the gamedesigninitiative at cornell university

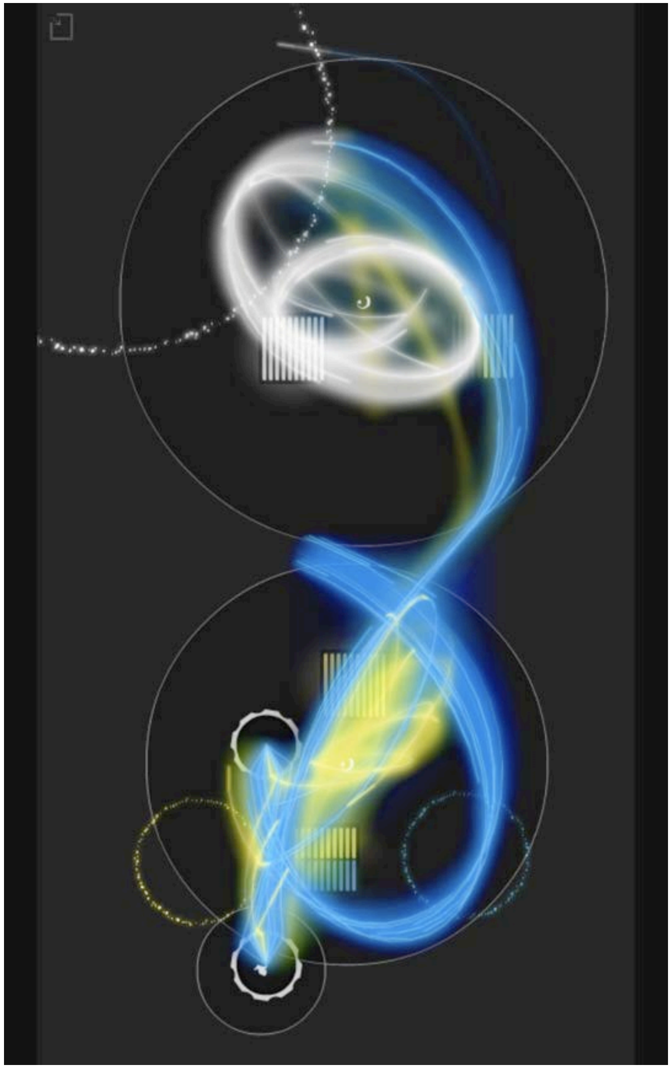
Game Audio

The Role of Audio in Games

Engagement

- **Entertains** the player
  - Music/Soundtrack
- Enhances the **realism**
  - Sound effects
- Establishes **atmosphere**
  - Ambient sounds

The Role of Audio in Games



Feedback

- **Indicate** off-screen action
  - Indicate player should move
- **Highlight** on-screen action
  - Call attention to an NPC
- Increase **reaction** time
  - Players react to sound faster

History of Sound in Games

Basic Sounds

- Arcade games
- Early handhelds
- Early consoles

Early Sounds: *Wizard of Wor*

History of Sound in Games

Basic Sounds → Recorded Sound Samples

Sample = pre-recorded audio

- Arcade games
- Early handhelds
- Early consoles

- Starts w/ MIDI
- 5th generation (Playstation)
- Early PCs

History of Sound in Games

Basic Sounds → **Recorded Sound Samples** → **Some Variability of Samples**

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- Early consoles

Recorded Sound Samples

- Starts w/ MIDI
- 5th generation (Playstation)
- Early PCs

Some Variability of Samples

- Sample selection
- Volume
- Pitch
- Stereo pan

History of Sound in Games

Basic Sounds → Recorded Sound Samples → Some Variability of Samples → More Variability of Samples

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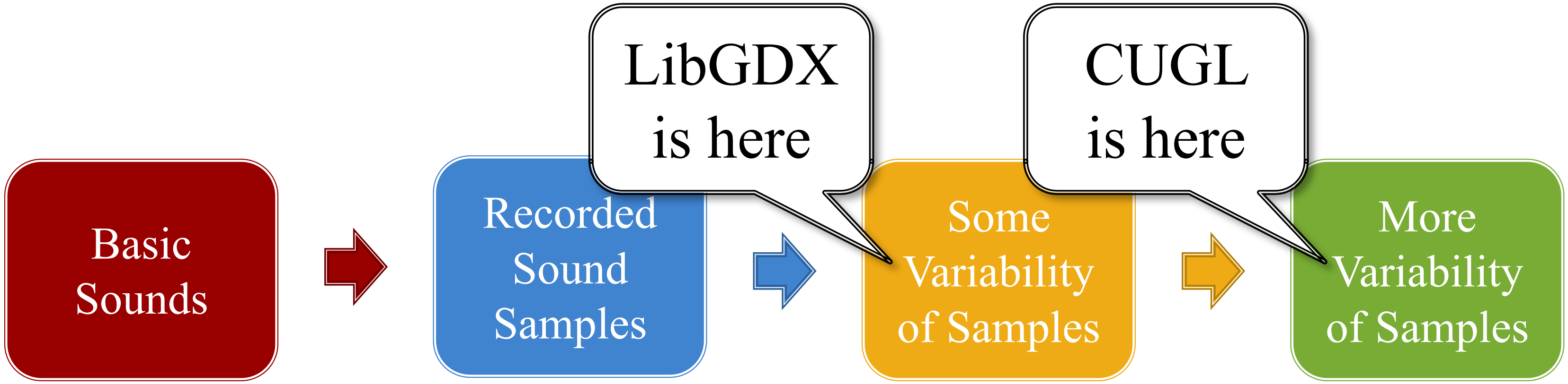
- Multiple samples
- Reverb models
- Sound filters
- Surround sound

History of Sound in Games



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The Technical Challenges

- Sound **formats** are not (really) cross-platform
  - It is not as easy as choosing MP3
  - Different platforms favor different formats
- Sound playback **APIs** are not standardized
  - LibGDX & CUGL are layered over many APIs
  - Behavior is not the same on all platforms
- Sound playback crosses **frame boundaries**
  - Mixing sound with animation has challenges

