High Performance Robotics Edge Platform

Safe, private, cost-effective robot solution

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Dallas, Texas

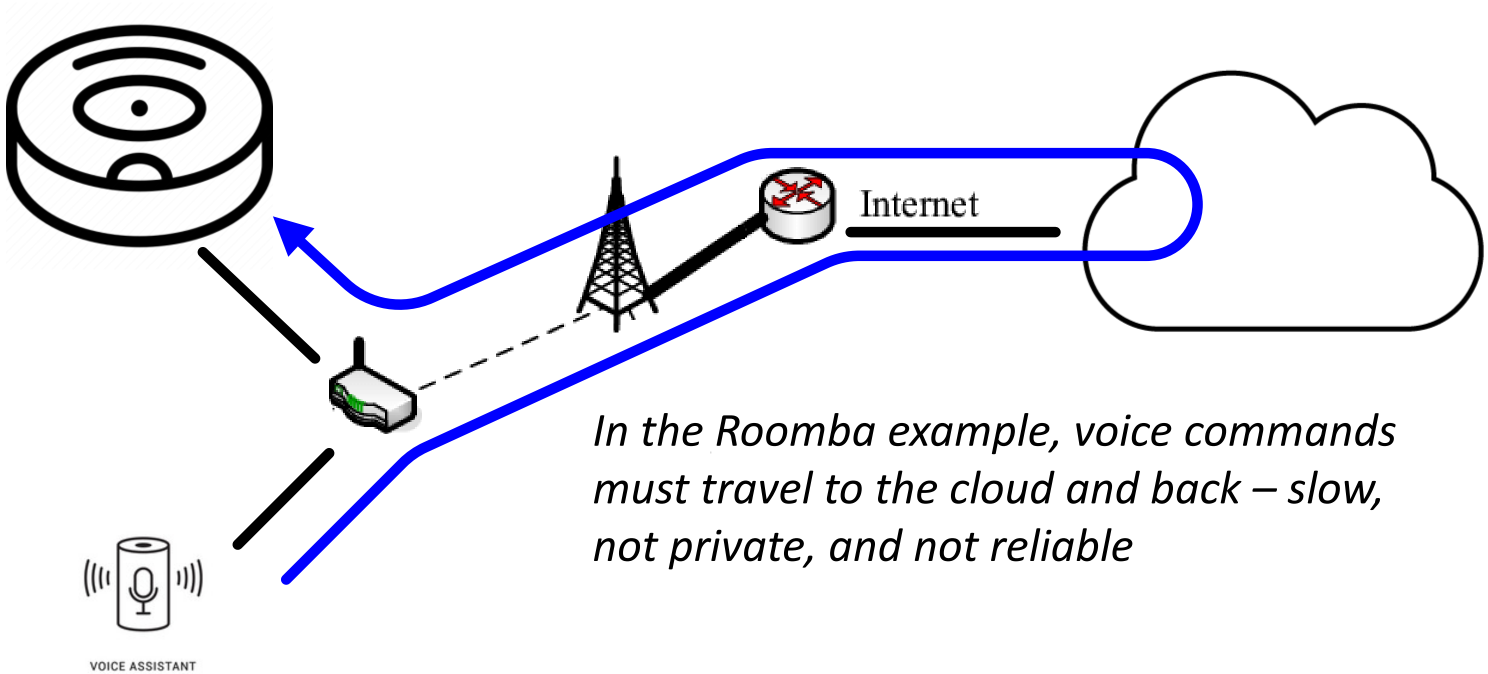
Contents

- **Problems and needs**
  - real-time voice commands: hands-free operation, safety / emergency concerns
  - data privacy
  - high performance onboard processing without costly cloud compute resources
- **Robotics edge platform**
  - software - EdgeStream™ platform optimized for high performance edge computing / analytics
  - hardware – extremely small form-factor, quad core server. Low SWaP [1]
  - containerized solution
- **Akraino SESS Blueprint**
  - Fujitsu + Ritsumeikan University
  - food preparation and handling robotics
- **Architecture and data flow**
  - core / thread data flow – ASR, vision, failure prediction
- **Signal processing**
  - noise reduction, sound classification
  - failure prediction
- **ASR [2]**
  - urgent commands, factory hands-free environment
  - 20K word vocabulary, high accuracy
- **Roomba lab demo**
  - robotics server and battery “dead bugged” on a Roomba
  - voice commands processed and sent to Roomba APIs via USB; e.g. “watch out !”, “stop”, “turn left”, etc

[1] Size, weight, and power consumption
[2] Automatic speech recognition

Cloud tradeoffs in High Performance Robotics

- **For high performance robotics the cloud …**
  - lacks provide data privacy (e.g. background conversations, proprietary sounds)
  - cannot guarantee real-time response in human safety / emergency situations
  - is cost-prohibitive as number of robots increases
- **… but the cloud enables**
  - management and orchestration of containerized robotics solutions
  - app interfaces – robot status and monitoring, telemetry, event notification



In the Roomba example, voice commands must travel to the cloud and back – slow, not private, and not reliable

High Performance Robotics Needs

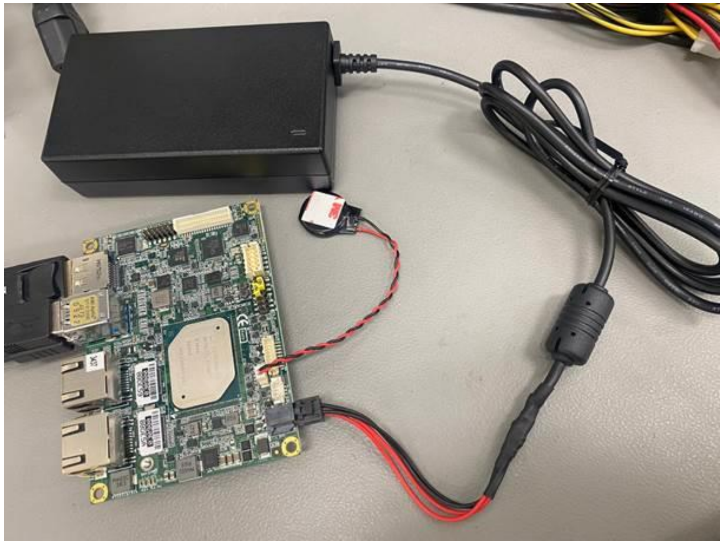
- **"Laws of robotics"**
  - respond to voice commands, especially urgent safety / emergency commands
  - avoid people, animals, property
  - do not break things and do not incur legal liability
  - operate with zero trust in cloud responses – ground situation takes precedence
- **Cannot be cost-prohibitive to scale**
  - as an example, AWS "Lightsail" compute instances cost $80 per month
    - dedicated, always available, high performance quad x86 cores with SSE
    - 8 GB mem, 320 GB SSD
  - cost adds up for multiple robots
- **Maintain strict data privacy**
  - background sounds, conversation are considered strictly private
  - data may be sent to cloud for AI training, within carefully defined policies
- **Ultra low SWaP**
  - under 50 W power consumption with no fans
  - extremely small form-factor
  - must be physically lightweight, especially for mobile robots

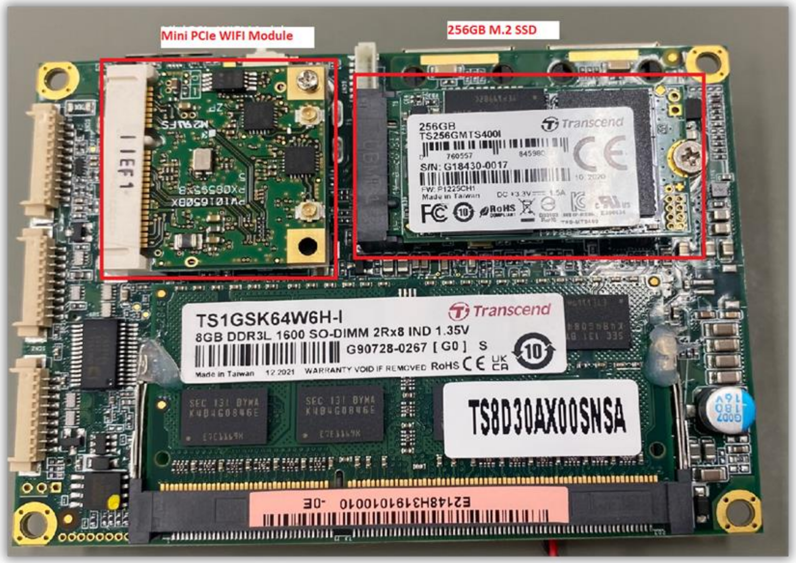
RobotHPC™ Edge Platform - Software

- **EdgeStream™ software**
  - optimized for per core high performance, low energy usage
  - deployed with telecom, lawful interception, and gov/mil customers
  - using quad-core Atom, handles up to three (3) concurrent ASR streams, three (3) far-field microphones for one stream, other sensor types, or some combination
- **Input from USB or RTP packets**
  - input from one more sensors, microphones
  - input from RTP packet streams (IP/UDP, sensors or microphones with Ethernet interface)
  - wide range of codecs supported (AMR, EVS, G711, etc) if needed
- **Development environment**
  - Ubuntu 20.04, gcc/g++ 9.x
  - command line, WinSCP, gdebug … standard Linux and WinX tools
- **Supports containerization and cloud orchestration**

RobotHPC™ Edge Platform - Hardware

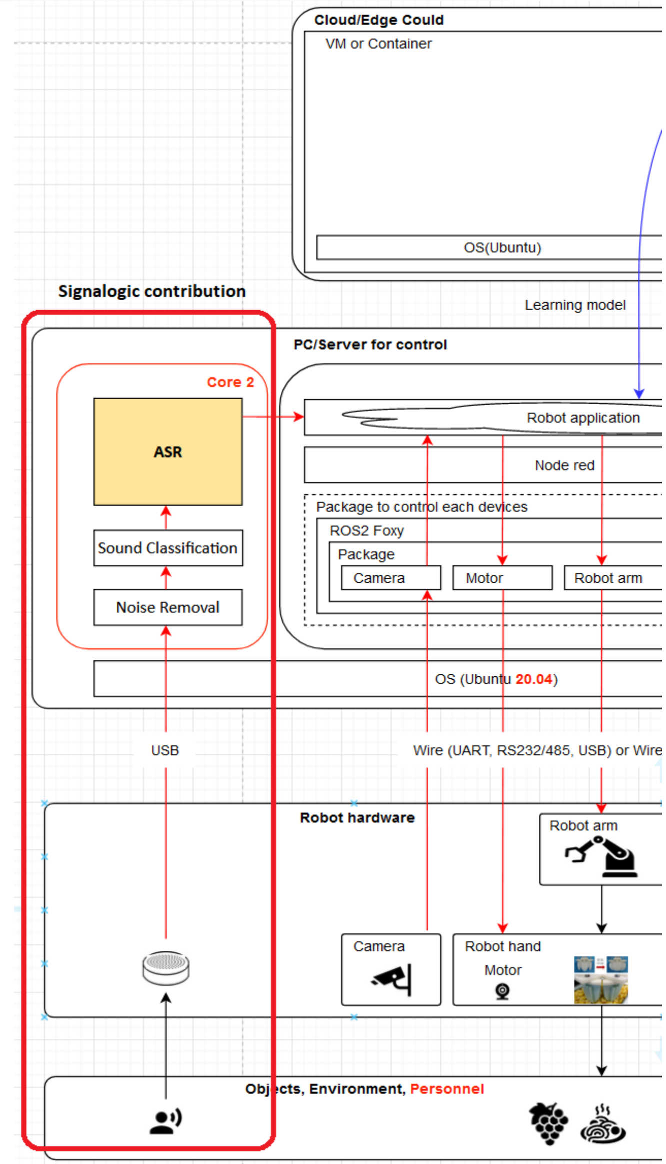
- **Ultra low SWaP server**
  - *small* pico ITX form-factor, 3.5” x 3.5”
  - *high performance* quad-core Atom (x5-E3940), 8 GB mem
  - *low power* ~50 W, no fans
  - *integrated* SSD, WiFi, HDMI, USB, etc.
- **Flexible, easy-to-use, standard**
  - code developed on x86 lab / cloud servers, runs on robotics servers w/o changes. No IDEs, JTAG emulators, or special “translation tools” / “porting tools” for AI frameworks
  - standard Linux (Ubuntu 20.04), gcc/g++, debug tools
  - 100% compatible with mainstream open source edge computing and analytics
  - 100% compatible with mainstream containerization and cloud orchestration





