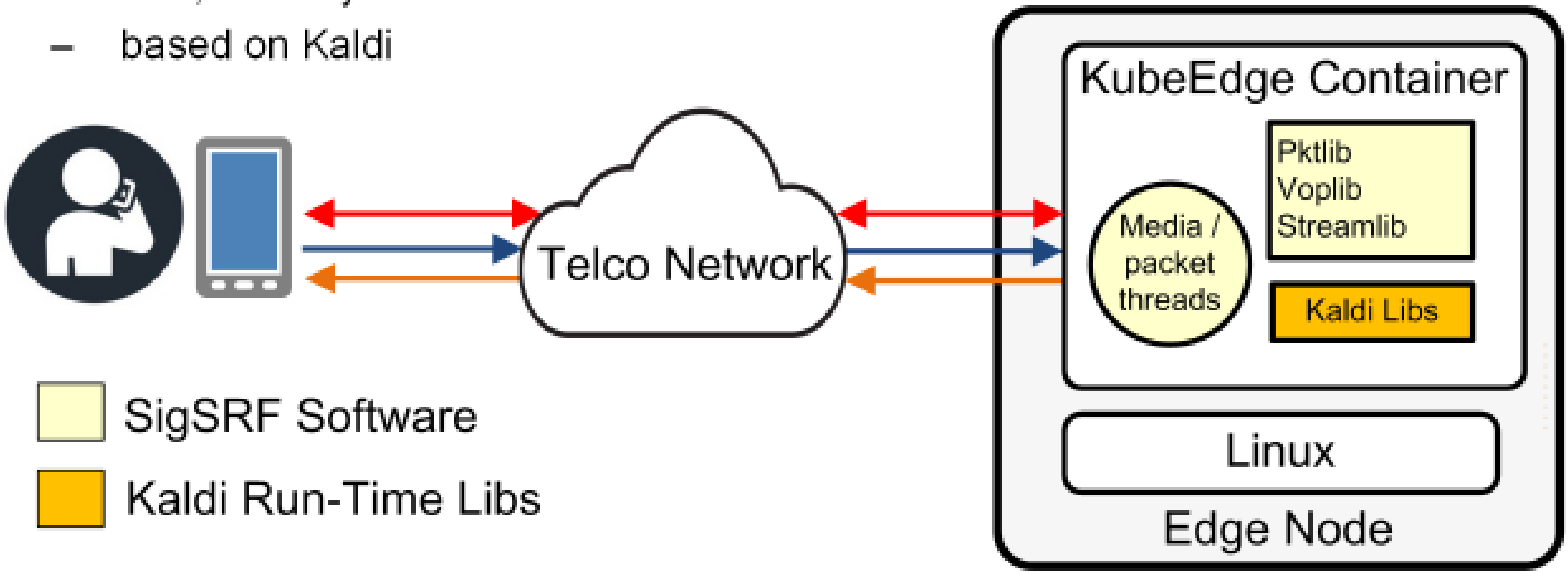
KubeEdge ASR Offloading

Signalogic, Aug 2020

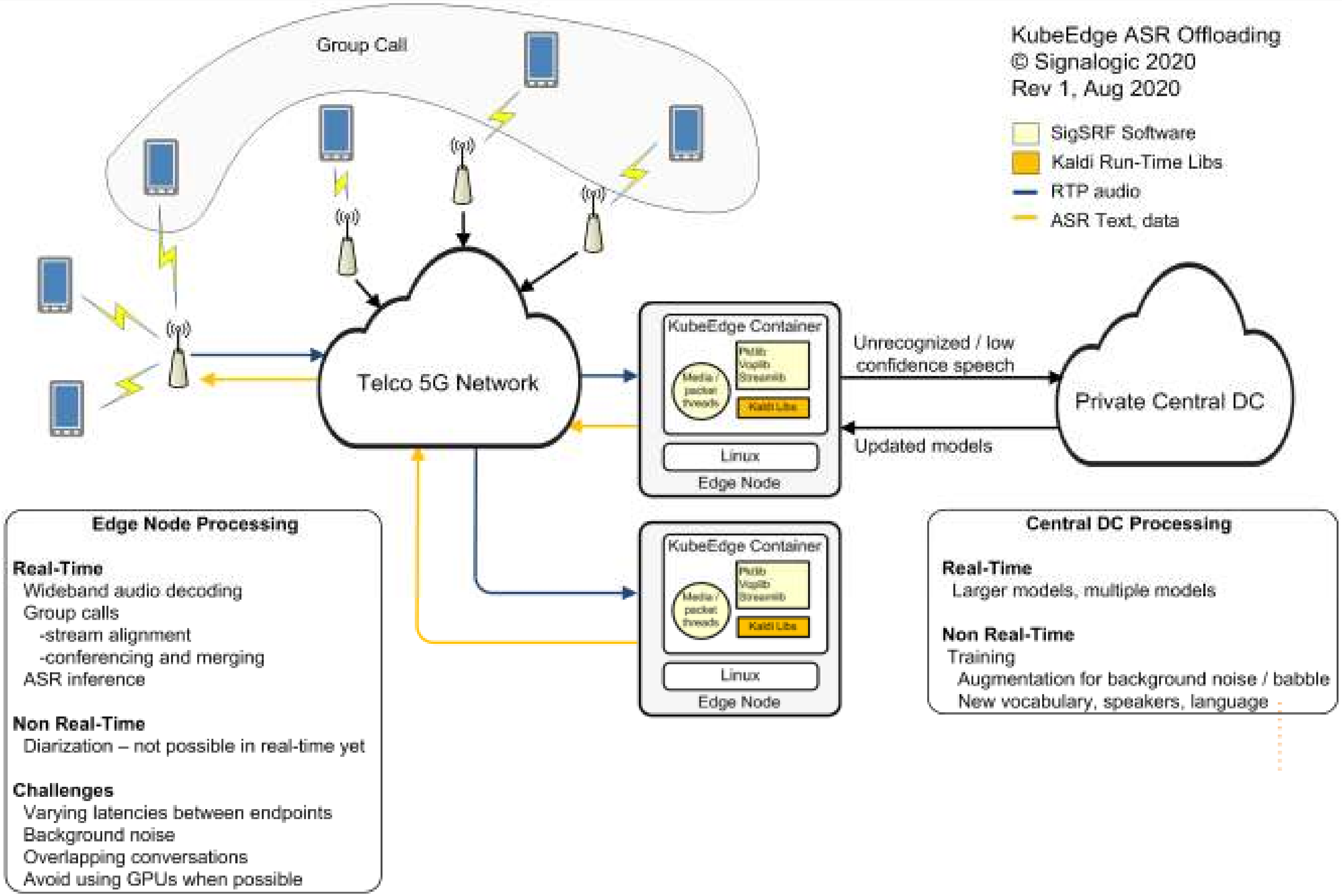
Contents

Overview

- **Signalogic is building an ASR offloading demo for KubeEdge**
- **SigSRF packet + media processing software**
  - SigSRF is deployed by telecoms, LEAs, and analytics customers worldwide
  - includes a robust packet interface, including wideband audio codecs, jitter buffer and packet loss handling, stream alignment, etc
- **ASR**
  - new, recently added
  - based on Kaldi



