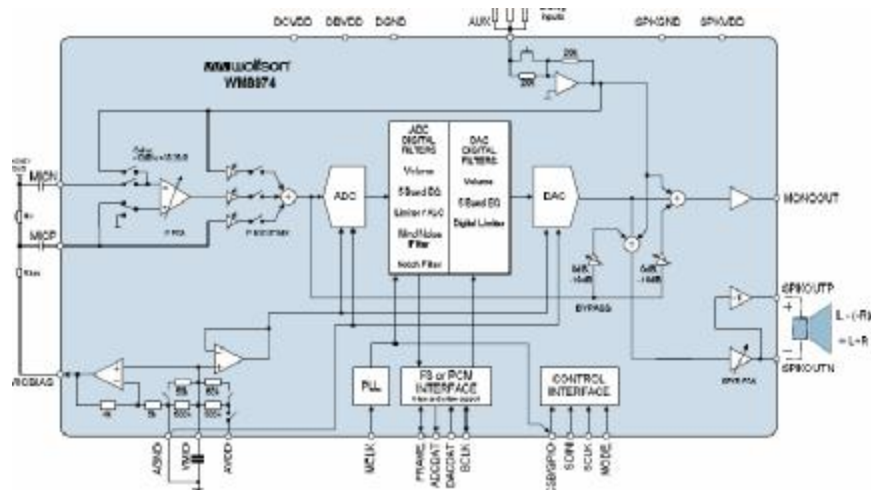
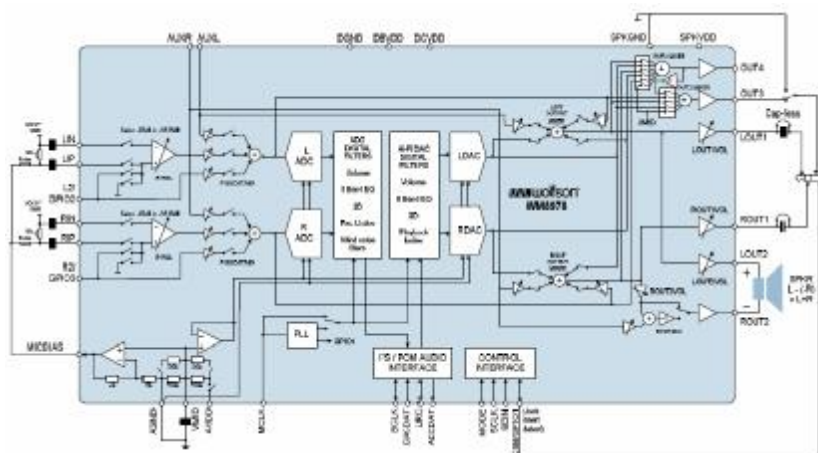
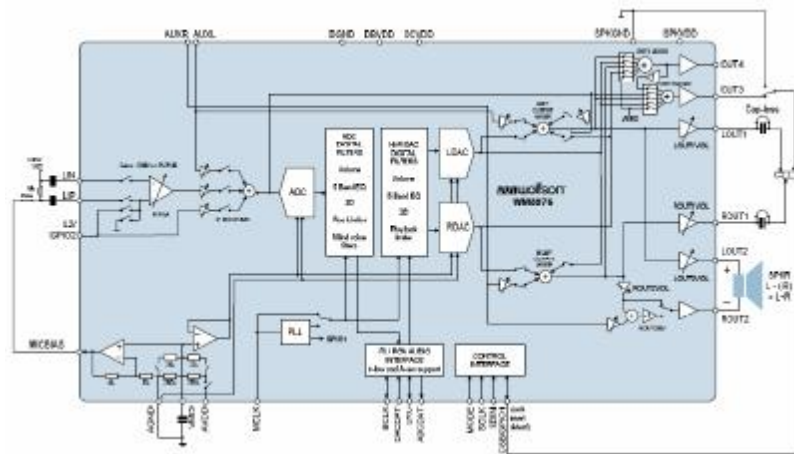
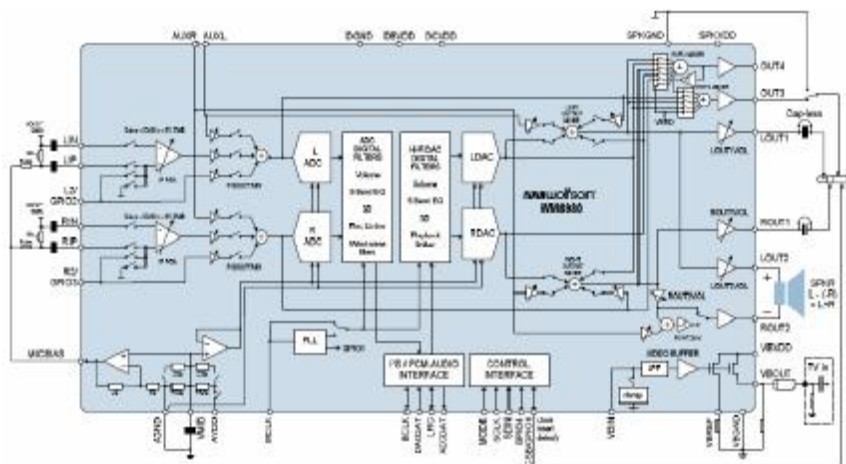
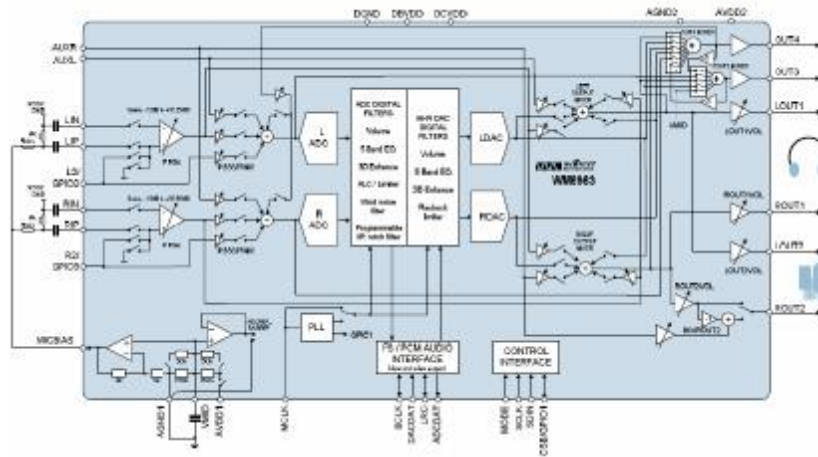


- 3st :
 - WM8974/WM8976/WM8978/WM8983
 - WM8980/WM8982
 - WM8985 – headphone class-d amplifier
 - WM8940/8941
 - WM8950/WM8951
- 4st :
 - WM8960/WM8956/WM8984 – Stereo class-d amplifier
- 2st :
 - WM8750/WM8753/WM8971/WM8751/WM8955
- AC97
 - WM9711/WM9712/WM9713/WM9714





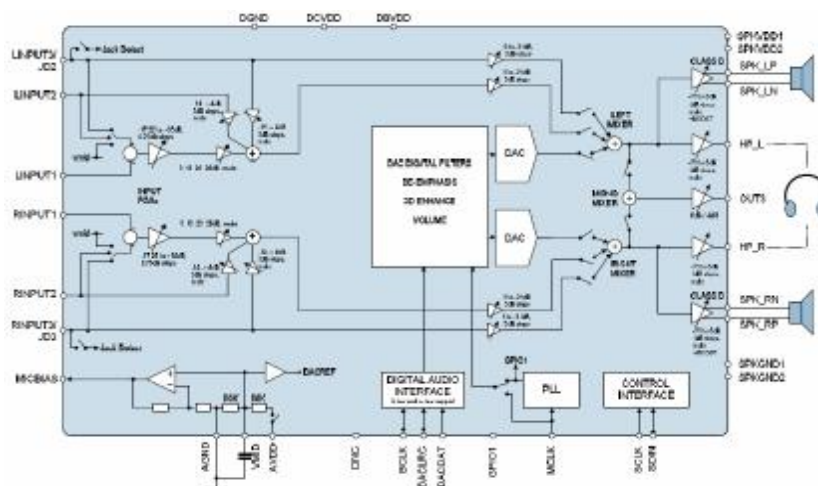
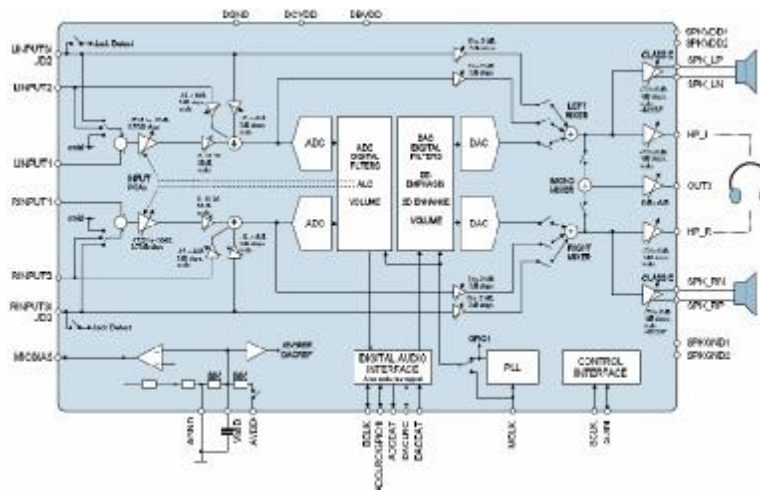


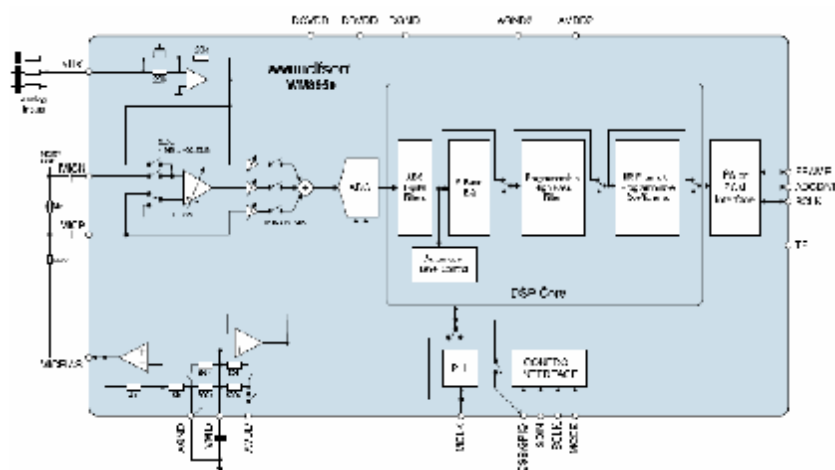
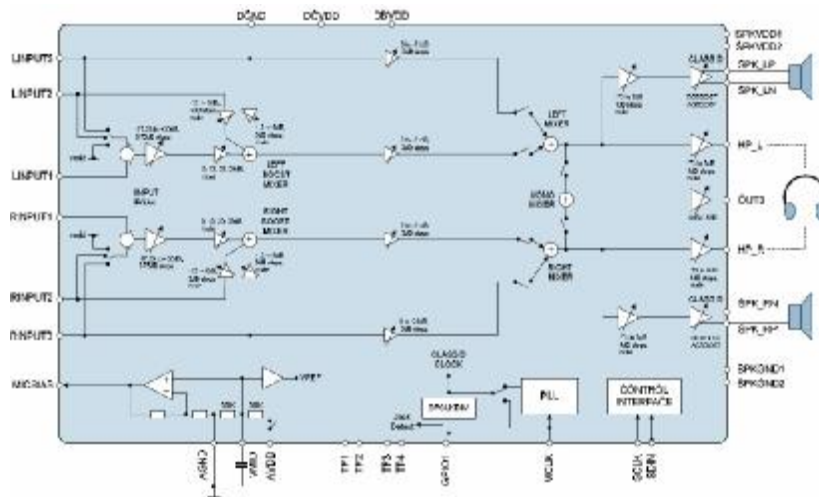