File Format vs Data Format

File Format

- The data storage format
  - Has data other than audio
- Many have many encodings
  - .caf holds MP3 *and* PCM
- **Examples:**
  - .mp3, .wav, .aiff
  - .aac, .mp4, .m4a (Apple)
  - .flac, .ogg (Linux)

Data Format

- The actual audio encoding
  - Basic audio codec
  - Bit rate (# of bits/unit time)
  - Sample rate
    (digitizes an analog signal)
- **Examples:**
  - MP3, Linear PCM
  - AAC, HE-AAC, ALAC
  - FLAC, Vorbis

Game Audio Formats

Format	Description	File Formats
Linear PCM	Completely uncompressed sound	.wav, .aiff
MP3	A popular compressed, lossy codec	.mp3, .wav
Vorbis	Xiph.org's alternative to MP3	.ogg
FLAC	Xiph.org's compressed, lossless codec	.flac, .ogg
MIDI	**NOT SOUND**; Data for an instrument	.midi
(HE-)AAC	A lossy codec, Apple's MP3 alternative	.aac, .mp4, .m4a
ALAC	Apple's lossless codec (but compressed)	.alac, .mp4, .m4a

MP3 historically avoided due to patent issues

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Supported in LibGDX

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Supported in CUGL

MP3 historically avoided due to patent issues

Which Formats Should You Choose?

- **Question 1:** Streaming or no streaming?
  - Audio gets large fast; music often streamed
  - But streaming creates overhead; bad for sound fx
  - Few engines support WAV streams (LibGDX & CUGL do)
- **Question 2:** Lossy or lossless compression?
  - Music can by lossy; sound fx not so much
  - Only FLAC and WAV are standard lossless
- **Question 3:** How many channels (speakers) needed?
  - Standard MP3 support is *stereo only*
  - Others support many channels (e.g. 7.1 surround)

Which Formats Should You Choose?

- **Question 1:** Streaming or no streaming?
  - Audio gets large fast; music often streamed
  - But
  - Fe
- **Ques**
  - Mu
  - On

Sound FX: Linear PCM/WAV

Music: OGG Vorbis

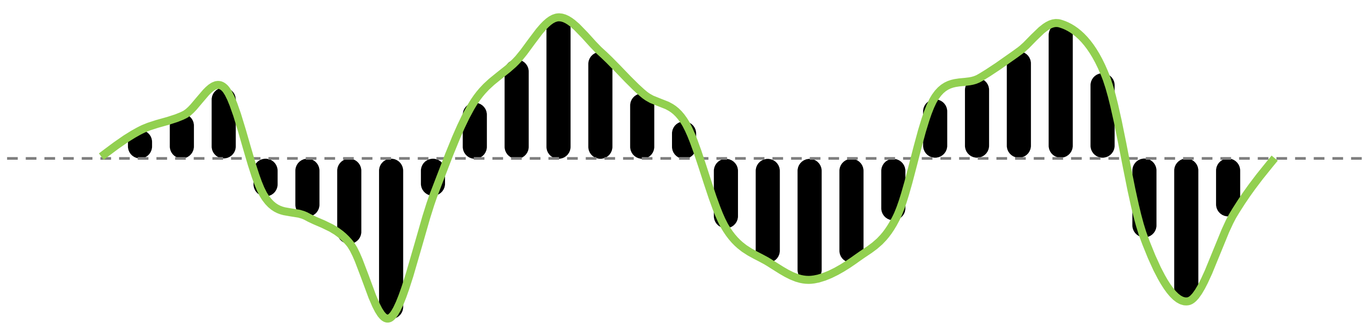
- **Question 3:** How many channels (speakers) needed?
  - Standard MP3 support is *stereo only*
  - Others support many channels (e.g. 7.1 surround)

Linear PCM Format

- Sound data is an array of **sample** values

0.5	0.2	-0.1	0.3	-0.5	0.0	-0.2	-0.2	0.0	-0.6	0.2	-0.3	0.4	0.0

- A sample is an **amplitude** of a sound wave



- Values are normalized -1.0 to 1.0 (so they are floats)

Linear PCM Format

- Sound data is an array of **sample** values

0.5	0.2	-0.1	0.3	-0.5	0.0	-0.2	-0.2	0.0	-0.6	0.2	-0.3	0.4	0.0

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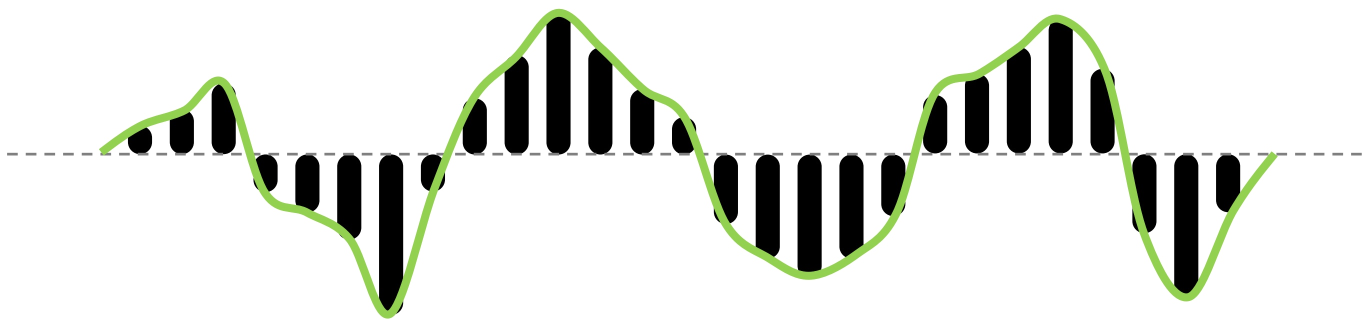
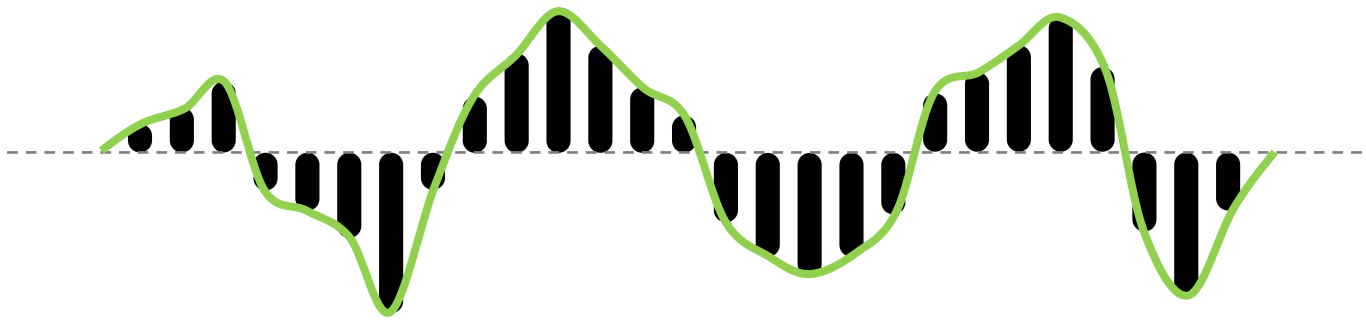


