0x10: receiving RTP packets  
Clock domain: PTPv2 0  
Address: 239.9.9.1/15  
Payload: 98 L24/48000/2

▶ SDP

# EUROPE- MITTWEIDA



PTP GM

PRODIGY.MP Audio I/O  
Interface with SRT Gateway



Mittweida, Germany Setup



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# MITTWEIDA SETUP



**01 - INPUT STREAM SETTINGS**

Activate Stream:

Stream input: Stream state offset max (samples): **19788**

Backup Stream: Stream state offset min (samples): 15815

Backup Stream Timeout: Stream state ip address src NIC 1: -

Stream name: Stream state ip address src NIC 2: 239.67.2.1

Stream state: -

Stream state messages: -

Stream state offset max (samples):

Stream state offset min (samples):

Stream state ip address src NIC 1:

Stream state ip address src NIC 2:

Offset fine:

Offset in samples (458.33 ms): **22000**

Start channel: 1

Discovery protocol: Manual configuration

Stream name (manual): Ottawa

Number of channels: 2

RTP payload ID: 93

Audio format: L24

Media offset: 0

NIC 1		NIC 2	
Dest IP address:	230.1.1.4	Dest IP address:	230.67.2.1
SSM (Source Specific Multicast):	-	SSM (Source Specific Multicast):	-
Src IP address:	0.0.0.0	Src IP address:	172.16.100.201
RTP dst port:	6054	RTP dst port:	6054
RTCP dst port:	6056	RTCP dst port:	6056

**01 - INPUT STREAM SETTINGS**

Activate Stream:

Stream input: Stream state offset max (samples): **14893**

Backup Stream: Stream state offset min (samples): 13009

Backup Stream Timeout: Stream state ip address src NIC 1: -

Stream name: Stream state ip address src NIC 2: 239.67.2.1

Stream state: -

Stream state messages: -

Stream state offset max (samples):

Stream state offset min (samples):

Stream state ip address src NIC 1:

Stream state ip address src NIC 2:

Offset fine:

Offset in samples (333.33 ms): **16000**

Start channel: 1

Discovery protocol: Manual configuration

Stream name (manual): Lausanne

Number of channels: 2

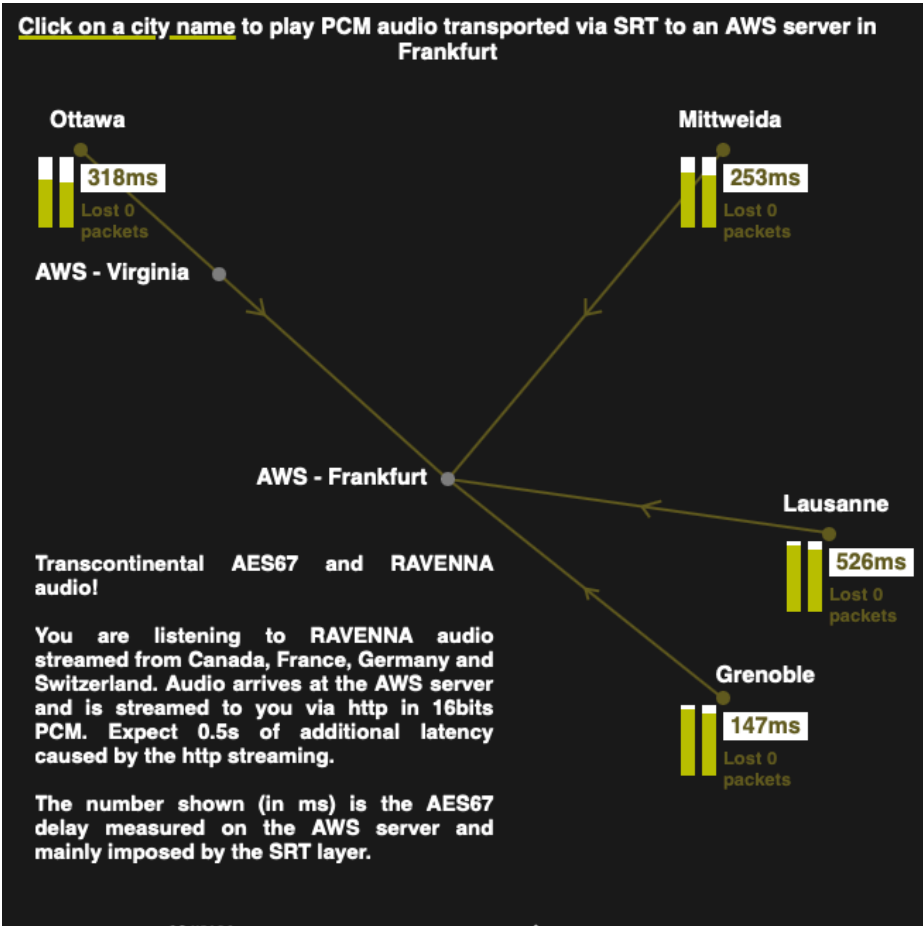
RTP payload ID: 93

Audio format: L24

Media offset: 0

NIC 1		NIC 2	
Dest IP address:	230.1.1.4	Dest IP address:	239.67.2.1
SSM (Source Specific Multicast):	-	SSM (Source Specific Multicast):	-
Src IP address:	0.0.0.0	Src IP address:	172.16.100.201
RTP dst port:	6054	RTP dst port:	6054
RTCP dst port:	6056	RTCP dst port:	6056

# GLOBAL AES67 OVER WAN DEMO WEB PAGE



- A special version of srt-live-transmit is used to send the AES67 payload to the local loop (on top of the regular behaviour)
- Utilizes a monitoring application created by Nicolas Sturmel to analyze the AES67 packets and stream the audio:  
<https://github.com/nicolassturmel/aes67-web-monitor>
- The audio is streamed with a one second buffer at 16bit / 48khz, just to save bandwidth



# LESSONS LEARNED



- “Local only” PTP synchronization locked to GPS works fine
- There is packet loss but this can be managed via SRT
- Latency ranged from 200 to 600 msec
- Manual connections using SDP files
- Manual tuning of link-offset required
- Receivers need to have deep buffers or mechanisms to compensate for the network delay



# FUTURE CONSIDERATIONS



- Transporting timing through the cloud
- RAVENNA Advertisements or NMOS
- Automated handling of link-offset
- Other techniques, FEC & ST2022-7, to manage packet loss
- RIST open standard instead of SRT



QUESTIONS ?



# THANK YOU!

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