Adding ASR to an Akraino Robot

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  – long “supply chain” from cloud to edge
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  – operate safely – don’t hurt anyone, don’t damage property. Operate with zero trust in cloud commands
  – operate with energy and size constraints
- **EdgeStream™ - Software**
  – optimized on per thread basis, one thread per core, no spinlocks
  – packet processing, media codecs, signal processing, inference
  – comparison with DeepStream and GStreamer
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  – pico ITX form factor: 3.5” x 3.5”
  – quad-core Atom, no fan
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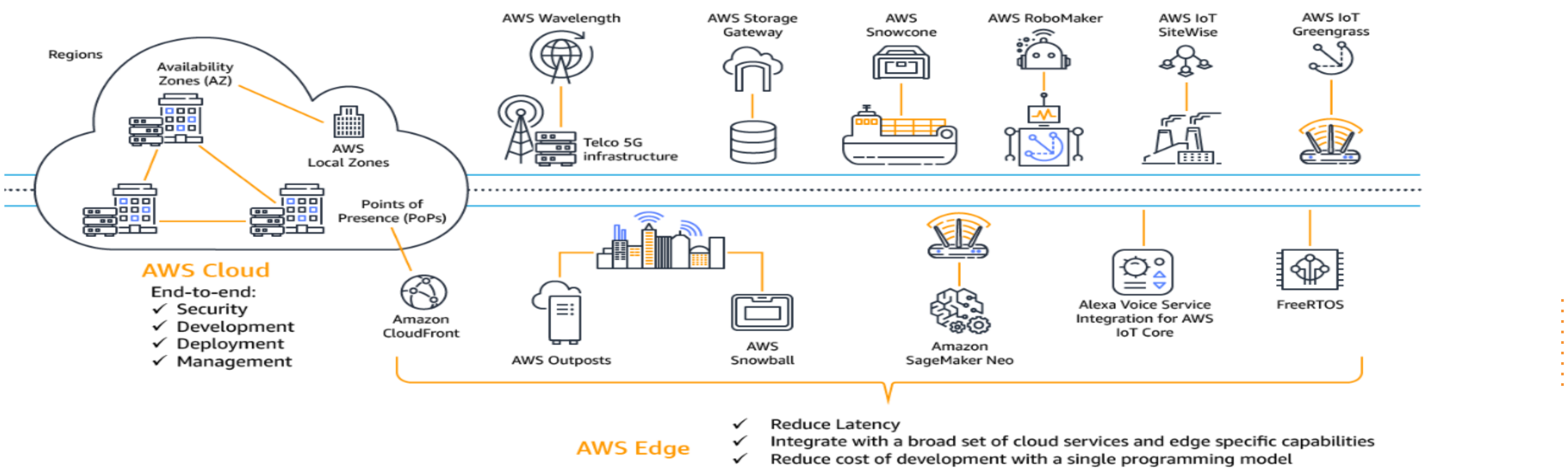
  – Source code

Executive Summary

- **Signalogic is adding ASR (automatic speech recognition) to an Akraino robotics blueprint**
  - key blueprint organizations: Fujitsu and Ritsumeikan University
  - https://wiki.akraino.org/display/AK/Robot+basic+architecture+based+on+SSES
- **Real-time with a 20K word vocabulary**
  - Signalogic software enables both high performance and energy efficient implementation
  - typical robot compute resources are insufficient for both background noise removal and real-time, high accuracy, high vocabulary ASR
  - the blueprint will use a Kaldi 20K word English vocabulary
- **Initial demo is a Roomba**
  - quad-core pico ITX board and battery “dead bugged” on a Roomba
  - voice commands are processed and sent to the Roomba’s API via USB; e.g. “stop”, “come back later”, “turn left”, etc.
- **Planning to include ASR in release 7 of the Fujitsu / Ritsumeikan blueprint**

Problems Facing Robotics ASR

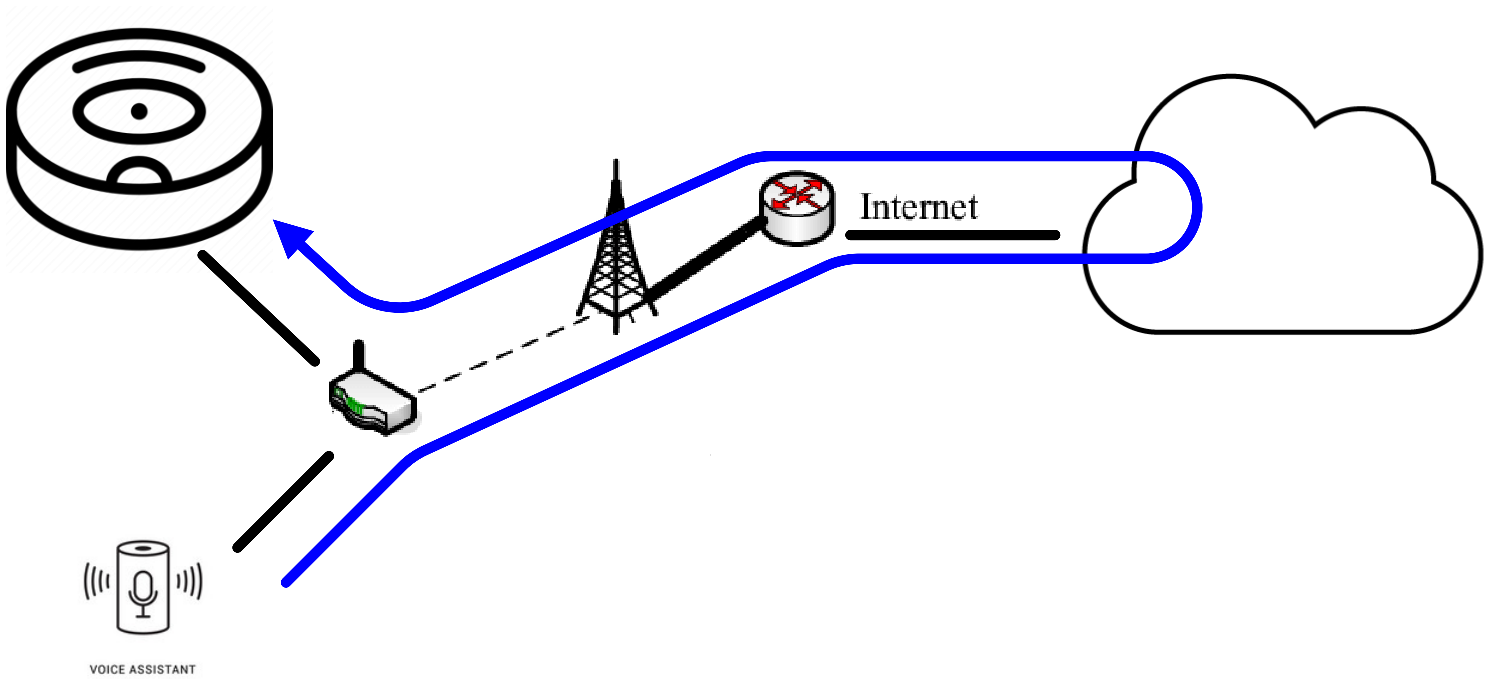
- **Robots are a quintessential edge application**
  - compute resources and power consumption are strictly limited. There is no "economy of scale"
  - operation must be continuous / fail-safe, regardless of intermittent internet connectivity
- **Human safety is paramount**
  - accuracy must be high – when someone says "stop" or "watch out" robots must immediately do exactly that
- **Edge data is private data**
  - Images and audio may contain identifying / proprietary information. ***Audio data cannot be sent to the cloud*** over a "long supply chain" of zones, regions, services



Source: AWS, 2020

Problems, example

- **Why talk to your roomba through the cloud ?**
  – the “centralized cloud” model is based on huge scale, data collection, analytics, command and control, and complexity
  – not based on energy and performance efficiency, safety, and data privacy



Robot Needs

- **Deal with environment at the edge**
  - respond to voice commands, especially urgent commands
  - vision - avoid people, pets, property. Don't break things
- **Operate safely - don't hurt anyone, don't damage property**
  - operate with zero trust in cloud commands – the measurable physical situation always takes precedence
  - actively pursue and detect malware
- **Share content with other edge nodes**
  - share private edge data with other private nodes
- **Energy and size constraints**
  - operate with 50 – 75 W power consumption with no fans
  - operate in extremely small-form factor
  - must be physically lightweight, especially for mobile robots