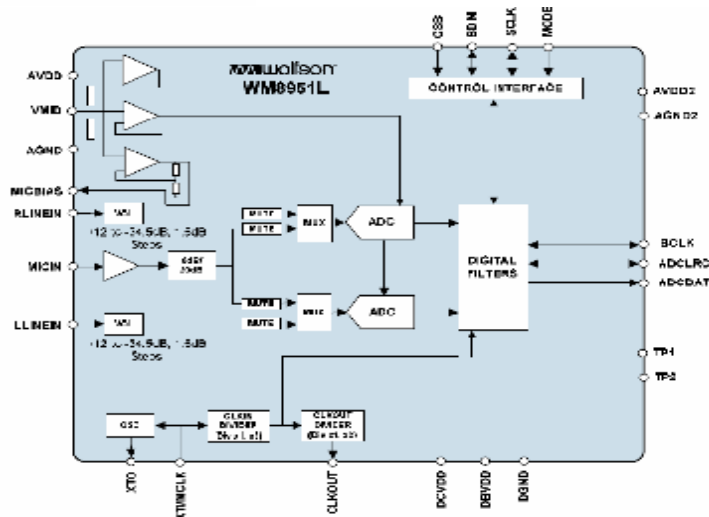
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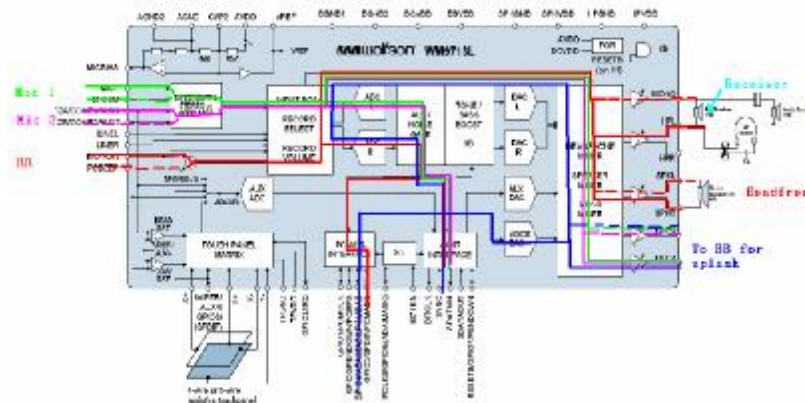






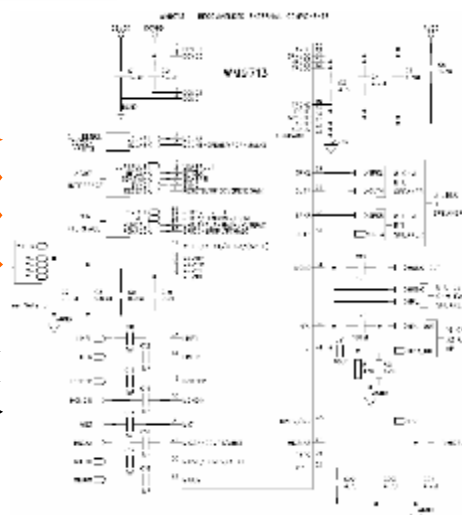
	WM9711	WM9712	WM9713	WM9714
Speaker driver	✓	✓	✓ (support stereo)	✓ (support stereo)
Headphone driver	✓	✓	✓	✓
Touch panel driver		✓	✓	
Battery measure	Comparator Only	✓	✓	✓
PLL			✓	✓
AUXADC		✓	✓	✓
AUXDAC			✓	✓
GPIO support	5	5	8	8
Package	7x7 QFN	7x7 QFN	7x7 QFN	7x7 QFN

- Recording for both communication (include Bluetooth headset)
- PDA phone、VOIP phone、Smartphone



INPUT

MCLK →
 I2S ↔
 PCM ↔
 Touch panel (4 or 5 wire) →
 BB (Differential) →
 Microphone 1 →
 Microphone 2 →
 Microphone 3 →
 FM/AM (Stereo) →
 Midi →



OUTPUT

Stereo Loudspeaker
 Mono out to BB
 Ear Speaker
 Headphone

Audio Codecs for Smartphones and PDAs

Focus on WM8753 and WM9713



Wolfson Audio Codecs for Smartphones and PDAs



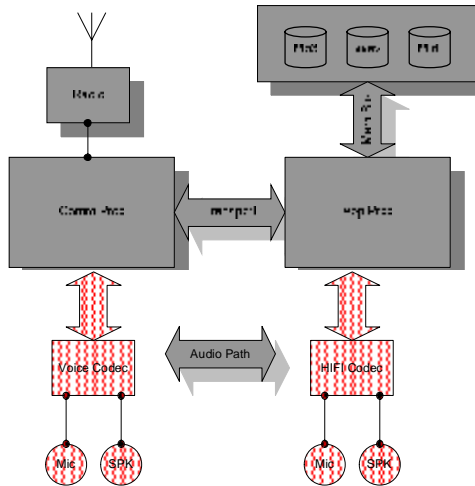


- **A Smartphone combines the features:**
 - Phone,
 - PDA,
 - Music Centre,
 - Camera,
 - Video
- **A smart phone runs an OS:**
 - Palm
 - Pocket PC, Smartphone, (Both Win CE variants),
 - Symbian
 - Linux



- Phone call
- Playback stored audio
- Record from mic or line in
- EQ functions
- FM radio
- Voice memo
- Record phone conversation
- Playback MP3 over the air
- On phone answering machine
- Music playback for greeting and ring tone

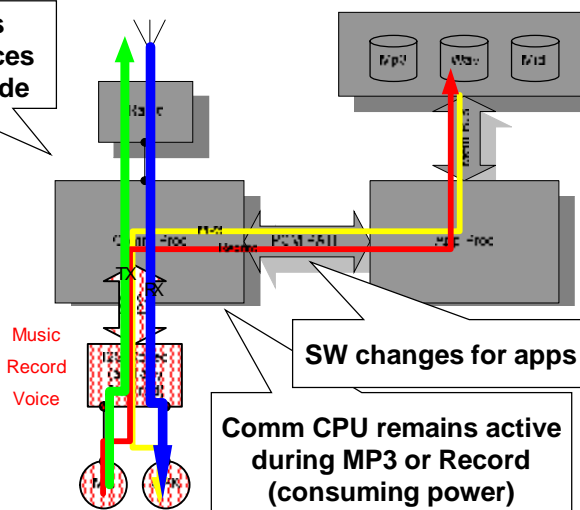
System Partitioning in a Smartphone



- Comms Processor handles all real time signal processing associated with baseband Radio and Voice processing.
- Apps Processor runs OS, PDA applications, MP3 and MPEG4 decode.
- Both processors require access to codec.
- What is the best system architecture for Audio?

Choice: Comm Side Codec

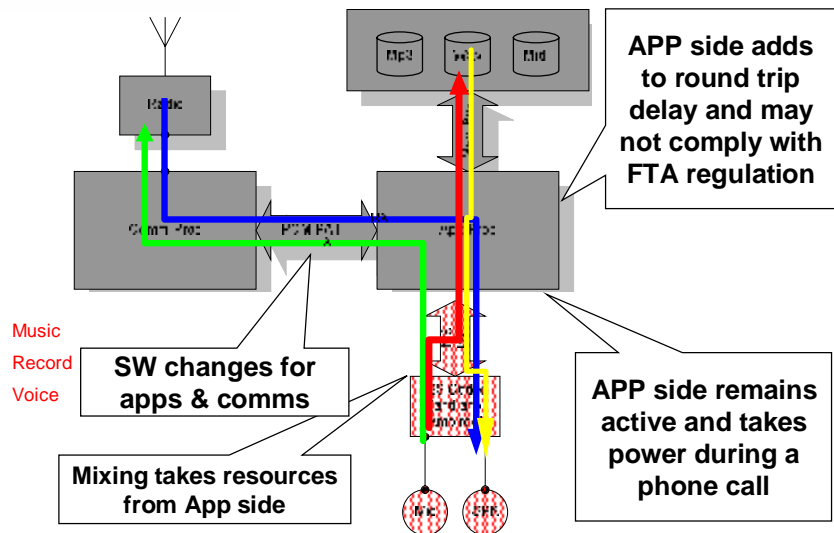
Mixing takes critical resources from Comm side



SW changes for apps & comms

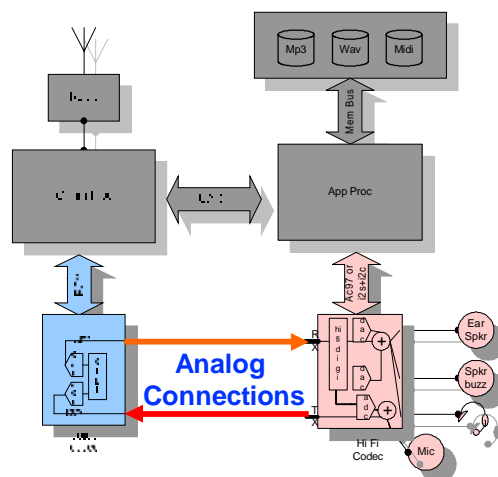
Comm CPU remains active during MP3 or Record (consuming power)

Choice: Applications side Codec

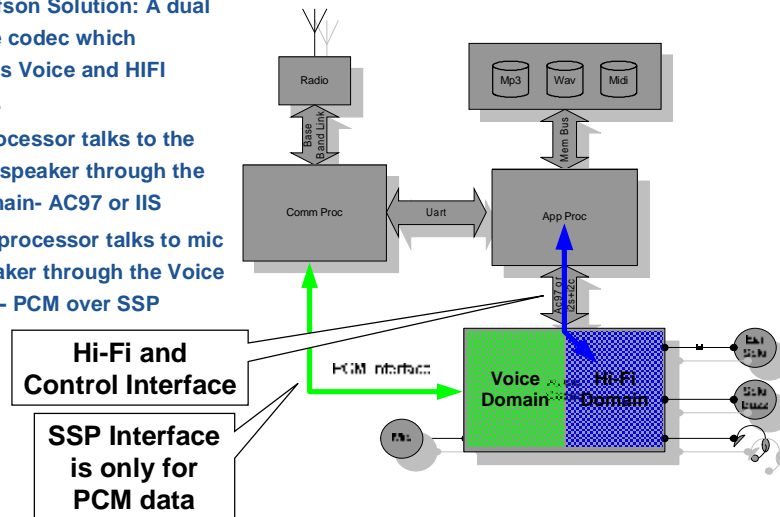


Choice: Two Codec's

- + Independent Streaming
- + Uses current parts
- + Fewer software and firmware changes
- Higher cost;
more Hardware
more real-estate
- Higher power
- More susceptible to TDMA Noise



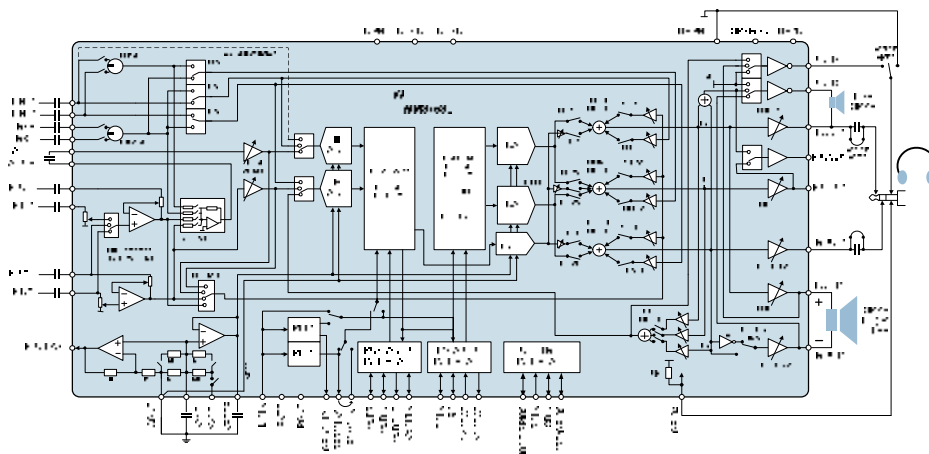
- The Wolfson Solution: A dual interface codec which combines Voice and HiFi features.
- Apps Processor talks to the mic and speaker through the HiFi domain- AC97 or IIS
- Comms processor talks to mic and speaker through the Voice domain – PCM over SSP