- Sometimes encoded as shorts or bytes MIN to MAX

Linear PCM Format

- Sound data is an array of **sample** values

0.5	0.2	-0.1	0.3	-0.5	0.0	-0.2	-0.2	0.0	-0.6	0.2	-0.3	0.4	0.0



- Magnitude of the amplitude is the volume
  - 0 is lowest volume (silence)
  - 1 is maximum volume of sound card
  - Multiply by number 0 to 1 to change global volume

Linear PCM Format

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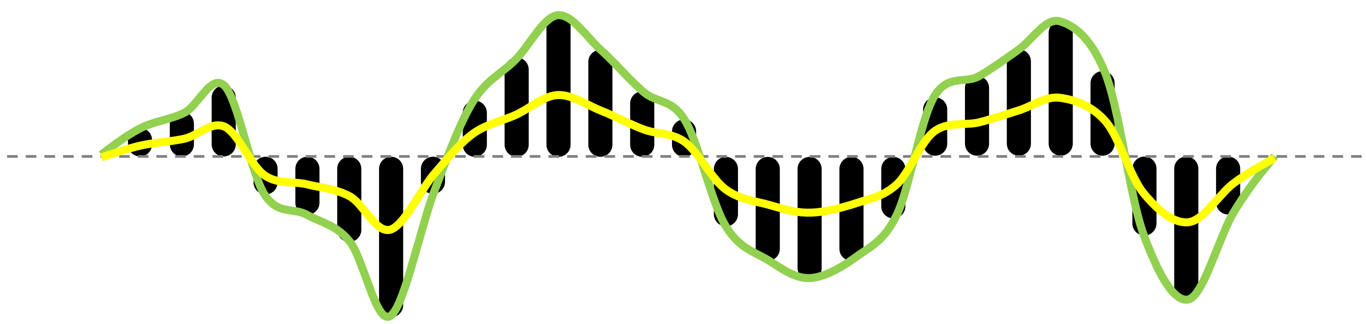
0.5	0.2	-0.1	0.3	-0.5	0.0	-0.2	-0.2	0.0	-0.6	0.2	-0.3	0.4	0.0



- Magnitude of the amplitude is the volume
  - 0 is lowest volume (silence)
  - 1 is maximum volume of sound card
  - Multiply by number 0 to 1 to change global volume

Linear PCM Format

- Samples are organized into (interleaved) **channels**

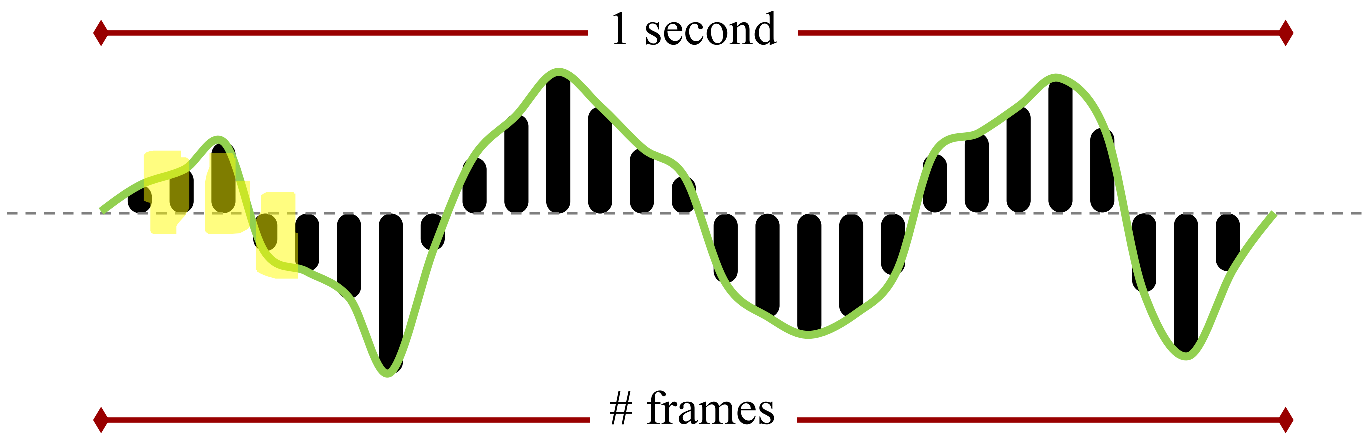
0.5	0.2	-0.1	0.3	-0.5	0.0	-0.2	-0.2	0.0	-0.6	0.2	-0.3	0.4	0.0

frame

- Each channel is essentially a **speaker**
  - Mono sound has one channel
  - Stereo sound has two channels
  - 7.1 surround sound is ***eight*** channels
- A **frame** is set of simultaneous samples
  - Each sample is in a separate frame

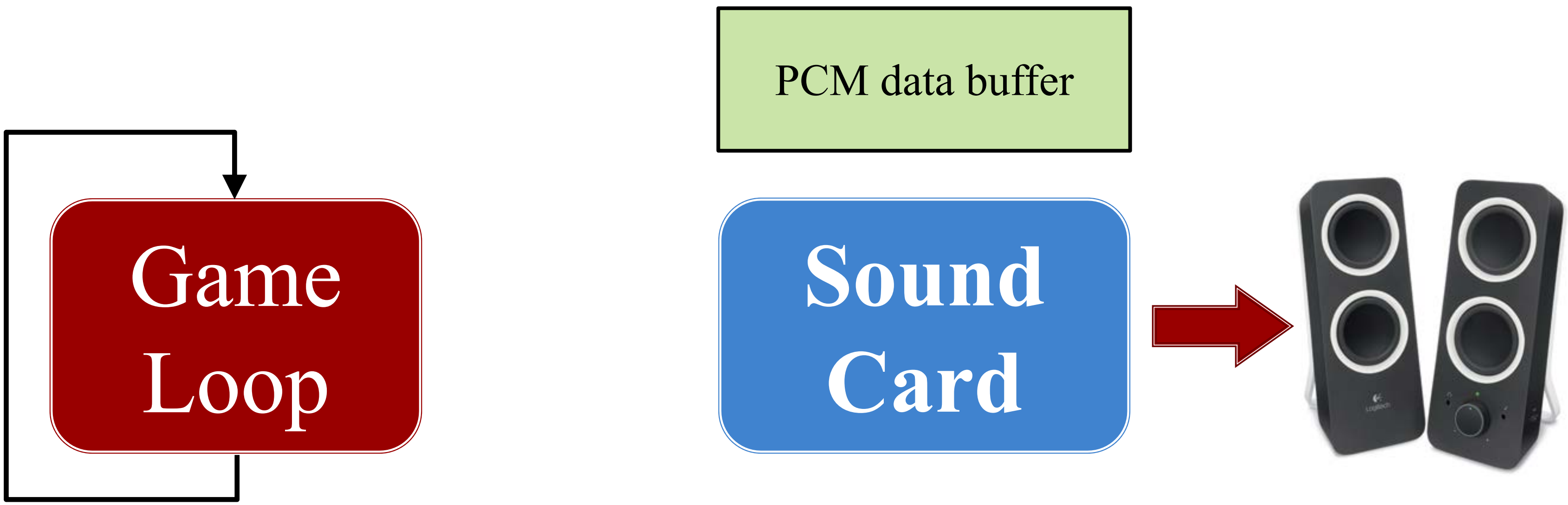
Linear PCM Format

- The sample rate is frames per second

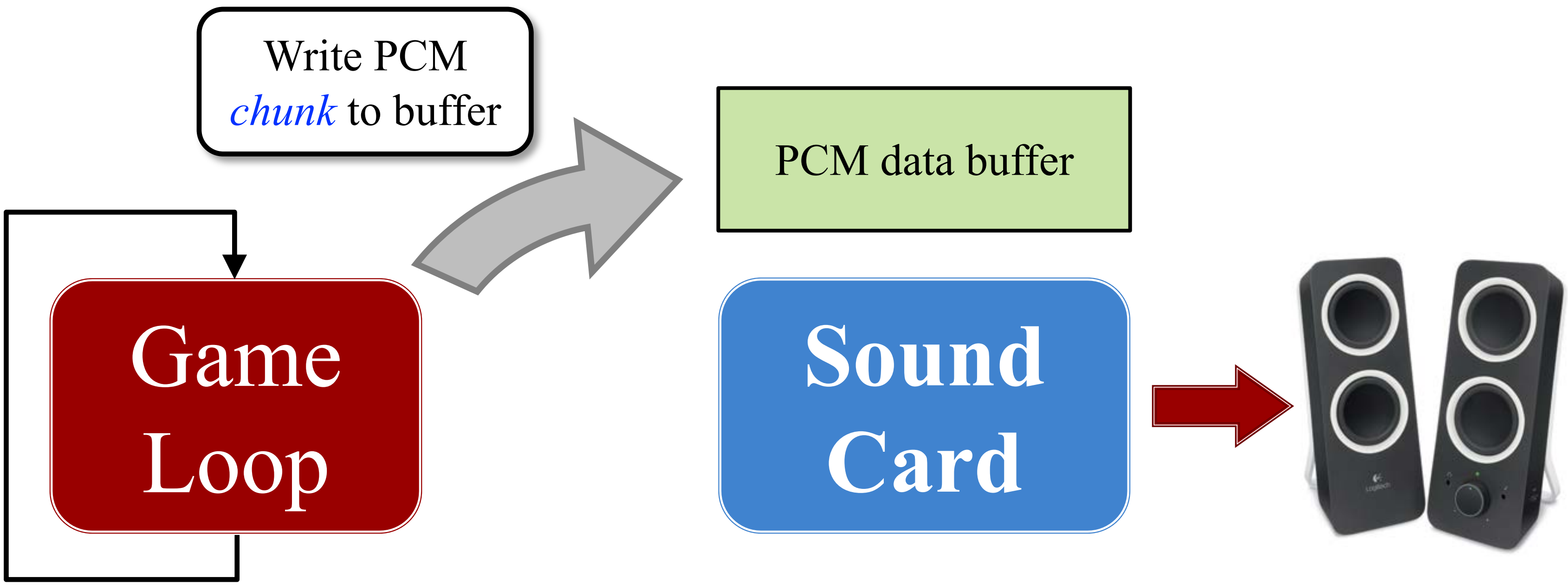


- **Example**: 0.5 seconds of stereo at 44.1 kHZ
  - 0.5 s * 44100 f/s = 22050 frames
  - 2 samples/frame * 22050 frames = 44100 samples
  - 4 bytes/sample * 44100 samples = 176.4 kBytes
- 1 minute of stereo CD sound is 21 MB!

Playing Sound Directly



Playing Sound Directly



Direct Sound in LibGDX: AudioDevice

- ```
  /**
   * Writes the array of float PCM samples to the audio device.
   *
   * This method blocks until they have been processed.
   */
  void writeSamples(float[] samples, int offset, int numSamples)
  ```

- ```
  /**
   * Writes array of 16-bit signed PCM samples to the audio device.
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   * This method blocks until they have been processed.
   */
  void writeSamples(short[] samples, int offset, int numSamples)
  ```

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  ```

  Requires separate *audio thread*

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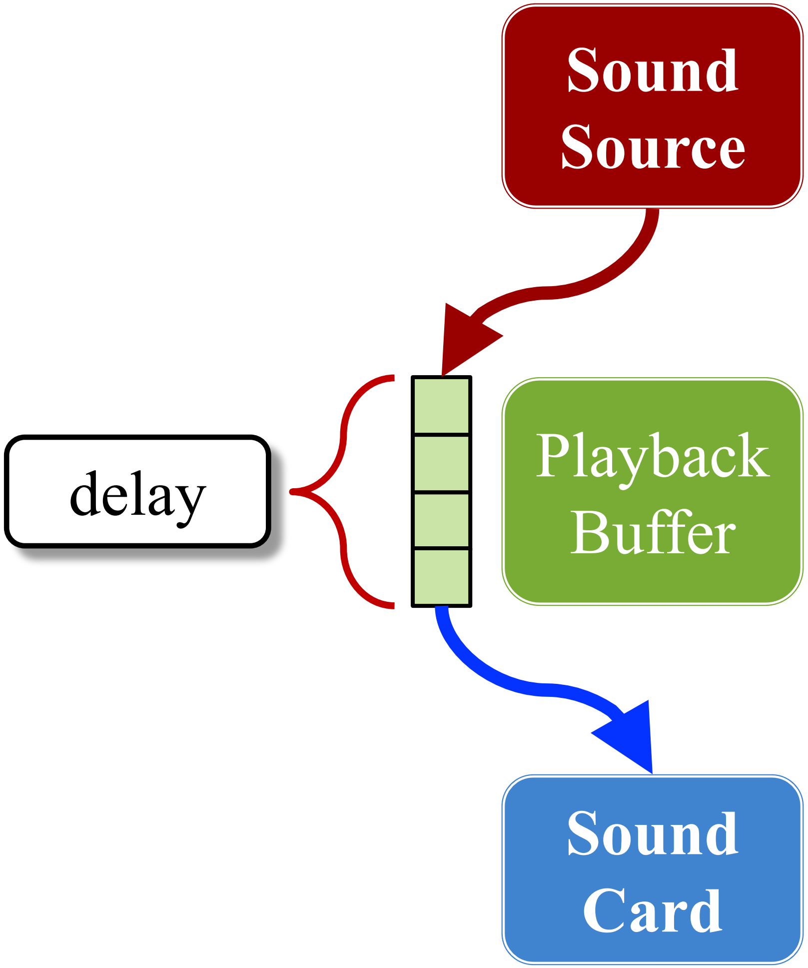
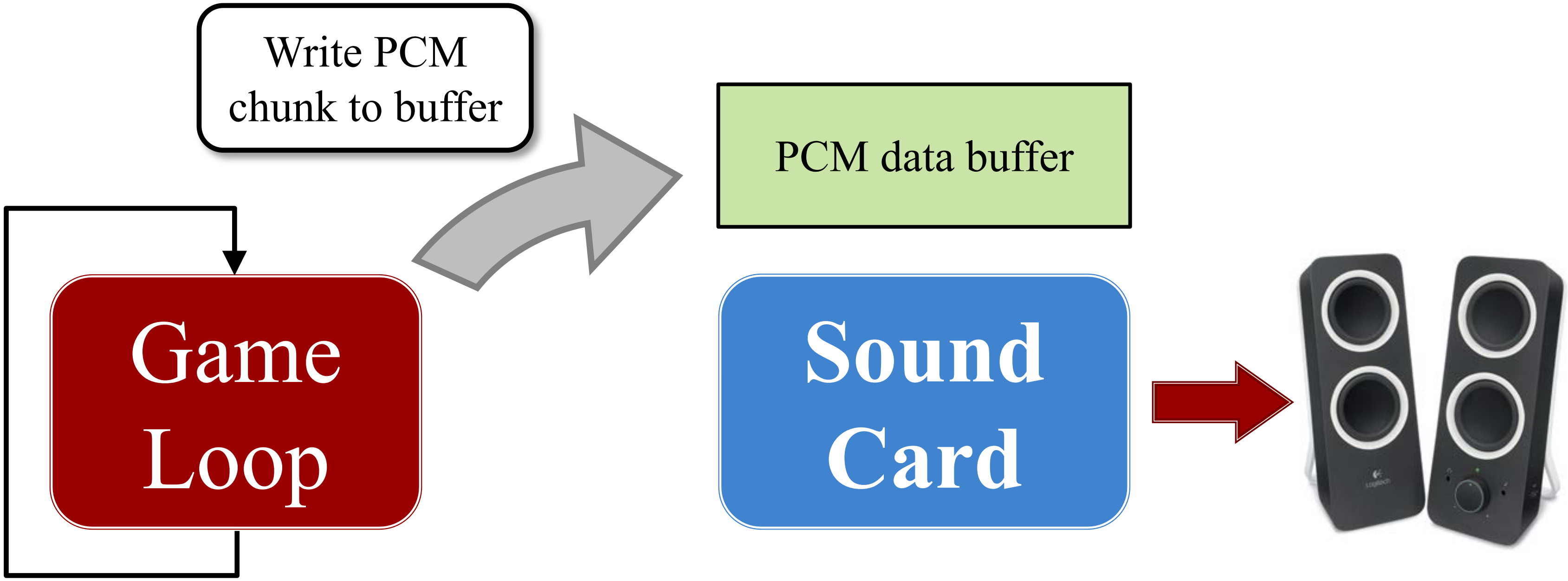
  ```

The Latency Problem

- Buffer is really a *queue*
  - Output from queue front
  - Playback writes to end
  - Creates a *playback delay*
- **Latency**: amount of delay
  - Some latency must exist
  - Okay if latency ≤ framerate
  - **Android latency is ~90 ms!**
- Buffering is a necessary evil
  - Keeps playback smooth
  - Allows real-time *effects*



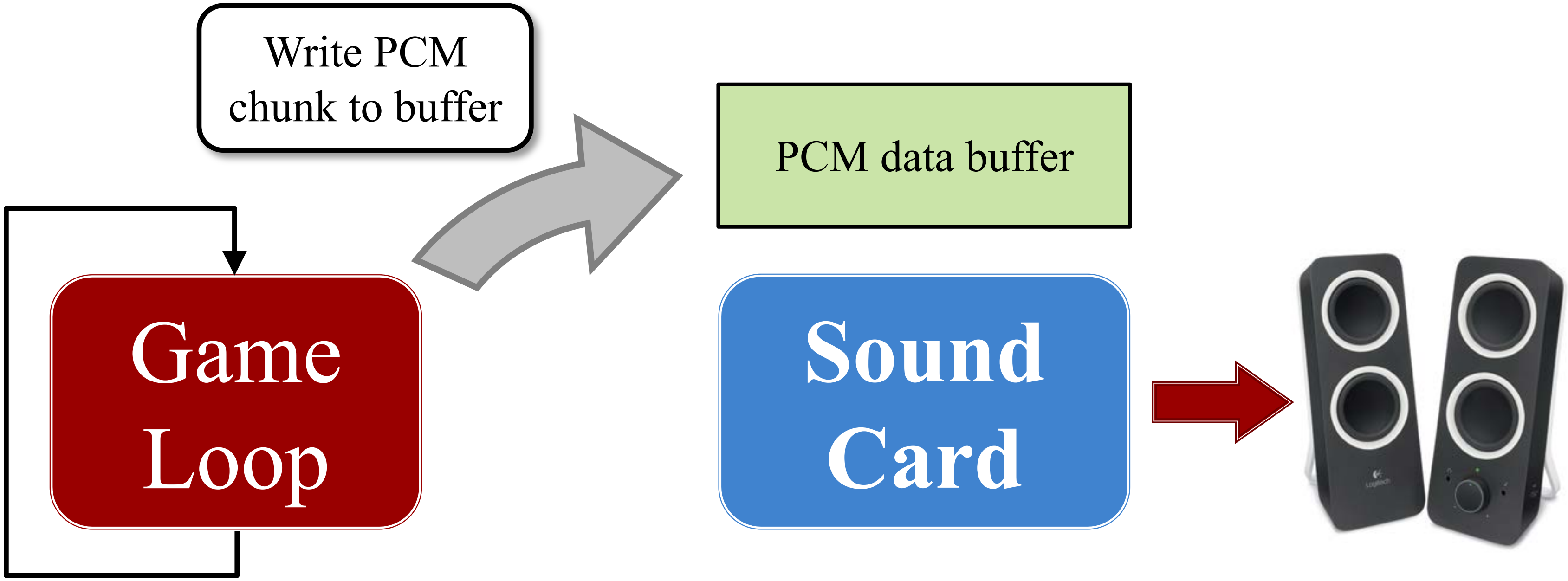
Playing Sound Directly



Choice of buffer size is important!

- **Too large**: *long* latency until next sound plays
- **Too small**: buffers swap too fast, causing audible pops

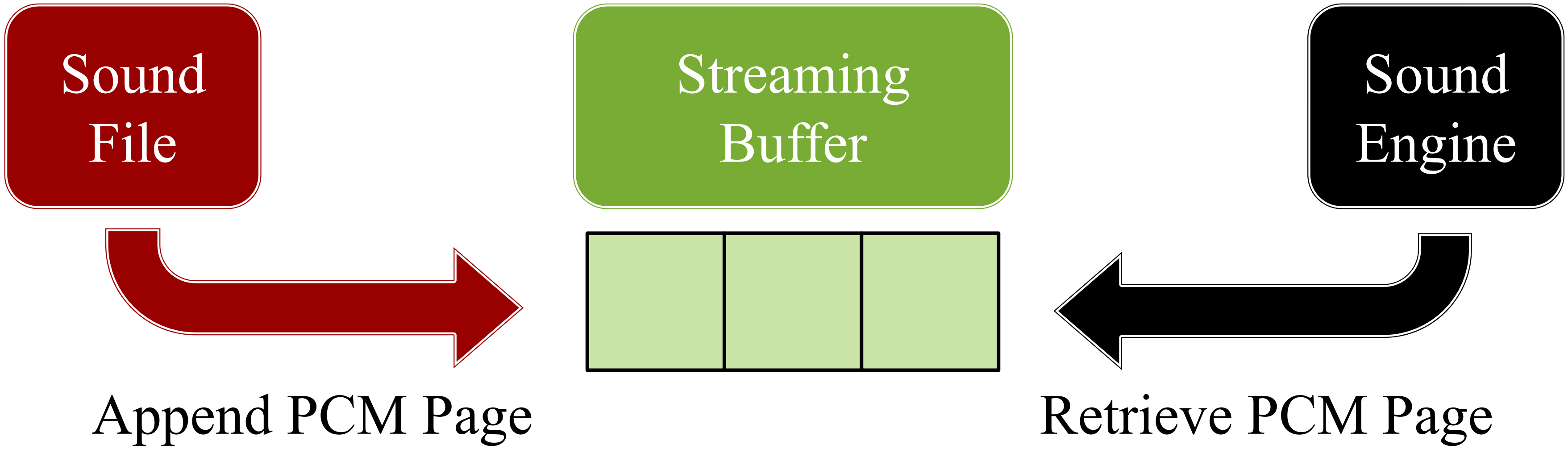
Playing Sound Directly



- Windows: 528 bytes (even if you ask for larger)
- MacOS, iOS: 512-1024 bytes (hardware varies)
- Android: 2048-4096 bytes (hardware varies)

How Streaming Works

- All sound cards **only** play PCM data
  - Other files (MP3 etc.) are decoded into PCM data
  - But the data is *paged-in* like memory in an OS
- Why LibGDX/CUGL can stream WAV files too!



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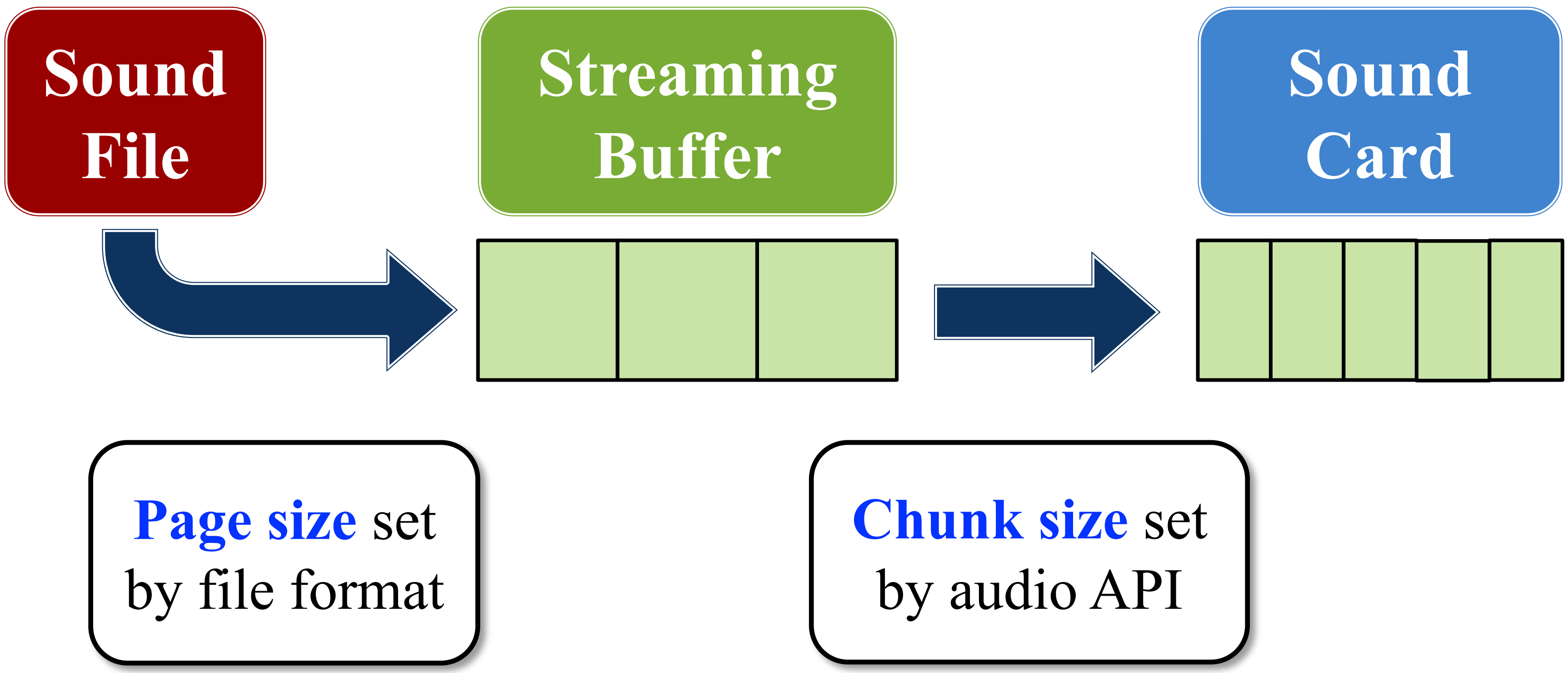
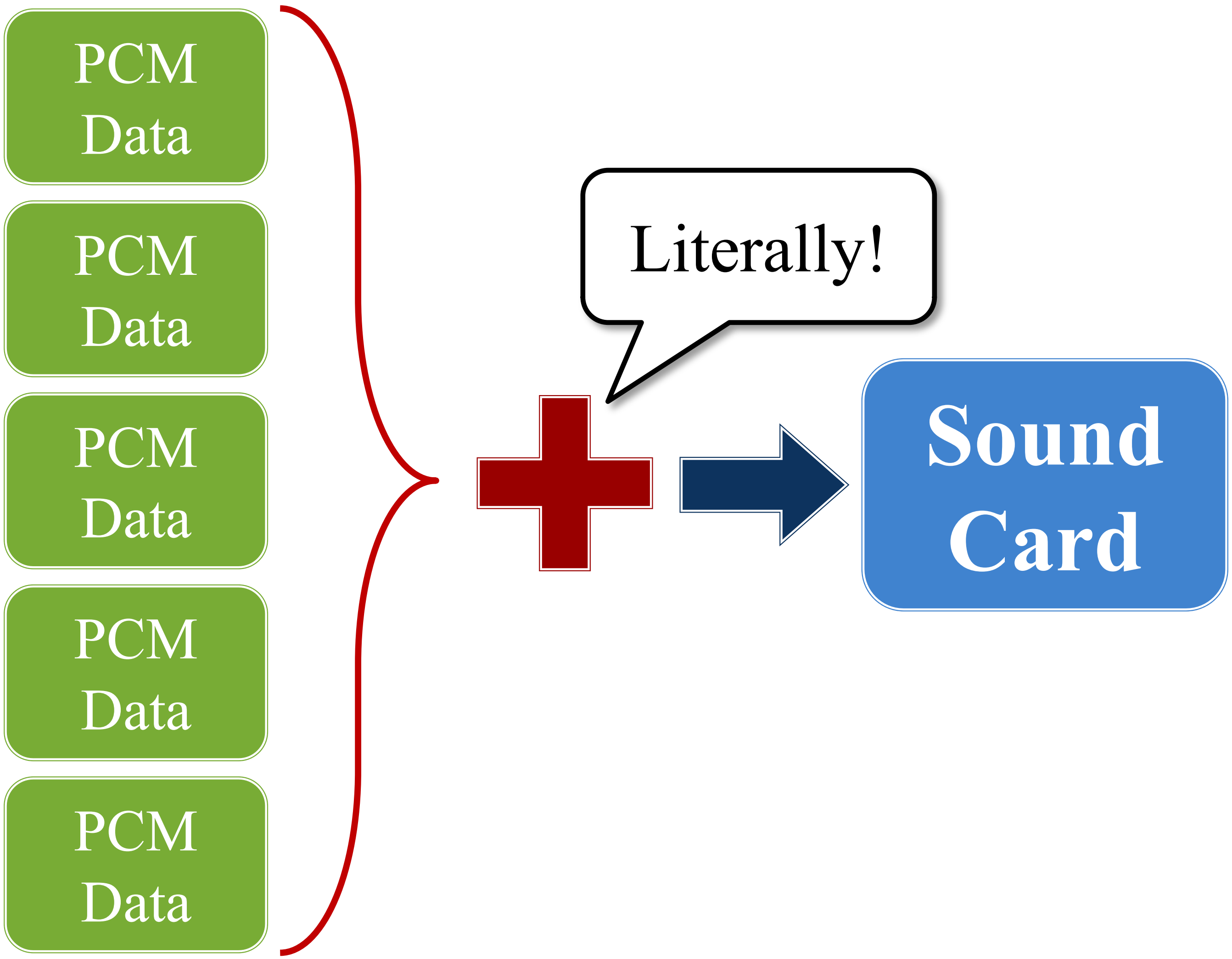
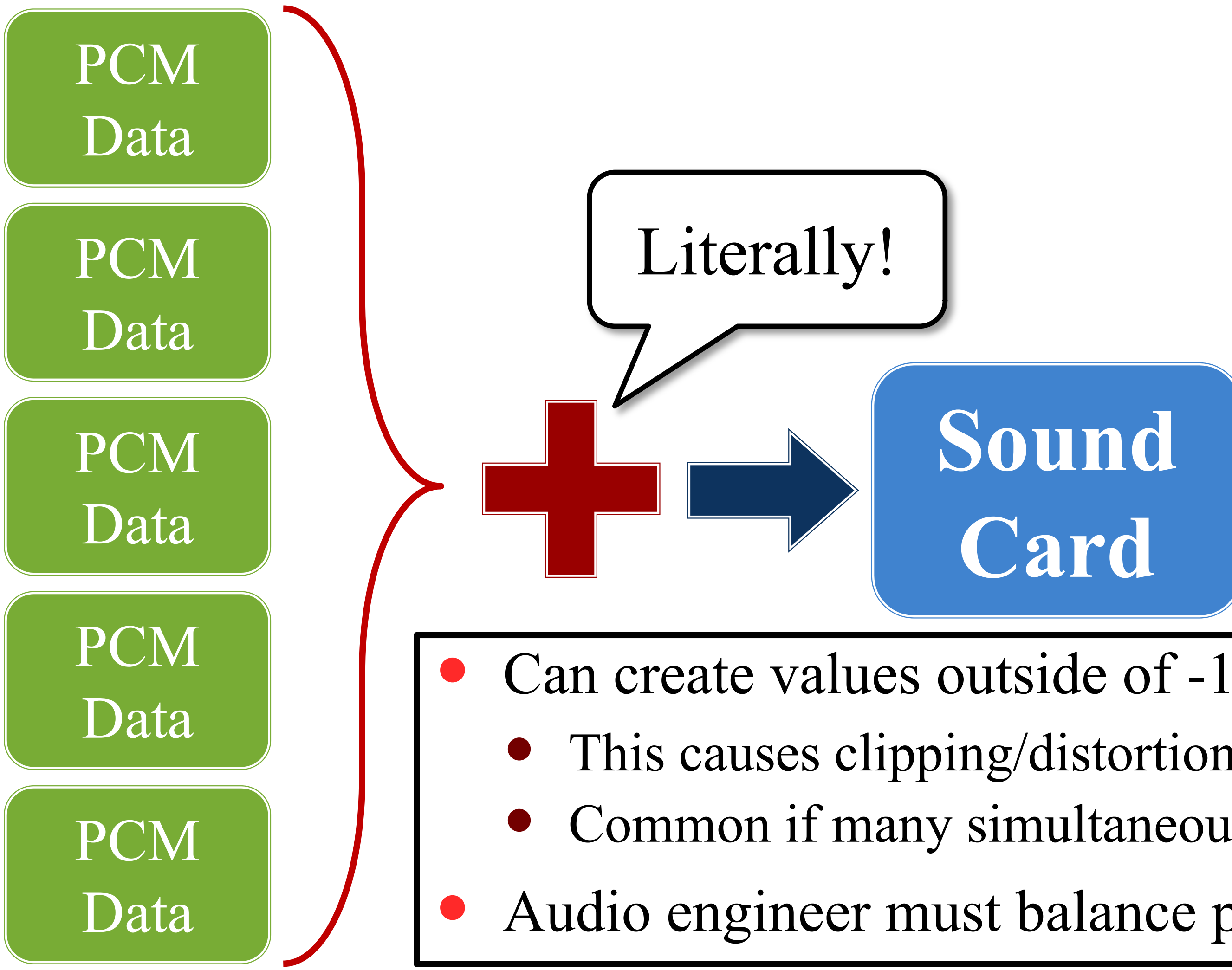


- **Sound**: Sound asset that is *preloaded* as full PCM
- **Music**: Sound asset that is *streamed* as PCM pages

Handling Multiple Sounds



Handling Multiple Sounds



- Can create values outside of -1 to 1
  - This causes clipping/distortion
  - Common if many simultaneous sounds
- Audio engineer must balance properly

Why is Mixing Hard?

- Playback may include **multiple sounds**
  - Sounds may play simultaneously (offset)
  - Simultaneous sounds may be same asset
  - Asset (source) vs. Instance (playback)
- Playback crosses **frame boundaries**
  - It may span multiple animation frames
  - Need to know when it stops playing
  - May need to stop (or pause) it early

We Want Something Simpler!

- Want ability to **play** and **track** sounds
  - Functions to load sound into card buffer
  - Functions to detect if sound has finished
- Want ability to **modify** active sounds
  - Functions for volume and pitch adjustment
  - Functions for stereo panning (e.g. left/right channels)
  - Functions to pause, resume, or loop sound
- Want ability to **mix** sounds together
  - Functions to add together sound data quickly
  - Background process for dynamic volume adjustment

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This is the purpose of a **sound engine**

Cross-Platform Sound Engines

- OpenAL
  - Created in 2000 by Loki Software for Linux
  - Was an attempt to make a sound standard
  - Loki went under; last stable release in 2005
  - