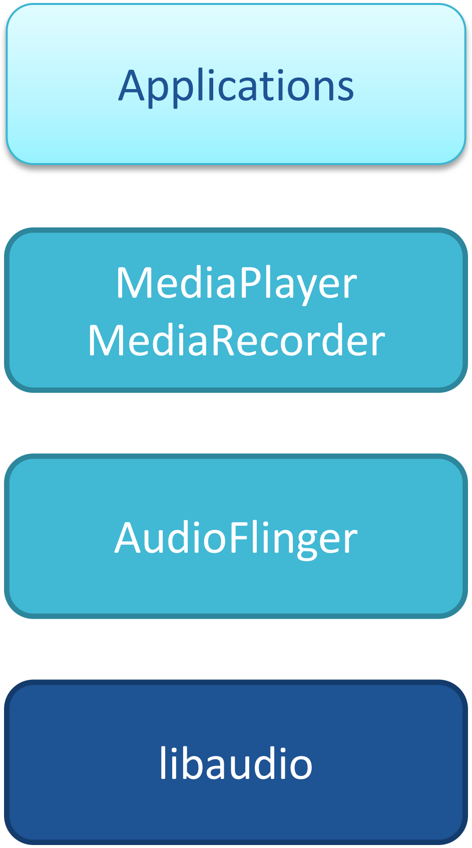
Linux Audio for Smartphones

System integration basics

Overview

- **Android audio stack overview**
- **Modern smartphone audio subsystems**
- **Traditional solutions**
- **ASoC – the Linux solution**
  - Design overview
  - Brief introduction to chip drivers
- **Walk through of system audio driver construction**
- **Debugging tips**
- **Future directions**

Android audio stack



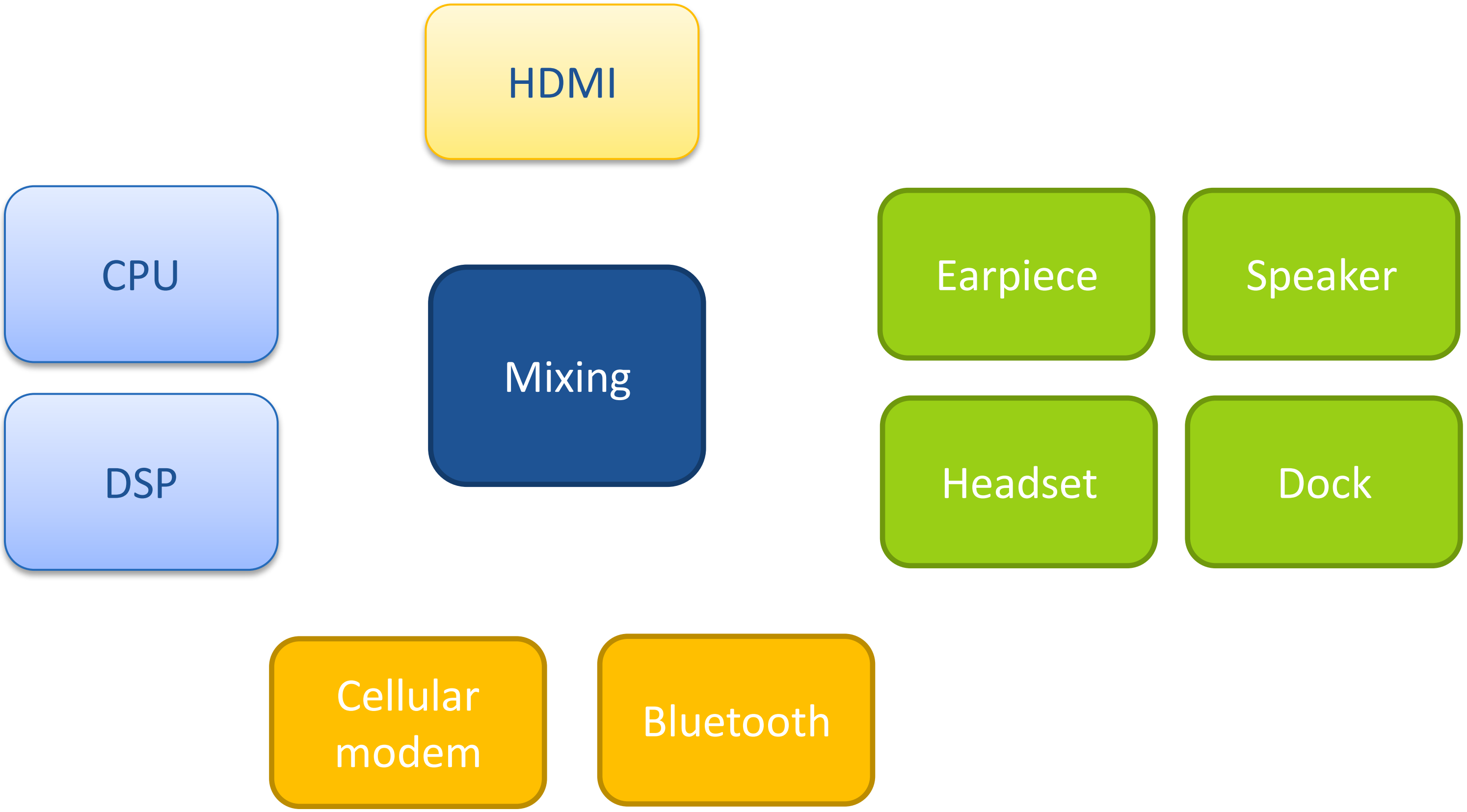
Applications

MediaPlayer
MediaRecorder

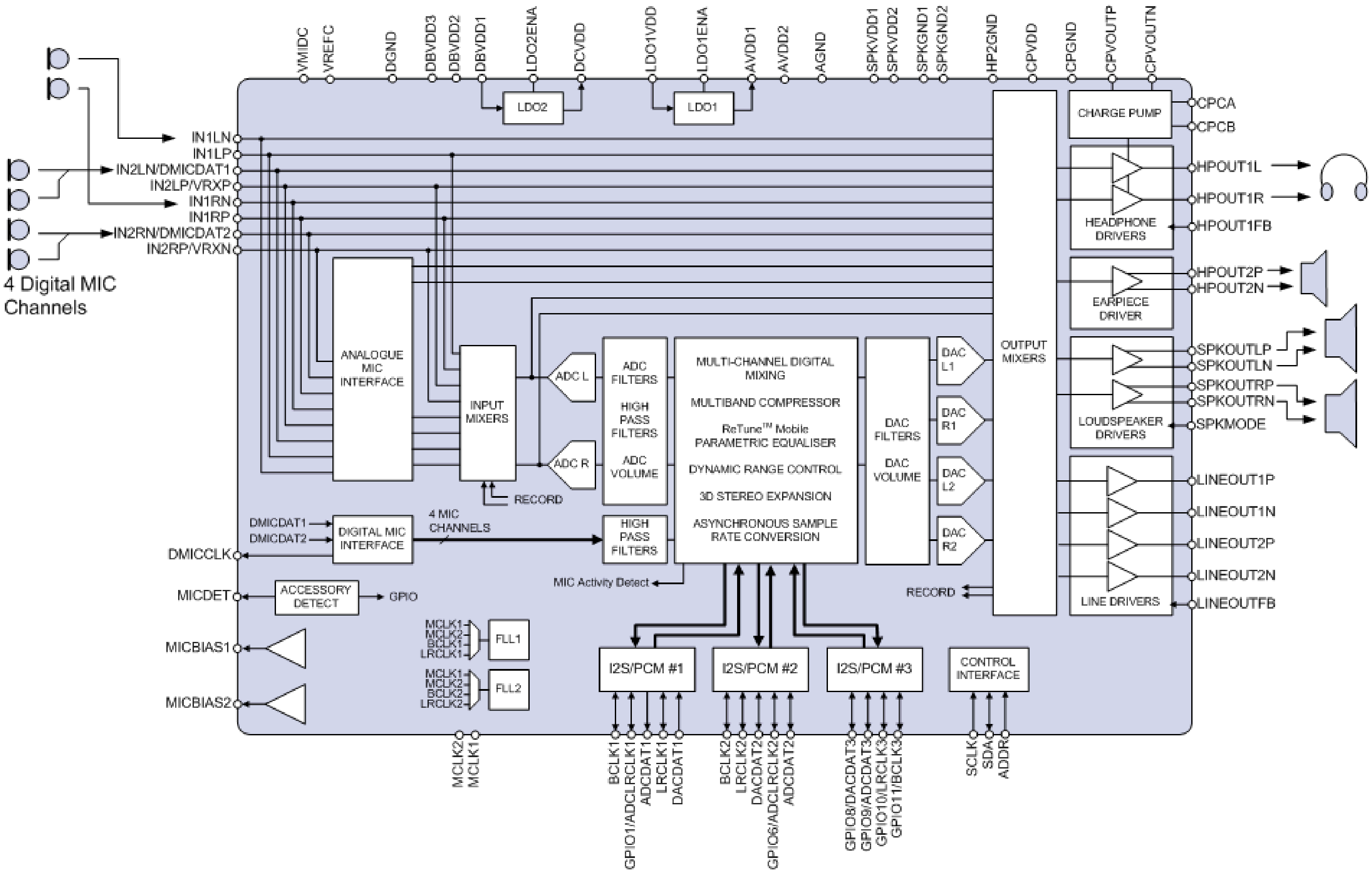
AudioFlinger

libaudio

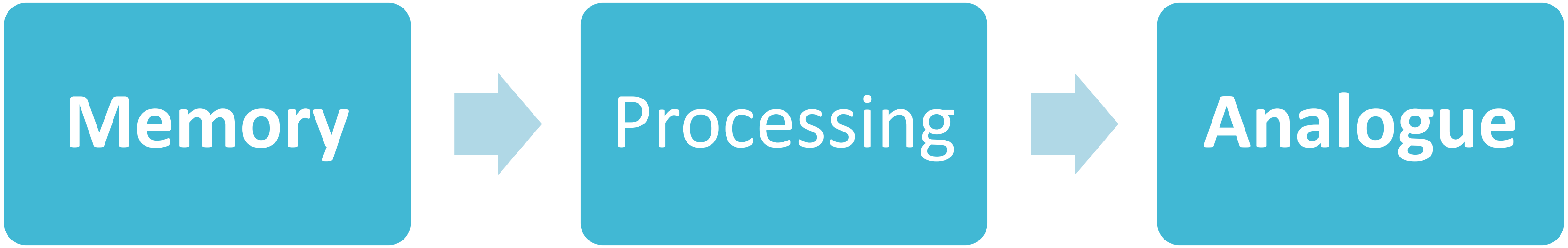
Modern smartphone audio subsystems



Flexibility and complexity



Traditional driver model



Memory

Processing

Analogue

Traditional driver model

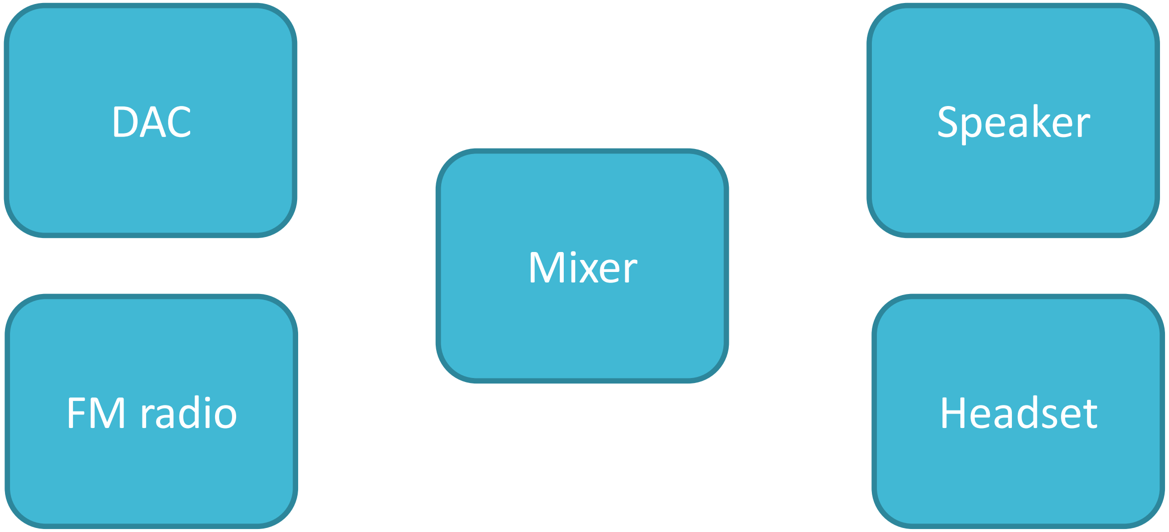
- **Monolithic driver for each card**
  - No structure for managing off-CPU hardware
  - Very little reuse
- **Tight coupling between application and kernel code**
  - Per-use case register settings
  - Detailed register level knowledge of components
- **Time consuming**

The Linux solution

- **ASoC embedded audio framework**
  - Merged since 2.6.21, April 2007
  - Provides standard ALSA interface to applications
- **Reusable drivers for each chip**
- **Minimal per-system drivers**
- **Use case configuration done by userspace**
- **Automatic and transparent power management**
- **More reuse, less coupling**

Dynamic Audio Power Management

- **Looks for audio paths connecting inputs to outputs**
- **Powers only components in an active path**
- **Automatically activates DACs and ADCs**

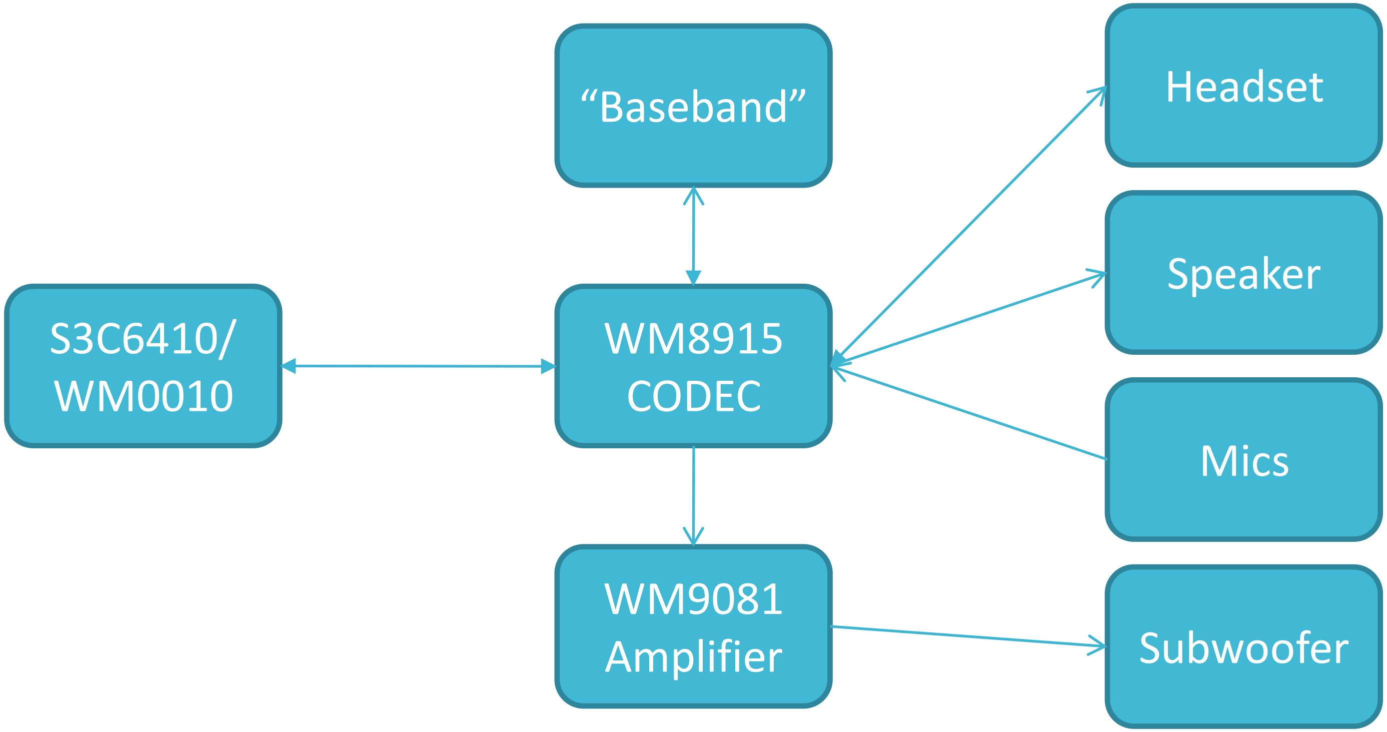


ASoC CODEC drivers

- **Four classes of control**
  - Audio processing controls (eg, volume, effects)
  - Audio routing controls (DAPM controls and routes)
  - Power controls (DAPM widgets, bias)
  - Stream control (Digital audio streaming)
- **Mostly direct mapping into register map**

  -
    ```
    SOC_DOUBLE_R_TLV("DAC1 Volume", WM8994_DAC1_LEFT_VOLUME,
    WM8994_DAC1_RIGHT_VOLUME, 1, 96, 0, digital_tlv),
    ```

Driver integration walkthrough



http://opensource.wolfsonmicro.com/content/speyside-audio

Application layer

- **AudioPolicyManager and AudioHardware**
  - platform/hardware/alsa_sound – Generic ALSA, asound.conf, LGPL
  - devices/samsung/crespo – Nexus S, hard coded, Apache licensed
- **Getting use cases**
  - Devices specified when streams are opened
  - setMode()
- **Applying use cases**
  - Run external utilities
  - Use asound.conf
  - Call raw ALSA control APIs
  - Apply settings with ALSA UCM
  - Using common base use cases helps

Debugging tips - DAPM

- **Data in debugfs**
  - CONFIG_DEBUG_FS
  - mount –t debugfs /dev/null /debug
- **codec_reg – Register map**
- **dapm_pop_time – log sequences**
- **dapm directory**

```
SPKL: Off in 0 out 1
 in "DAC2L" "DAC2L"
 out "static" "SPKL PGA"
```

- **Tools:**
  - git://git.opensource.wolfsonmicro.com/asoc-tools.git

Debugging tips - general

- Audio stuck – check clocking
- Silent audio – check volumes and mutes
- Use bypass paths to bisect
- Turn volumes up to maximum
- Make sure machine drivers check error codes
- Check kernel logs for errors
- 2.6.38 and later support trace points
    - http://www.sirena.org.uk/log/2011/01/22/tracing-asoc-with-trace-points/

Future work

- **Nicer handling of digital basebands**
- **Resolve headset detection API compatibility**
- **Greater use of DSP**
  - Enhanced features – ambient noise cancellation, beam forming, offloaded decompression, speaker compensation
  - Even more dynamic reconfiguration of the audio subsystem
- **Coefficient management and in-system calibration**
- **Use case development and management**
  - Media controller API
  - User interfaces for configuration development
- **ASoC conference, 4th-5th May**
  - http://www.slimlogic.co.uk/?p=268