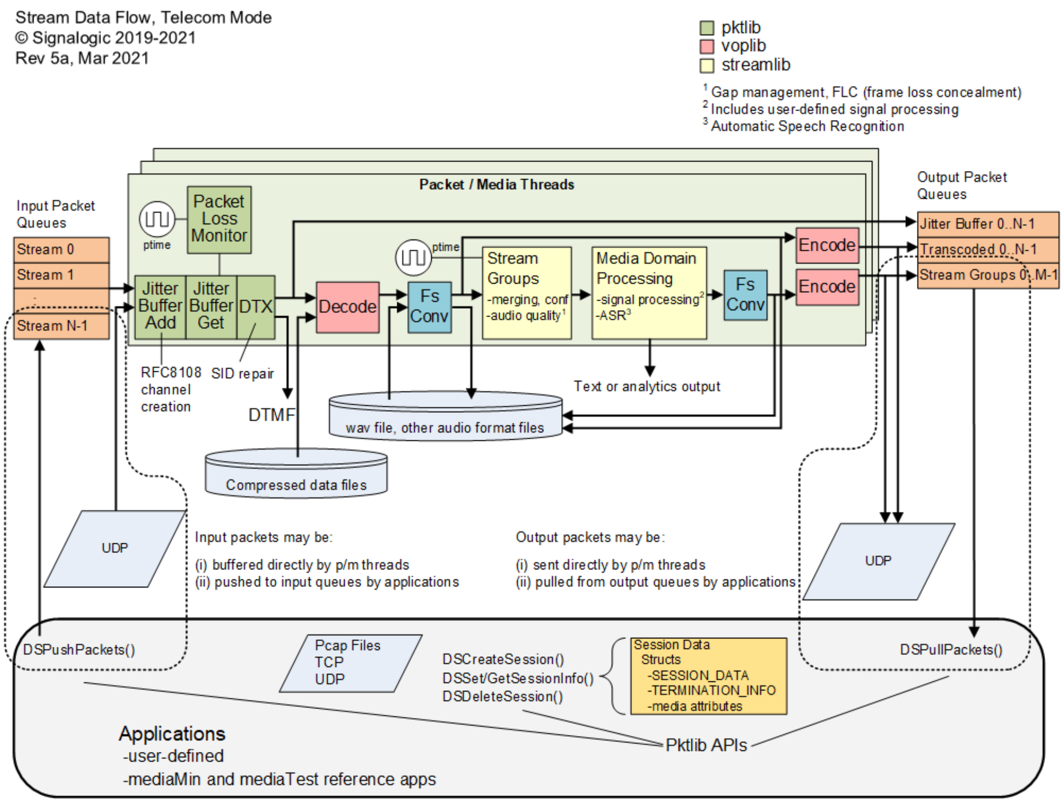
"Beat it roomba, come back later"

Software – EdgeStream™ Platform

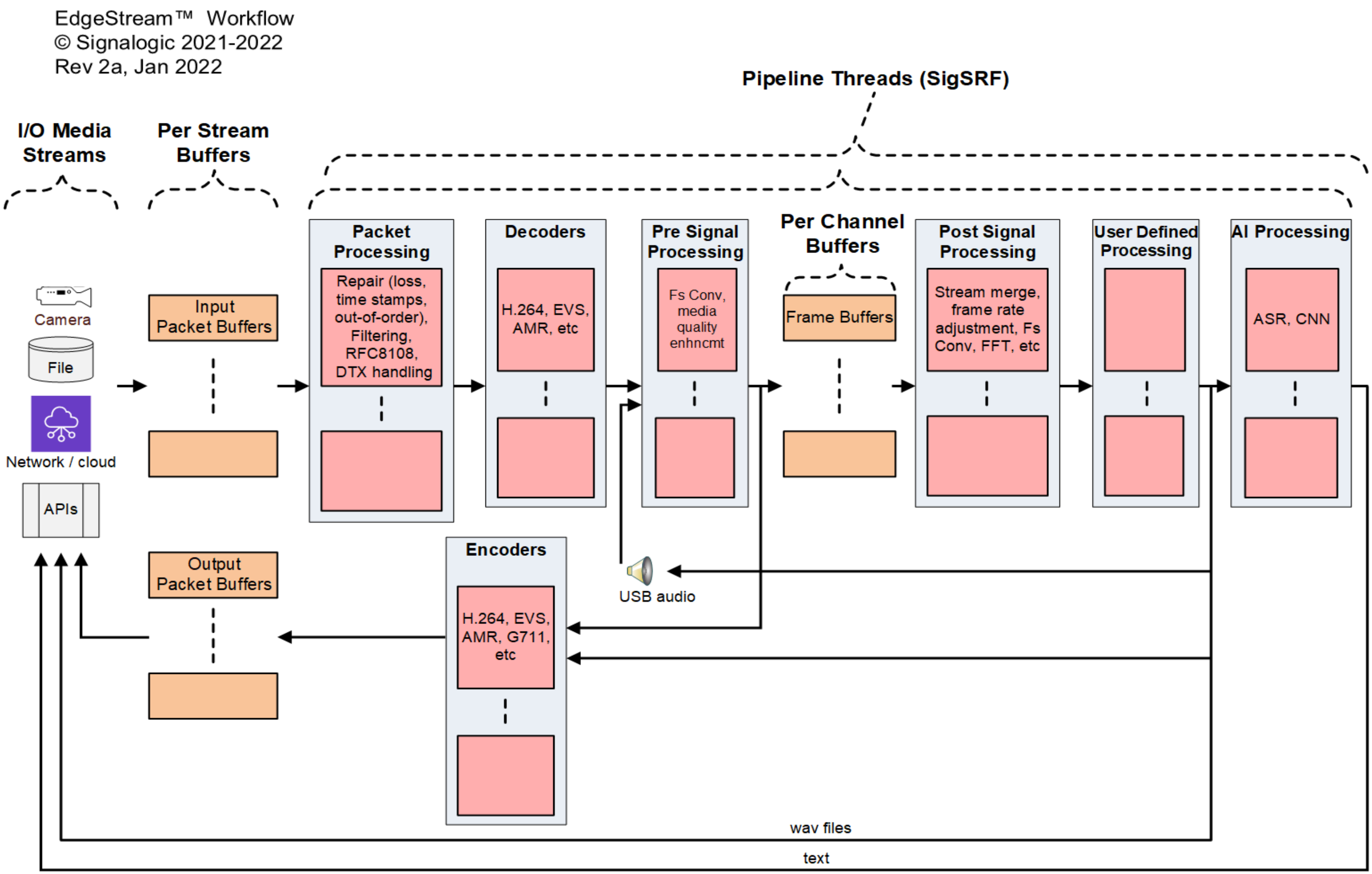
- **Input from USB audio or RTP packets**
  - input from one more microphones
  - input from RTP packet streams (IP/UDP, microphones with Ethernet interface). A wide range of codecs supported (AMR, EVS, G711, etc)
- **Pre-processing (e.g. background noise removal)**
  - environment background noise
  - robot noise, e.g. motors, wheels, brushes
- **Application specific processing if needed**
  - lawful intelligence / interception
  - telecom
  - application-specific signal processing
- **ASR**
- **Output commands to connected devices as needed**
  - translate recognized text to command APIs