SSES Akraino Blueprint

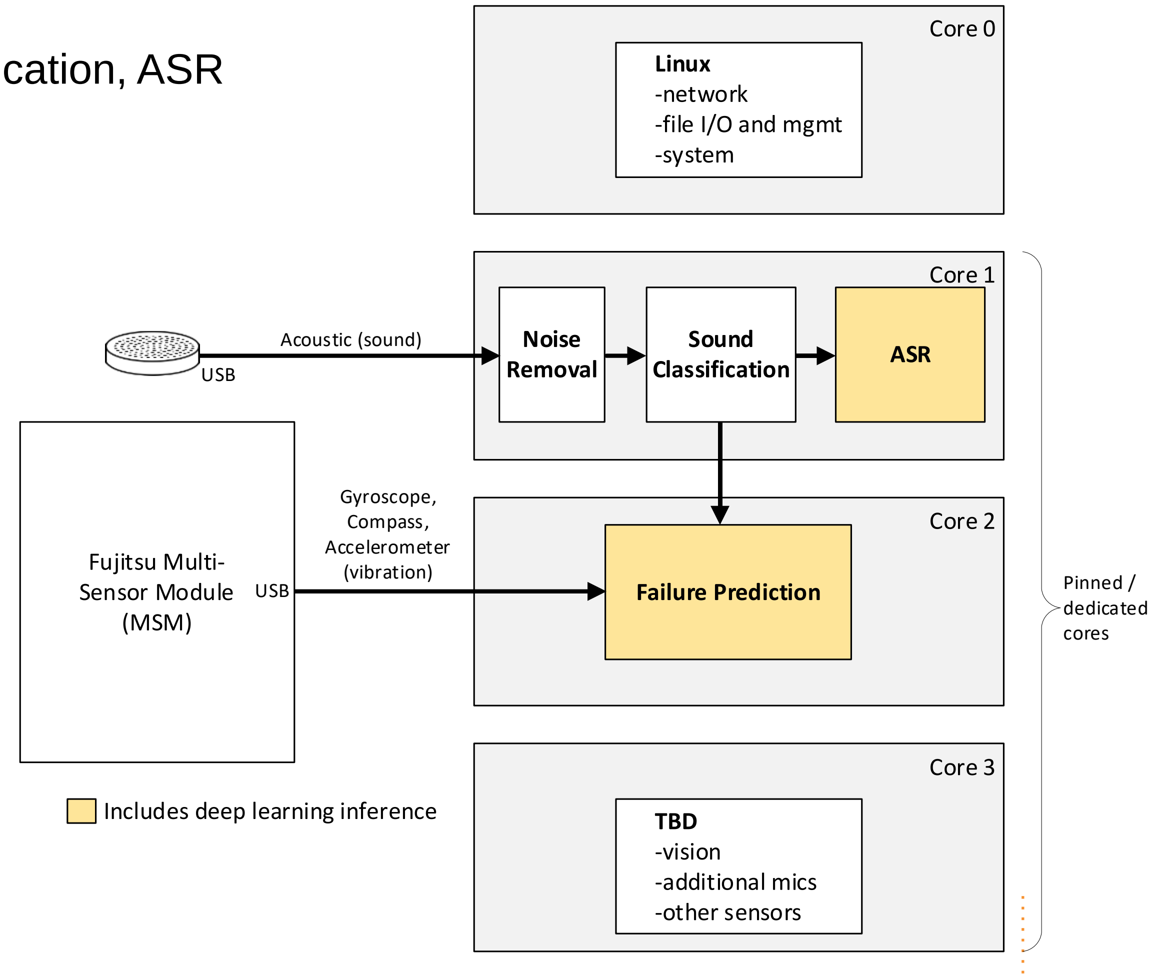
- **Fujitsu + Ritsumeikan University**
  - food preparation and handling for manufacturing and enterprise customers
  - hands-free operation – wet environment, workers wearing gloves
  - safety and emergency needs
  - robots include x86 based small servers
- **Signalogic contribution**
  - speech recognition – high performance, high vocabulary ASR
  - optimized solution, consumes one (1) x86 core from available pool of cores



SSES architecture diagram showing RobotHPC™ data flow

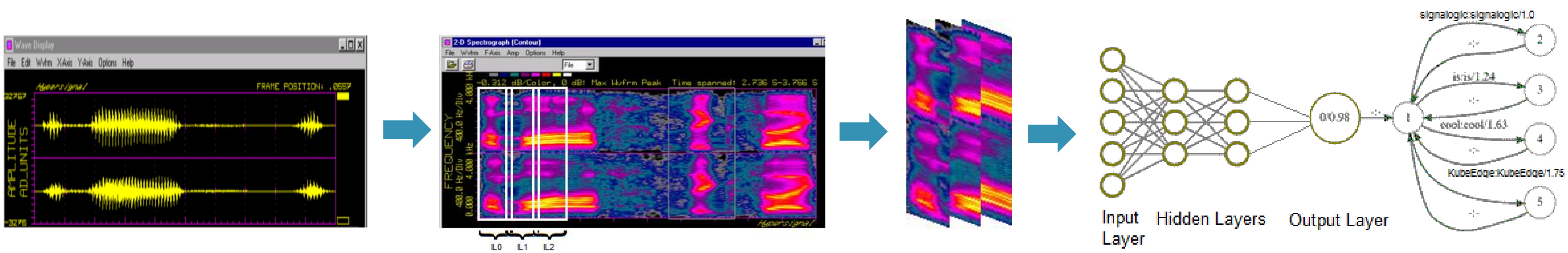
Core / Thread Data Flow

- **Pinned x86 cores**
  - noise removal, sound classification, ASR
  - failure prediction
  - TBD, as needed
  - one core reserved for Linux
- **Each core runs end-to-end thread**
  - real-time
  - including input and output
- **EdgeStream™ software**
  - makes this type of real-time architecture possible – no thread slicing
  - no spinlocks
  - data passed between cores with lock-free buffers



ASR Data Flow

- **Wideband audio input**
  - 16 kHz sampling rate, 16-bit data
  - input from one or more USB microphones, A/D converters, etc
- **Frequency domain “images”**
  - formed by overlap FFT analysis of incoming time series data
  - each FFT frame output is similar to cochlea in human ears
  - groups of FFT frames form images
  - successive images are called “TDNN” (time delayed DNN [3]), similar to series of CNN [4]
  - xMM [5] applied to DNN outputs



time series | 2D spectrograph | sliding overlapped frames | DNN | xMM

[3] Deep neural network
[4] Convolutional neural network
[5] Hidden Markov model, Gaussian mixed model

Signal Processing

- **Noise removal**
  - spectral subtraction and other methods minimize environment background noise
  - target robot noise, e.g. motors, wheels, brushes
  - detect and minimize incidental background conversation
- **Sound classification**
  - voice, sound, noise
- **Failure prediction**
  - incorporate multiple sensors (vibration, acceleration, acoustic)
  - apply deep learning methods
  - recognize “sequences of sounds”, similar to language sentences

ASR

- **Capabilities**
  - urgent / safety voice commands; e.g. “stop”, “watch out”
  - operating commands ; e.g. “change mode”, “discard”
  - large vocabulary, context-awareness provides error resistance, reducing false positives
  - currently running in real-time on robotics edge platform in the lab (x5-E3940 Atom)
- **Deep learning architecture**
  - combines xMM and DNN technology
  - based on Kaldi open source, which is used by Alexa, Google Home, and Cortana
- **Processing intensive**
  - real-time on one (1) x5-E3940 core, but not by a large margin (RTF [6] approx 2)
- **Fast development / debug / test**
  - develop, debug, and test on any x86 (with SSE), run on the robotics platform. No conversion to “internal representation”, model translation tools, etc needed

[6] Real-time factor

Roomba Demo Objectives

- **Edge computing high performance**
  - real-time
  - respond to voice commands, especially urgent commands
  - vision - avoid people, pets, property. Don't break things
- **Cloud independence**
  - operate with zero trust in cloud commands – on-the-ground situation takes precedence
  - factory background conversation, sounds are strictly private
  - cloud data storage, training only under well-defined, controlled conditions
- **Extremely low SWaP**
  - operate with ~50 W power consumption, no fans
  - ultra small form-factor
  - physically lightweight, especially suitable for mobile robots

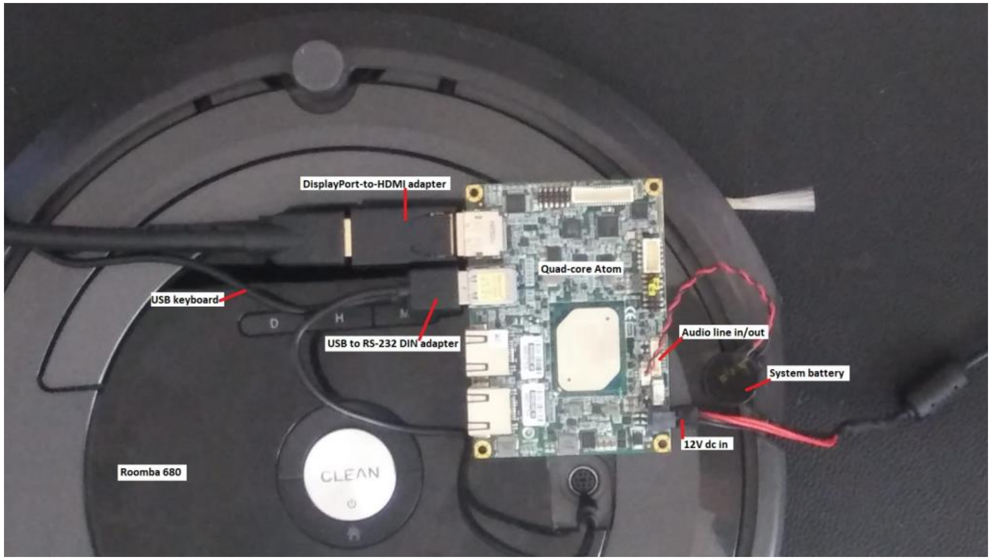
"Not now roomba"

Roomba Lab Demo

- **Robotics edge platform**
  - “dead bugged” onto Roomba 680
  - quad-core Atom server, no fans
  - 12V dc (battery or AC adapter)
  - USB interface to Roomba RS-232
  - line-in or USB microphone

- **Lab interface**
  - standard server tools used for development, test, and debug – WinSCP, cmd line, gdebug, etc.
  - for hands-on lab work, HDMI monitor, keyboard available

- **Roomba real-time voice control**
  - real-time ASR, 20k word vocabulary
  - translate recognized ASR text to Roomba command APIs



Roomba lab demo

Github and Docker Hub

- **Akraino**
  - SSES Blueprint wiki page: https://wiki.akraino.org/display/AK/CPS+Robot+Blueprint+family
- **Github**
  - SigSRF software page: https://github.com/signalogic/SigSRF_SDK
  - example command lines for reference apps and demos
  - documentation
- **Docker Hub**
  - ready-to-run Ubuntu and CentOS containers https://hub.docker.com/u/signalogic
- **Demos and reference apps**
  - ready-to-run containers on Docker Hub, installation Rar packages on Github
  - help with installing and running demos available over Skype (no charge)
- **Source code**
  - developed entirely in US
  - no dependencies on 3rd party libraries

Thanks !

