

Unfolding the mystery of automotive audio applications **for not** using Zephyr and Linux PREEMPT\_RT

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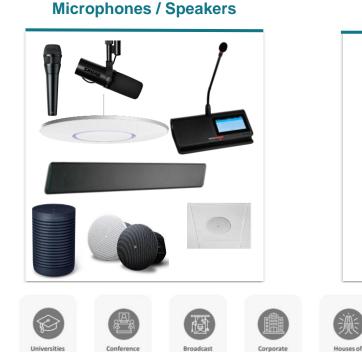


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#### **Audio Industry** – An overview of what we are talking about ?

Arenas and

Stadiums



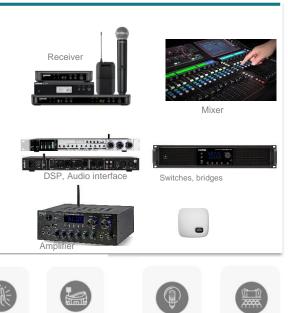
Studios

Campuses

Worship

Rooms

Receivers, DSP, Mixers, Switches, bridges, Amp



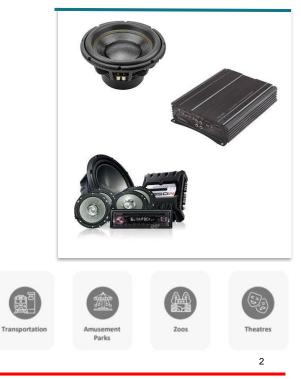
Recording

Studios

Conference

Centres

#### **Car Audio and Amplifiers**



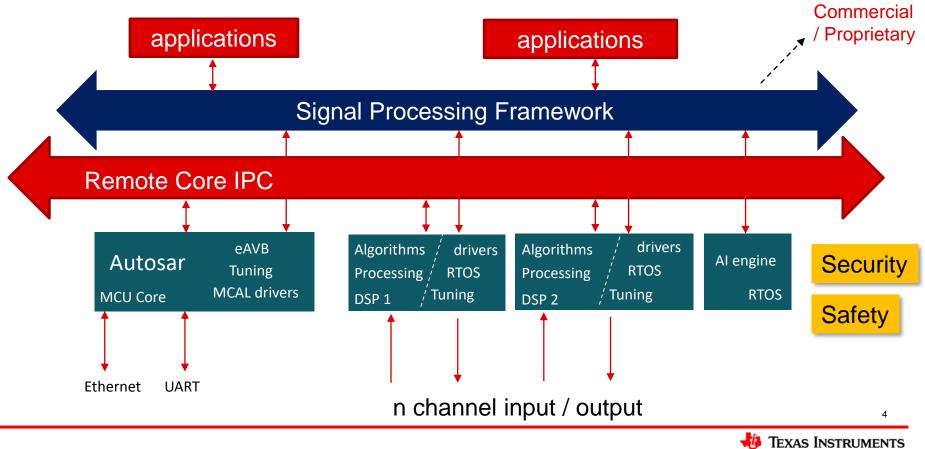


# **Scope of the session**

- Share with you the Software stack required for the key audio centric solutions.
- Tools that are required to validate, standardize, configure and benchmark.
- What are latency requirements and other expectations can RT Linux or Zephyr meet those.
- Can we build the required software stack on Zephyr or Linux.
- What are the safety constraints imposed for qualifying for the safety certifications (if any).
- Security, over the air upgrade, streaming media over the network do they have to be proprietary ?.
- How do we make the software scalable to cater to wide range of audio equipment ?



# **Typical Software Stack (Auto Audio)**



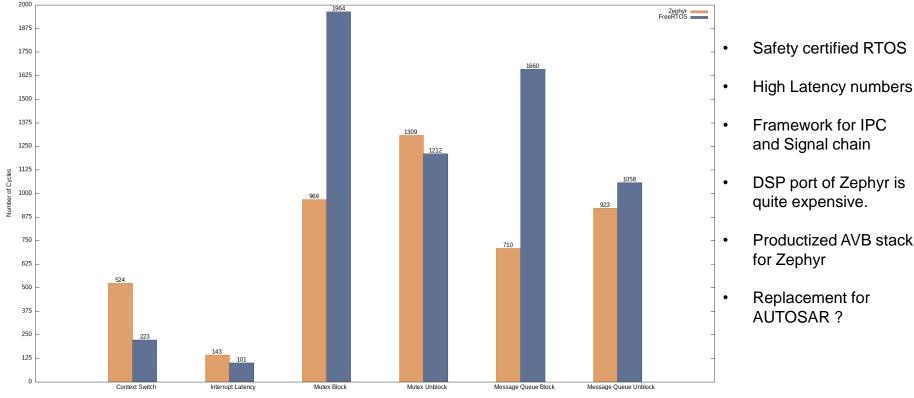
# Key care about

- Low Latency :
  - Typical Jack to Jack expected Latency numbers are 750 to 900msec for 32 channels at 48 kHz.
- Real time processing
  - Handling of live data from network interfaces and pre and post processing media streams. Runtime Tuning of parameters.
- High memory Throughput.
  - High capacity buffers using shared memory are passed to and from AVB cores to DSPs.
- Scalable Media processing
  - One core to multi-core.
  - ARM based processing to DSP based processing.
- Al enabled interfaces.
  - Both bring your data and bring your model support easy interface layers.
- UI based Tools to customize and optimize the processing.
  - Auto generated code and optimized signal chain framework.



# **Zephyr for Audio use cases**

Average Execution Times of Kernel Services (1000 Measurements)



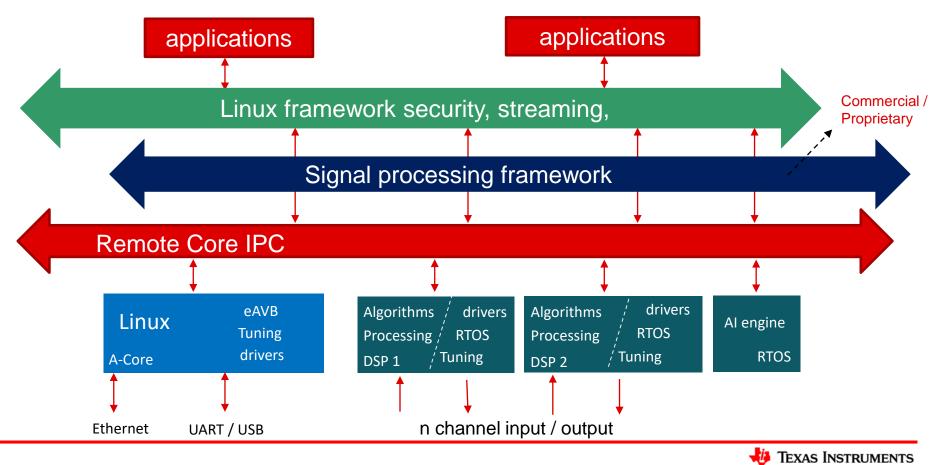
Courtesy : Measuring Real-Time Operating System Performance – Part II: Comparing FreeRTOS vs. Zephyr | Method Park by UL 6



# **General purpose - professional audio**



## **Typical Software Stack**



# Key care about

- Low Latency :
  - Typical Jack to Jack expected Latency numbers are 0.25 to 2 milli seconds.
- Key processing elements in public stack
  - Beamforming
  - Automatic mixers
  - Gain controls
  - Acoustic Echo cancelers (AEC)
  - Nosie reduction (NR)
  - Equalizer (EQ)
  - Audio conversion
  - PoE or PoE+ speaker w/ integrated amp
- UI based Tools to measure performance and optimize the processing.
  - Standardized performance measurement techniques and tools
- Scalable Standardized opens source software stack
  - One stack fits all applications security, AI enabled, network streaming, low power
  - Flexibility to leverage commercially purchased/acquired tools, frameworks where and when required.

# Linux for Audio use cases

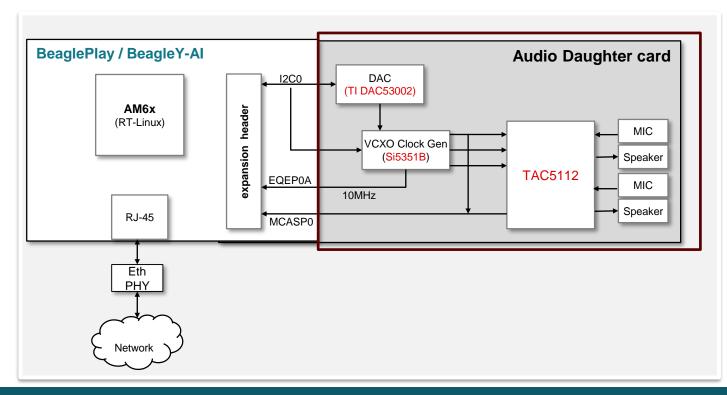
- Productize any of the Audio servers: *PulseAudio*, *JACK* and *PipeWire*.
  - JACK has been favorable to provide real-time, low-latency connections for both audio and MIDI. Will this be shortly replaced with PipeWire ?
- PREEMPT\_RT and cyclic tests :
  - a slight addition of user space service or kernel config update results in drastic change in cyclic test numbers.
- PREEMPT\_RT and Power Management :
  - Power plays an important role when the devices are battery operated but the latency and deterministic response is important as well. How can they co-exist ?
- Ease of use and Standardization
  - Selection of audio server, frameworks and fine tuning isn't standardized. A simple distro that fits all audio requirements could have been easy.
- Missing standardized signal processing framework
  - Runtime attach of accelerators and DSPs to enable remote core processing of media streams.
- Public versions of AVB stacks aren't community maintained
  - OpenAVB last patch submitted 5 years ago.
- Security while media streaming
  - Multiple options available, no clarity on audio use cases



### Let's collaborate



#### Audio HAT for AM62x/AM67x based BeagleBoard platforms



- In design phase, to be launched by BeagleBoard foundation by end of December 2024.
- Enables open source community to collaborate on audio software stack development.

#### Common Audio card that works across BeagleBoard and header on TI starter kits



#### **Collaborate & develop an Audio Stack with Beagle Audio Board :**

- Optimize latency and meet the performance requirements
  - Maintain Preempt-RT patches with kernel & userspace configs for audio configurations.
- Identify and optimize the open source audio stacks
  - Jack, NetJack2, pipewire
- Introduce OpenVx based signal chain framework for audio application
  - Remote core management and offload of processing media stream to DSPs.
- Revisit Opensource OpenAVB project
  - Enable community to revisit open source version of AVB stack.
- Collaborate to port open source audio solutions to beagle board platforms.
  - RAVENNA : is a solution for real-time distribution of audio and other media content in IP-based network environments.
  - AES67 : port and optimize the stack for low latency applications.
  - Identify the appropriate protocol stack to support security enabled streaming of real-time audio over network interfaces.
  - Explore sound open firmware : https://www.sofproject.org



## References

- <u>AWE Core OS 8.B.19 Documentation: AWE Core OS Integration Guide</u> (dspconcepts.com)
- <u>An Introduction to AVB Networking | PreSonus</u>
- <u>RAVENNA IP-based Networking Technology RAVENNA Network (ravennanetwork.com)</u>



## **Credits**

Thanks to the following team members for helping in populating this info :

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