ASR Offloading

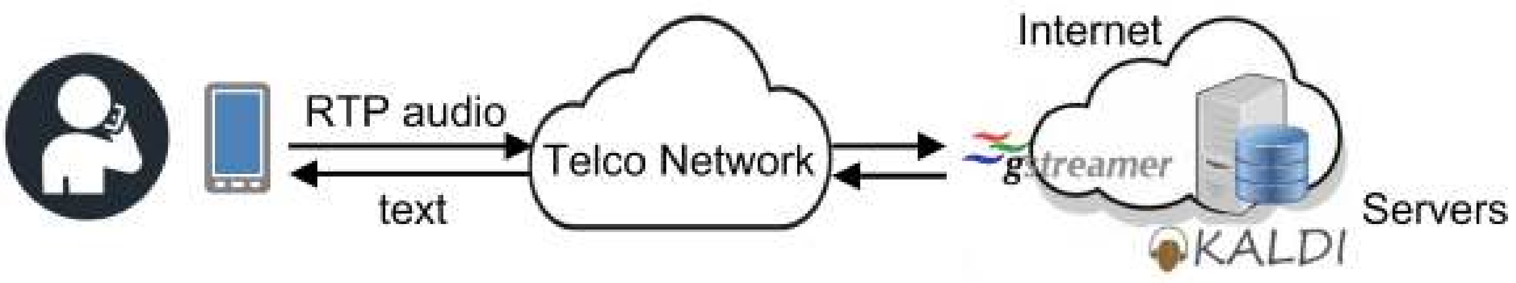


Demo Capability

- **ASR based on Kaldi's mini-librispeech model**
  - subset of librispeech model, which has 200k word vocabulary (English)
  - trained with fewer hours, producing a smaller model easier to use for development and testing
  - demo uses pre-trained x-vectors and i-vectors – no training required
- **SigSRF packet + media software**
  - codecs - AMR, AMR-WB, EVS, G729, G726
  - RFCs - child streams (8108), DTMF (4733), 7198, others
  - concurrent sessions - 8 (demo subset of 512)
  - packet handling - jitter buffer, DTX, packet loss mitigation
- **Call groups (one or more endpoints)**
  - conferencing, merging, deduplication
  - ASR is applied to call group output

