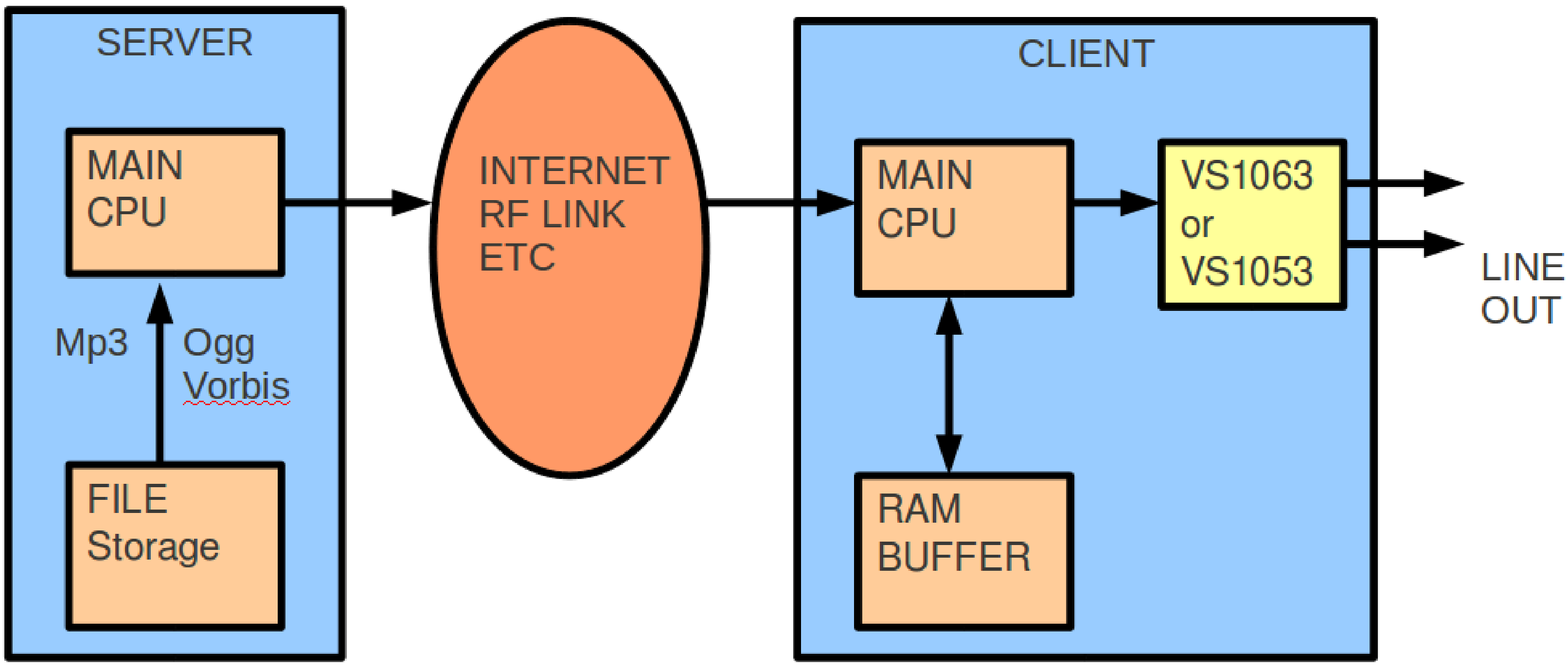
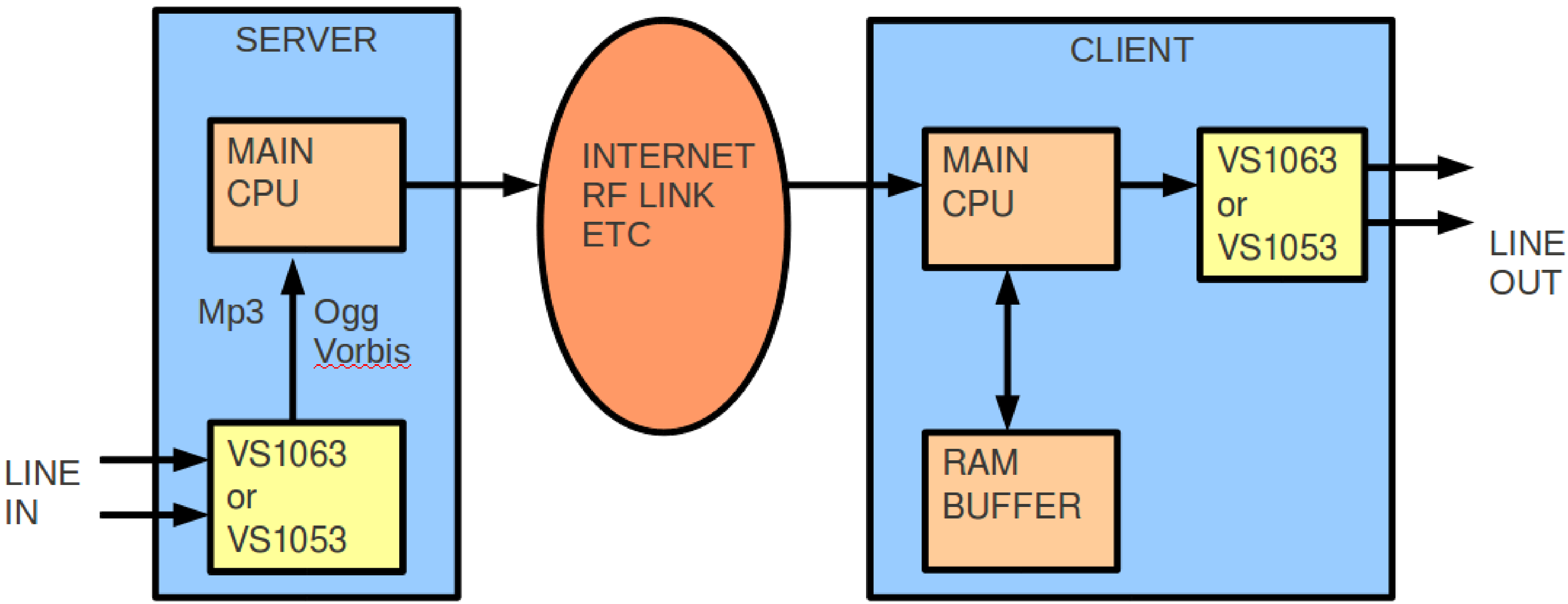
VLSI Solution Oy
Internet Streaming Presentation