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

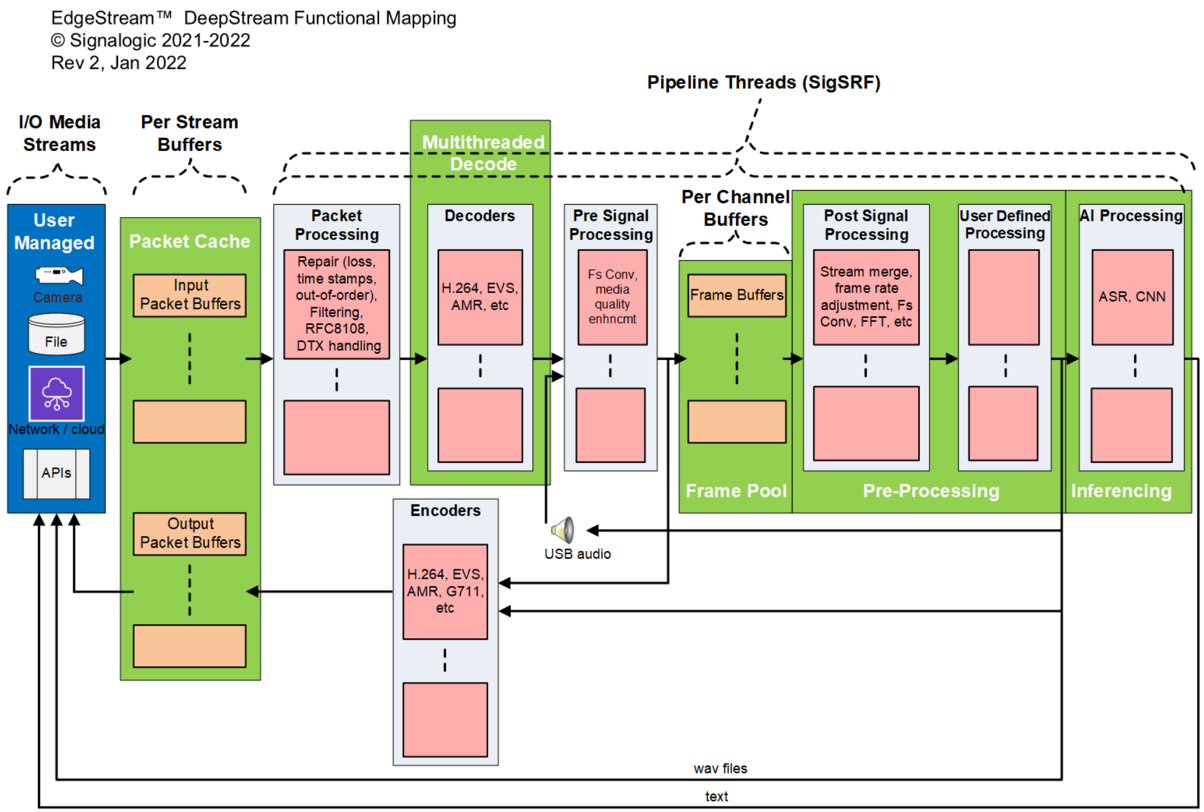


EdgeStream Workflow



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

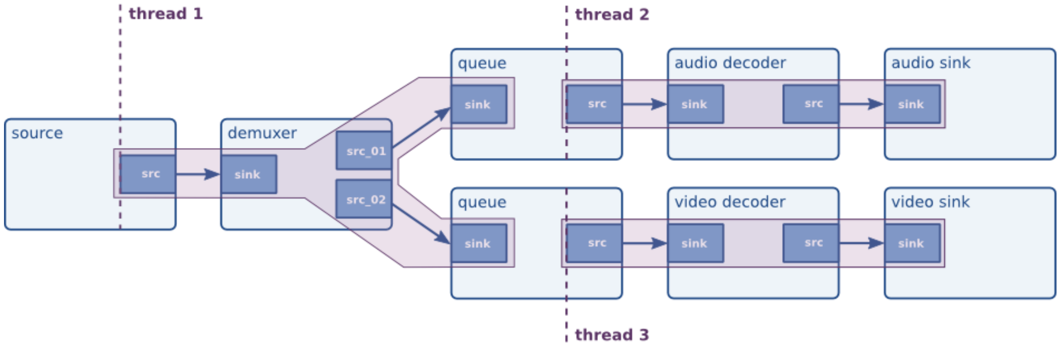


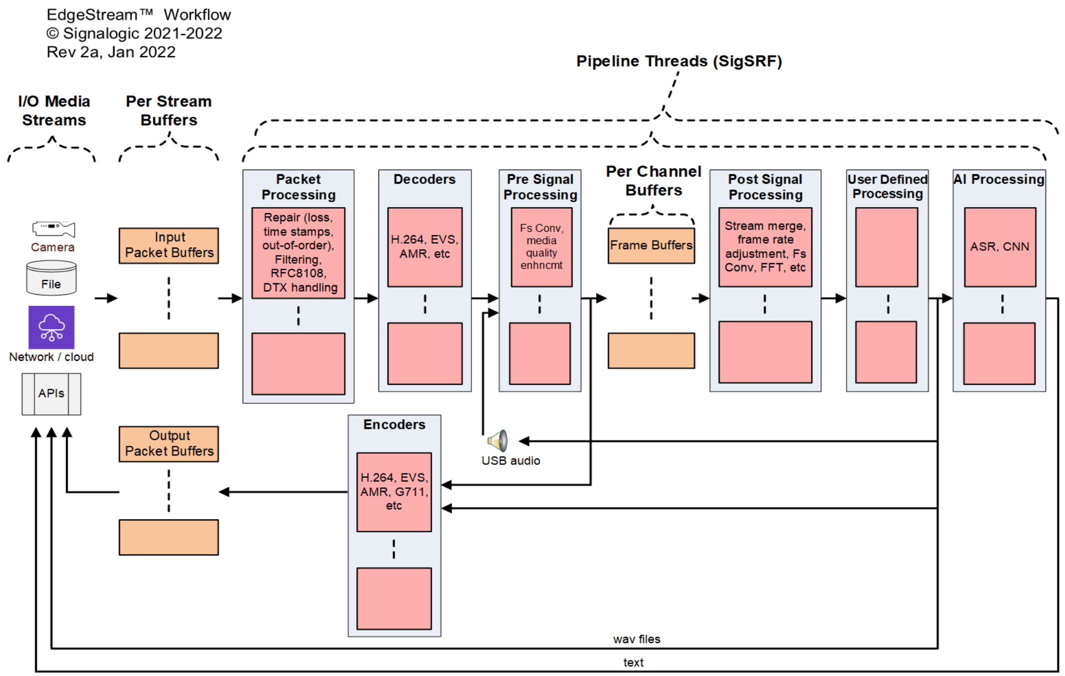
DeepStream is a trademark of Nvidia Corp

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

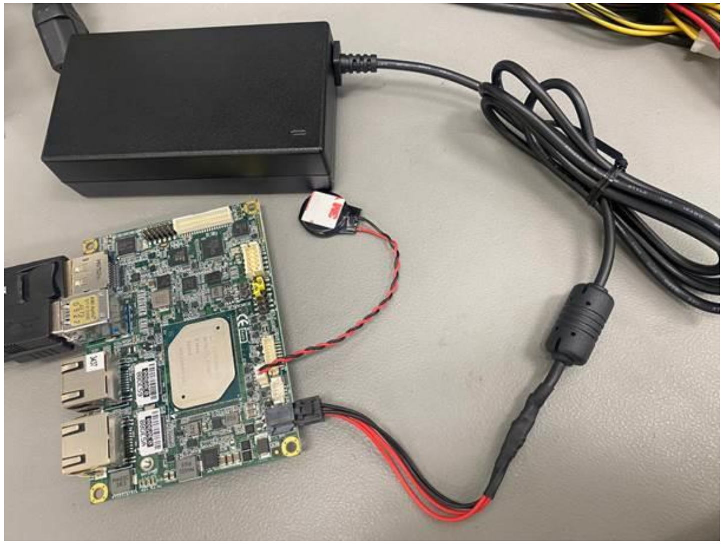
Example GStreamer Workflow

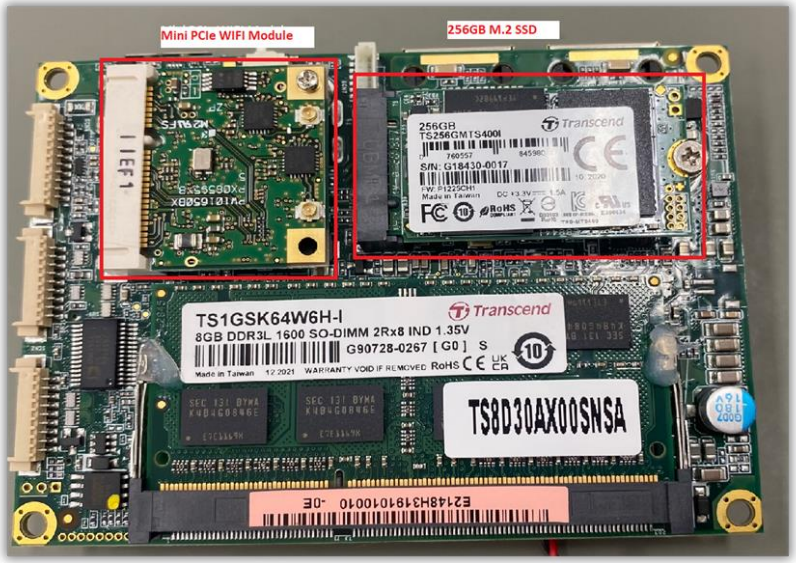




Hardware – EdgeStream™ Platform

- **Pico ITX form-factor board**
  - 3.5” x 3.5” form-factor
  - quad-core Atom, 10 to 20 W power consumption
  - integrated SSD, WiFi, HDMI, etc.
  - no fans
  - boots Ubuntu 20.04
- **Essentially a “robot server”**
  - straightforward to develop code on lab and cloud servers, then run on robot servers
  - lab / cloud servers can simulate robot servers by controlling number of cores and clock rate
  - handles up to three (3) concurrent ASR streams, or three (3) far-field microphones for one stream, or a combination





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/

Github and Docker Hub

- **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 !

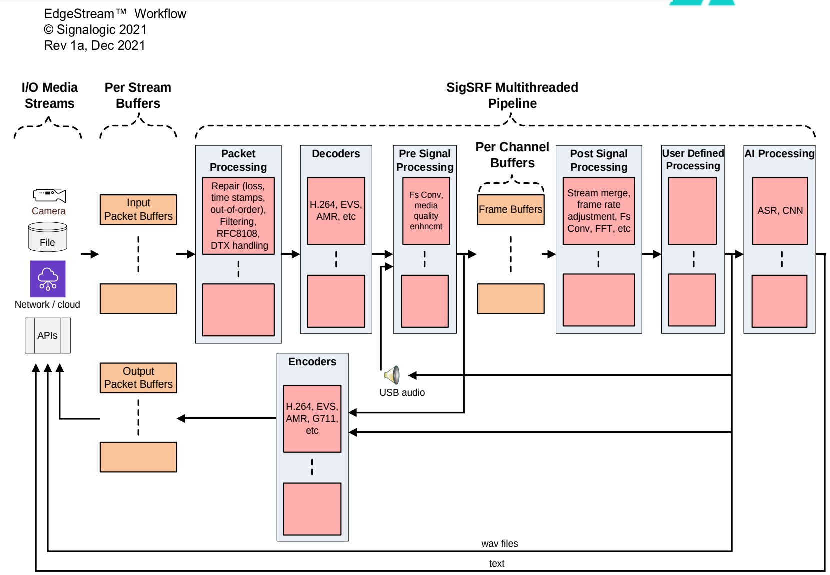
- **Q&A**
- **Follow-up questions / comments: info@signalogic.com**
- **Web page: https://signalogic.com/edgestream**

Supplemental

- **Following slides are background info …**

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

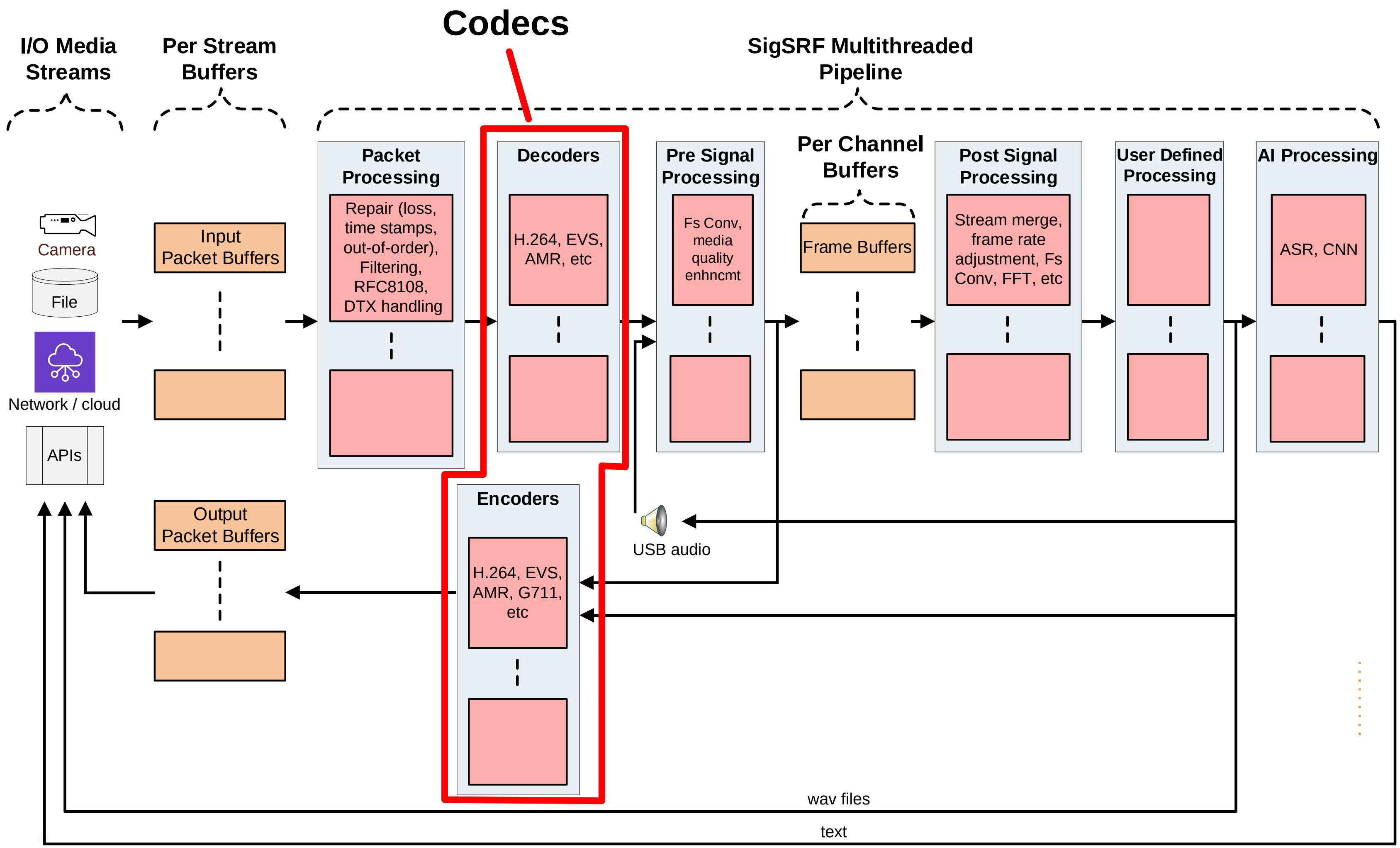


- **EdgeStream™ applications**
  - reference apps for customer-defined development
  - also used as-is by many of our customers. Most common: telecom, LI, and ASR
  - key features
    - dynamic session creation
    - packet push/pull API interface with SigSRF libs
    - multiple streams from multiple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

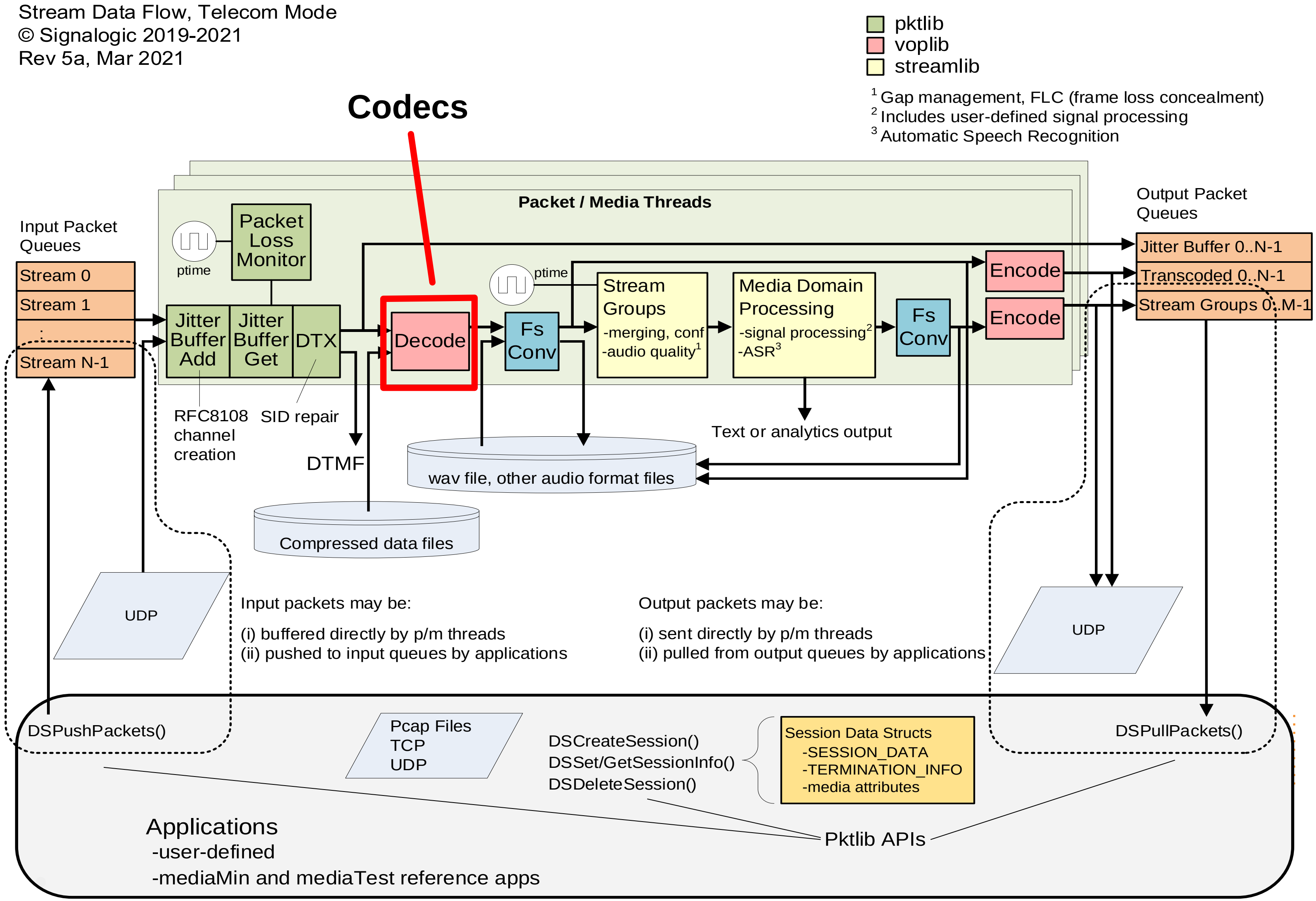
Overview – Pipeline Flow

EdgeStream™ Workflow

Rev 1a, Dec 2021



Overview – Per Thread Data Flow



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 = 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
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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
```
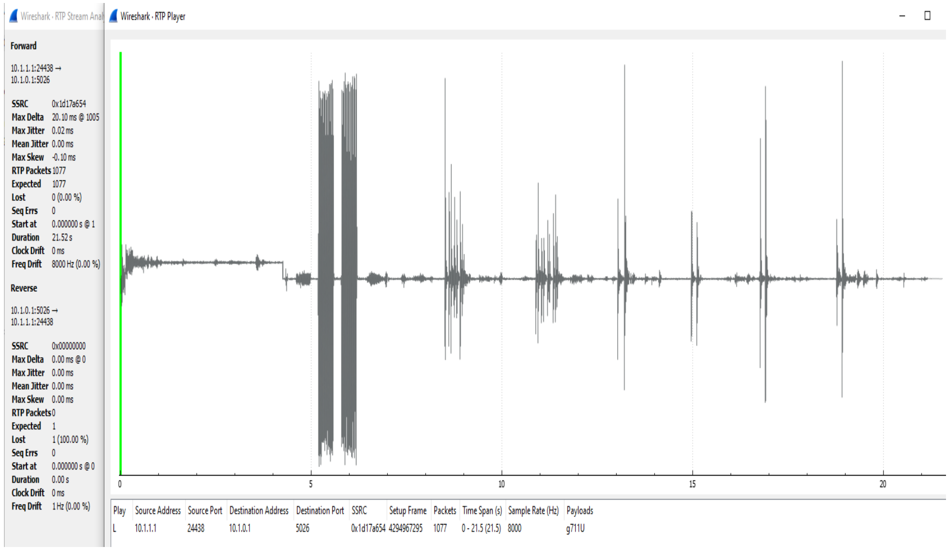

High amount of ooo (out-of-order) example

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**

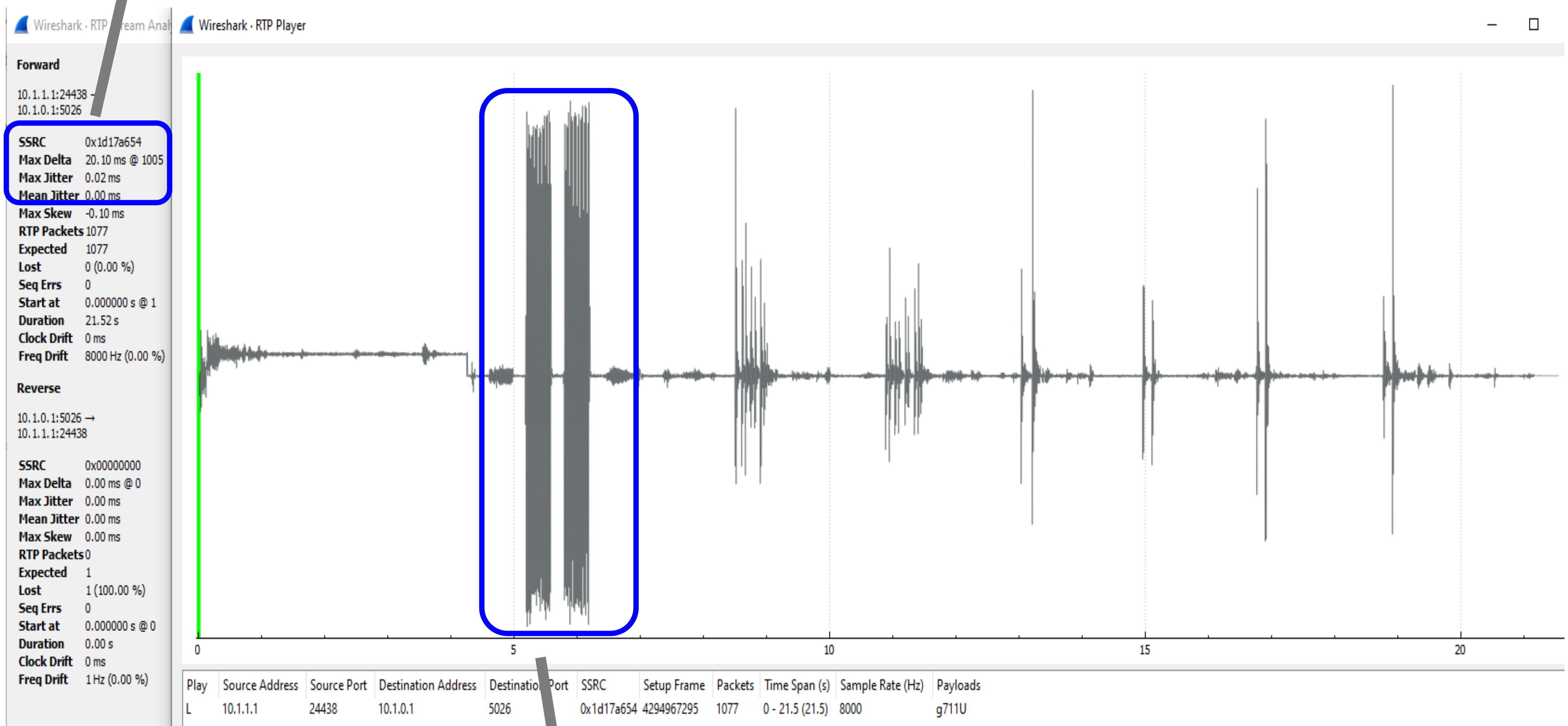


Real-Time Streaming Output Example

Signalogic

AKRAINO

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

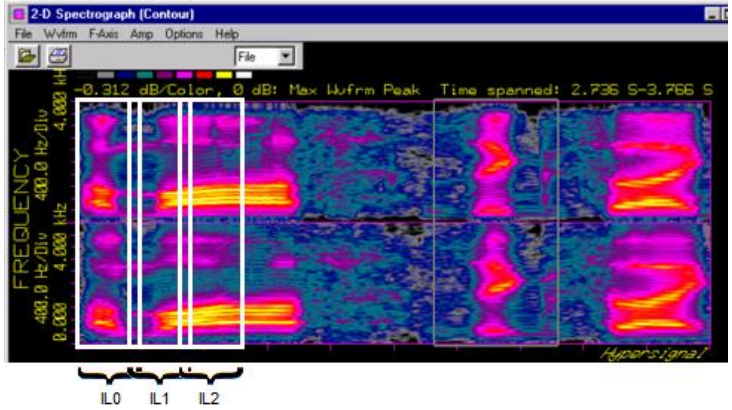
Functionality – Codec API

```
/* 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

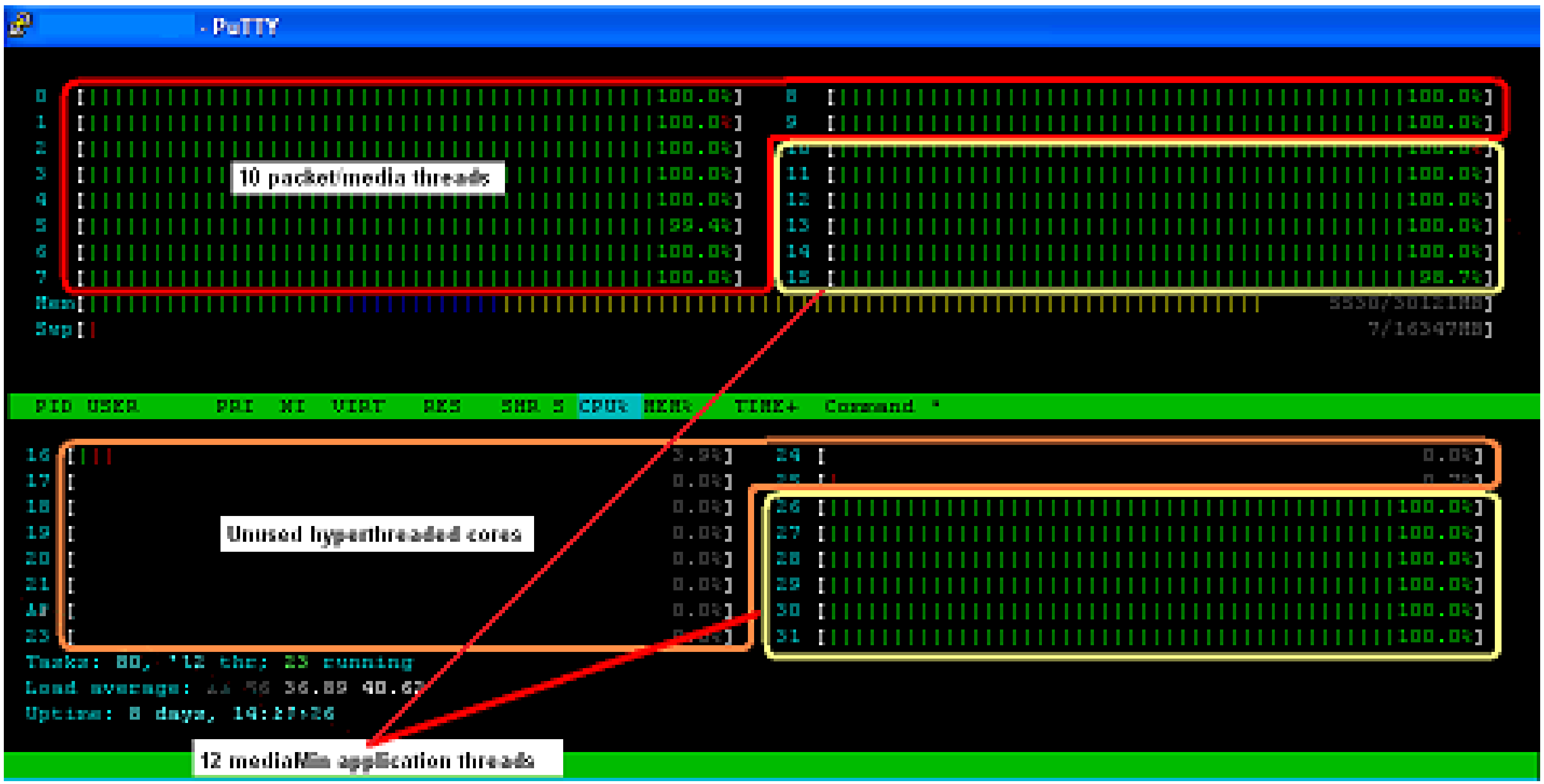
Capacity

- **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

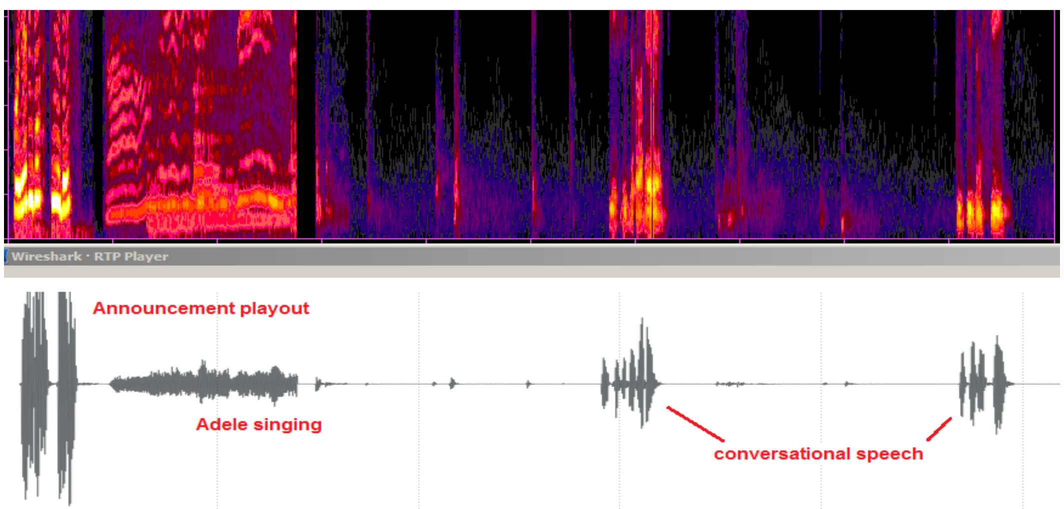
[1] Data Plane Development Kit – refers to non-Linux x86 cores dedicated to packet processing

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

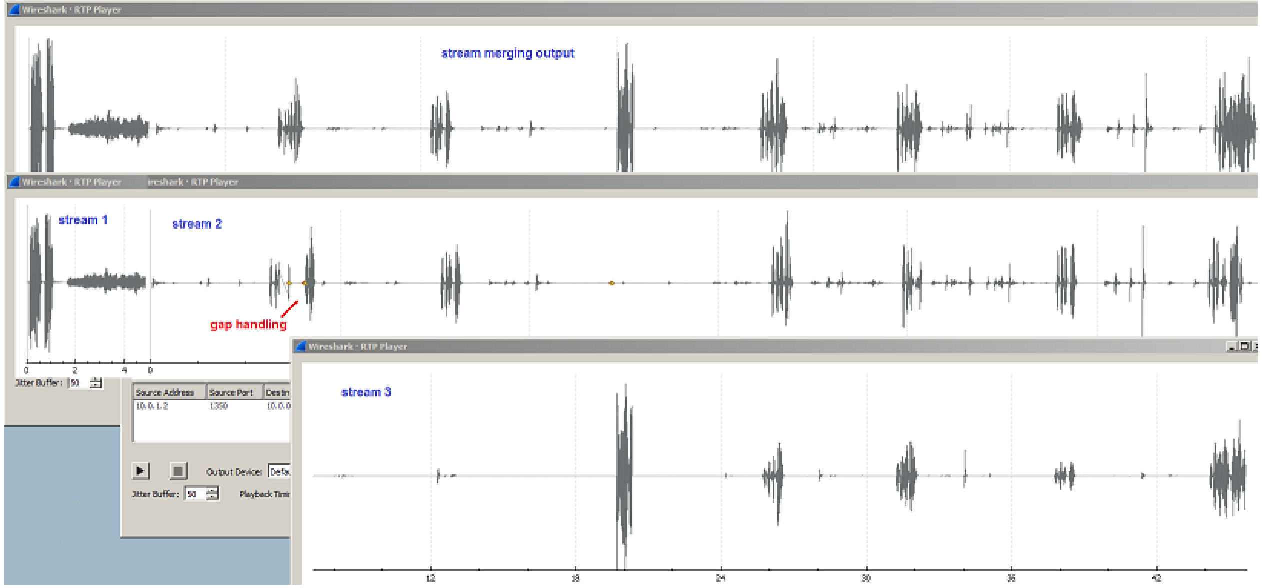
- **Certain customers obsess over audio quality**
  - we have observed customers using metronomes and whale sounds to verify timing and frequency integrity when testing endpoints
- **“No sound left behind”**
  - we enhance audio quality by detecting and repairing:
    - packet problems (lost packets, 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