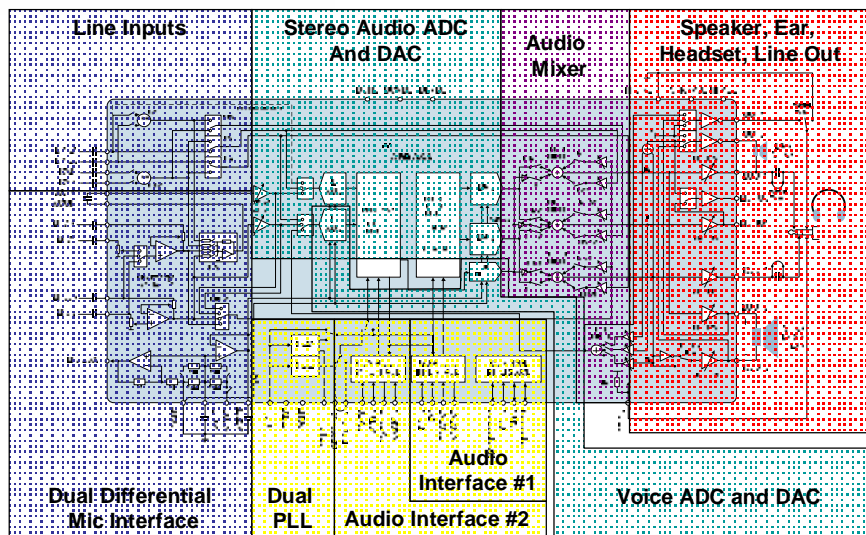


- **Reduce system power consumption and extend battery lifetime**
 - Apps and comms side can have separate and optimised power schemes
- **Audio Mixing in analog domain on Codec**
 - Advanced audio features can be added with no processor cycle overhead
- **Reduce BOM and component cost**
 - Integrated and comprehensive solution
- **Partitioning into Comms and Apps sections, simplifies design.**
 - Maximises design re-use, few DSP code changes, reduced development time.

- WM8753
- Stereo HIFI CODEC
- Telephony Codec
- **IIS** and PCM Audio interfaces
- Audio Mixer
- PLL
- Differential Mic and Line inputs
- Speaker, headphone and line out

- WM9713
- Stereo HIFI codec
- Telephony codec
- **AC97** and PCM Audio interfaces
- Audio Mixer
- PLL
- Differential Mic and Line inputs
- Speaker, Headphone and Line Out
- **Touch Panel**



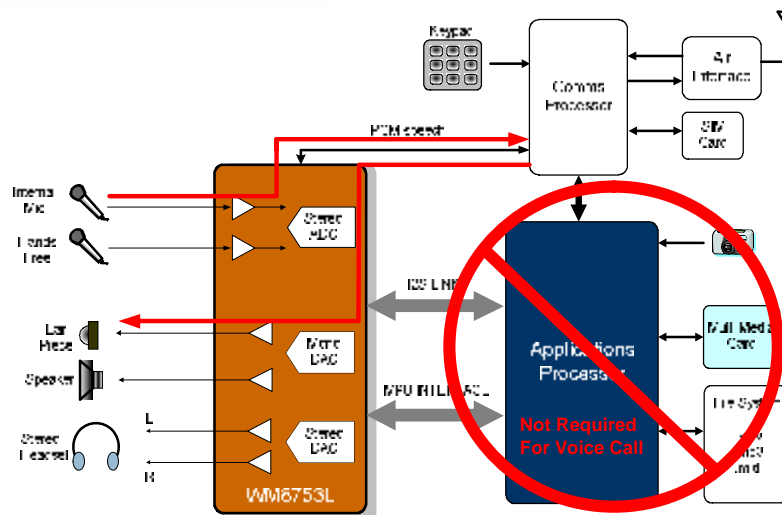
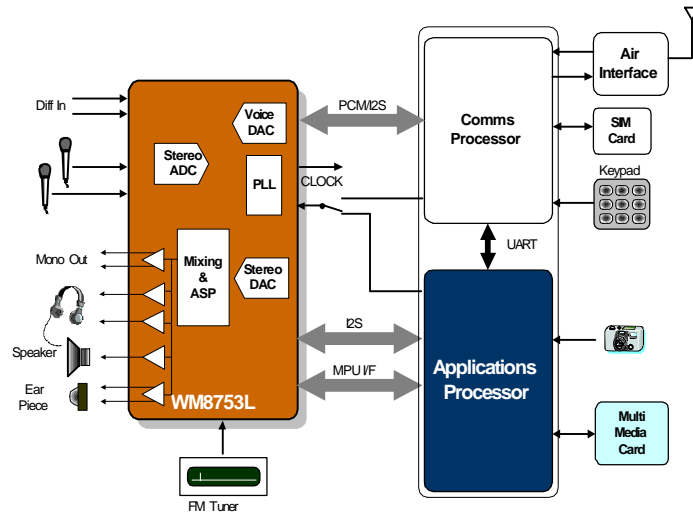


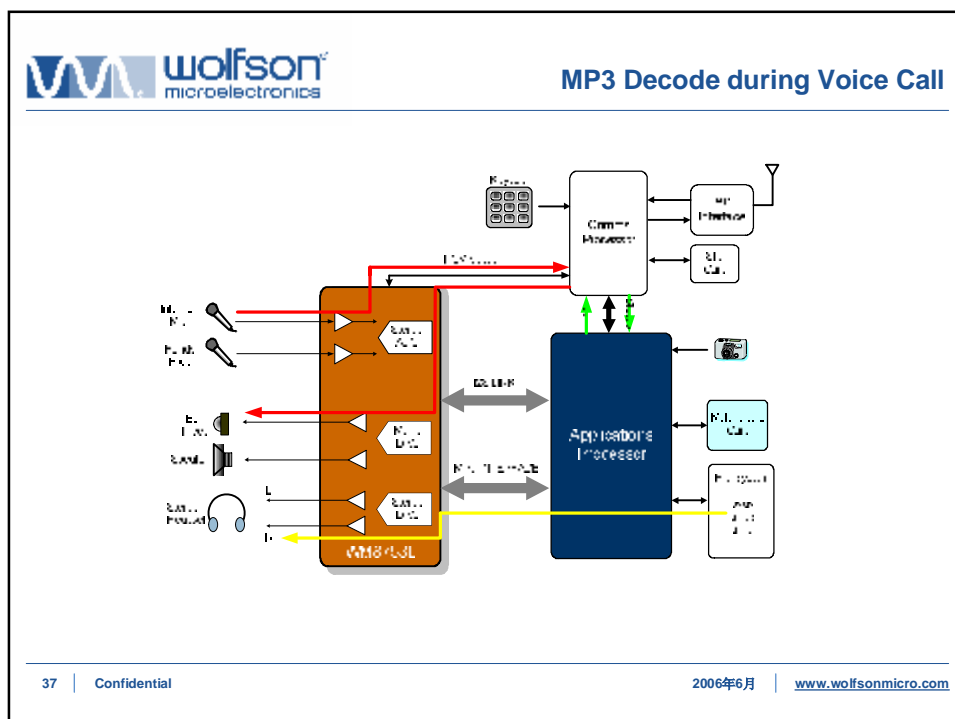
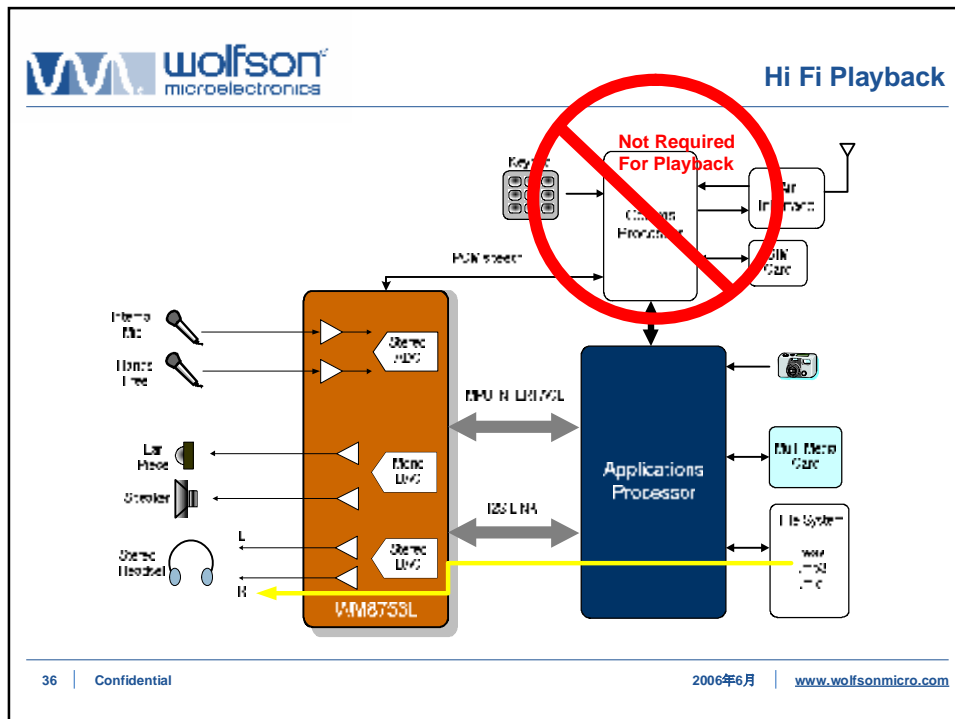
- The WM8753L is a highly featured Dual Codec designed for combined telephony and Hi-Fi audio applications
- Contains stereo ADC, stereo DAC and separate mono DAC
- Provides inputs and mixing to integrate MP3, speech, and other audio sources
- Drives headphones, ear speaker and external speaker via power amplifier
- Supports simultaneous music and speech during voice call through independent dual interface architecture
- Small size, ease of use, ultra low power consumption and efficient design
- Applications include
 - Smartphone
 - Multi media phone
 - Personal Digital Assistant

Device Description

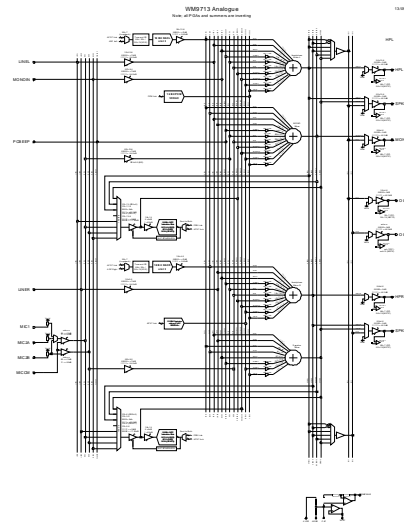
- Lowest power consumption in class:
 - 7mW Stereo playback @1.8V
 - <6mW for PCM codec operation @1.8V
 - 20mW all-on @1.8V
- Only CODEC which runs at 1.8V (analogue), 1.42V (digital) for extended battery life
- I²S digital audio interface and PCM voice interface, both master or slave able
 - Standard and robust interfaces enabling connection to any processors
- Choice of package for maximum flexibility:
 - 5mm x 5mm 52-ball BGA or
 - 7mm x 7mm 48-pin QFN
- 2 x ADC in both Hi-Fi or Voice modes; allow DSP Dual Microphone based noise cancellation
- Includes dual PLLs to allow multiple input reference clock options
- Handsfree differential connection support

- Includes all functions for a telephony codec
- Can render MP3/WMA/WAV while the comms is powered down
- Play HIFI sound during a call
- Different samples rates can be sent to the codec and are mixed
- Record voice memos during a call
- Playback audio over the air and over the speaker
- Apps can playback beeps and tones during a call
- Mix in Audio from an FM tuner
- Karaoke on your phone
- An Answering Machine on your phone

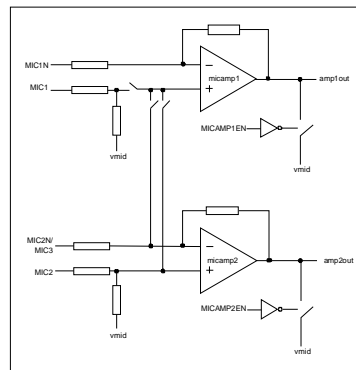
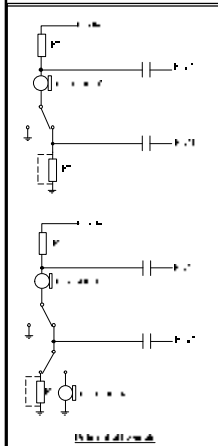




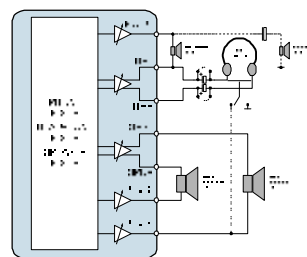
- Codec includes mixing fn and routing
- Mixing is done on the codec in the analog domain
- During a call apps may play audio to codec, mixing is done on the codec
- No extra processor cycles require – power saved
- Audio at different samples rates can be mixed on the codec – no need for sample rate matching – power saved.



- Dual Interface allows voice codec to be powered down during audio playback. Comms proc can be powered down.
- During a phone call, the Apps processor can be powered down.
- Mixing is done on the codec in the analog domain, no processor cycles are required for Audio mixing.
- ADC and DAC are optimised for function. Voice for 8k and audio for 44.1/48k. This saves power.

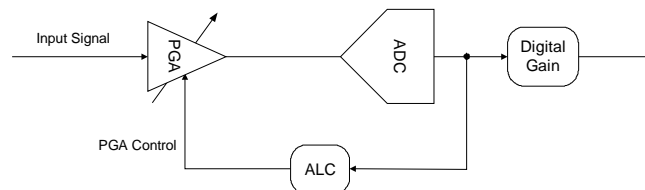


- Flexible input allowing connection of :
 - 3 Single ended mic's
 - OR
 - 2 Differentially configured mic's



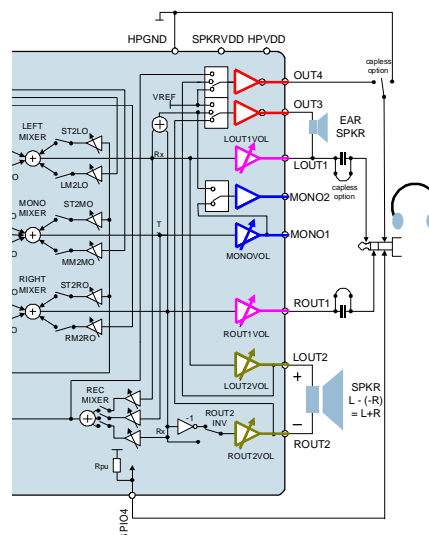
CONTROL	MIC1IN	MIC2IN	MIC1EN	MIC2EN	MIC1OUT	MIC2OUT	MIC1AMP	MIC2AMP	MIC1EN	MIC2EN	MIC1OUT	MIC2OUT	MIC1AMP	MIC2AMP	MIC1EN	MIC2EN	MIC1OUT	MIC2OUT	MIC1AMP	MIC2AMP
1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
2	2	2	2	2	2	2	2	2	2	2	2	2	2	2	2	2	2	2	2	2
3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3
4	4	4	4	4	4	4	4	4	4	4	4	4	4	4	4	4	4	4	4	4
5	5	5	5	5	5	5	5	5	5	5	5	5	5	5	5	5	5	5	5	5
6	6	6	6	6	6	6	6	6	6	6	6	6	6	6	6	6	6	6	6	6
7	7	7	7	7	7	7	7	7	7	7	7	7	7	7	7	7	7	7	7	7
8	8	8	8	8	8	8	8	8	8	8	8	8	8	8	8	8	8	8	8	8
9	9	9	9	9	9	9	9	9	9	9	9	9	9	9	9	9	9	9	9	9
10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10

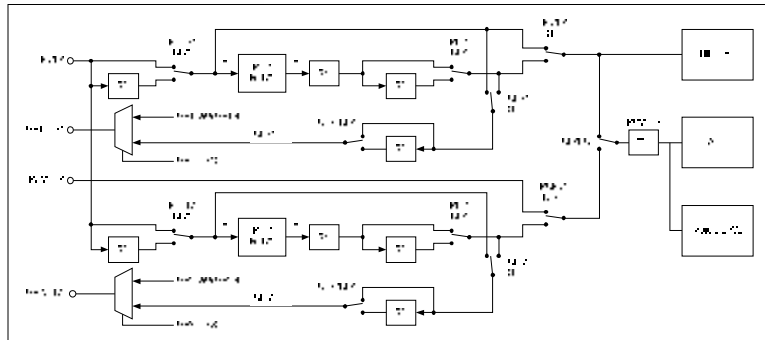
- **What is the Portable ALC (Automatic Level Control)?**
 - Automatic control of input PGAs (gain or attenuation) to maintain an optimum recording level
 - Many user definable parameters including Target Level and Max Gain



- **Application Examples**
 - Recording a conversation between two people who are at different distances from the mic
 - In DSCs or Digital Camcorders the ALC can be used to maintain a useful signal level for recording in environments where signal levels are expected to change significantly

- **L/ROUT1:**
 - can drive a 16Ω or 32Ω headphone or line out
 - -73 to +6dB gain
 - Zero-cross detect
 - Gain update bit
- **L/ROUT2:**
 - can drive a 8Ω speaker, 16 Ω or 32Ω headphone or line out
 - -73 to +6dB gain
 - Zero-cross detect
 - Gain update bit
- **OUT3/4:**
 - can drive a 16Ω or 32Ω headphone, stereo line out or buffered Vmid
- **MONO1/2:**
 - Can drive a line out
 - -73 to +6dB gain (MONO1)
 - Zero-cross detect (MONO1)
 - Gain update bit (MONO1)





- A Dual PLL generates the clocks for the Voice and HiFi codec.
- Very flexible clocking scheme
- Complex to set up
- Individual PLL's can be used for each interface
- PLL setup software to be provided

- High Quality Digital Audio requires a high quality reference clock @ 12.288MHz or 24.576MHz.
- Smartphones often don't have such a clock available either not the right frequency or jitter spec / frequency tolerance is too high.
- The PLL solves this problem elegantly and at low cost to the end user.
- No dedicated XTAL required, can even operate from 13MHz GSM reference clock.
- PLL multiplies or divides the clock to the correct frequency and reduces the Jitter bandwidth
- The WM8753 uses a fractional N PLL

- WM8753
- Stereo HIFI CODEC
- Telephony Codec
- IIS and PCM Audio interfaces
- Audio Mixer
- PLL
- Differential Mic and Line inputs
- Speaker, headphone and line out

- WM9713
- Stereo HIFI codec
- Telephony codec
- AC97 and PCM Audio interfaces
- Audio Mixer
- PLL
- Differential Mic and Line inputs
- Speaker, Headphone and Line Out
- Touch Panel

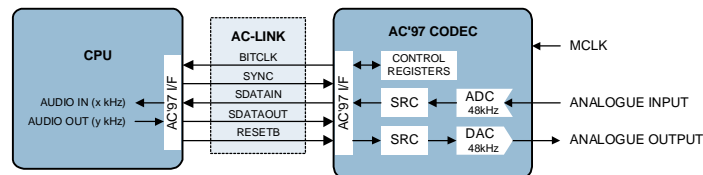
Fundamental Differences

AC'97

- Can carry audio, control, GPIO, CPU interrupts and other data (e.g. touchpanel)
- Full split-rate support is built into the AC'97 standard (Rev 2.1 or higher)
- Interface speed fixed at 12.288MHz regardless of sample rates

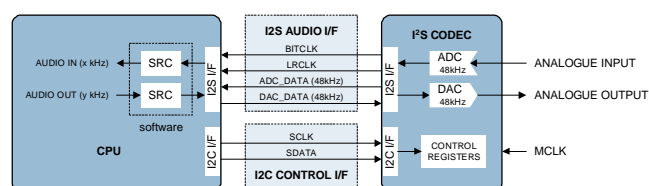
I2S

- Audio-only interface, requires separate control interface such as I2C (2 additional pins)
- Full split-rate support is difficult to achieve (see following slides)
- Interface speed(s) proportional to sample rate(s)



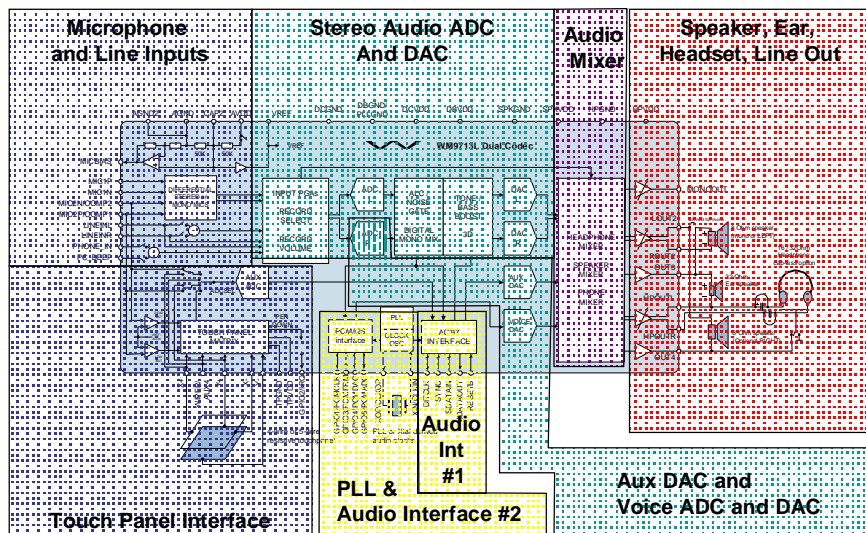
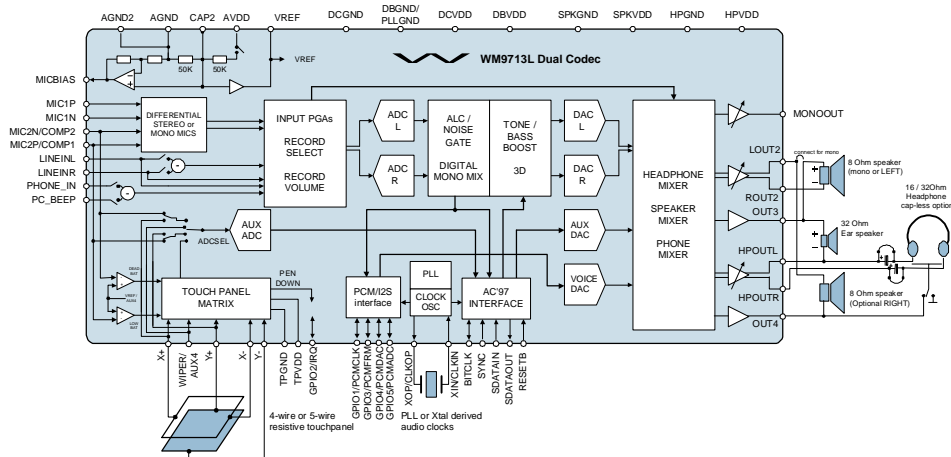
AC'97 interface

- **One interface for audio and control data**
- **Sample Rate Conversion (SRC) performed by codec**
- **No CPU overhead**
- **The fastest (therefore power-hungriest) signal is BITCLK, 12.288 MHz**
- **5 pins total**



I2S interface with software SRC

- **Two separate interfaces for audio and control data**
- **Sample Rate Conversion (SRC) in software significantly increases CPU loading**
- **Power consumption in the interface may be marginally lower than AC'97 because BITCLK is slower**
- **6 pins total**