- **Q&A**
- **Follow-up questions / comments: info@signalogic.com**
- **Web page: https://signalogic.com/edgestream**

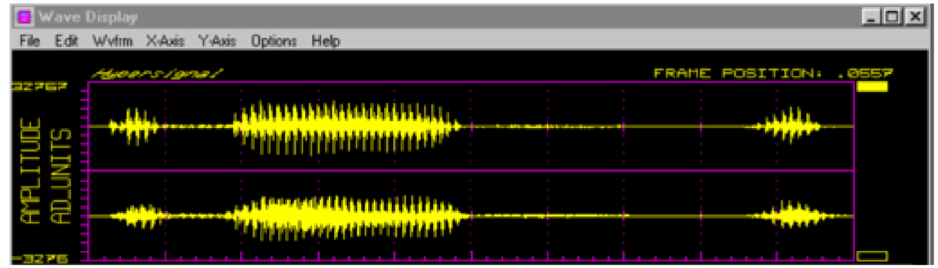
Supplemental

- **Following slides are background info …**

ASR Basics

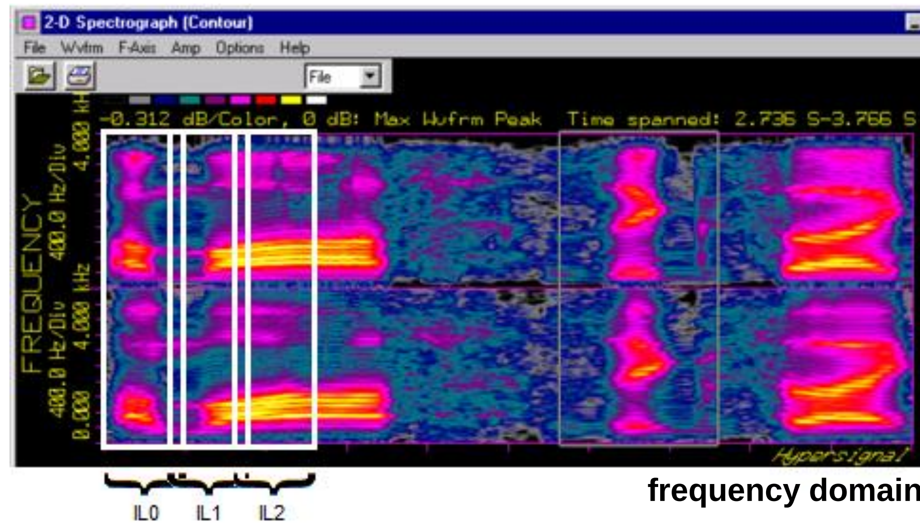
- **Deep Learning Architecture**
  - combines previous generation xMM[1] technology with DNNs (Deep Neural Networks)
  - relies on extensive training and "augmentation" methods
  - Kaldi open source is basis for Alexa, Google Home, and Cortana

- **Frequency domain "images"**
  - formed by sliding FFT analysis of incoming time series data. Each FFT frame output is similar to cochlea in human ears
  - groups of FFT frames form images
  - successive images are called "TDNN" (time delayed DNN), similar to series of CNNs[2]



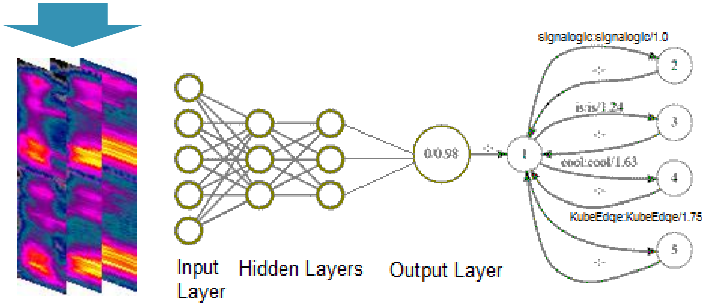
time domain (time series)

Sliding FFT



frequency domain

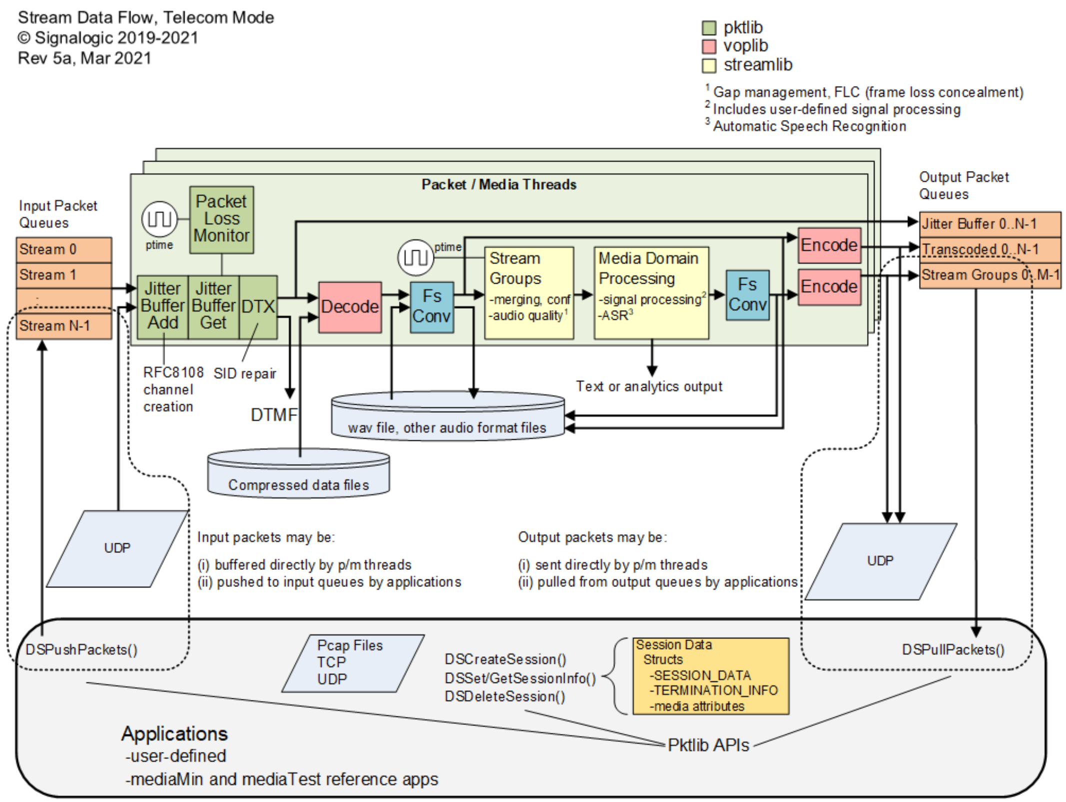
DNN Input Layers (ILn)



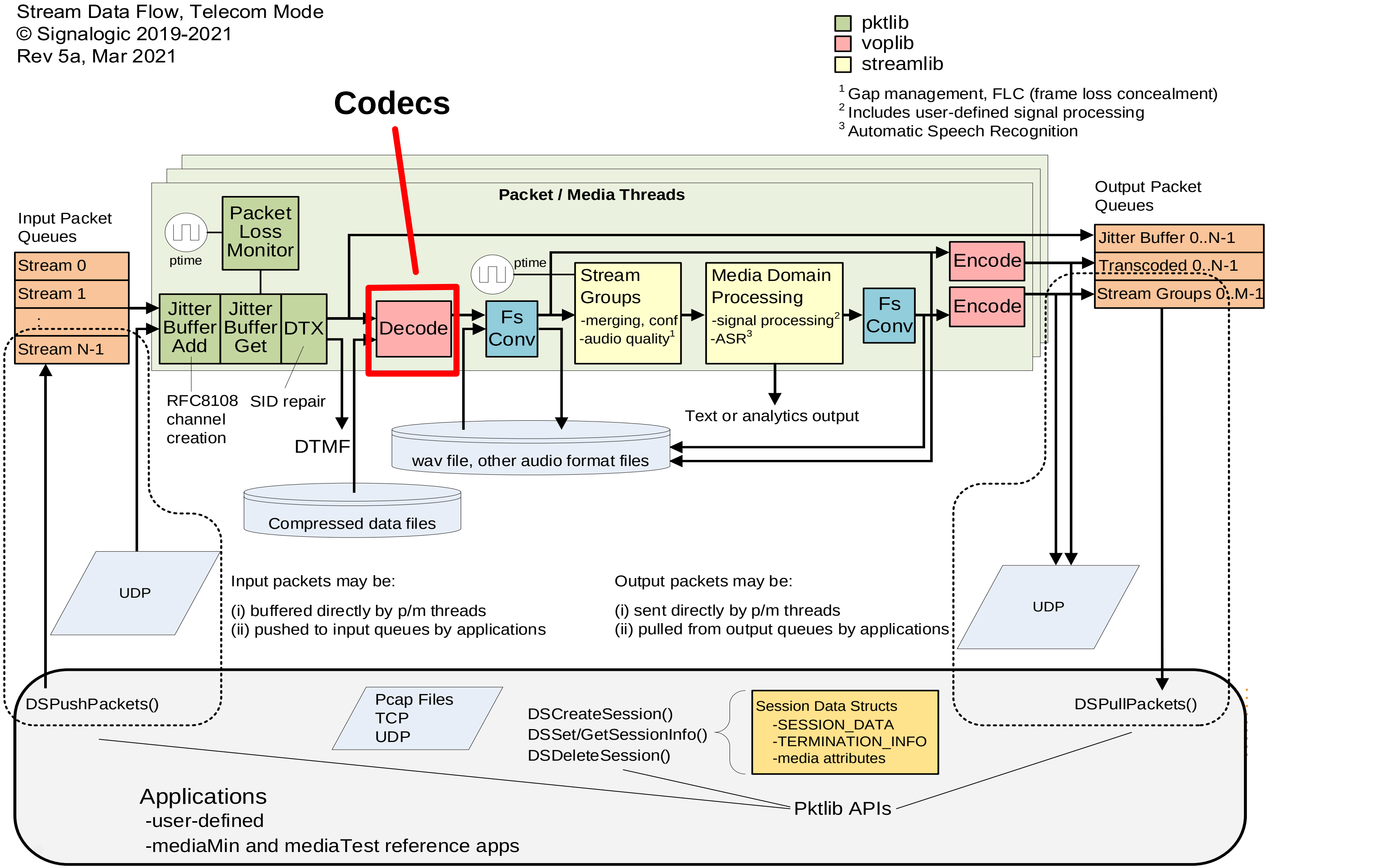
[1] Hidden Markov Model, Gaussian Mixed Model, [2] Convolutional Neural Network

EdgeStream Data Flow

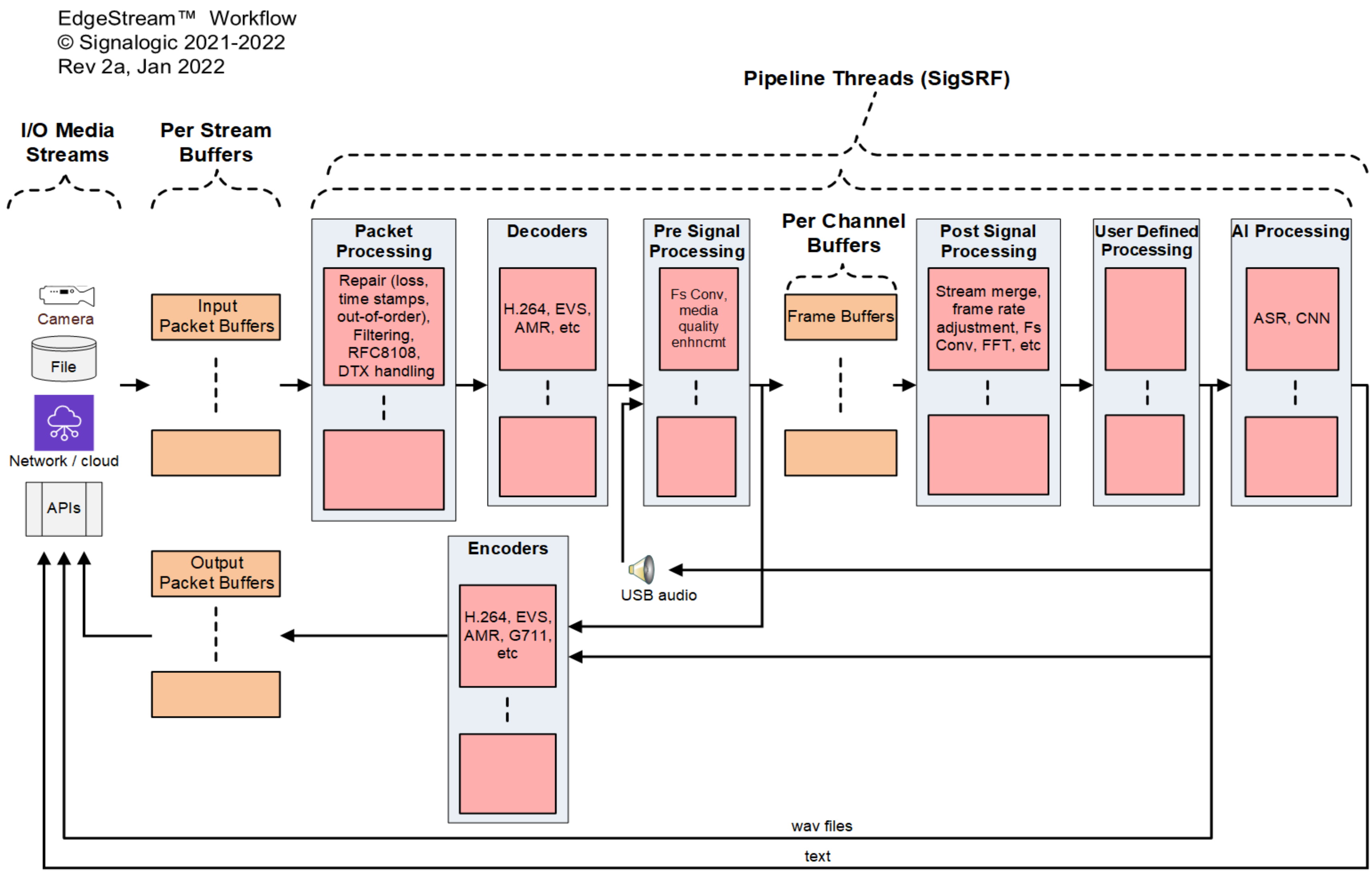
- **Per core data flow**
  - one thread per core
  - no spinlocks
  - precise control over power consumption
- **Real-time workflow**
  - packet handling
  - media codecs
  - signal processing
  - user-defined processing
  - inference
- **Hardware acceleration**
  - DirectCore® option
  - x86 and Arm options supported



Overview – Per Thread Data Flow



EdgeStream Workflow

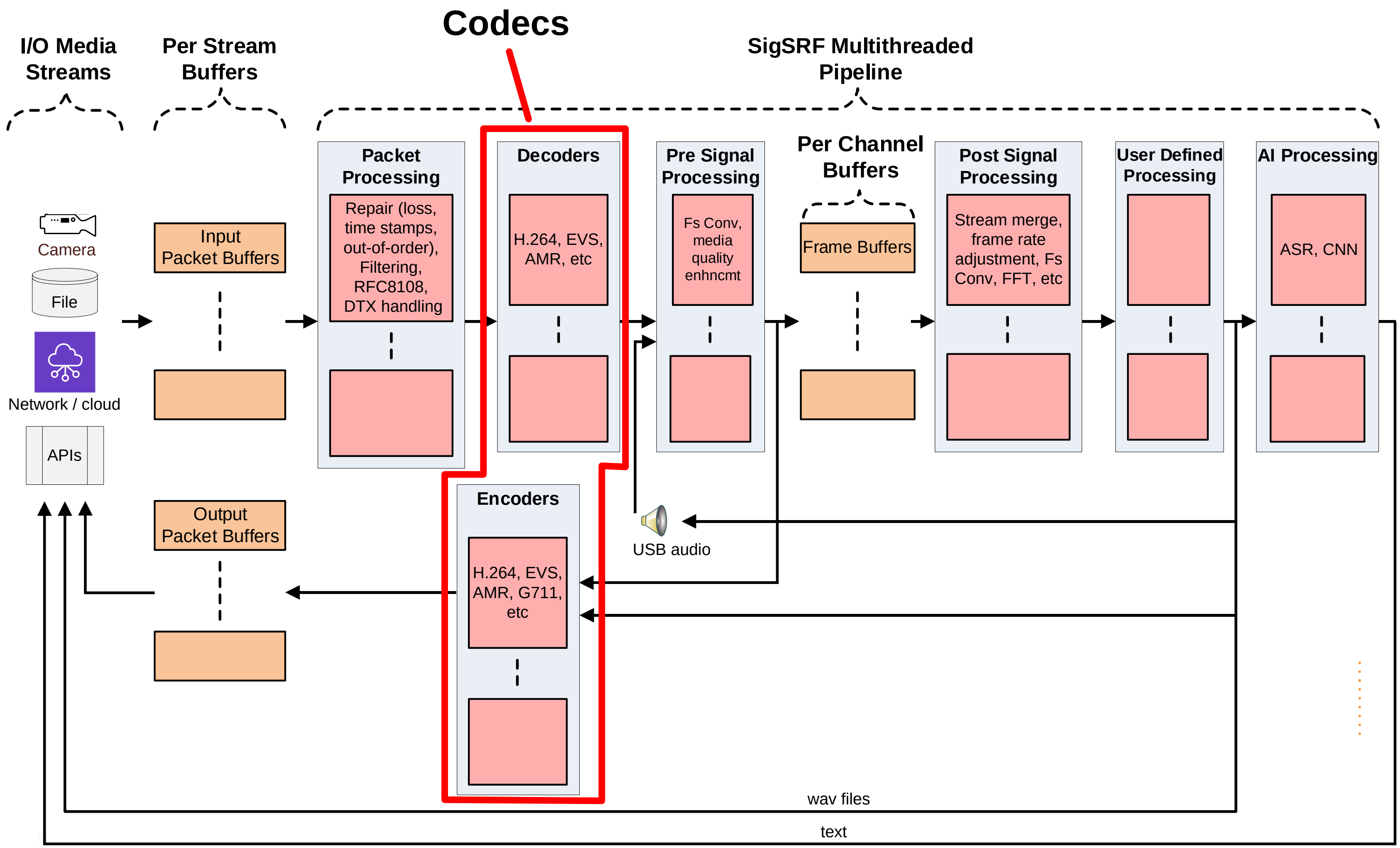


Overview – Pipeline Flow

Signalogic

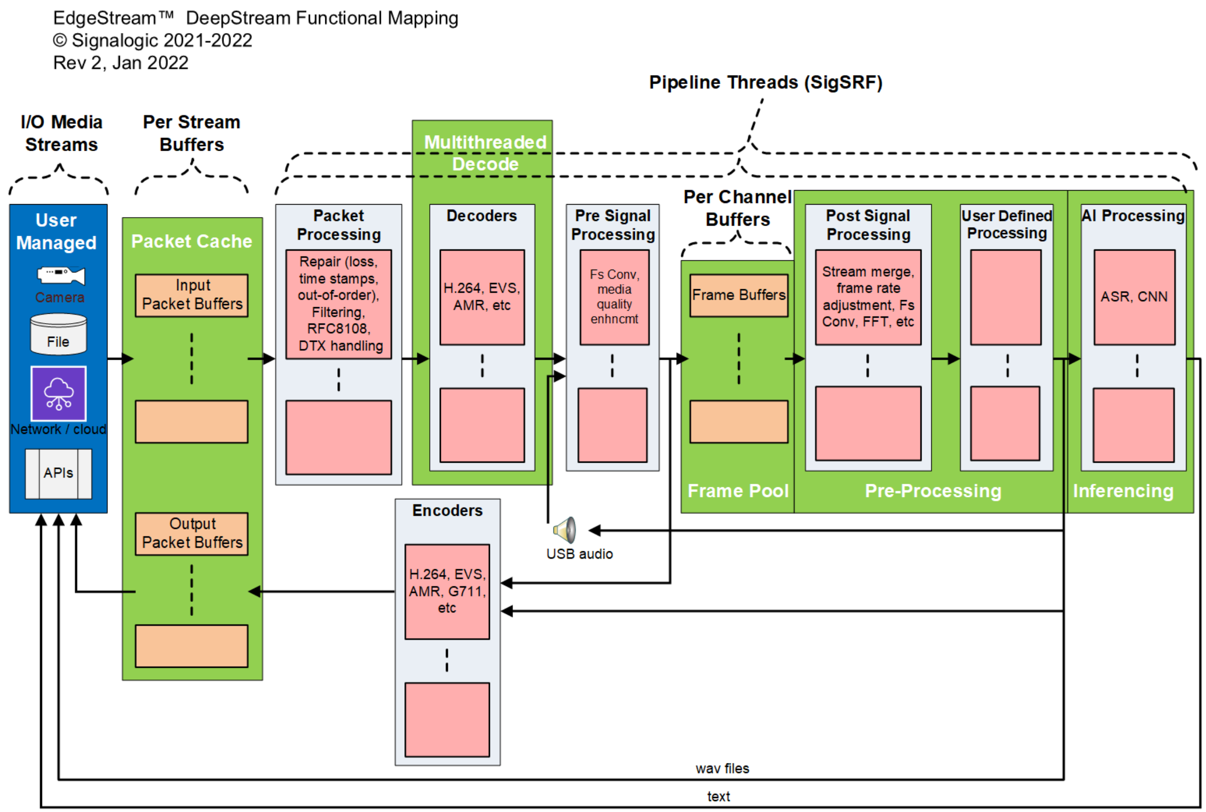
EdgeStream™ Workflow

Rev 1a, Dec 2021



Comparison with DeepStream

- **Packet Processing**
  - EdgeStream provides telecom grade packet processing, including…
  - loss repair
  - 500+ out-of-order handling
  - support for encapsulated protocols
  - multiple RFCs
  - logging
- **Media**
  - includes encoders in addition to decoders
- **Signal processing**
  - more user-defined insertion points

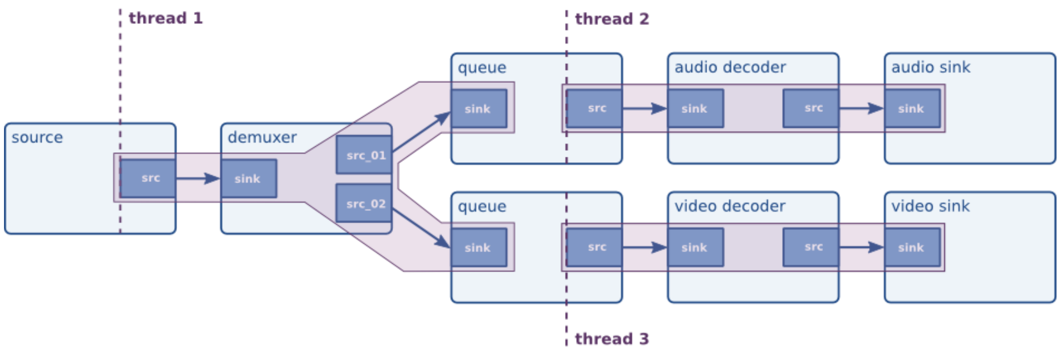


DeepStream is a trademark of Nvidia Corp

Comparison with GStreamer

- **Thread architecture**
  - EdgeStream allocates one workflow thread per core ("unified thread")
  - GStreamer uses a thread slicing architecture – flexible but requires spinlocks
- **Packet Processing**
  - EdgeStream provides telecom grade packet processing, including …
  - loss repair
  - 500+ out-of-order handling
  - supports encapsulated protocols
  - multiple RFCs
  - logging

Example GStreamer Workflow



EdgeStream™ Workflow
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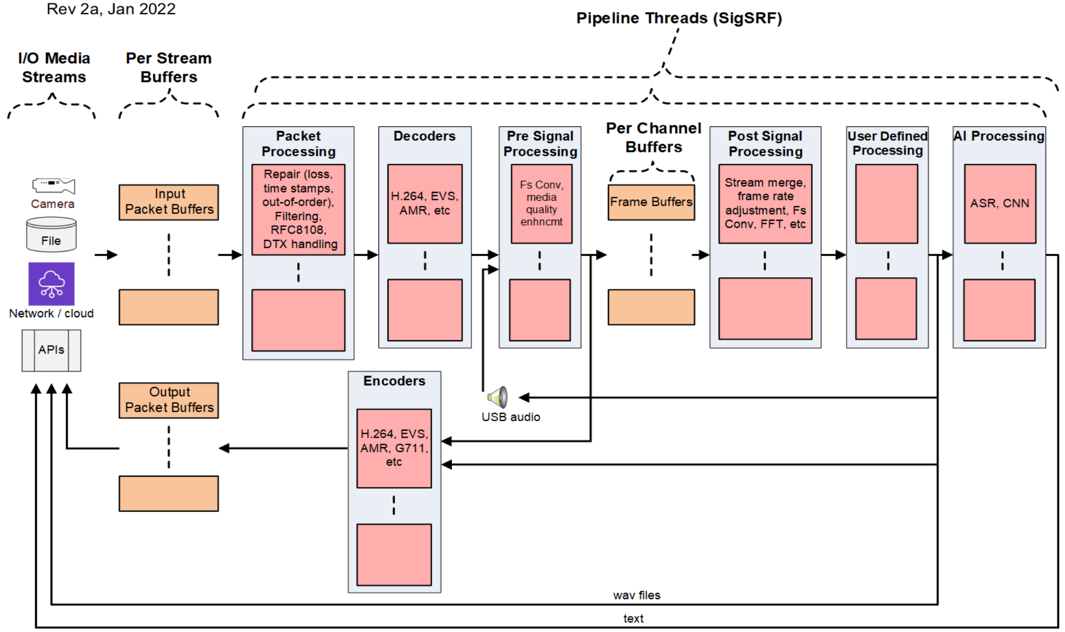
Rev 2a, Jan 2022



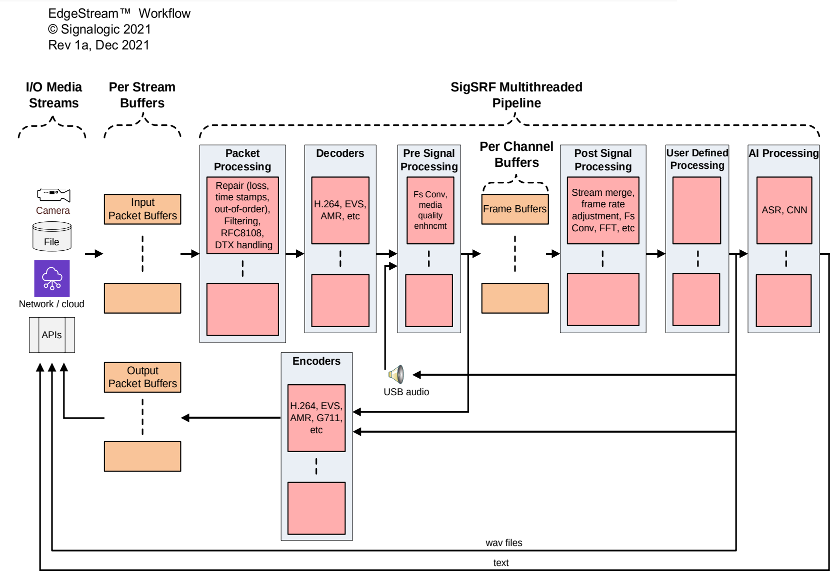
EdgeStream Deployments

- **Asia**
  - Japan
  - India (ISRO)
  - Australia
  - New Zealand (OpenLI [1] support)
- **Europe**
  - Germany
  - Italy
  - Czech Republic
- **North America**
  - AFRL
  - Raytheon
  - Boeing

[1] OpenLI is "Open Lawful Intercept" for CSPs. More info at https://openli.nz/

Software Overview

- **SigSRF libraries**
  - codecs
    - VoLTE (EVS, AMR-NB, AMR-WB)
    - legacy (G729, EVRC, GSM, etc)
    - mil/gov (MELPe)
  - packet procesing
    - media/SID packet repair (out-of-order, packet loss, RTP timestamps)
    - timing reconstruction of missing/damaged arrival timestamps
    - child streams (RFC8108)
  - frame processing
    - “stream groups” can be defined for related streams
    - per-stream correction for overrun, underrun, gaps, bursts
    - accurate time-aligned merging / mixing of multiple endpoints
    - high capacity – multiple concurrent streams



- **EdgeStream™ applications**
  - reference apps for customer-defined development
  - also used as-is by many of our customers. Most common: telecom, LI, and ASR
  - key features
    - dynamic session creation
    - packet push/pull API interface with SigSRF libs
    - multiple streams from multiple sources
    - flexible command line – similar to ffmpeg or sox

Functionality – Packet Processing

- **Media quality – packet level**
  - media/SID packet repair
    - out-of-order (ooo)
    - packet loss
    - RTP timestamps
  - child streams (RFC8108)
  - timing reconstruction for missing/damaged packet arrival timestamps
- **Huge levels of ooo handled**
  - to support TCP encapsulated UDP/RTP, for example lawful interception apps implementing ETSI protocols
- **Packet logging / tracing**
  - per stream packet logging
  - timestamp reconciliation
  - individual packet tracing

```
Ingress Packet info for SSRC = 0xbad52e64, first seq num = 3, last
seq num = 651 ...

Seq num 4 ooo 3         timestamp = 1280, pkt len = 6 SID
Seq num 3 ooo 4         timestamp = 960, pkt len = 61
Seq num 5               timestamp = 3840, pkt len = 6 SID
Seq num 6               timestamp = 6400, pkt len = 6 SID
Seq num 7               timestamp = 8960, pkt len = 6 SID
Seq num 8               timestamp = 11520, pkt len = 6 SID
Seq num 9               timestamp = 14080, pkt len = 6 SID
Seq num 10               timestamp = 16640, pkt len = 6 SID
Seq num 12 ooo 11        timestamp = 18560, pkt len = 61
Seq num 15 ooo 12        timestamp = 19520, pkt len = 61
Seq num 11 ooo 13        timestamp = 18240, pkt len = 61
Seq num 13 ooo 14        timestamp = 18880, pkt len = 61
Seq num 14 ooo 15        timestamp = 19200, pkt len = 61
Seq num 18 ooo 16        timestamp = 20480, pkt len = 61
Seq num 19 ooo 17        timestamp = 20800, pkt len = 61
Seq num 16 ooo 18        timestamp = 19840, pkt len = 61
Seq num 21 ooo 19        timestamp = 21440, pkt len = 6 SID
Seq num 23 ooo 20        timestamp = 23680, pkt len = 61
Seq num 24 ooo 21        timestamp = 24000, pkt len = 61
Seq num 25 ooo 22        timestamp = 24320, pkt len = 61
Seq num 27 ooo 23        timestamp = 24960, pkt len = 61
Seq num 28 ooo 24        timestamp = 25280, pkt len = 61
Seq num 31 ooo 25        timestamp = 26240, pkt len = 61
Seq num 32 ooo 26        timestamp = 26560, pkt len = 61
Seq num 34 ooo 27        timestamp = 27200, pkt len = 61
Seq num 17 ooo 28        timestamp = 20160, pkt len = 61
Seq num 20 ooo 29        timestamp = 21120, pkt len = 61
Seq num 22 ooo 30        timestamp = 23360, pkt len = 61
Seq num 26 ooo 31        timestamp = 24640, pkt len = 61
Seq num 29 ooo 32        timestamp = 25600, pkt len = 61
Seq num 30 ooo 33        timestamp = 25920, pkt len = 61
Seq num 33 ooo 34        timestamp = 26880, pkt len = 61
Seq num 35               timestamp = 27520, pkt len = 61
Seq num 37 ooo 36        timestamp = 28160, pkt len = 61
Seq num 38 ooo 37        timestamp = 28480, pkt len = 61
Seq num 40 ooo 38        timestamp = 29120, pkt len = 61
Seq num 42 ooo 39        timestamp = 29760, pkt len = 61
Seq num 44 ooo 40        timestamp = 30400, pkt len = 61
Seq num 46 ooo 41        timestamp = 31040, pkt len = 61
Seq num 36 ooo 42        timestamp = 27840, pkt len = 61
Seq num 48 ooo 43        timestamp = 31680, pkt len = 61
Seq num 39 ooo 44        timestamp = 28800, pkt len = 61
Seq num 41 ooo 45        timestamp = 29440, pkt len = 61
Seq num 50 ooo 46        timestamp = 32320, pkt len = 61
Seq num 53 ooo 47        timestamp = 33280, pkt len = 61
Seq num 55 ooo 48        timestamp = 33920, pkt len = 61
Seq num 43 ooo 49        timestamp = 30080, pkt len = 61
Seq num 57 ooo 50        timestamp = 34560, pkt len = 61
```

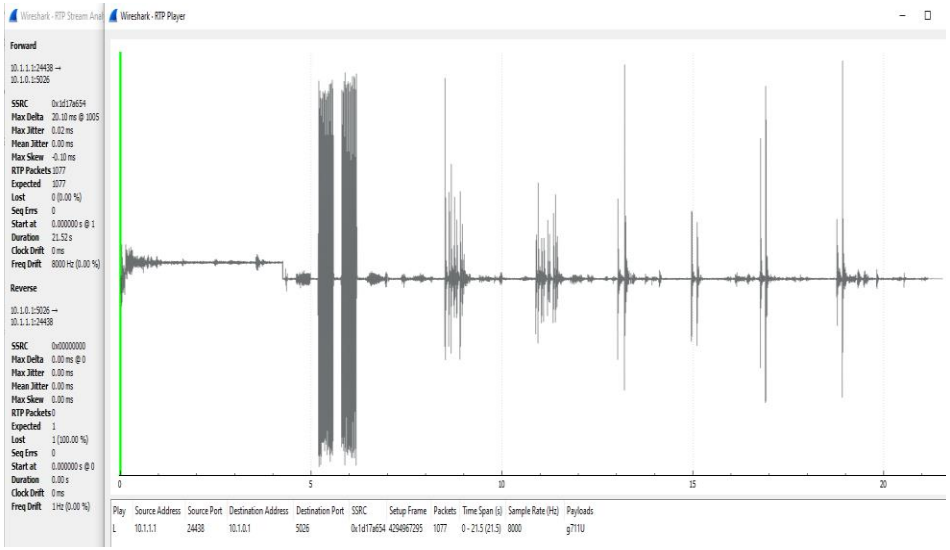
Packet Log Excerpt

```
Ingress Packet info for SSRC = 0xbad52e64, first seq num = 3, last
seq num = 651 ...

Seq num 4 ooo 3        timestamp = 1280, pkt len = 6 SID
Seq num 3 ooo 4        timestamp = 960, pkt len = 61
Seq num 5              timestamp = 3840, pkt len = 6 SID
Seq num 6              timestamp = 6400, pkt len = 6 SID
Seq num 7              timestamp = 8960, pkt len = 6 SID
Seq num 8              timestamp = 11520, pkt len = 6 SID
Seq num 9              timestamp = 14080, pkt len = 6 SID
Seq num 10              timestamp = 16640, pkt len = 6 SID
Seq num 12 ooo 11       timestamp = 18560, pkt len = 61
Seq num 15 ooo 12       timestamp = 19520, pkt len = 61
Seq num 11 ooo 13       timestamp = [illegible]240, pkt len = 61
Seq num 13 ooo 14       timestamp = 18880, pkt len = 61
Seq num 14 ooo 15       timestamp = 19200, pkt len = 61
Seq num 18 ooo 16       timestamp = 20480, pkt len = 61
Seq num 19 ooo 17       timestamp = 20800, pkt len = 61
Seq num 16 ooo 18       timestamp = 19840, pkt len = 61
Seq num 21 ooo 19       timestamp = 21440, pkt len = 6 SID
Seq num 23 ooo 20       timestamp = 23680, pkt len = 61
Seq num 24 ooo 21       timestamp = 24000, pkt len = 61
Seq num 25 ooo 22       timestamp = 24320, pkt len = 61
Seq num 27 ooo 23       timestamp = 24960, pkt len = 61
Seq num 28 ooo 24       timestamp = 25280, pkt len = 61
Seq num 31 ooo 25       timestamp = 26240, pkt len = 61
Seq num 32 ooo 26       timestamp = 26560, pkt len = 61
Seq num 34 ooo 27       timestamp = 27200, pkt len = 61
Seq num 17 ooo 28       timestamp = 20160, pkt len = 61
Seq num 20 ooo 29       timestamp = 21120, pkt len = 61
Seq num 22 ooo 30       timestamp = 23360, pkt len = 61
Seq num 26 ooo 31       timestamp = 24640, pkt len = 61
Seq num 29 ooo 32       timestamp = 25600, pkt len = 61
Seq num 30 ooo 33       timestamp = 25920, pkt len = 61
Seq num 33 ooo 34       timestamp = 26880, pkt len = 61
Seq num 35              timestamp = 27520, pkt len = 61
Seq num 37 ooo 36       timestamp = 28160, pkt len = 61
Seq num 38 ooo 37       timestamp = 28480, pkt len = 61
Seq num 40 ooo 38       timestamp = 29120, pkt len = 61
Seq num 42 ooo 39       timestamp = 29760, pkt len = 61
Seq num 44 ooo 40       timestamp = 30400, pkt len = 61
Seq num 46 ooo 41       timestamp = 31040, pkt len = 61
Seq num 36 ooo 42       timestamp = 27840, pkt len = 61
Seq num 48 ooo 43       timestamp = 31680, pkt len = 61
Seq num 39 ooo 44       timestamp = 28800, pkt len = 61
Seq num 41 ooo 45       timestamp = 29440, pkt len = 61
Seq num 50 ooo 46       timestamp = 32320, pkt len = 61
Seq num 53 ooo 47       timestamp = 33280, pkt len = 61
Seq num 55 ooo 48       timestamp = 33920, pkt len = 61
Seq num 43 ooo 49       timestamp = 30080, pkt len = 61
Seq num 57 ooo 50       timestamp = 34560, pkt len = 61
```

High amount of ooo (out-of-order) example

Functionality – Frame Processing

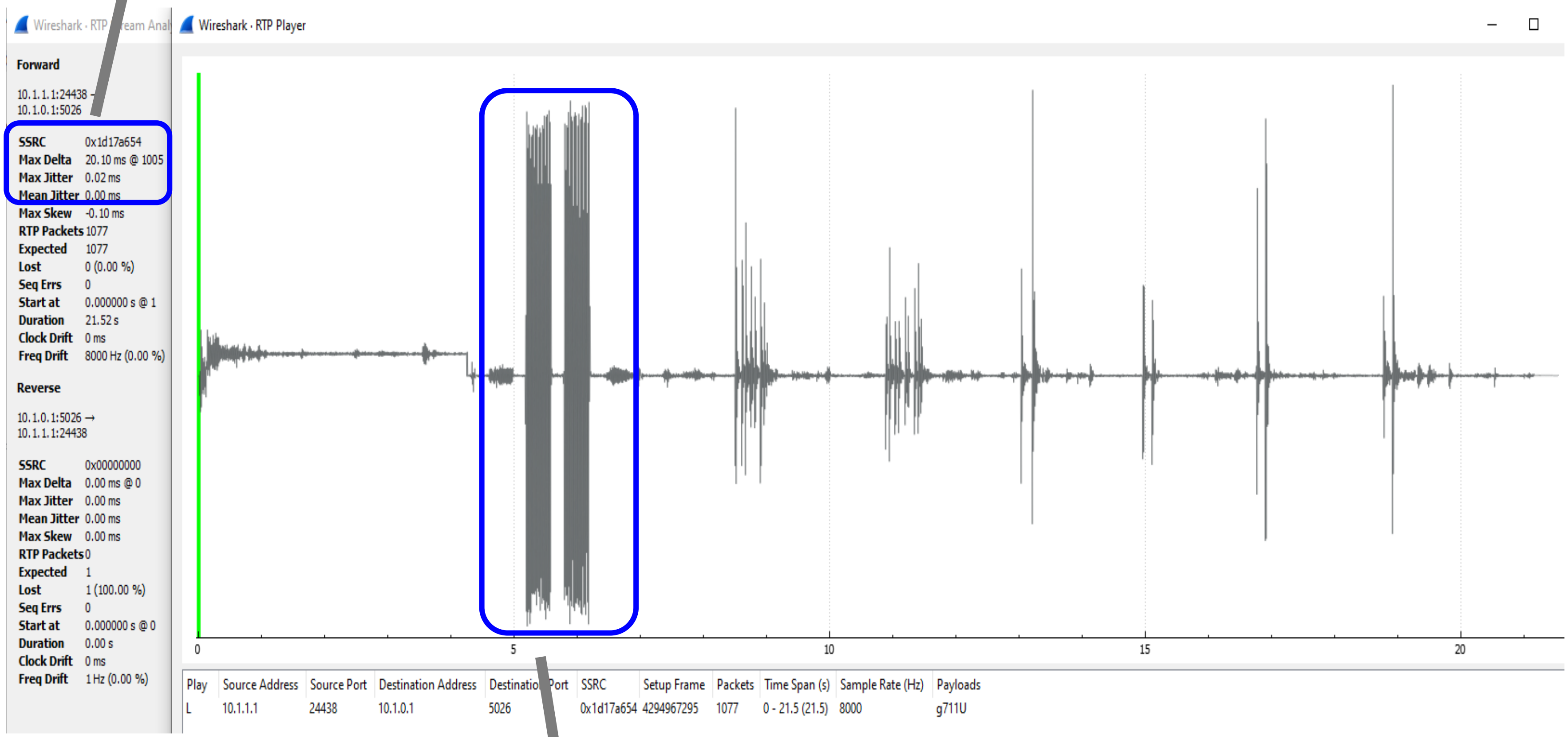
- **Decoded packet audio data**
  - buffered as frames (see Overview diagrams)
  - signal processing
- **Media quality – frame level**
  - "stream groups" can be defined for streams related in some way
  - per-stream correction for overrun, underrun, gaps, bursts
  - accurate time-aligned merging / mixing of multiple endpoints



- **Real-time output streaming**
  - some applications require real-time output, either per-stream or merged between related streams, typically in G711 format
  - high intelligibility required – all streams fully merged (non-overlapped) and non-duplicated *as if all endpoints are in the same room*
- **High capacity – multiple concurrent streams**

Real-Time Streaming Output Example

Reliable packet delta, no jitter over 1000s of hours of streaming



Child streams example - early media (ring tones)

Functionality – Applications

- **Dynamic and static session creation**
  - sessions created and codecs detected on-the-fly using (i) RTP only (ii) SIP invite packets (iii) .sdp files, or pre-set using static session config files
  - RTP only uses heuristic codec type detection
- **Packet push/pull interface to SigSRF libs**
  - reference application examples
  - Packet pull includes transcoded output, real-time streaming output
- **Event logging**
  - critical, major, minor, info, debug levels
  - includes alerts for thread pre-emption, queue starvation, and other performance / data related conditions
  - per-stream stats (i) on-demand, (ii) when streams close
- **Arrival timestamp reconstruction**
  - if needed due to missing / damaged arrival timestamps
  - algorithms based on queue balancing, decoded frame rate estimation

Event Log Example

```
00:00:00.000.011 INFO: DSConfigPktlib() uflags = 0x7
  P/M thread capacity  max sessions = 51, max groups = 17
  Event log            path = openli-voip-example_event_log_am.txt, uLogLevel = 8, uEventLogMode = 0x32, flush size = 1024, max size not set
  Debug                uDebugMode = 0x0, uPktStatsLogging = 0xd, uEnableDataObjectStats = 0x1
  Screen output        uPrintfLevel = 5, uPrintfControl = 0
  Energy saver         p/m thread energy saver inactivity time = 30000 msec, sleep time = 1000 usec
  Alarms               DSPushPackets packet cutoff alarm elapsed time not set, p/m thread preemption alarm elapsed time = 40 (msec)
00:00:00.000.721 INFO: DSConfigVoplib() voplib and codecs initialized, flags = 0x1d
00:00:00.000.749 INFO: DSConfigStreamlib() stream groups initialized
00:00:00.000.834 INFO: DSAssignPlatform() system CPU architecture supports rdtscp instruction, TSC integrity monitoring enabled
00:00:00.000.953 INFO: DSOpenPcap() opened pcap input file: ../pcaps/openli-voip-example.pcap
00:00:00.008.396 INFO: DSConfigMediaService() says setpriority() set Niceness to -15 for pkt/media thread 0
00:00:00.008.418 INFO: initializing packet/media thread 0, uFlags = 0x1180101, threadid = 0x7f320f34a700, total num pkt/med threads = 1
00:00:00.058.474 mediaMin INFO: SIP invite found, dst port = 43333, pyld len = 1994, len = 717, rem = 1979, start = 8, index = 0
o=02825591554 0 0 IN IP4 192.168.1.73
c=IN IP4 192.168.1.73
m=audio 5000 RTP/AVP 9 0 8 101
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=extmap:1 urn:ietf:params:rtp-hdrext:csrc-audio-level
a=zrtp-hash:1.10 1c812535e276bf518418c4146a20fd56e715704da9c591ae32d58ee6fed6d40f
m=video 5002 RTP/AVP 96 99
a=recvonly
a=rtpmap:96 H264/90000
a=fmtp:96 profile-level-id=4DE01f;packetization-mode=1
a=imageattr:96 send * recv [x=[0-1920],y=[0-1080]]
a=rtpmap:99 H264/90000
a=fmtp:99 profile-level-id=4DE01f
a=imageattr:99 send * recv [x=[0-1920],y=[0-1080]]
a=zrtp-hash:1.10 c1a98e15f12937b9cad2488c6091468f7610efeefa59863c77d827669b913f38
00:00:00.058.644 INFO: DSFindDerStream() found HI interception point ID 10g-dev1, tag = 0x86, len = 8, dest port = 43332, pyld len = 1448, pyld ofs = 52
00:00:00.058.727 mediaMin INFO: Creating dynamic session 1, input #1, SDP specified codec type G711a, auto-detected bitrate 64000, stream group openli-
voip-example. Creation packet info: IP ver 4, ssrc = 0x14a50012, seq num = 32584, payload type 8, pkt len 200, RTP payload size 160, cat 0
00:00:00.058.781 INFO: DSCreateSession() created stream group "openli-voip-example", idx = 0, owner session = 0, status = 1
```

Dynamic session creation

Functionality – Codecs

- **Multithreaded**
  - original 3GPP source modifications
    - instance create, delete, modify implemented using XDAIS standard
    - global data moved into per-instance “state structs”
  - API interface
    - voplib shared object (.so) library, C/C++ applications include voplib.h
    - DSCodecCreate returns a codec handle, usable with DSCodecEncode and DSCodecDecode
    - also with various codec-related APIs. Some examples:
      - DSGetCodecSampleRate, DSGetCodecBitRate, DSGetCodecRawFrameSize, DSGetCodecCodedFrameSize, DSGetCodecInfo, DSGetSampleRateValue, DSGetPayloadSize, etc
- **Optimization**
  - compiler optimizations
  - pragmas
  - XDAIS standard requires all memory allocation done up-front, so no real-time mallocs or spin-locks
- **Testing**
  - unit / functional testing – mediaTest app, with audio I/O (wav and other audio format files, USB audio)
  - capacity / stress testing – mediaMin app, with application packet push/pull APIs, pcap files, UDP port I/O)
  - system testing – using mediaMin app, highlighted in “Overview” slides
  - bit-exactness testing – comparison of floating-point reference vectors

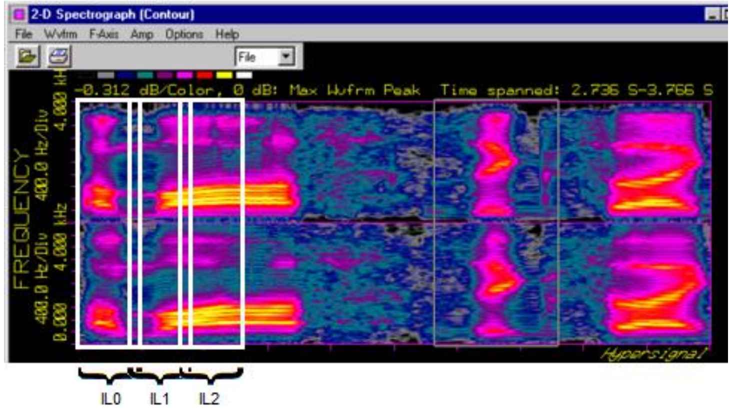
Functionality – Codec API

```
/* codec instance definitions and APIs */

  HCODEC DSCodecCreate(void* pCodecInfo, unsigned int uFlags);  /* if DS_CC_USE_TERMINFO flag is given, pCodecInfo is interpreted as TERMINATION_INFO* (shared_include/session.h), otherwise as
CODEC_PARAMS* (above) */

  void DSCodecDelete(HCODEC hCodec);

  int DSCodecEncode(HCODEC           hCodec,
                    unsigned int     uFlags,
                    uint8_t*         inData,
                    uint8_t*         outData,
                    uint32_t         in_frameSize,
                    CODEC_OUTARGS*   pOutArgs);

  int DSCodecDecode(HCODEC           hCodec,
                    unsigned int     uFlags,
                    uint8_t*         inData,
                    uint8_t*         outData,
                    uint32_t         in_frameSize,  /* in bytes */
                    CODEC_OUTARGS*   pOutArgs);

  typedef struct  {  /* CODEC_ENC_PARAMS */

/* generic items */

   int bitRate;
   int samplingRate;          /* most codecs are based on a fixed sampling rate so this is used only for advanced codecs such as EVS and Opus */
   float frameSize;           /* amount of data (in msec) processed by the codec per frame, for example 20 msec for AMR or EVS, 22.5 msec for MELPe, etc */

           :
           :

/* EVS, Opus, other advanced codec items */

   int sid_update_interval;   /* interval between SID frames when DTX is enabled */
   int rf_enable;             /* channel-aware mode (for EVS only supported at 13.2 kbps) */
   int fec_indicator;         /* for EVS, LO = 0, HI = 1 */
   int fec_offset;            /* for EVS, 2, 3, 5, or 7 in number of frames */
   int bandwidth_limit;       /* for EVS, typically set to SWB or FB */

           :
           :

  } CODEC_ENC_PARAMS;

  typedef struct {  /* CODEC_DEC_PARAMS */

/* generic items */

   int bitRate;               /* bitrate may not be used for codecs that can derive it from payload contents */
   int samplingRate;          /* not used for most codecs */
   float frameSize;           /* amount of data (in msec) processed by the codec per frame, for example 20 msec for AMR or EVS, 22.5 msec for MELPe, etc */

           :
           :

  } CODEC_DEC_PARAMS;
```

- **voplib.h**
  - excerpt shown here
  - available on Github page
  - C/C++ compatible

Functionality - Customer-Specific

- **Customers often ask us to incorporate / develop specific signal processing. Some examples:**
  - “deduplication” due to multiple copies of the same endpoint (with different latencies)
  - removing room echo / reverb
  - reducing background noise
- **Typically a substantial impact on performance**
- **Speech recognition (ASR)**
  - training is ultra sensitive to small changes in audio characteristics
  - production systems are trained with wide variety of “augmentations”, including background noise and babble, loud and quiet speech, frequency warping, etc.
  - preprocessing to normalize speech input decreases reliance on augmentation training and increases accuracy



  - major impact on performance; for real-time applications, concurrent streams may be reduced 10x

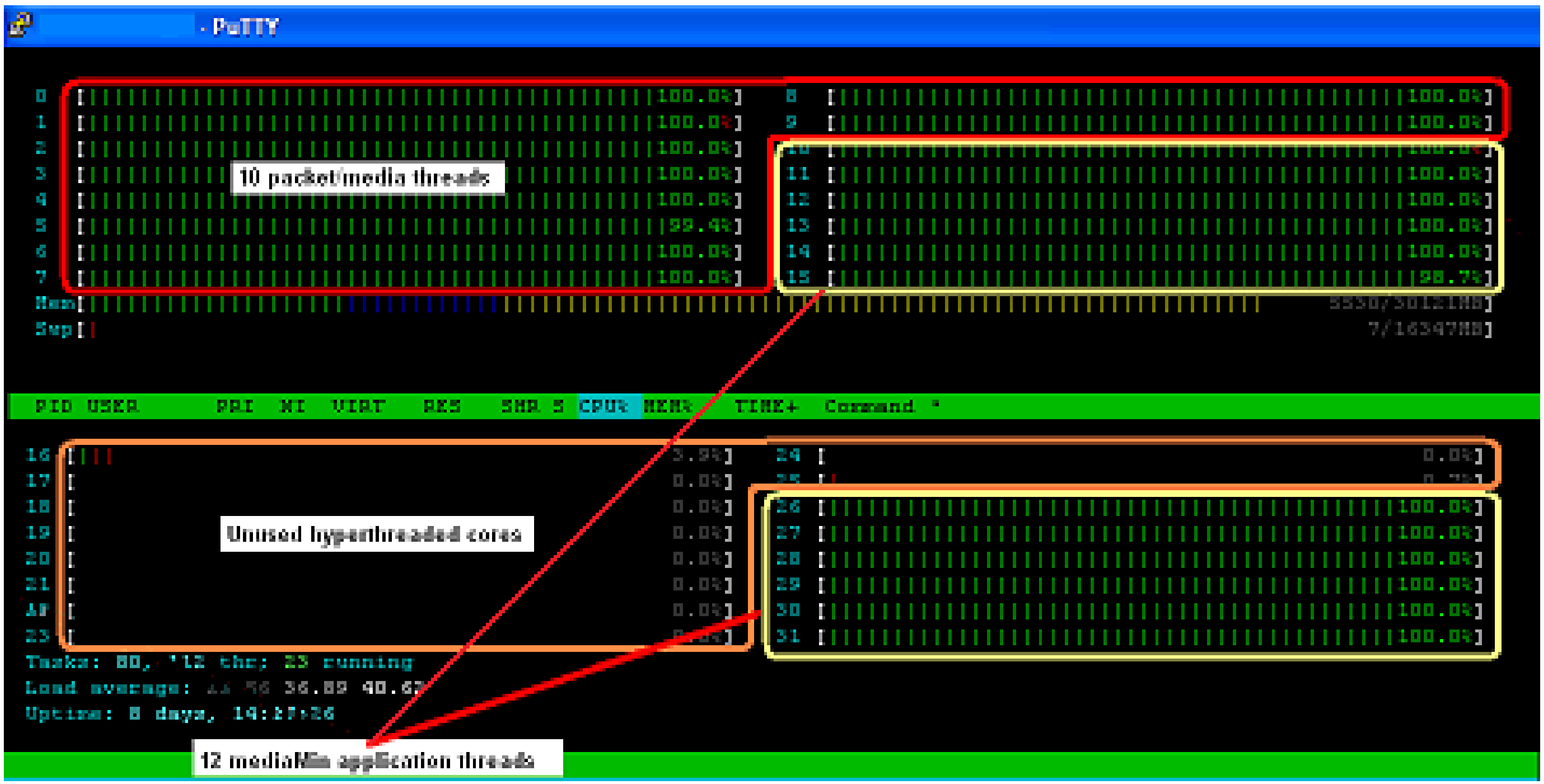
Capacity

- **Performance optimized per box / VM / container**
  - for specified core type and clock rate, we spec a max number of concurrent streams per core. For codecs sample rate and bitrate also specified
  - extensive use of htop to analyze and verify
  - we observe telecom norms – Signalogic has a long history of applications coded for high capacity, real-time performance
- **Codecs**
  - in addition to core type and clock rate, sample rate and bitrate must also be specified
  - https://www.signalogic.com/evs_codec has a Capacity Figure table for EVS on x86

Capacity, cont.

- **Extensive use of htop and to analyze / debug core usage**
  - hyperthreading must be disabled
  - stream groups must not cross core boundaries
  - look for memory leaks

htop screen capture showing
packet/media threads
application threads
disabled hyperthread cores



Capacity, cont.

- **Optimized for Linux**
  - Linux poses performance challenges - not deterministic, not an RTOS
  - carriers and LEAs understand “software defined solutions” are not deterministic, but still expect high capacity / reliability
  - software detects and alarms “thread preemption” – possible performance impairment due to Linux housekeeping and other user applications
- **GPUs and DPDK [1] may or may not be helpful**
  - GPUs
    - don’t help with packet processing
    - only “matrix expressible” operations can be easily accelerated
    - can help with some codecs, but accelerating an entire codec is labor-intensive and requires hand-coding
  - DPDK
    - useful when combined with high-rate packet I/O hardware
    - for PCIe accelerator cards, each x86 core needs a dedicated lane to avoid thread locks

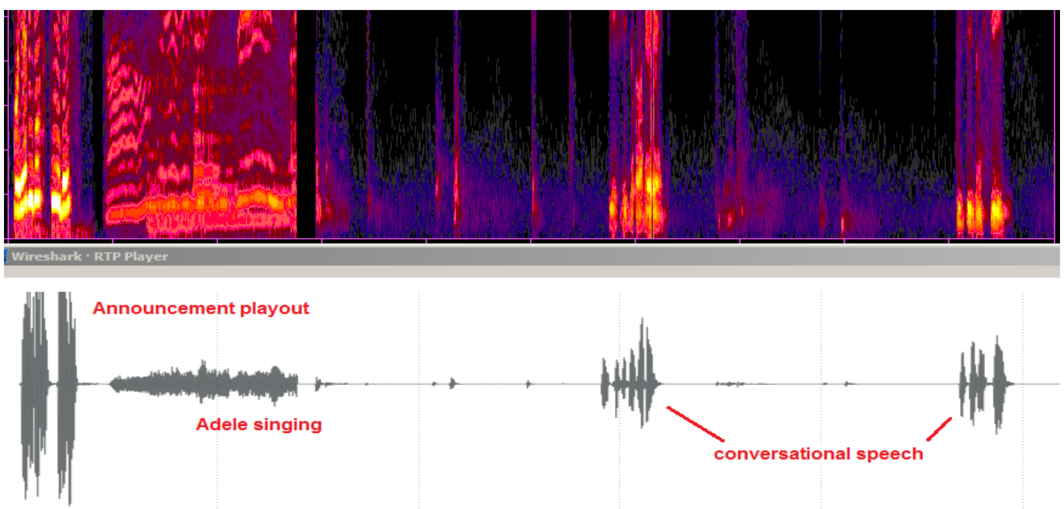
[1] Data Plane Development Kit – refers to non-Linux x86 cores dedicated to packet processing

Reliability and Testing

- **Carriers and LEAs obsess about reliability**
  - very long calls are common. All possible packet and audio data buffers and wrap conditions that could occur must be tested
  - as with capacity, we pay attention to telecom requirements. “5 9s” up time is a minimum
- **Customers run stress tests for weeks at a time**
  - we run stress tests continuously for 6+ months
  - tests include pcaps with artificial wraps, 10x packet push rates, deliberate thread preemptions, more
  - tests run at max capacity ratings
  - currently we run tests on Ubuntu 12.04 gcc++ 4.6.4 thru 20.04 gcc++ 9.3.0. Testing can be provided on CentOS systems as needed
- **Extensive use of htop and valgrind**
  - thorough and painstaking search for memory leaks
- **Software is designed for high reliability**
  - profiling and performance monitoring
  - alarms include data flow anomalies, thread preemption
  - event and packet logging
  - telemetry

Audio Quality

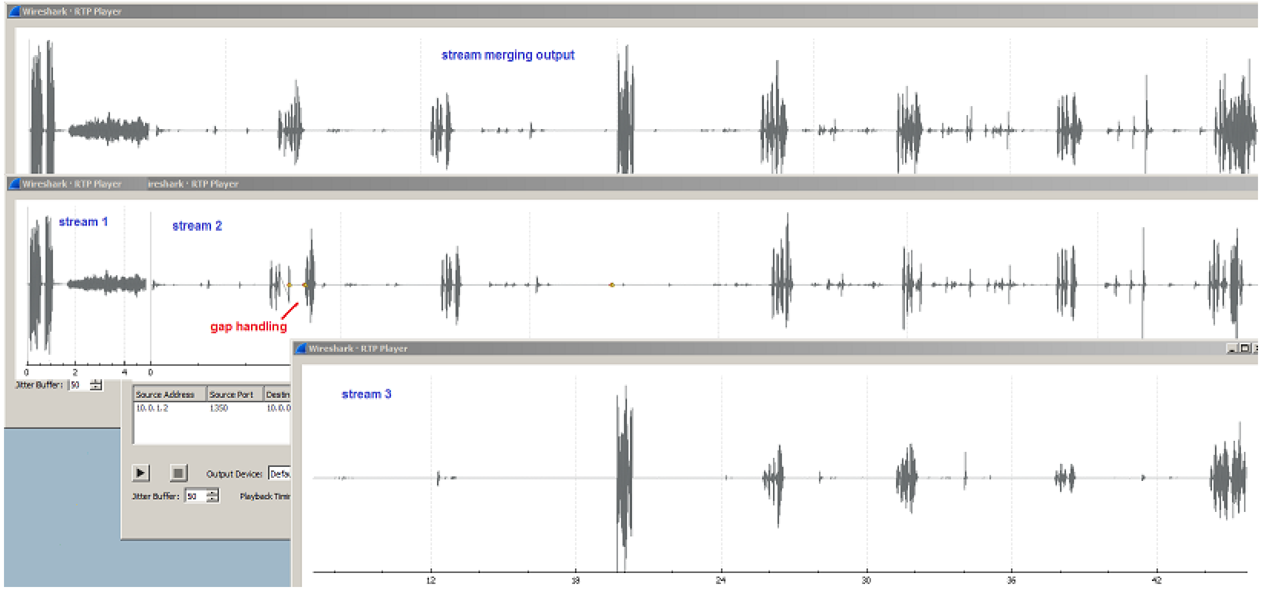
- **Certain customers obsess over audio quality**
  - we have observed customers using metronomes and whale sounds to verify timing and frequency integrity when testing endpoints
- **“No sound left behind”**
  - we enhance audio quality by detecting and repairing:
    - packet problems (lost packets, out-of-order, gaps, bursts)
    - stream timing (overrun, underrun, child streams)
- **Debug capability to identify root cause (CSP, cloud, or vendor)**
  - audio quality is complex and subjective; ability to identify root cause is crucial



