From Synthesis to Sampling

Evolution of PC Audio

Proskauer—Yakovlev
Overview

• Role of an audio interface
• History of audio interfaces
  – Shift from synthesized audio to sampled audio
• Modern audio devices
  – Entirely sample-based
  – Move toward integrated onboard audio
  – External DACs and amps
Role of an Audio Interface

• Perform audio recording and playback processing
• Move the load of audio processing away from the CPU
• Signal conversion
  – DAC for playback
  – ADC for recording
Synthesis

- Synthesis is the process of electrically creating sound
- There are many modes of synthesis
- Frequency Modulation (FM) synthesis
  - The frequency of a carrier wave is modulated by another oscillator
  - Complex and unique waveforms can be produced
  - Particularly good at metallic and percussive sounds
- Sample-based synthesis
  - Similar to synthesis using the standard waveforms but the seed waves are sampled sounds or instruments
Sampling and Pulse Code Modulation

- Analog signals are continuous and have a specific value at every given moment in time.
- This type of data is not conducive to the way computers operate.
- Sampling is the process of approximating an analog signal by reading the value at a given interval.
- The quality of the sample is determined by the sample rate and the bit depth (resolution).
- Pulse Code Modulation (PCM) is a method to digitally represent sampled audio.
PC Speaker (1981)

- First appeared in the IBM 5051 as a means of indicating motherboard error codes
- Produced only a single square wave at a given frequency
- Entirely CPU controlled, required lots of CPU performance to operate
- Rapidly switching between frequencies could give the illusion of multiple voices
PC Speaker PWM

- Pulse-width Modulation could be used to reproduce rudimentary sampled audio
- The Programmable Interval Timer (PIT) would be configured to produce a PWM signal at a frequency higher than the speaker could reproduce
- Adjusting the pulse width created a DC offset, extruding the speaker a specific amount
- 6 bits of freedom
- Very CPU intensive, required nearly all the CPU time
  - Graphics would lock up while audio was playing
PCjr/Tandy 1000 (1983)

- PCjr was IBM’s first step into the home micro market
- Included a Texas Instruments SN76496 programmable sound generator (PSG)
  - Capable of producing 3 square wave channels and one noise channel
  - 16 volume levels for each channel
  - Similar to the MOS Technologies SID chips on the Commodore 64
AdLib (1987)

- First major improvement in PC audio
- Featured the Yamaha YM3812 (OPL2) FM chip
  - 6 melodic voices with 5 percussion voices, or 9 melodic voices
  - 2 FM operators
  - Full ADSR envelope for each voice
- Very high support in games from the time
Creative Music System / Game Blaster (1988)

- First attempt from Creative Labs to produce a card
- Featured two Philips SAA1099 PSGs
  - 12 channels of square waves
- Marketed through RadioShack
- Did not sell well since the AdLib card offered far better music quality
Sound Blaster (1989)

- Designed to directly compete with the AdLib card
- First card to feature sampling
  - 22.05kHz 8-bit mono playback
  - 13kHz 8-bit mono recording
- Included the same Yamaha music chip as the AdLib
- Commercial success
  - Integrated game port
  - Massive library of supported games
  - Pushed AdLib out of the market within 2 years
Roland MT-32 (1987)

- External MIDI synthesizer module
  - 32 voices
  - 128 instruments
  - 30 percussion
  - 10 digital effects
- Not originally intended for use in PC gaming
  - Very cost prohibitive, required MIDI interface card (typically the MPU-401)
  - Was created by Roland for amateur musicians as a budget external synthesizer
- Best sounding audio solution at the time, supported by games of the time
Gravis Ultrasound (1991)

- First PC card to support sample-based synthesis
- Gravis GF1
  - 256kB RAM
  - 32 voices
  - 16-bit playback
  - 8-bit record
  - 44.1kHz
  - Stereo
- Supported hardware mixing, which was usually performed by the CPU
- Did not contain FM synthesis chip, used the GF1 to emulate the OPL2
AC’97 (1997)

• Audio codec developed by Intel in response to ubiquity of sound cards
• Digital controller (DC97)
  – Built into southbridge of motherboard
  – Interfaced with CPU and system
• AC-Link
  – Digital interface between DC97 and the codec
• Codec
  – Third party chip that interfaced with the AC-Link and contained the DAC components
Intel High Definition Audio (2004)

- Successor to AC’97
- Same controller/link(codec) architecture as AC’97
- Specs
  - 15 input, 15 output streams
  - 16 PCM channels/stream
  - 8-32 bit sample resolution
  - 6-192kHz sample rate
  - Audio jack detection and retasking
Sound Cards Today

- **Onboard solutions have sufficient isolation to prevent EMI.**
  - Placed far from other components
- **Discrete sound cards are less common**
- **Discrete cards typically use PCIe ports**
  - Offer more channels
  - Support for higher impedance headphones
  - Less EMI than onboard audio
External DACs and Amplifiers

• Used typically in music production and professional audio work
  – Situated outside of a computer
  – Offer lowest possible EMI for clean sound
  – Amplifiers may be necessary for high impedance speakers
  – Minimizes THD
Conclusion

• Early computers were not powerful enough to reproduce sampled audio
  – Synthesis was used to offload the audio generation workload off of the CPU

• Improvements in memory, processing power, and architecture bits allowed for dedicated sound cards to reproduce sampled audio

• Continued CPU performance improvements allowed audio generation to return to the CPU and motherboard-based chips
References

Audio Codec ‘97, Intel Corporation, Revision 2.3, April, 2002.


High Definition Audio Specification, Intel Corporation, Revision 1.0a, June 17, 2010.