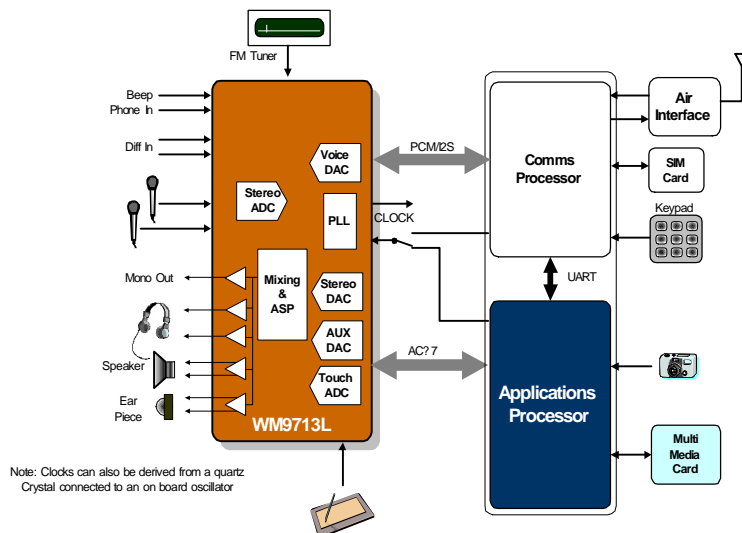


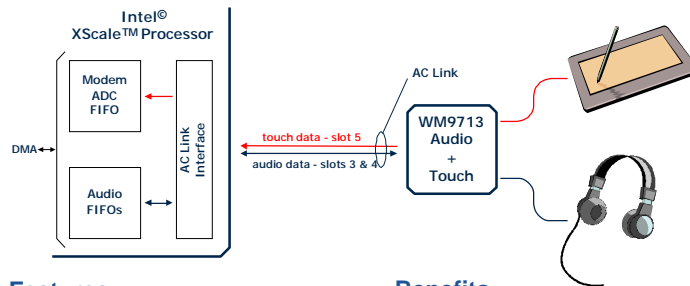
- Low power PCM codec operation (<6mW)
- Low power consumption, (as little as 7mW Stereo playback)
- Designed to run on low supply voltages (analog 1.8V min, digital 1.42V min)
- AC'97 control interface
- AC'97 digital audio and touch interface and also PCM voice interface, with master/slave option on voice I/F only
- 2 x ADC in both hi-fi or Voice modes; allow DSP Dual Microphone based noise cancellation
- Includes PLL to allow multiple input reference clock options inc 13MHz
- Supports all WinCE sample rates,
 - 8kHz, 11.025kHz, 12kHz, 16kHz, 22.05kHz, 24kHz, 32kHz, 44.1kHz, 48kHz

- Complete Multi-Media Stereo CODEC and touch screen controller for wireless telephone applications
- Uses WM9712 architecture whilst adding voice capability and ability to work in multi-time domain systems
- Reduces cost by minimising number of components
- Provides adaptable clocking architecture
- Provides performance advantage over any other alternative solutions
- Allows the best power management and comprehensive design with chipset flexibility

WM9713 is an enhanced WM9712 with additional features

- Added voice DAC suitable for PCM CODEC function
 - Use existing one or two ADCs for voice digitisation
 - Add PCM interface onto GPIO pins
 - Support 8ks/s plus 16ks/s and higher sample rates (WB-AMR)
- Adds additional OUT4 analogue output to allow stereo speaker support (2 x .5W drivers)
- Adds PLL to support 13MHz (or any 4 ~ 48MHz) clock inputs
- Enhanced mic preamp flexibility to support dual differential mics
- Enhanced touch digitiser support
 - Support streaming with aux conversion to registers
 - Three register for X,Y, Aux samples
- Highly advanced power management
 - Ultra low power; meets and beats I2S power performance
 - Unique power modes
- Software compatibility





Features

- Sampling performed automatically using on chip timer
- Pen data streamed over AC Link slot 5 (modem ADC)
- DMA used to retrieve data

Benefits

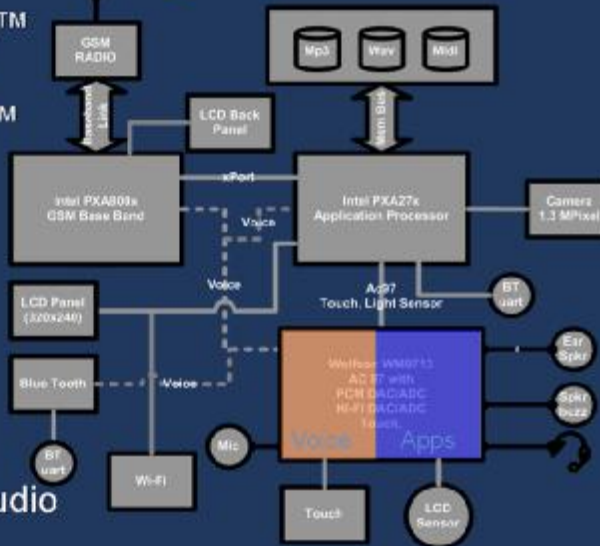
- Reduced CPU overhead (from 40% to 2%)
- Higher sampling rates possible
- Easily programmable sample rates

- **Zoar is a new cellular concept design by Intel**
 - 3 radios: GSM/GPRS, BT and Wi-Fi
 - Supports PocketPC and Smartphone
 - Presented during the '04 3GSM conference
- **Reveals the PCM interface;**
 - One Codec connects to single Controller at a given time (others get out of the way)
 - **Managed Point to Point for Voice;**
 - One set of wires
 - Two end point
 - Others tri-state
 - SW protocol required
 - Works with PCM



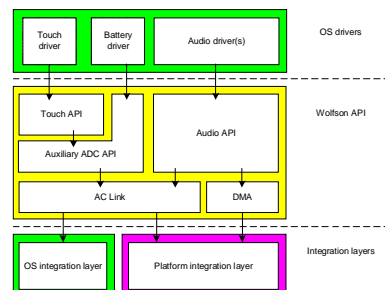
Zoar: Block Diagram

- Intel® XScale™ PXA27x CPU
- Intel® XScale™ PXA800x Baseband
- 1.3 MPixel Camera
- GSM, Wi-Fi, & BlueTooth
- Touch
- Hi-Fi / voice Audio



Support from Wolfson

- Evaluation Platform
- Software Drivers
- Dedicated team of experienced SW and HW engineers

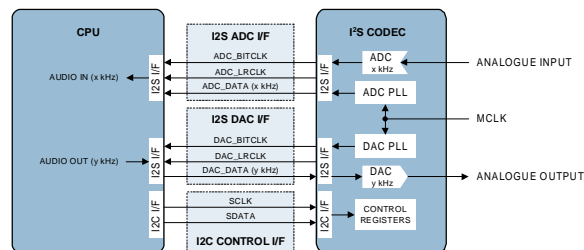


- Wolfson has 25 years experience of providing high quality Audio solutions to professional, consumer and portable markets.
- Wolfson has developed a family of Codecs specifically designed to meet all the Audio needs of a smartphone
- WM codecs offer low power, high integration and enable a feature rich phone
- 9713 and 8753 devices are highly integrated codecs which combine high quality full audio playback (of MP3), ringtone, with a dedicated codec for voice.
- WM is unique in offering a dual interface codec which integrates simply with existing designs, enables overall system power to significantly reduced, simplifies hardware and software design.

Audio Codecs for Smartphones and PDAs

Focus on WM8753 and WM9713





I2S interface without SRC

- Three separate interfaces for ADC, DAC and control
- Integration of ADC and DAC running at different rates is likely to increase noise
- Two PLLs add cost and increase power consumption
- Two separate BITCLK LRCLK and pins increase power consumption
- 8 pins total

AC'97 offers:









- Audio and control interface in one
- Hi-Fi quality devices available now
- Full split-rate support built in
- No CPU software overhead
- Low Power consumption
 - lower than I2S if AC-Link is also used to transfer non-audio data (see next slide)
 - comparable to I2S schemes in audio-only system
- Lowest pin count

AC'97 additional advantages:

- Every AC'97 frame consists of 12 time-multiplexed data slots with 20 bits each
- Slots 1&2 for control register read / write
- Slots 3&4 for left/right channel audio data
- Slots 5-11 intended for additional audio channels, unused in stereo system
- Slot 12 intended for GPIO and interrupt functions
- Total spare bandwidth in slots 5 to 11 = 7 slots * 20 bits * 48 kHz = 6.72 Mbit/s
- This bandwidth can be used at no extra cost, without adding pins and without increasing power consumption in the interface.
- Applications: integrated touchpanel interface, GPIO, keypad scanner, backlight control, etc.
- Interrupt sources in PDA: pen-down, key press, battery alarm, headphone detect



- Wolfson has complete portfolio of products designed for the next generation of multi-media and smartphone mobile phones
- The products combine the highest quality audio with PCM telephony processing, touch panel interfaces, mixed-signal functions and other digital signal processing

<p>AC'97</p> 	  
<p>PS</p> 	  

Hi-Fi CODEC

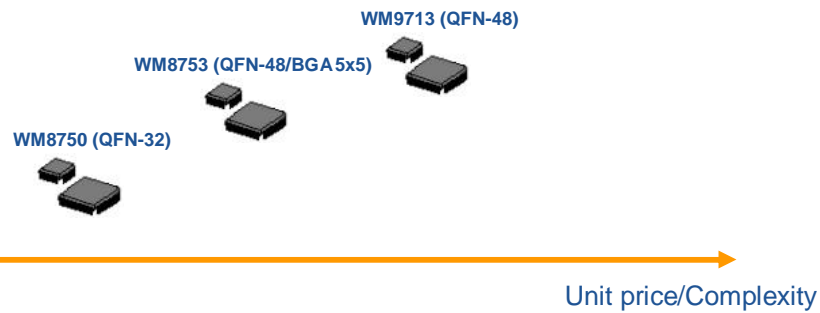
- WM8750L: I2S audio CODEC with headphone and speaker driver
- WM8955L: I2S audio DAC with headphone, speaker driver and PLL
- WM9712L: AC'97 audio CODEC with touch screen controller

Dual CODEC Architecture

- WM8753L: PS audio CODEC with headphone, speaker driver, PLL and PCM voice CODEC
- WM9713L: AC'97 audio CODEC with Touch Screen (TS) controller and PCM voice CODEC

Features

- WM9713 – AC'97 Dual Codec with touch screen
- WM8753 – Dual Codec; Stereo CODEC plus voice DAC
- WM8750 – Stereo CODEC with Speaker Driver

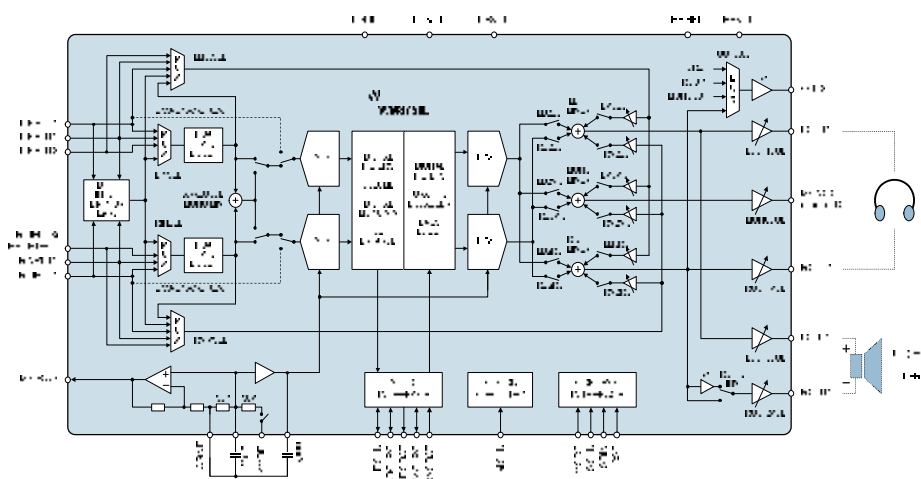


Device Description

- A low power audio sub-system that integrates mixing, converters, headphone and speaker drivers suitable for MP3, voice/phone, DSC and DVC applications
- Low power consumption (down to 7mW for stereo audio playback)
- Designed to run on low supply voltages (analog 1.8V min, digital 1.42V min)
- I²S digital audio interface
- Supports *all* WinCE sample rates
 - 8kHz, 11.025kHz, 12kHz, 16kHz, 22.05kHz, 24kHz, 32kHz, 44.1kHz, 48kHz
- USB (24MHz or 12MHz) clocking scheme
- 2 or 3/4 Wire MPU interface
- 5mm x 5mm, 32-pin QFN package saves board space