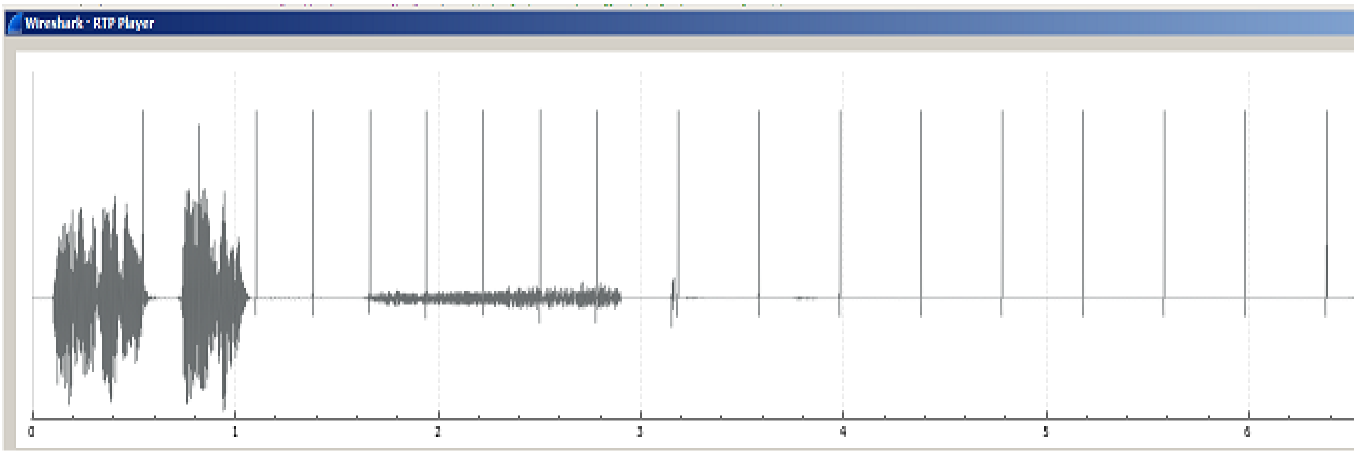
- **Encapsulation artifacts**
  - encapsulation packet rate may be very different than original audio RTP packet rate - slow, fast, variable. We've seen up to ±15%
  - extreme bursts of ooo (out-of-order) packets, 20-50 packets not uncommon
- **Streams not time-aligned**
  - artifacts and child streams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

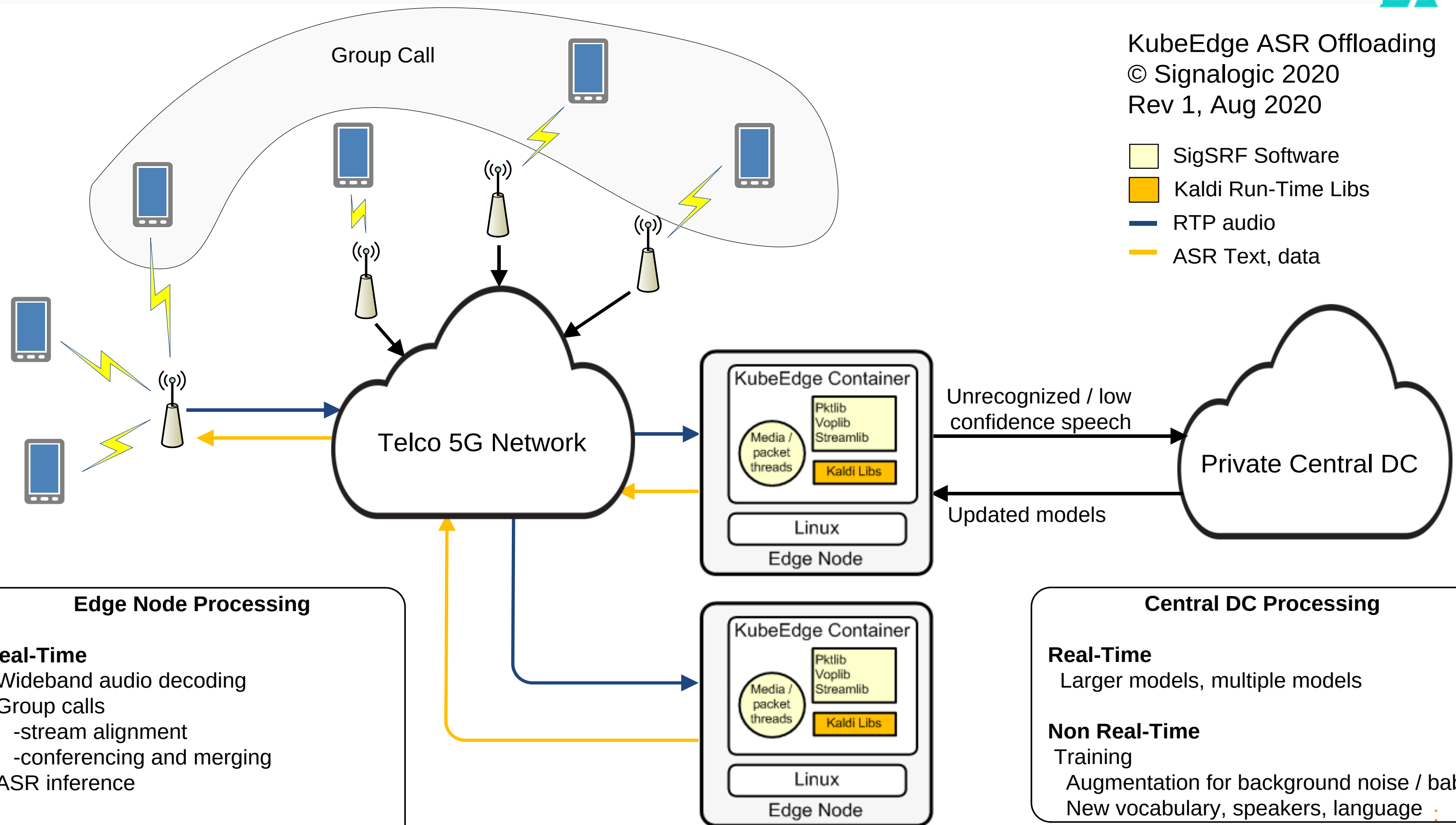
- **Content analysis and signal processing**
  - artifact detection
  - background noise reduction
  - detecting and avoiding conversation overlap (correcting time alignment between streams in a stream group)
  - stream deduplication
- **Content recognition**
  - speech recognition
  - speaker identification
  - 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

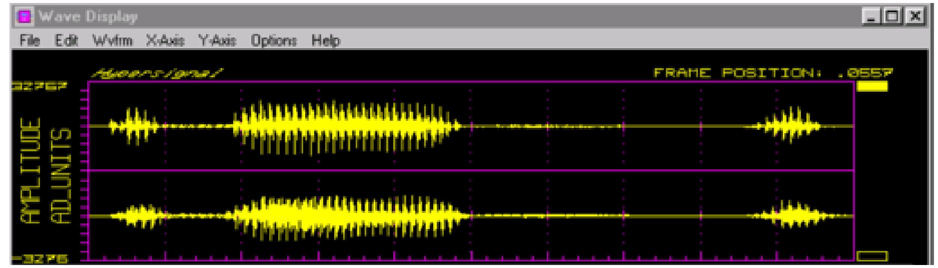
Central DC Processing

Real-Time
- Larger models, multiple models

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

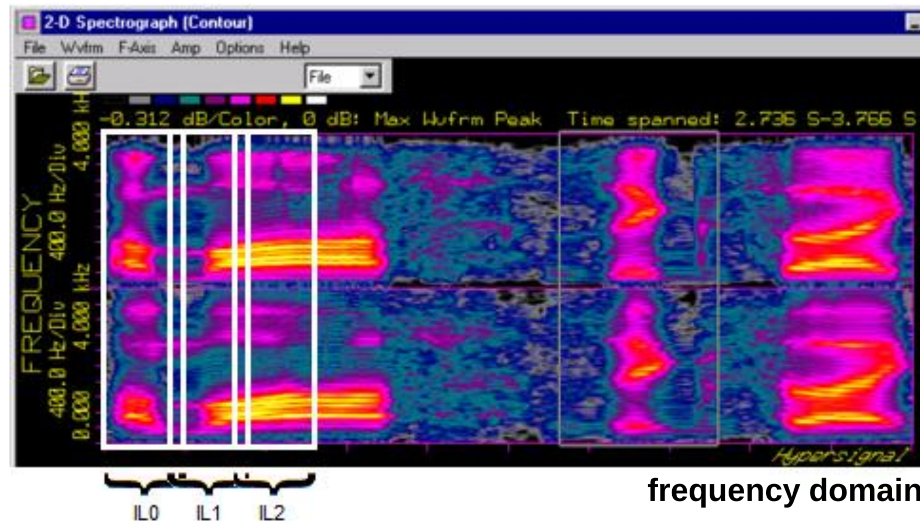
ASR Basics

- **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]