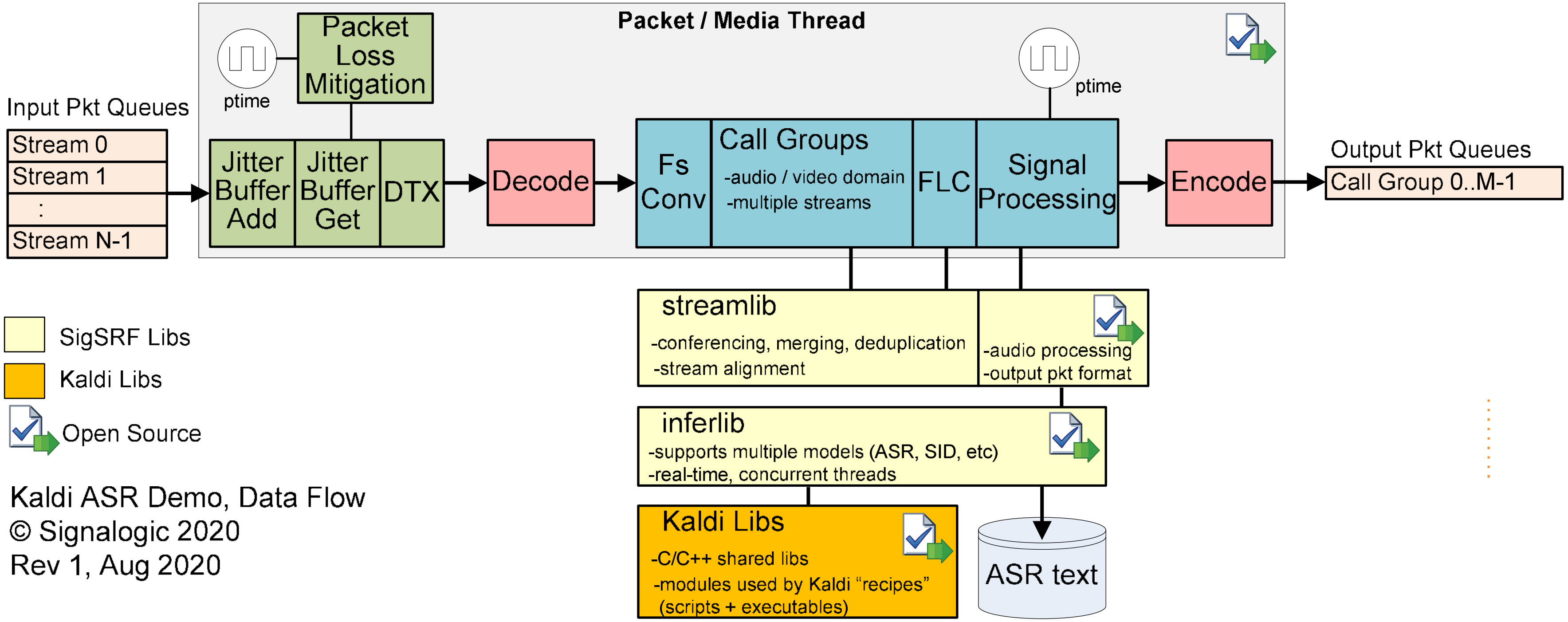
Kaldi Interface

- **Kaldi real-time inference is called "online decoding"**
  - Kaldi run-time inference expects raw 16-bit audio chunks. They recommend receiving audio as
    - raw audio over TCP/IP
    - via RTP audio packets received and decoded by GStreamer
- **Kaldi needs wideband audio**
  - for accuracy benchmarks
  - training augmentation, R&D work, published results based on wideband audio
- **GStreamer is weak in telecom / wideband audio support**
  - no support for EVS, AMR-WB support is weak for concurrent threads, reliability
  - lack of advanced handling for packet loss, stream alignment between multiple streams within a call, stream gaps (call waiting, music on hold), etc
  - no support for RFC8108 (multiple streams from one endpoint)



Kaldi Interface, Data Flow

- **SigSRF replaces GStreamer**
  - minimum REST APIs required – session create/delete/modify
  - session create can be specific or give IP:port and let SigSRF auto-detect codec, bitrate, ptime, etc from RTP data flow
  - packet input via UDP, pcap input for R&D, testing purposes
- **inferlib**
  - we added an interface library that in turn interfaces to Kaldi run-time libs



Kaldi ASR Demo, Data Flow
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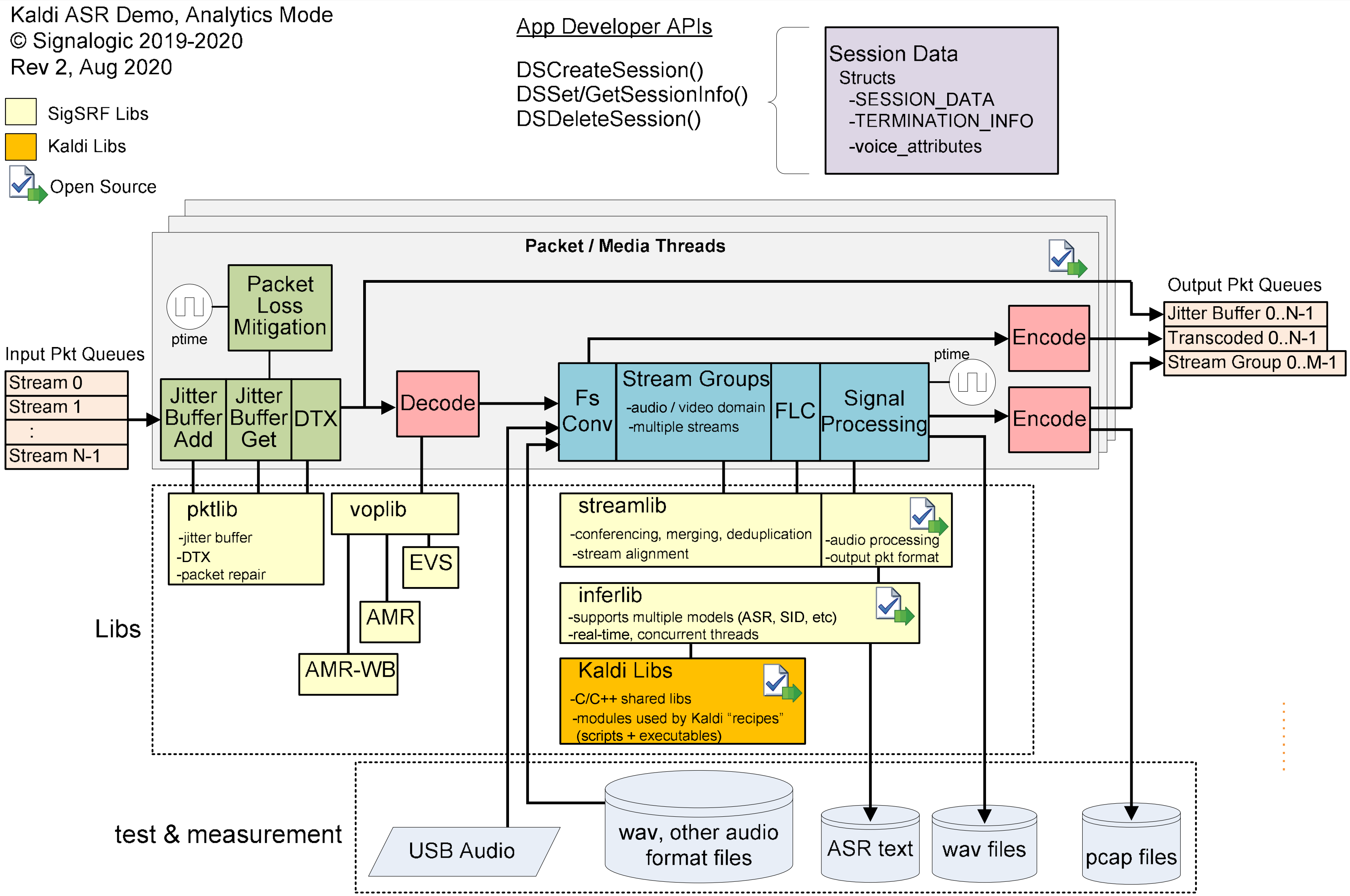