Audio CODEC

- 400mW speaker drive capability for dynamic or piezo speakers (8R, 3.3V supply)
 - Reduces component count
- Headphone driver with headset detection
- Possible to drive speaker, headphone and line out independently
- 3 Stereo input pairs, every pair can be configured as microphone inputs, line inputs or a differential microphone input, with or without ALC
- Enhanced audio signal processing: tone control, ALC, bass boost, 3D enhancement
- Flexible source/driver selection and powerful mixing capability optimised for smart phone, DSC and DVC applications

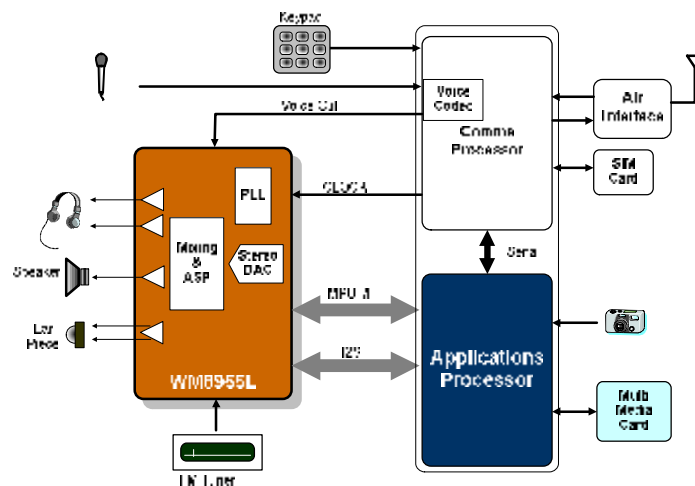
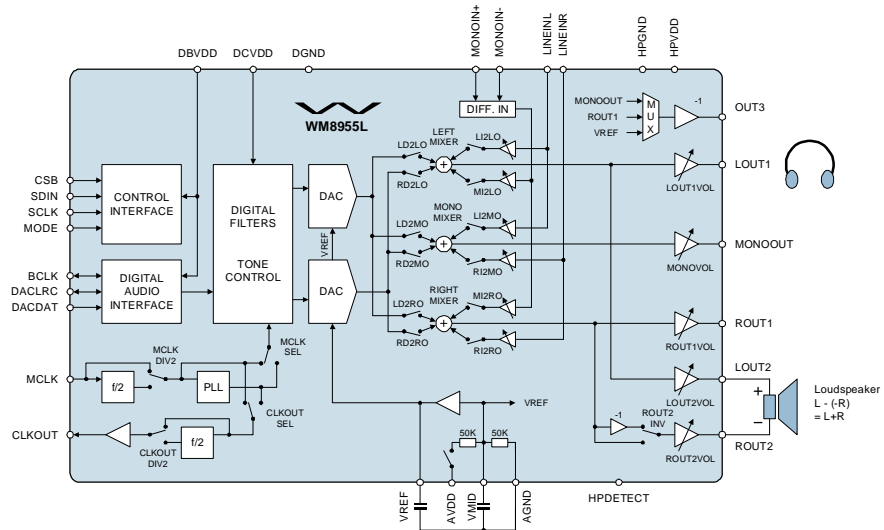


Device Description

- CD-quality stereo audio DAC
- 400mW BTL speaker drive capability (8Ω, 3.3V supply)
- Headphone driver with headset detection
- Enhanced audio signal processing: tone control, adaptive bass boost
- Flexible source/driver selection and powerful mixing capability optimised for smart phone
- Differential mono input and output for reduced noise in designs

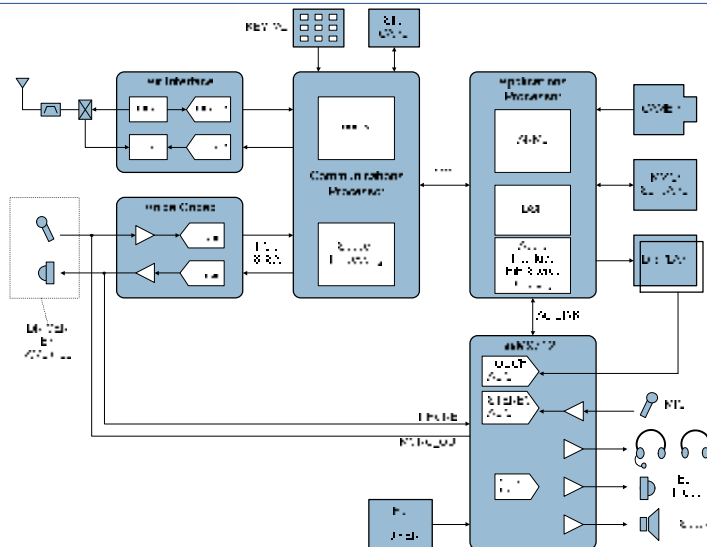
On-chip Phase-Locked Loop (PLL)

- Saves external crystal oscillator
- Generates audio clock (e.g. $256 \times f_s$) from external system clock
 - GSM phones: 13 / 26 MHz
 - CDMA phones: 19.2 / 19.68 / 19.8 MHz
 - PDC phones: 14.4 MHz
 - MPEG-4 video systems: 27 MHz
- OR Generates external clocks from an existing audio or USB clock



Device Description

- The WM9712L is a single chip solution for audio and touch screen interfaces
- Designed to reduce power, cost, and space
- Target applications
 - PDA
 - Smart phone
 - Tablet PC

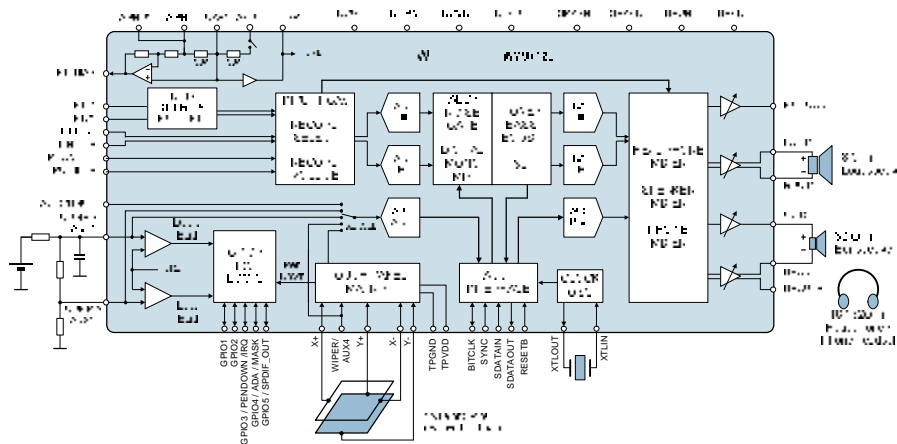


Device Description

- High quality stereo codec
- Stereo headphone driver with headset detection
- Line out and mono out
- Flexible mixing of audio signals
- On-chip speaker drivers reducing component count
 - Speaker: 300mW, 0.3%THD into 8 Ohm, 3.3V supply
 - Ear speaker: 30mW into 32 Ohm, 3.3V supply
- Stereo, mono or differential mono microphone input with Automatic Level Control
- Tone Control, Bass Boost and 3D (implemented in digital domain, reducing power)
- Auxiliary DAC for alerts/ polyphonic ring tones/ etc

Device Description Cont'd

- Power consumption reduced by up to 50% through re-design and technology change
 - Voltage rails 1.8V – 3.6V
 - Enhanced internal power management
- 4 and 5 wire touch-screen interface
 - Pen down detection with programmable sensitivity
 - Streaming touch data capability to reduce CPU loading
- Battery monitor inputs: programmable interrupts for low battery and dead battery
- 5 GPIO pins for control/monitor
- Digital Interface
 - AC'97 compliant data interface
 - S/PDIF audio output
 - Sample rate conversion



Low supply voltages

- Digital core : 1.8V ~ 3.6V
- Digital I/O : 1.8V ~ 3.6V
- Analogue : 1.8V ~ 3.6V

Power Management

- Audio DAC, audio ADC, mixer, touchscreen interface and AC-Link interface can all be powered down separately

Power consumption at 2.5V (all supplies):

- Sleep (all functions disabled): 1uW
- Sleep (pen down detector and GPIO interrupt active): 10uW
- Touchpanel only: 7.5mW
- Stereo headphone DAC playback only: 29mW
- Phone call function (using headphone): 4.5mW
- All-on (worst case): 45mW

- **XScale drivers**
 - WM9712L drivers are available for most OS
 - More highly featured driver is under continual development
- **Evaluation tools**
 - Evaluation board connects to Lubbock board
- **Operating systems**
 - Initial driver development for WinCE 3.0 & 4.0
- **Platforms**
 - Other platforms such as AMD Alchemy, Transmeta and others supported to varying degrees

Summary

- Input configurations and outputs fit into PDA and Smartphone architectures
- Fully integrated speaker and headphone drivers
- Audio mixing provides flexible audio paths
- Auxiliary DAC for ring tones etc.
- Enhanced audio signal processing
- Streaming touch data reduces processor overhead
- 4 or 5 wire touchscreen support supporting all touch screen solutions
- Low power with flexible management

New WM8753 and WM9713 multimedia/smart-phone CODECs

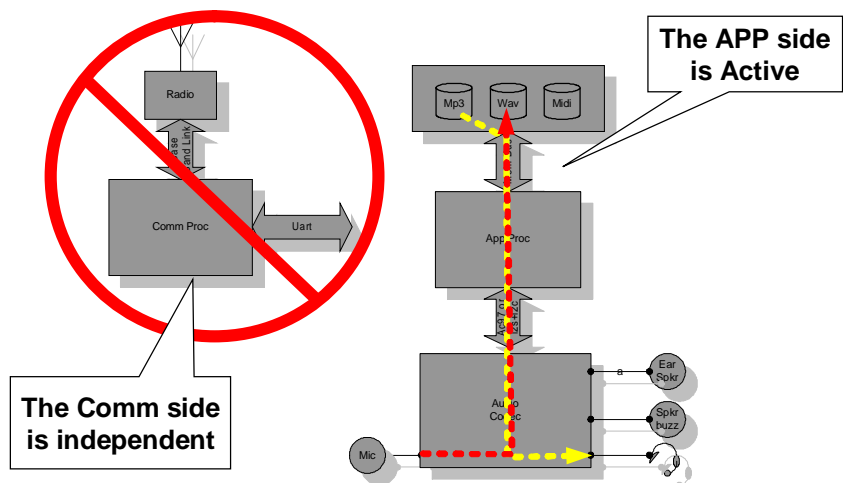
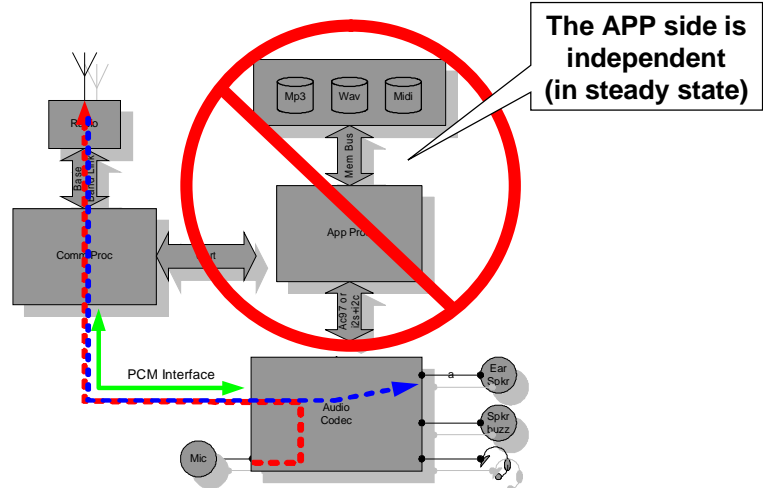
- WM8750L and WM9712L cover both PDA and smart-phone applications
- WM8753L and WM9713L add PCM voice CODEC functionality

WM8753 = WM8750 + PLL + PCM Voice CODEC + Auto Detection

WM9713 = WM9712 + PLL + PCM Voice CODEC + Auto Detection

- Voice CODEC to G712 specification
- PLL included to allow 13/26MHz clock input (and any 4 ~ 48MHz)
- Two ADCs allow for DSP Dual Microphone based noise cancellation
- Differential microphone and line inputs

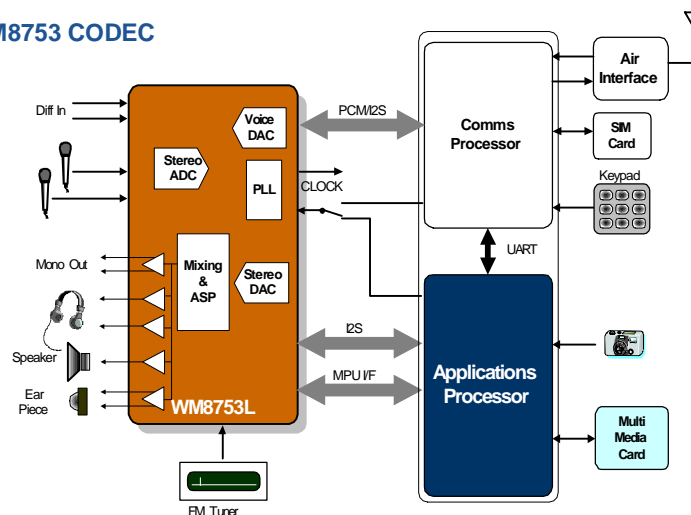
- Reduce overall power consumption and extend battery lifetime
 - Extended Talk Time as well as Standby Time
 - Apps side power scheme should not dictate Comm side power scheme and vice versa
 - Further optimize Comm side power scheme
- Reduce components and redundancy
 - Integrated and comprehensive solution
- GSM, FTA Compliance for all audio aspects
 - Minimum overall induced delay to the voice path (always <143.9mS)
- Simplified integration for min. TTM
- Max reuse from previous designs and No FW (DSP code) changes
 - Basically inherits the actual code of any previous design
- Align with any future solutions for reuse
 - Practically applies to any Apps side Comm side architecture solution



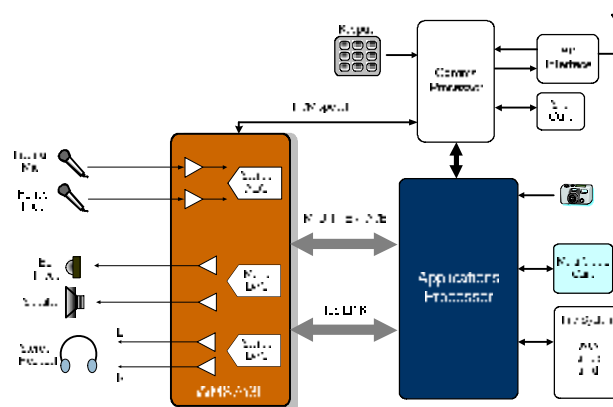
- The WM8753L is a highly featured dual codec designed for combined telephony and Hi-Fi audio applications
- Contains stereo ADC, stereo DAC and separate mono DAC
- Provides inputs and mixing to integrate MP3, speech, and other audio sources
- Drives headphones, ear speaker and external speaker via power amplifier
- Supports simultaneous music and speech during voice call
- Small size, ease of use, low power consumption and efficient design
- Applications include:
 - Smartphone
 - Multi media phone
 - Personal Digital Assistant

- **400mW speaker drive capability (8R, 3.3V supply)**
 - Reduces component count
- Headphone driver with headset detection
- Possible to drive speaker, headphone and line out independently
- Total of eight inputs
 - Can be configured as microphone inputs, line inputs or differential microphone inputs (up to two), with or without ALC
- Enhanced audio signal processing: tone control, ALC, bass boost, 3D enhancement
- Optimised for 8KHz (PCM speech) and 48KHz (Hi-Fi) operation
- Flexible source/driver selection and powerful mixing capability optimised for smartphone

- **Lowest power consumption in class:**
 - 7mW Stereo playback @1.8V
 - <6mW for PCM codec operation @1.8V
 - 20mW all-on @1.8V
- **Only codec which runs at 1.8V (analogue), 1.42V (digital)**
 - Extends battery life
- **I²S digital audio interface**
 - Standard interface enabling connection to most processors
- **Supports all PocketPC sample rates**
 - 8kHz, 11.025kHz, 12kHz, 16kHz, 22.05kHz, 24kHz, 32kHz, 44.1kHz, 48kHz
- **USB (24MHz or 12MHz) clocking scheme**
- **2 or 3 Wire MPU interface**
 - Simple control of device functions
- **Choice of package for maximum flexibility:**
 - 5mm x 5mm 52-pin BGA or
 - 7mm x 7mm 48-pin QFN

WM8753 CODEC


- **Reduce driver complexity and CPU loading**
- **No Hi-Fi PCM streams between apps and comms processors**
 - Multiple audio streams mixed by apps processor so that only one stream is sent across I2S
- **No need for mixer in comms sub-system**
- **Speech and Hi-Fi audio mixed in terminal in analogue domain**
- **Speech and audio mixture can be played back over air interface in real-time**
 - Karaoke
 - Personal commentary on soundtrack (e.g. sport)



- **High integration level**
 - Simplifies design and implementation
 - Analogue mixers reduce CPU loading
- **Highly suited to GSM/GPRS solutions with digital I,Q**
 - No need for analogue interface IC
 - Consistent with Intel PCA architecture
- **Both speech and Hi-Fi codec are part of the Apps processor subsystem**
 - Consistent with 3G view of audio as an application
 - Apps and comms power management can be kept separate and optimised independently
- **Simplified driver development**
 - Apps processor driver implements all codec functions
 - Comms processor driver only needs to stream speech to the PCM codec

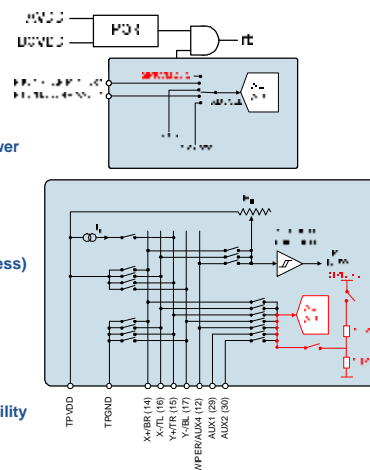
- The WM8753L is designed for smartphone applications
- Allows dual operation and mixing of speech and Hi-Fi audio
- Board space is minimised through high levels of integration
- Power consumption has been reduced to extend battery life
- Evaluation boards and software drivers are available to reduce time to market
- The WM8753L is part of a range of products allowing efficient, cost effective development with a minimum of time



New Features for WM9713

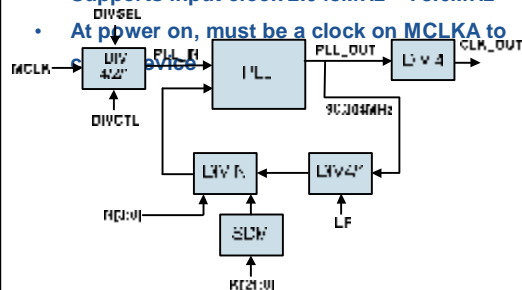
Additional Features

- **Power On Reset**
 - Device will Power On in a default OFF condition
 - DCVDD & AVDD
 - RESETB (AC97 I/F)
 - If either method occurs, WM9713 will default power down
 - "Warm Rest" only method to wake AC97 link
- **MICBIAS Current Detect Added**
 - 1/2 Mics connected and Short mic (mic button press)
 - Selectable Threshold Detection
 - Configurable GPIO Trigger
- **GPIO's**
 - 8 Possible GPIO'S
 - Addition due to multi-function pins – more flexibility
- **BMON**
 - No dedicated pin
 - Can measure SPKVDD supply pin

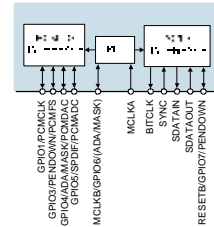


PLL

- PLL Included
- No Crystal Oscillator
- 2 Possible Input Clock Sources (MCLKA and/or MCLKB)
- Supports input clock 2.048MHz – 78.6MHz
- At power on, must be a clock on MCLKA to



- PLL Architecture exclusive to WM9713
 - PLL_OUT in the region of 98.304MHz for optimal performance
 - $98.304/4 = 24.576\text{MHz}$ system clock
 - N = Integer & K = Fractional
Divide
- 25 May 2004 | www.wolfson.com
- LE Divide for low input clocks

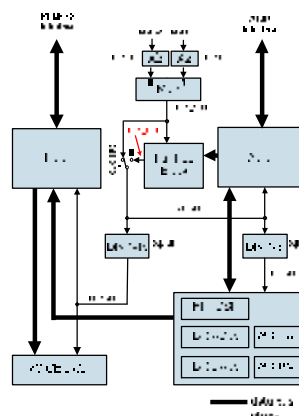
98 | www.wolfsonmicro.com

Divide 25 May 2004 | www.wolfsonmicro.com

25 May 2004

Internal Clock Architecture

- 2 possible clock inputs
- Internal clock from input or PLL output
- $\text{BITCLK} = \text{AC97 CLK} / 2$
- **HIFI (stereo adc / dac) & PCM CLK have separate Division**
 - Therefore can be run slower

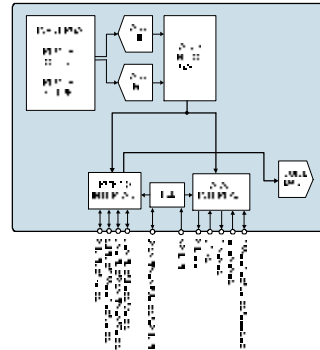


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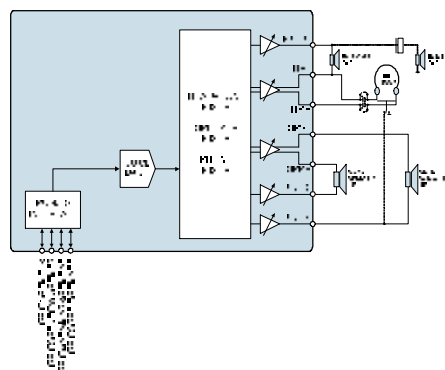
PCM / I2S Interface

- GPIO pins from the WM9712 now used as multifunction pins.
- Interface in addition to AC97 interface
- Connects to Voice DAC and Stereo ADC
- Supports the same interfaces as the WM8753
 - 7 Formats
- ADC data common to both PCM/I2S and AC97



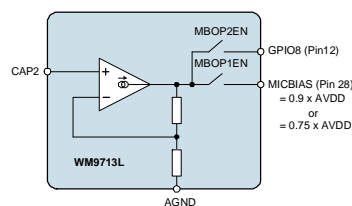
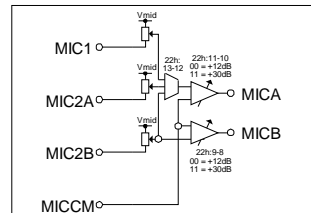
Voice DAC

- 16 Bit Mono DAC
- Smart phone Applications
- Playback of Rx voice signals
- Typical 8k/s, will support up to 48k/s
- Full analogue mixing capability



Microphone Configuration

- **2 Microphone Pre Amplifiers**
 - MICA
 - MICB
- **MICA Input Mux**
 - MIC1 or MIC2A or MIC2B
- **MICB fixed Input – MIC2B**
- **Input Structure Supports variety of configurations**
- **Pre Amps have common positive input (MICCM)**
- **Device Has 2 possible MICBIAS outputs**
 - Pin 28 (Main)
 - Pin 12 (Secondary function)



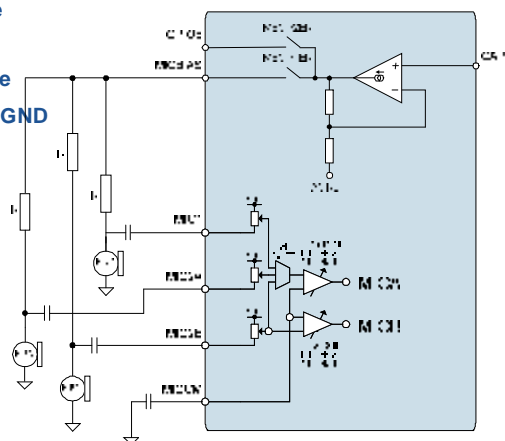
102 | Input Supports 2 Main Input Configurations

2006年6月 | www.wolfsonmicro.com

- 3 x Single Ended

Single Ended Configuration

- **Supports connection of 3 possible single ended microphones**
- **Only 2 can be input at any one time**
- **MICCM pin must be ac coupled to GND**

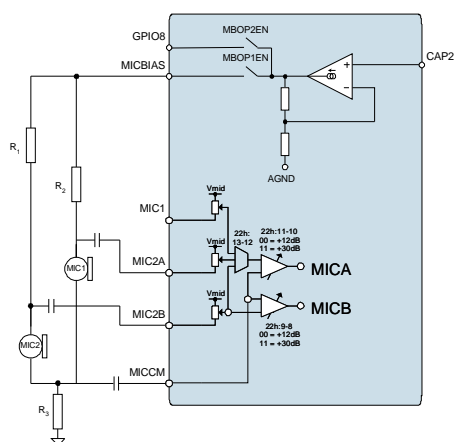


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Dual Microphone Configuration

- Stereo Microphone or Noise Cancellation
- MIC1 – MIC2A to MICCM
- MIC2 – MIC2B to MICCM
- R1 and R2 are microphone bias resistors
- R3 should be selected from experimental results, depending on microphones type

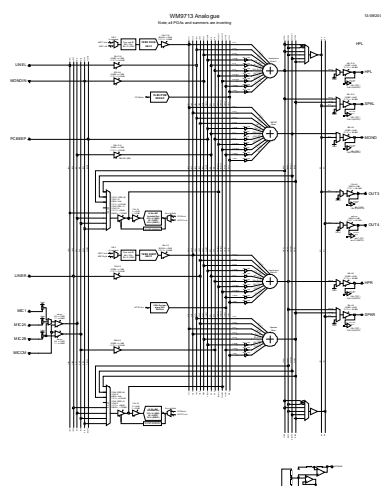


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Internal Analogue Mixing Paths

- Improved Mixing Capability from WM9712
- Mixing improves output drive options
- Inclusion of INV1 and INV2 to support stereo BTL and differential outputs
- Voice DAC included
- All PGA's and Summers are invertors
- MICA and MICB to all analogue mixers
- Power registers codes added

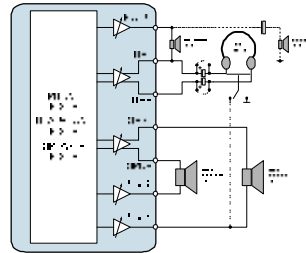


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Analogue Output Configuration

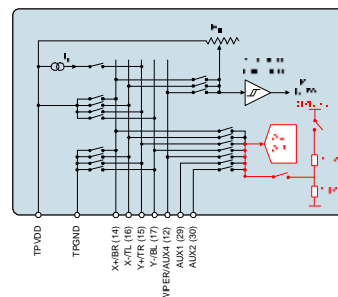
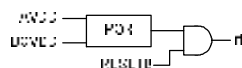
- Flexible and improved output configuration
- Only one additional output
 - OUT4
- Improved mixing allows more flexible configuration
- Stereo BTL (previously mono)
- Load Drive of each important
 - HPL/R, MONO => 16R
 - SPKL/R, OUT3/4 => 8R



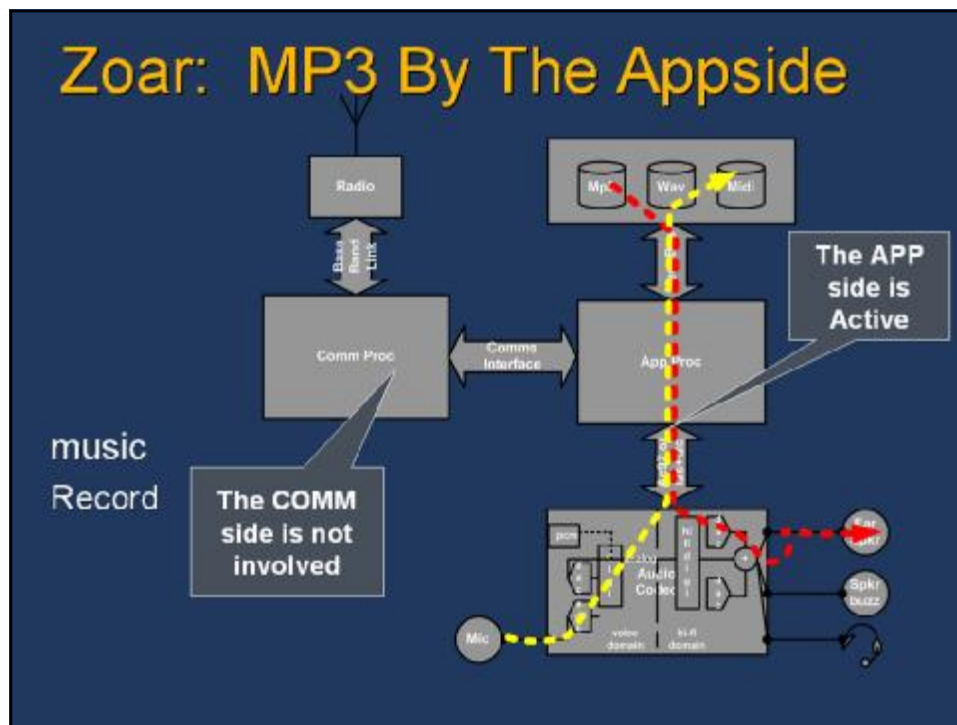
Output	Input	Gain	Impedance	Power	THD	Frequency	Notes
HPL	IN1	1	16R	1W	0.1%	20kHz	
HPR	IN2	1	16R	1W	0.1%	20kHz	
MONO	IN3	1	16R	1W	0.1%	20kHz	
SPKL	IN4	1	8R	0.5W	0.1%	20kHz	
SPKR	IN5	1	8R	0.5W	0.1%	20kHz	
OUT3	IN6	1	8R	0.5W	0.1%	20kHz	
OUT4	IN7	1	8R	0.5W	0.1%	20kHz	

Additional Features

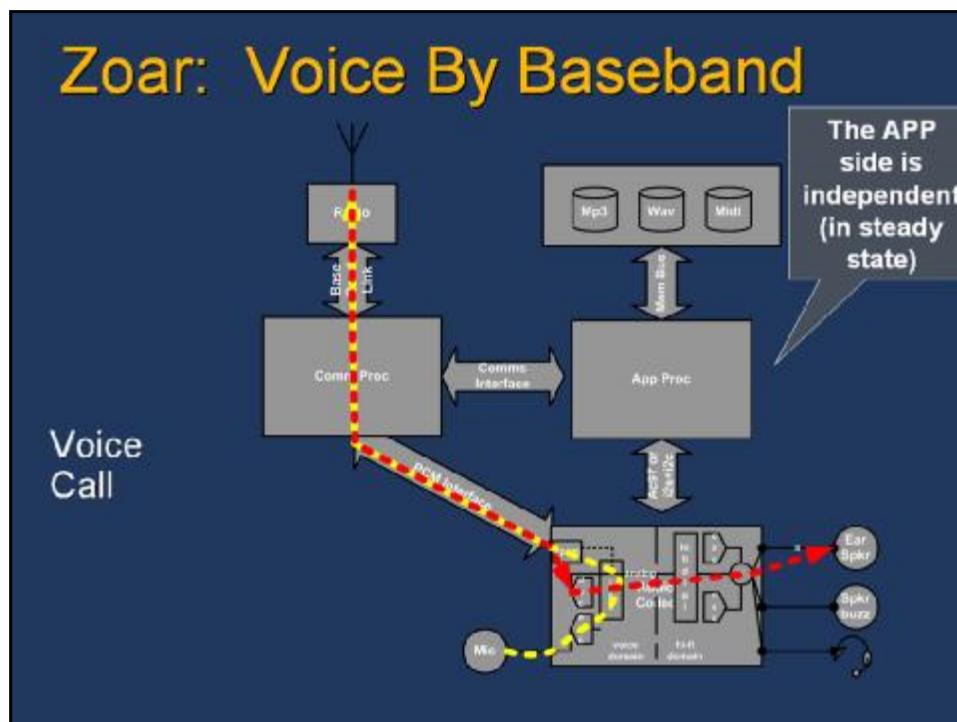
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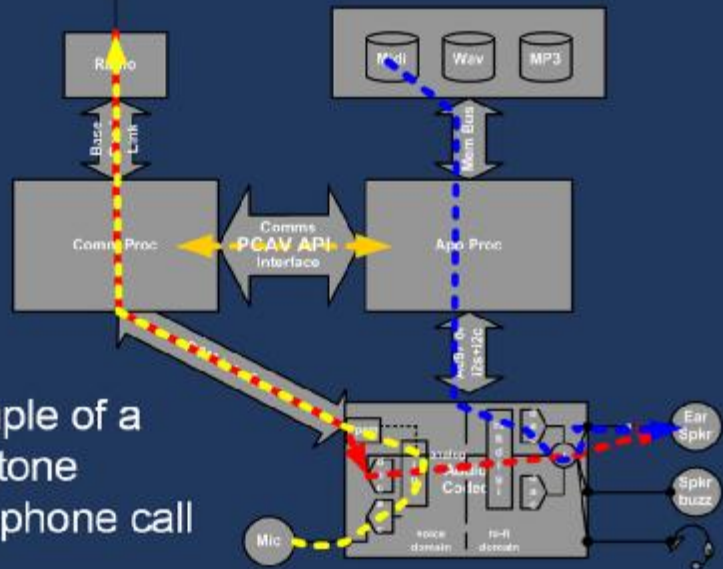
Zoar: MP3 By The Appside



Zoar: Voice By Baseband



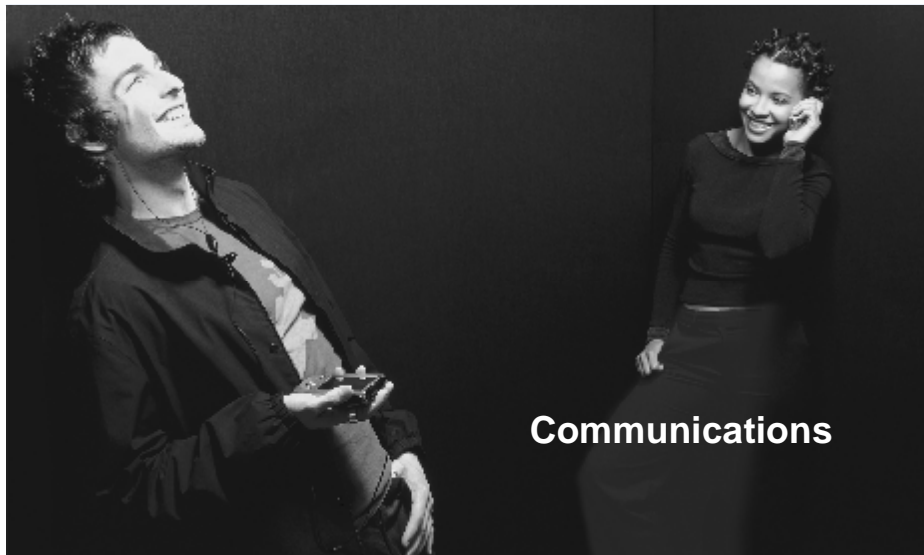
Zoar: Both Sides Do Audio



An example of a warning tone during a phone call



bringing digital technology to life



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