Frequency domain analysis and corresponding Wireshark packet capture

Audio Quality Challenges

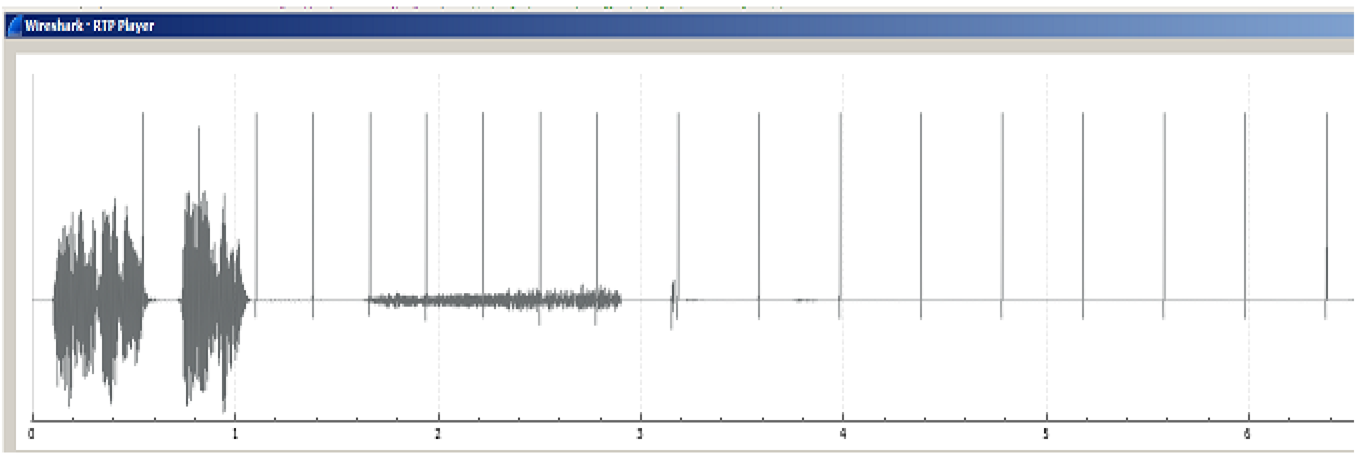
- **Encapsulation artifacts**
  - encapsulation packet rate may be very different than original audio RTP packet rate - slow, fast, variable. We've seen up to ±15%
  - extreme bursts of ooo (out-of-order) packets, 20-50 packets not uncommon
- **Streams not time-aligned**
  - artifacts and child streams distributed unevenly between streams
  - media playout servers are particularly bad offenders



Multiple Wireshark captures showing stream merging of 3 endpoints

Audio Quality Verification and Debug

- **Test case verification**
  - analysis and debug tools can pinpoint whether it's CSP, cloud,or handset issue
  - visual audio markers can be enabled to verify timing, frame repair, etc. Different types of markers are supported



Wireshark screen capture showing audio markers inserted by software

Media Content Processing

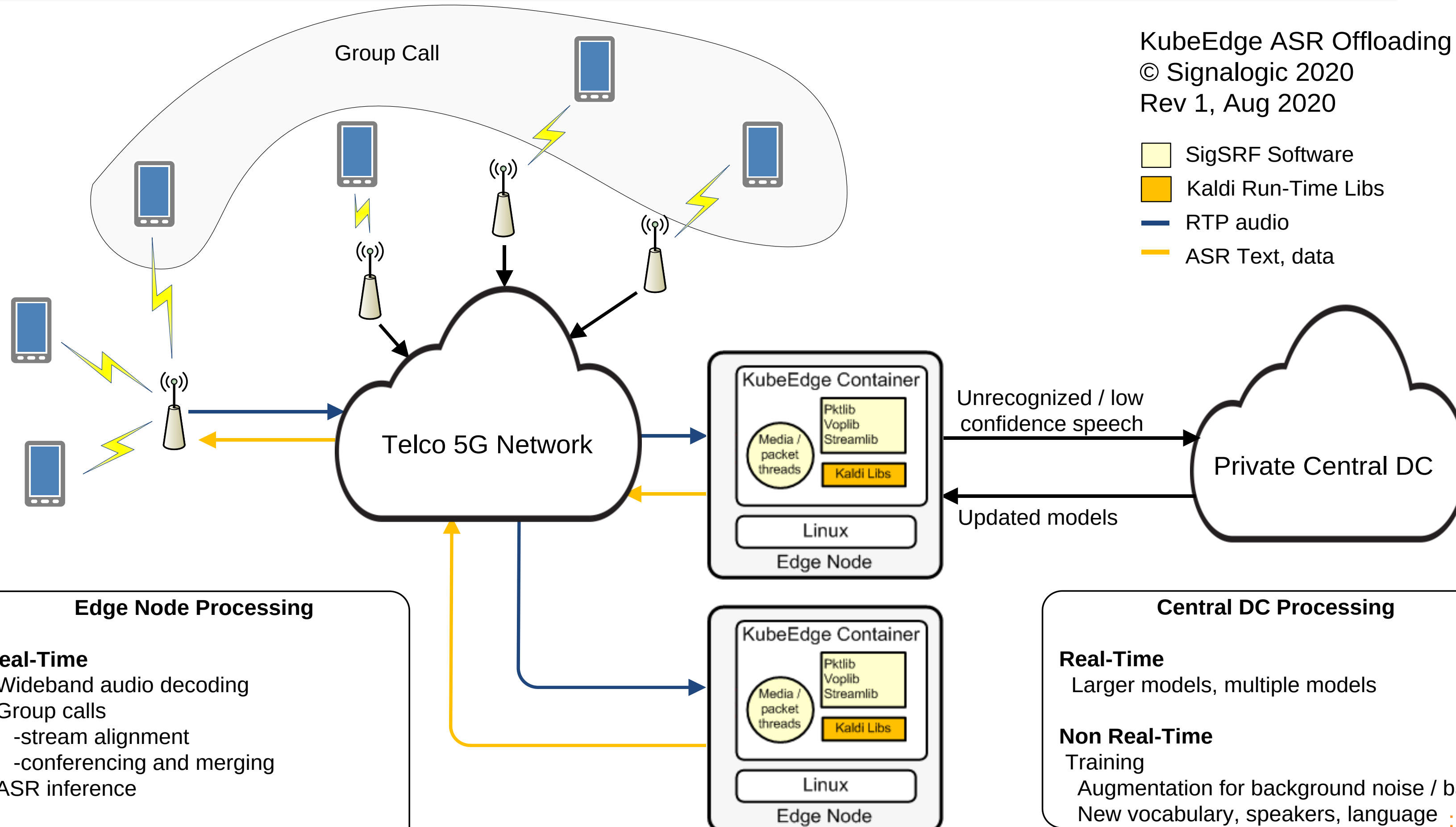
- **Content analysis and signal processing**
  - artifact detection
  - background noise reduction
  - detecting and avoiding conversation overlap (correcting time alignment between streams in a stream group)
  - stream deduplication
- **Content recognition**
  - speech recognition
  - speaker identification
  - we use Kaldi open source
  - requires tradeoff between capacity and real-time processing
- **RTP malware detection**
  - malware payloads can hide in codec packets
  - no way to differentiate “ordinary bad voice” from “deliberate bad voice” without extensive analysis of fully decoded packets

Current R&D

- **Edge Computing**
  - ongoing PoCs and LF Edge blueprints demonstrating hybrid cloud, enhanced privacy / security
  - many telecom carriers do not trust security in public clouds
- **ASR (Automatic Speech Recognition)**
  - can be done in real-time, but substantially less capacity
  - not yet in real-time: individual speaker identification and transcription, known as "diarization"
  - potential to reduce workloads, accurately alert on "conversations of interest"
  - open source accuracy only a few % WER[2] more than proprietary code bases
- **Telecom migration to public cloud**
  - containerized solutions needed
  - LI is a particular problem due to encryption requirements
  - allow CICD[1], for example improving ASR accuracy with "on the fly" training based on collected data

[1] Continuous Integration, Continuous Deployment
[2] WER = Word Error Rate

Edge Computing + Containerization



Edge Node Processing

Real-Time
- Wideband audio decoding
- Group calls
  - -stream alignment
  - -conferencing and merging
- ASR inference

Non Real-Time
- Diarization – not possible in real-time yet

Challenges
- Varying latencies between endpoints
- Background noise
- Overlapping conversations
- Avoid using GPUs when possible

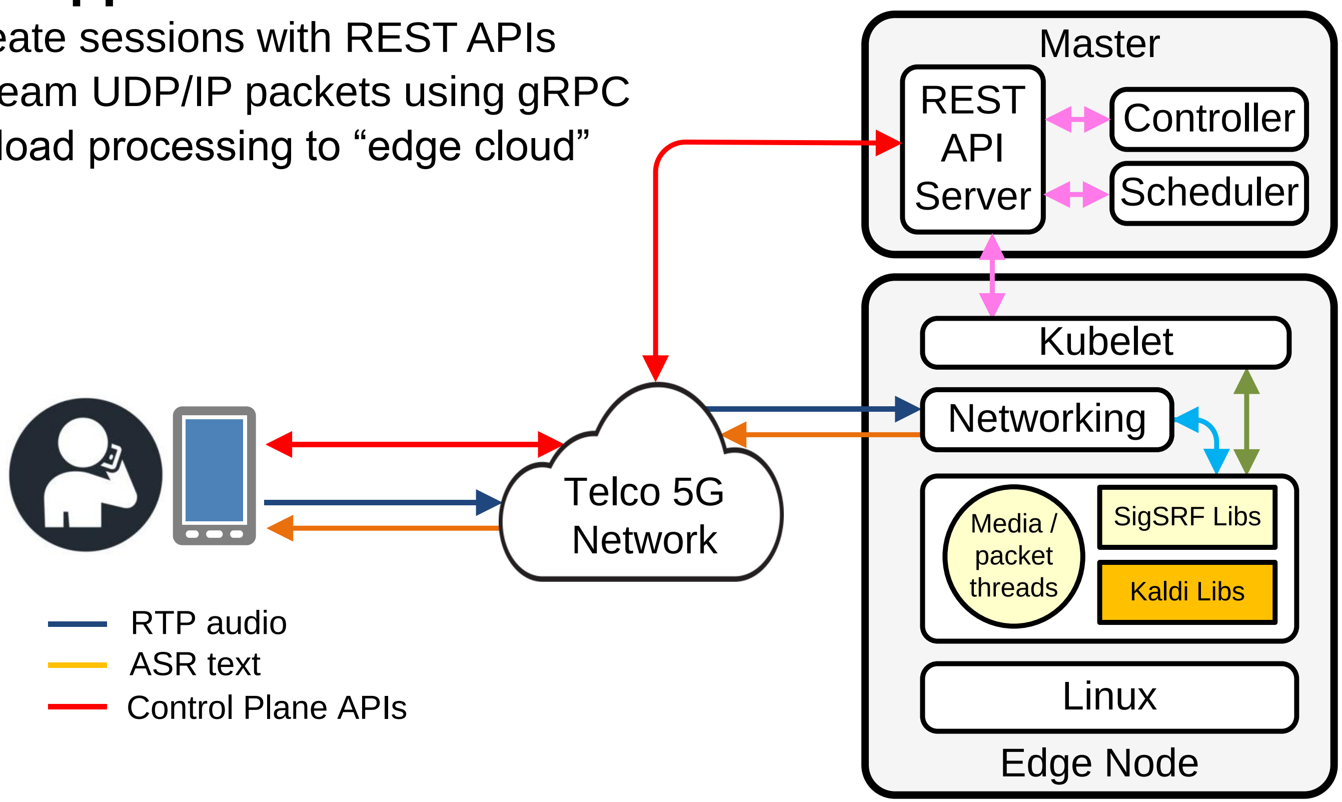
Central DC Processing

Real-Time
- Larger models, multiple models

Non Real-Time
- Training
  - Augmentation for background noise / babble
  - New vocabulary, speakers, language

Containers and Kubernetes

- **Packet + media + ASR inside container**
  - minimum 2 x86 cores, 32 GB mem, 1 TB HDD can handle 32 sessions
  - a session is wideband decode (e.g. EVS), jitter buffer, stream merging up to 8 stream groups, G711 pcap output, wideband wav file output
  - scales up linearly with more cores
- **Field apps**
  - create sessions with REST APIs
  - stream UDP/IP packets using gRPC
  - offload processing to "edge cloud"



LI Perspective

- **ETSI LI Terminology: CC mediation (communication content), HI2 and HI3 (Handover Interfaces)**
- **Packet Handling**
  - Jitter buffer, packet repair, rate adjustment
- **Media**
  - Decoding (AMR, AMR-WB, EVS, more), stream alignment
- **Signal Processing**
  - Stream merging, conferencing, speech recognition

ETSI Diagram – Vendor Software Mapping
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