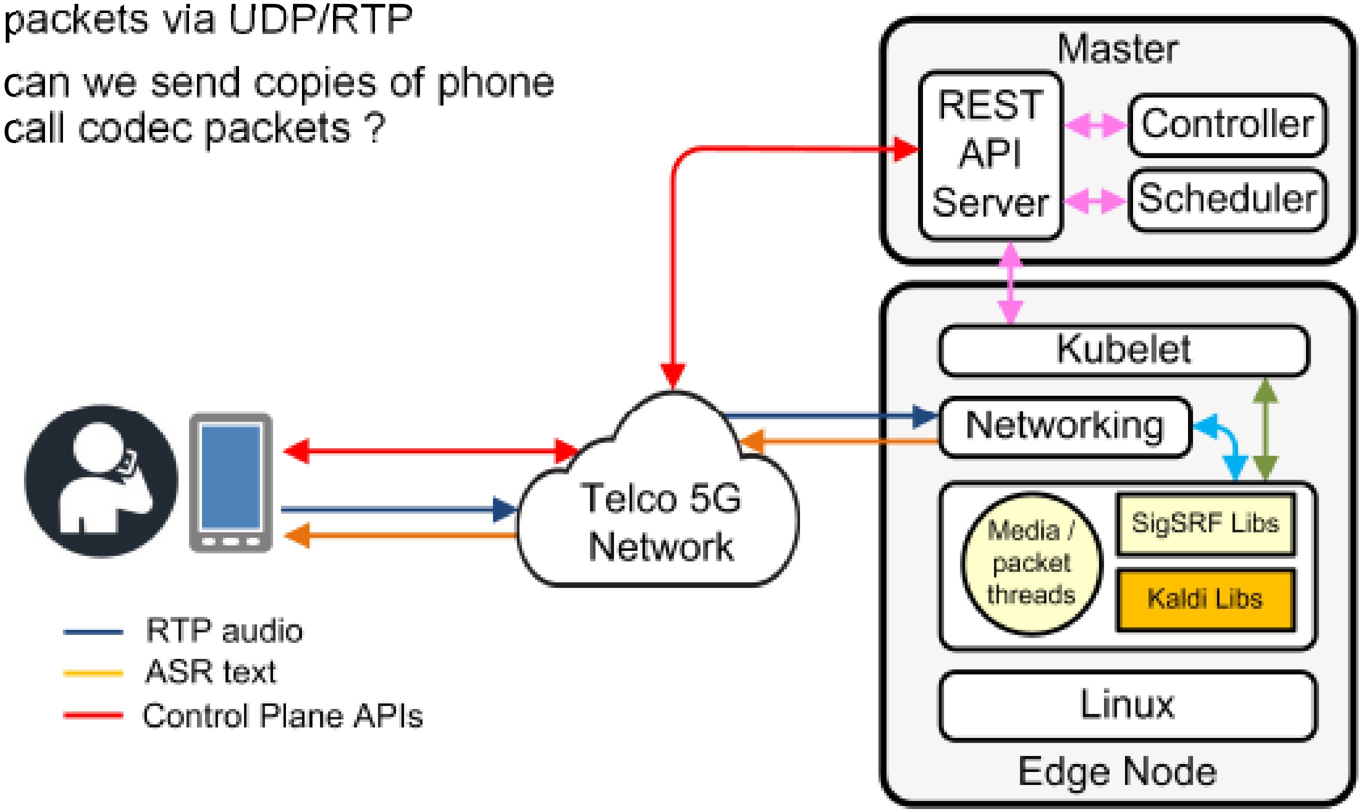
Kaldi Interface, Software Architecture



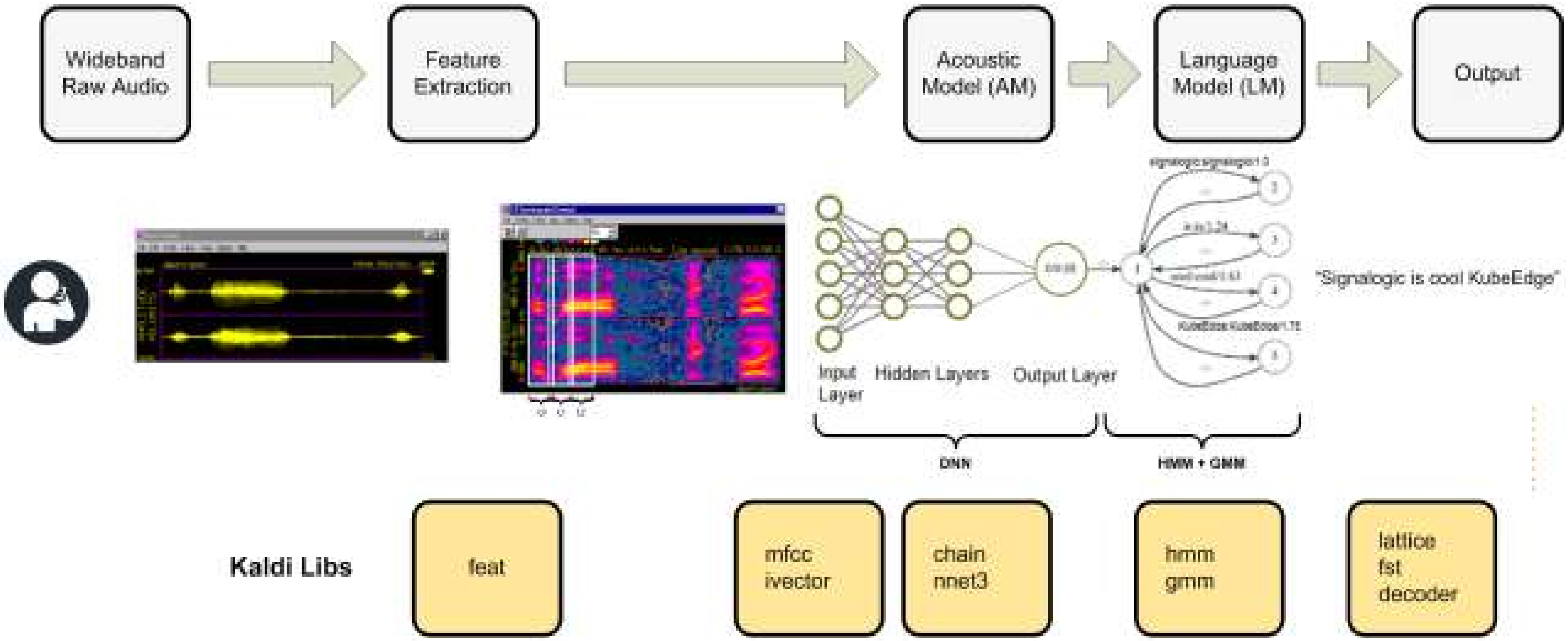
KubeEdge Integration

- **SigSRF and Kaldi libs inside KubeEdge container**
  - minimum 4 x86 cores, 32 GB mem, 1 TB HDD
- **Mobile device app**
  - creates ASR sessions with REST APIs
  - push-to-talk, send codec output packets via UDP/RTP
  - can we send copies of phone call codec packets ?



Kaldi Info: Run-Time Inference

- **One end-to-end thread on one Xeon x86 core**
  - input is 16-bit raw audio, output is ASR text (plus log, stat files, etc)
  - they maintain a sweet spot of about 2x RTF (real-time factor). They don't use OpenMP, TBB, or other HPC multicore methods
  - ARM cores can be used, but support on Kaldi user groups is limited
- **Main Kaldi contributors are focused on state-of-the-art R&D**
  - DNN and HMM architecture, improved training are priorities, not performance
  - not focused on concurrent streams, high capacity, reliability, etc

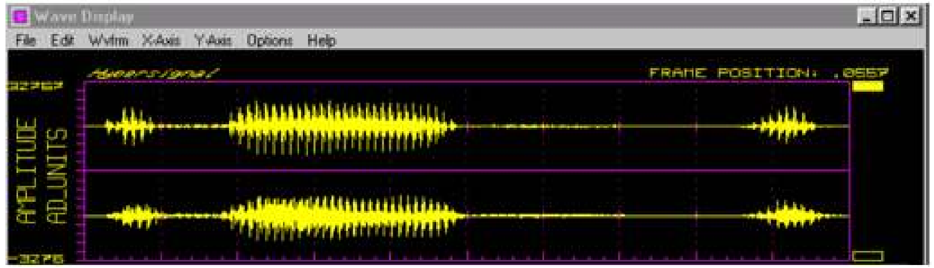


Kaldi Info: Integration

- **Kaldi is its own framework**
  - main Kaldi contributors are working on PyTorch support
  - partially supports TensorFlow, but main contributors no longer working on it
  - no support for Caffe, MXnet, etc
- **To integrate Kaldi into production applications takes effort**
  - developer interface is based on Linux shell scripts, so we tracked inference scripts + binaries, to find necessary APIs supported by inferlib
  - if you ask questions on kaldi-asr.org about improving performance, reducing model size, concurrent threads, etc you will get general advice only
- **Acceleration**
  - GPUs are supported by Nvidia tech personnel on kaldi-asr.org
  - also seems to be the case for OpenVINO (Intel)

Kaldi Info: Architecture, DNNs

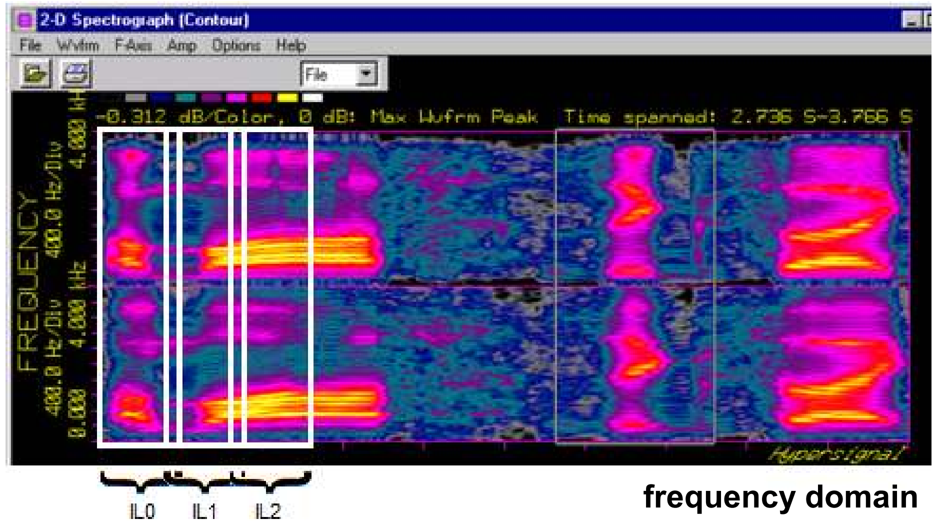
- **Architecture**
  - uses "chain" models: DNN[1] + xMM[2]
  - AM (acoustic model) recognizes phonemes
  - phonemes vary depending on context, so "tri-phones" are used
  - LM (language model) recognizes words as tri-phone combinations



time domain (time series)

Sliding FFT

- **DNN frequency domain data**
  - 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[3]



frequency domain

- **Training**
  - DNNs saved as "x-vectors" and "i-vectors"
  - HMM / GMMs saved as FSTs[4]

DNN Input Layers (ILn)

[1] Deep Neural Network, [2] Hidden Markov Model, Gaussian Mixed Model, [3] Convolutional Neural Network, [4] Finite State Transducer