June 2012

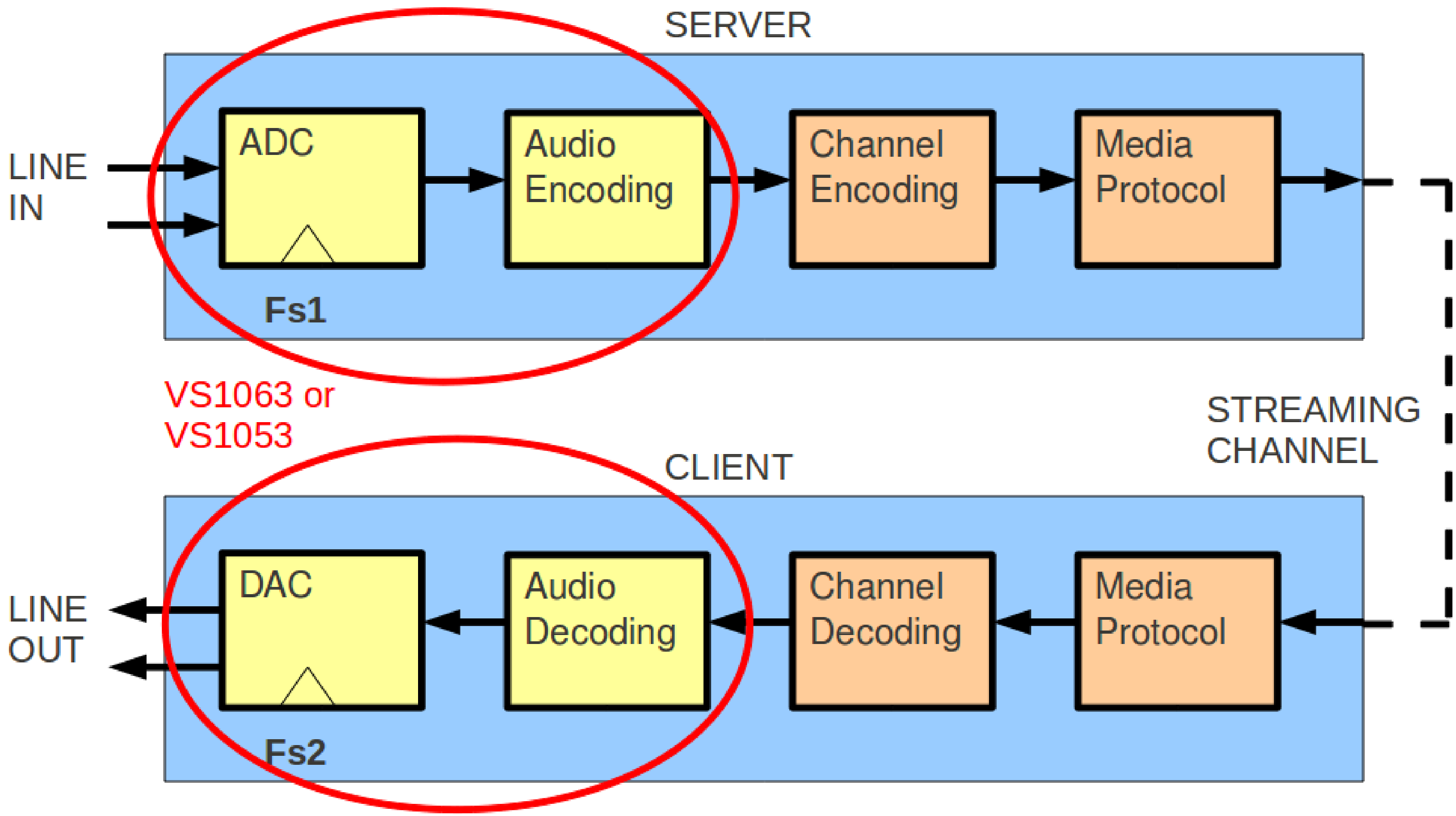
Streaming – File based server / client(s)