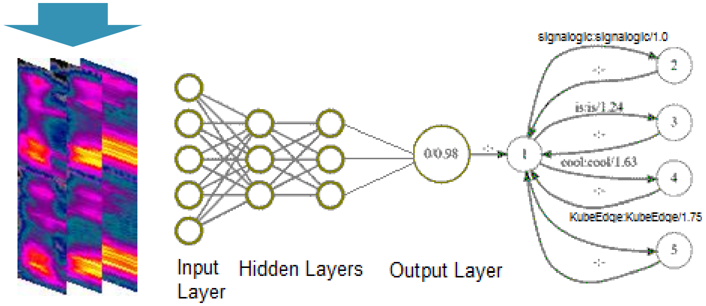
time domain (time series)

Sliding FFT



frequency domain

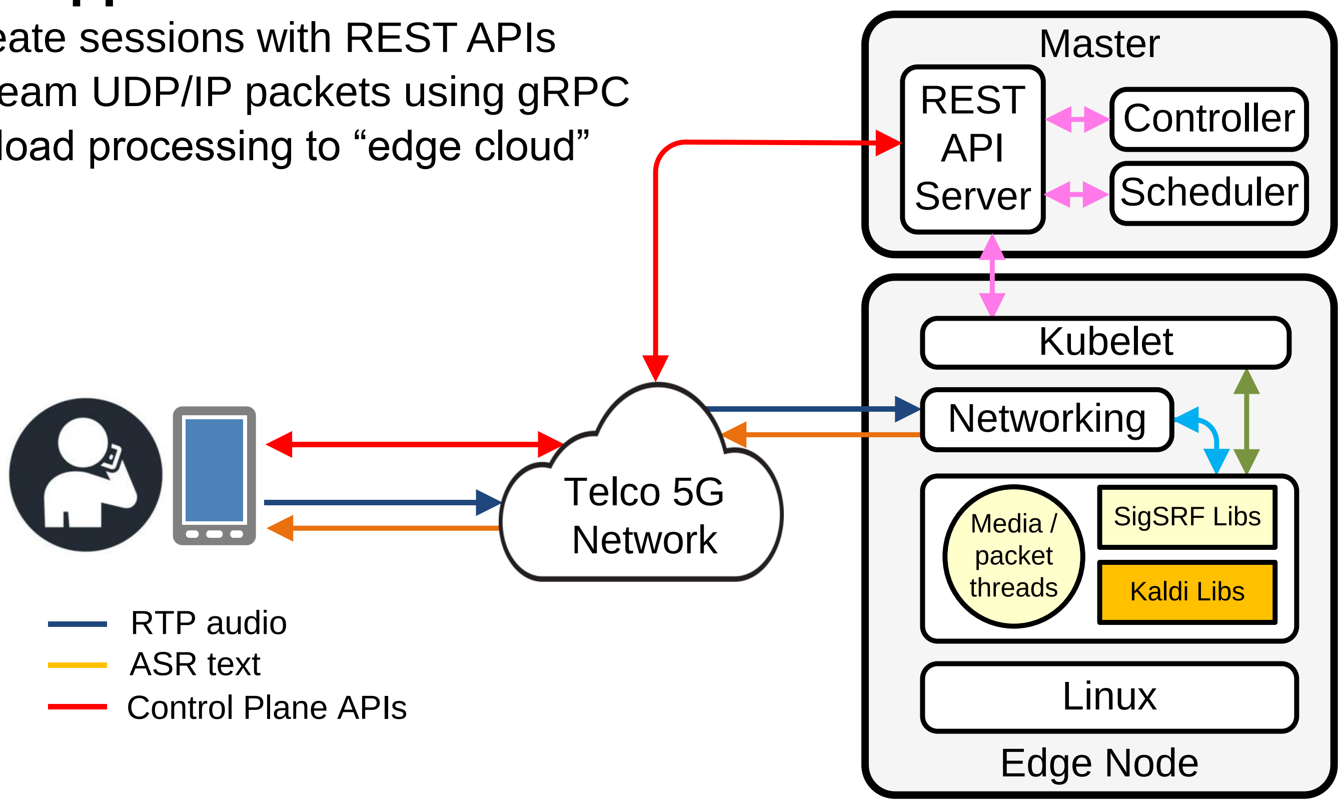
DNN Input Layers (ILn)



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

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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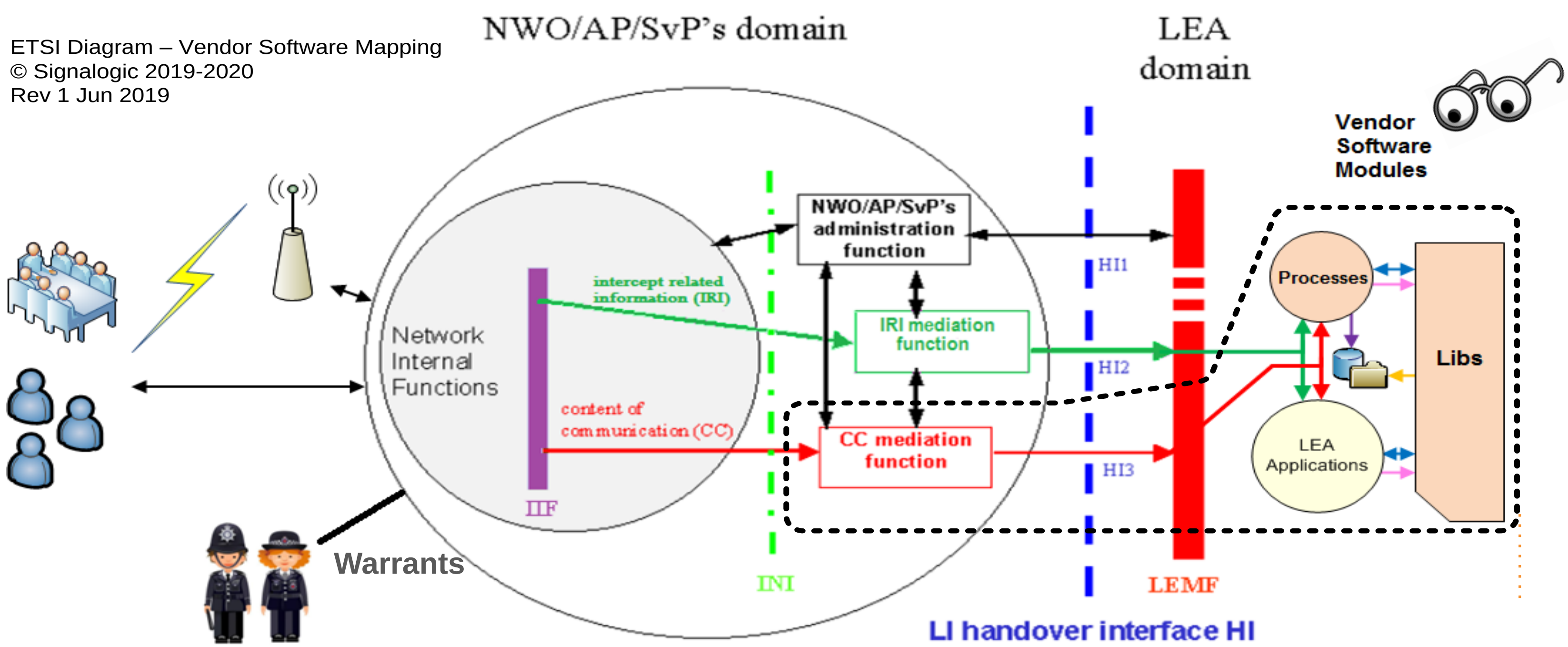

IIF: internal interception function
INI: internal network interface

HI1: administrative information
HI2: intercept related information
HI3: content of communication