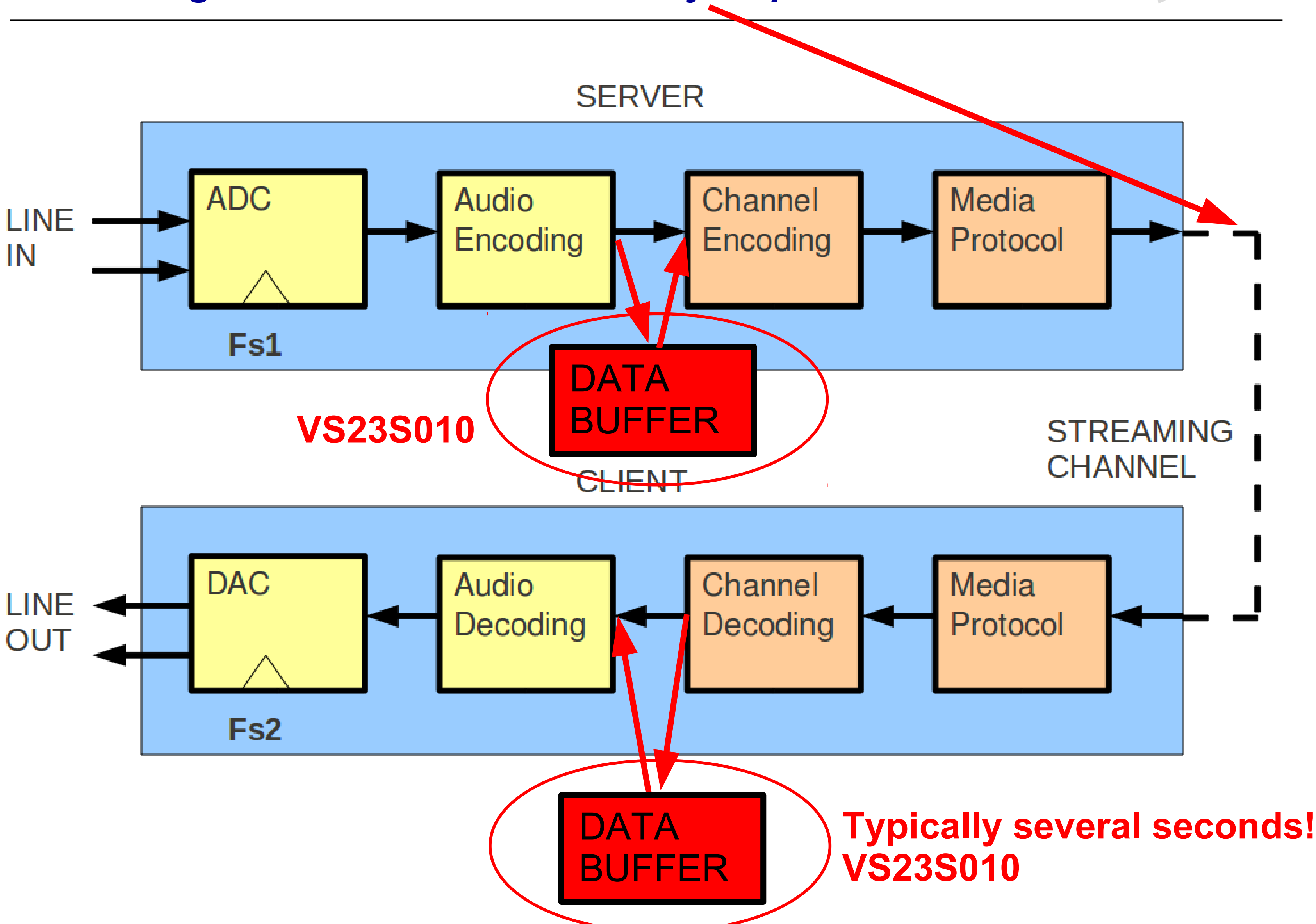
Streaming – Real-time server / client(s)



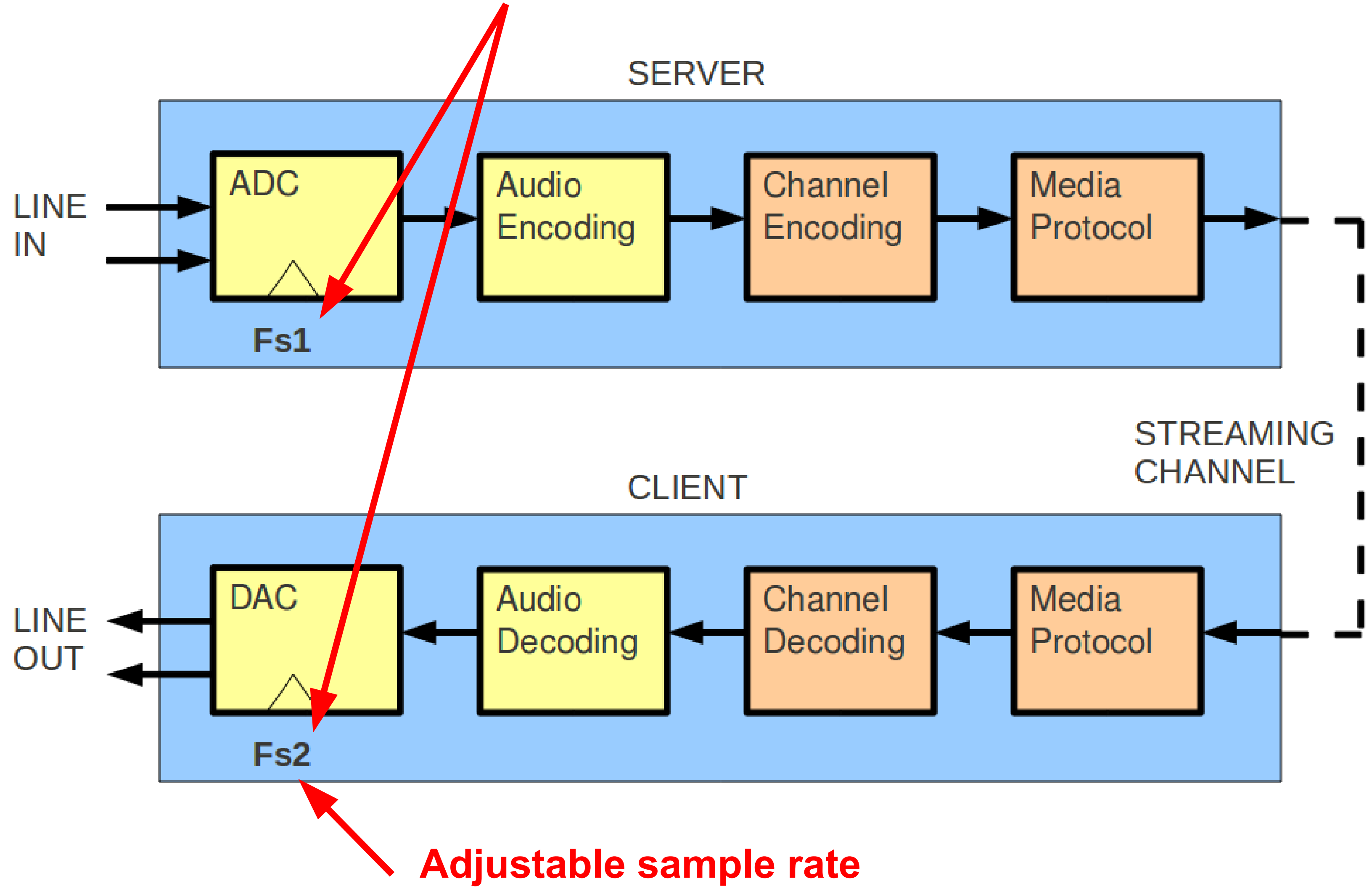
Streaming – Basic Block Diagram from Server to Client



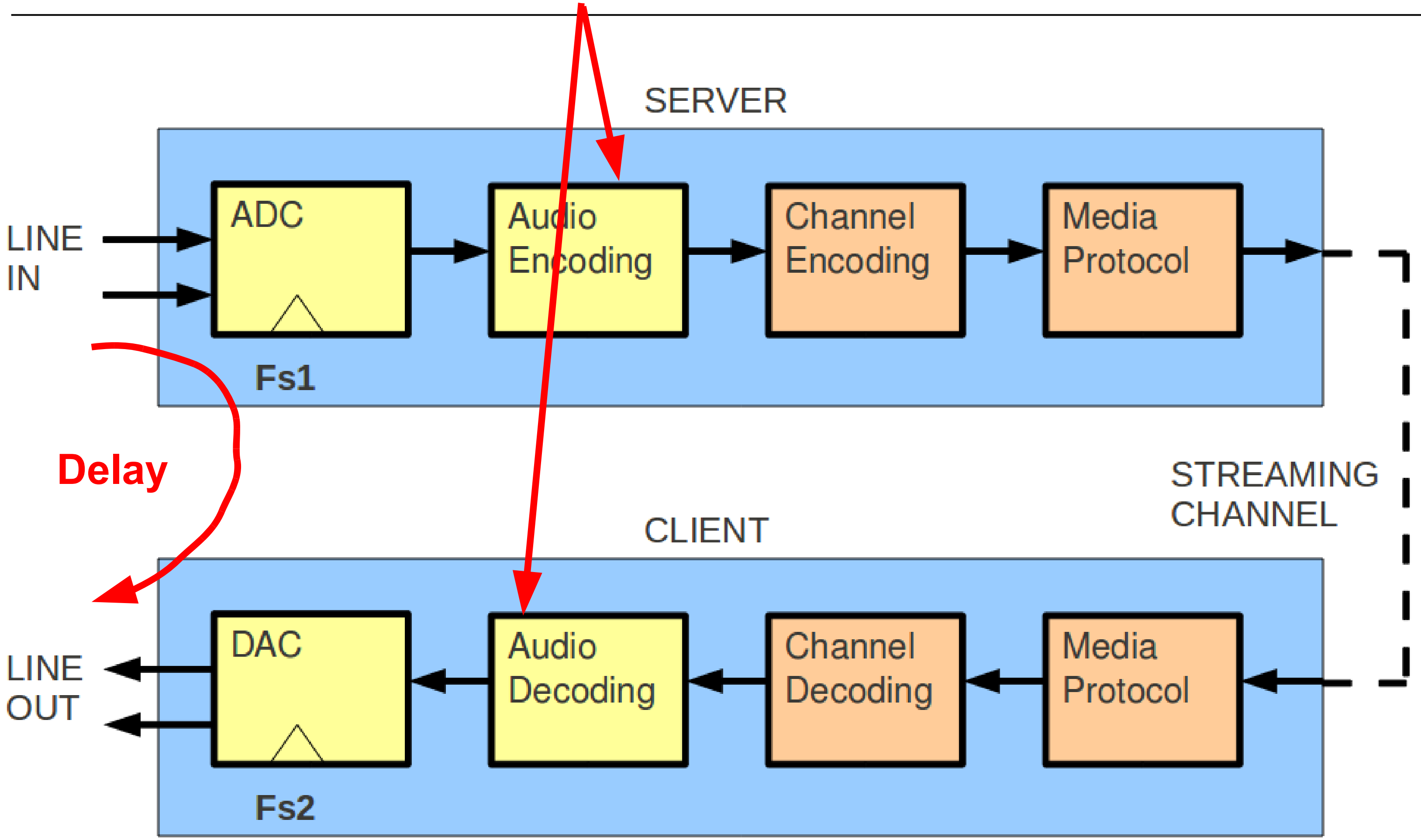
Streaming – Issue 1: Network delay unpredictable



Streaming – Issue 2: Sample rates are not identical



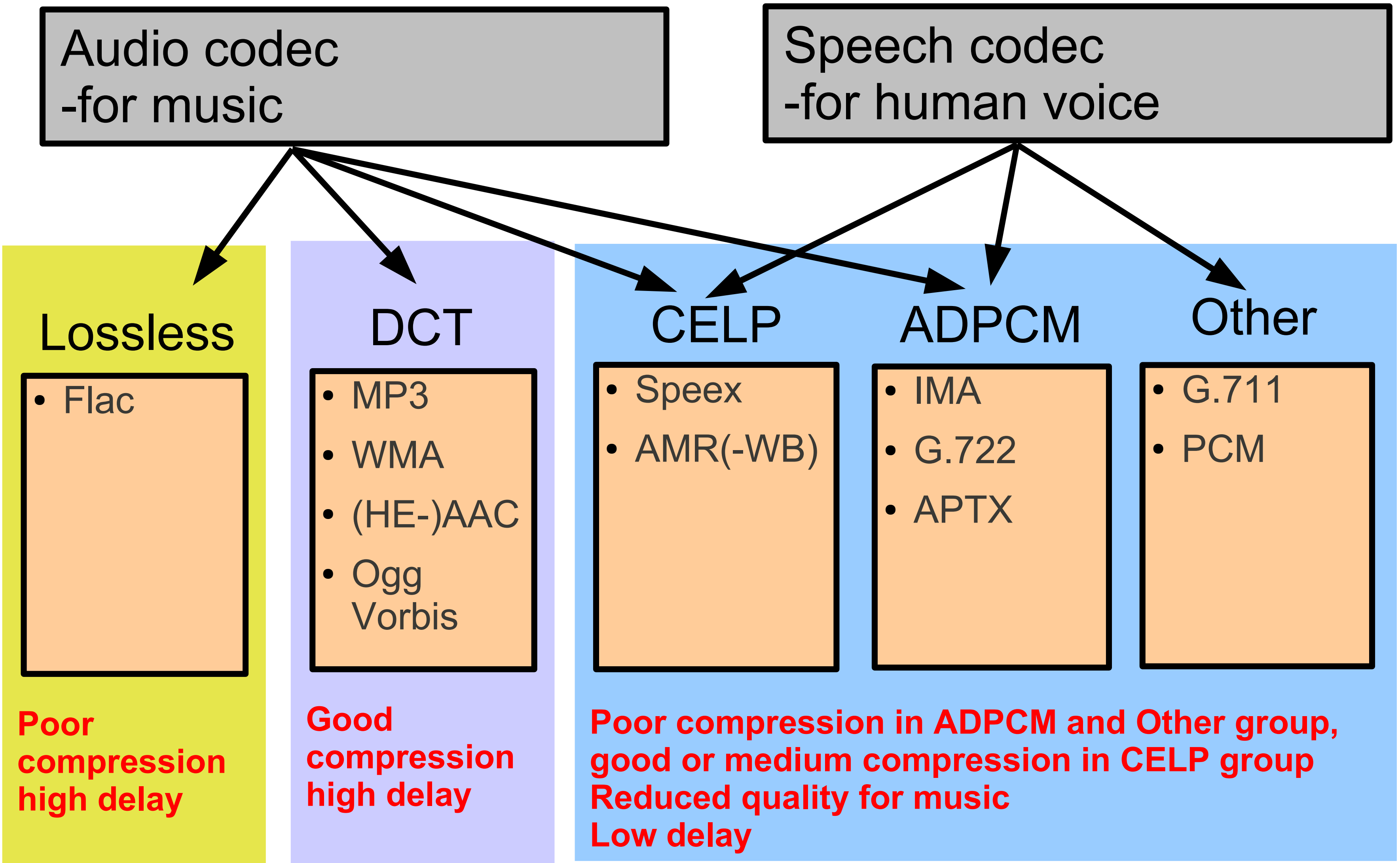
Streaming – Issue 3: Compression vs coding delay



Streaming – Issue 3: Compression vs delay

- Audio compression is based on a psychoacoustic model of human hearing system, which requires analysis of the frequency components of the audio
- The analysis filter of the model uses DCT
- The DCT requires a Data Buffer
- The Data Buffer causes a delay
- Encoding – decoding path has twice the delay of Data Buffer
- **There is a relationship between COMPRESSION EFFICIENCY and CODING DELAY**
- **GOOD COMPRESSION REQUIRES A LONG BUFFER**
- **THE LONG BUFFER RESULTS IN A LONG DELAY !**

Streaming – Issue 3: Compression vs delay



Audio Encoder - Decoder delays of VS1063

Fs [Hz]	PCM G.711 G.722 [ms]	IMA [ms]	MP3 [ms]	Ogg Vorbis [ms]
48000	3	14	36	124
44100	3	15	40	135
32000	3	19	54	185
24000	3	25	48	125
22050	3	26	52	140
16000	3	35	72	190
12000	3	46	96	250
11025	3	49	105	270
8000	3	66	144	200

Fs is sample rate
The value in the table is the total delay when using one VS1063 for encoding and another VS1063 for decoding

Technical Challenges

CHALLENGE	SOLUTION
Bandwidth of streaming channel	Use high compression **Ogg Vorbis**
Low delay	Avoid using DCT based high compression codecs
HiFi quality and low delay	Use **PCM** codec
HiFi quality, medium delay and good compression	Use high sample rate **mp3**
Variable streaming delay in the streaming channel	Use **buffer memory** in the receiver
Sample rate skew of the transmitter and receiver	Use buffer memory and **control receiver's sample rate** to match the average sample rate
Bit errors of streaming channel	Use **channel coding** for bit error correction and **robust decoder**
Low Power consumption	Use dedicated low-power chip (not PC)!

VLSI's Solutions for streaming - Encoding

Encoding Capability

	VS1063	VS1053	VS1003	VS1011
Mp3	Yes			
Ogg Vorbis	Yes	Plugin		
IMA ADPCM	Yes	Yes	Yes	
G.722	Yes *	Plugin **		
G711	Yes *	Plugin **		
PCM	Yes *	Yes	Yes	

* Full duplex operation
** to be available in the near future

VLSI's Solutions for Streaming - Decoding

Decoding Capability

	VS1063	VS1053	VS1003	VS1011
Flac	Yes	Plugin		
Mp3	Yes	Yes	Yes	Yes
WMA	Yes	Yes	Yes	
AAC	Yes	Yes		
HE-AAC	Yes	Yes		
Ogg Vorbis	Yes	Yes		
IMA ADPCM	Yes *	Yes	Yes	Yes
G.722	Yes *	Plugin		
G711	Yes *	Plugin		
PCM	Yes *	Yes	Yes	Yes
Midi		Yes	Yes	

* Full duplex operation

VS1063 or VS1053?

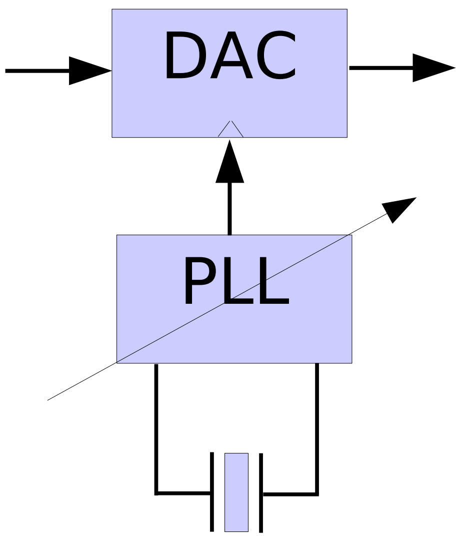
Use VS1063 when you need

- mp3 encoder
- Enhanced SNR performance mp3 decoder
- Sample clock (SRC) very fine tuning support
- Full-duplex G.711 or G.722

For other purposes the VS1053 is less expensive

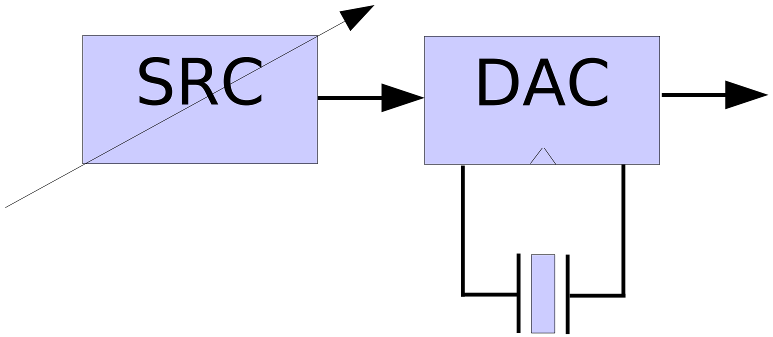
Highlights of VS1063 and VS1053 – Sample rate conversion

CONVENTIONAL



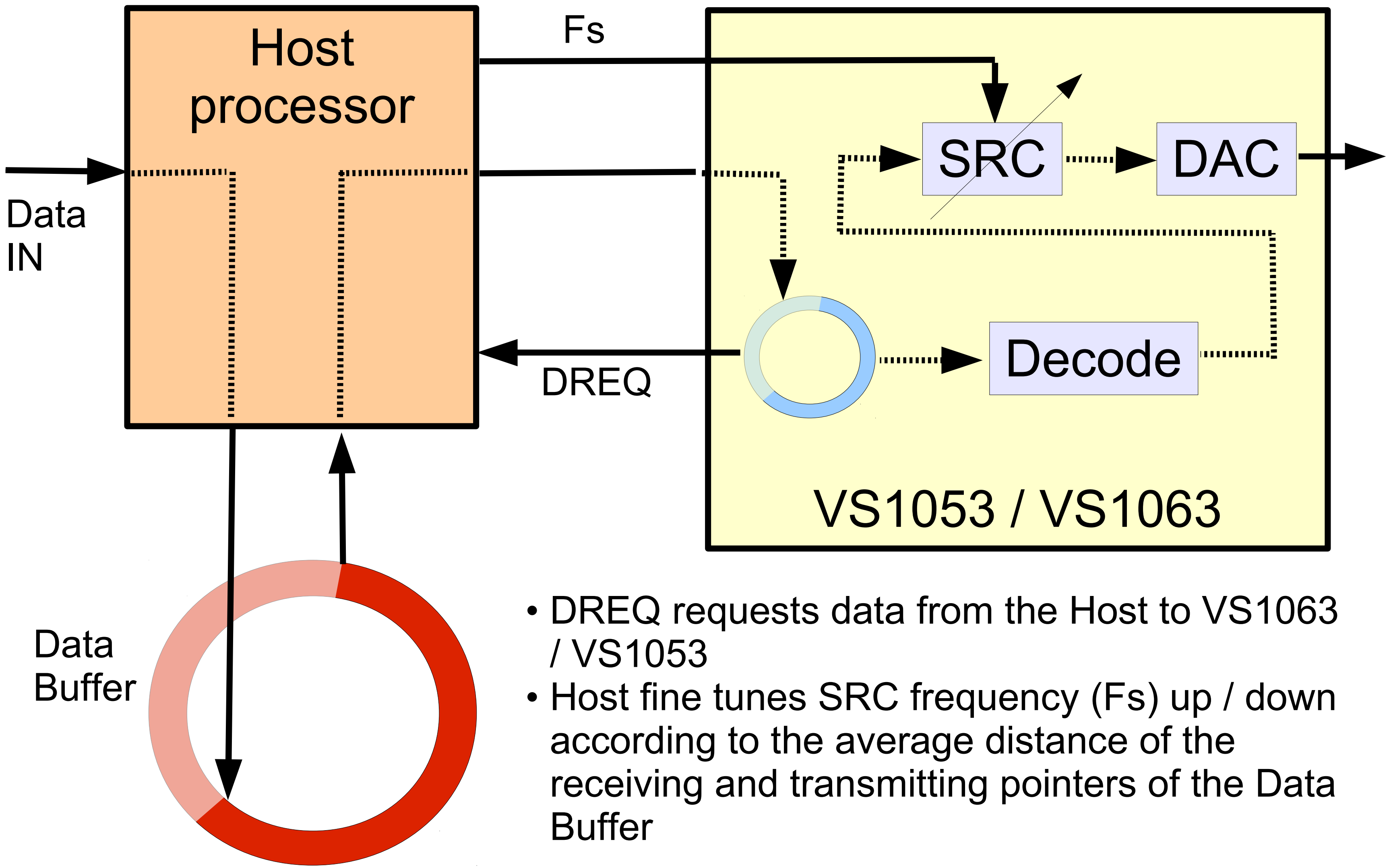
Conventional way to tune sample rate
=> possibly audible jitter from PLL

VS1053/VS1063



VS1063/VS1053 uses finely adjustable digital Sample Rate Converter (SRC) and constant sample rate DAC
=> crystal clear sound

Highlights of VS1063 and VS1053 – Sample rate conversion



- DREQ requests data from the Host to VS1063 / VS1053
- Host fine tunes SRC frequency (Fs) up / down according to the average distance of the receiving and transmitting pointers of the Data Buffer

Highlights of VS1063 and VS1053 – Ogg Vorbis Recording

- Input from microphone or line-in (stereo)
- Line input typ. SNR > 90 dB and THD < 0.005%
- Supports 11 quality settings (below some examples)
  - Voice: mono 15kbit/s @ 8kHz
  - Wide band voice: mono 28kbit/s @ 16kHz
  - HiFi voice: mono 87kbit/s @ 44.1kHz
  - Music: stereo 135kbit/s @ 44.1 kHz
- VS1053 (plugin), VS1063 (in ROM)

Highlights of VS1063 – Mp3 Recording

- Input from microphone or line-in (stereo)
- Line input typ. SNR > 90 dB and THD < 0.005%
- Supports 11 quality settings of VBR or CBR
- Supports all mp3 sample rates
- Available in VS1063 only

Highlights of VS1063 and VS1053 – DSP functions

Several DSP functions are included in VS1063

- EarSpeaker (also in VS1053)
- VU Meter
- AD Mixer, PCM Mixer
- 5-channel parametric EQ
- Speed shifter

IRAM for custom DSP functions

DSP functions designed by VLSI are added/updated as plugins

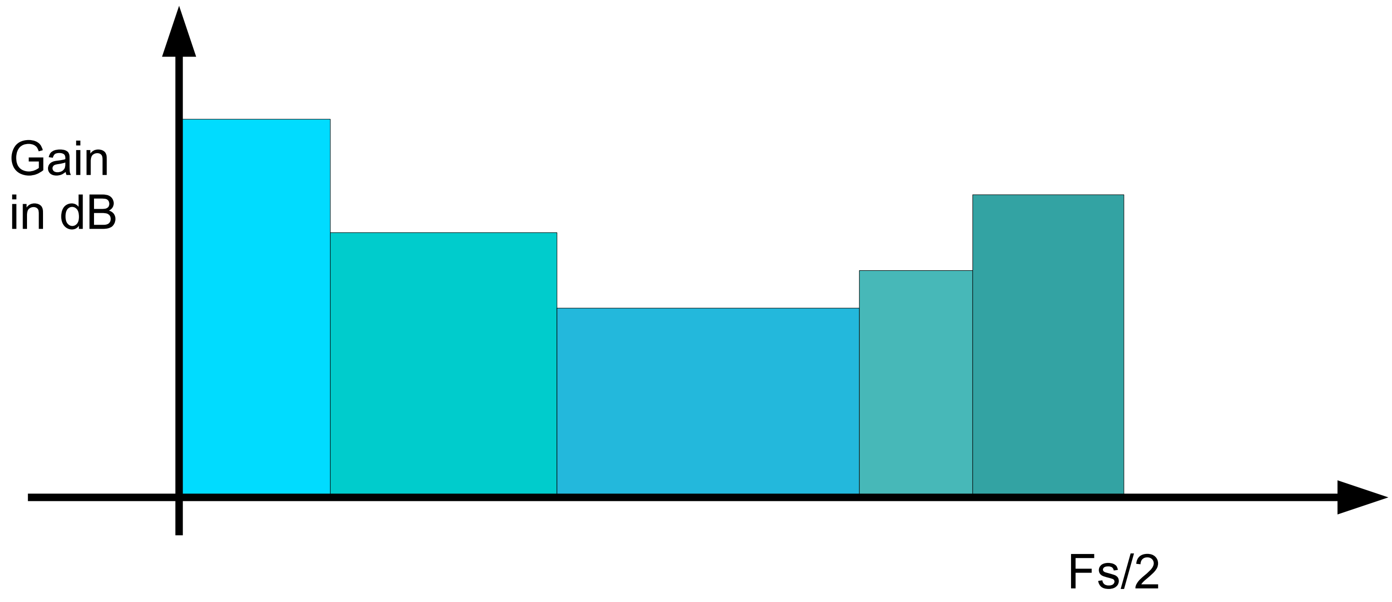
Highlights of VS1063 and VS1053 – EarSpeaker Technology

DSP simulates the Room acoustics to shift the headphone sound outside the listener's head

=>

- Natural sounding 3-dimensional sound
- Realistic stereo image
- No listening fatigue

Highlights of VS1063 and VS1053 – Parametric EQ



- 5 Bands, start and stop band can be set individually
- Gain programmed in dB for each band
- Synthesis of the filter coefficients done by the chip!
- In VS1063 only

VS1063 and VS1053 - Summary

Large range of codecs supported
- All major audio codecs supported, best codec selection on the market

Easy to use
- Sample rate fine tuning support
- SPI interface for data and control communication
- Mp3 licenses included

High performance analog hardware
- Integrated DAC, ADC, Headphone driver

DSP Processing included
- EarSpeaker, VU Meter, Mixer, 5 channel EQ, Speed Shifter

Customization
- IDE, plugins

Support
- KITs and Boards, customer support team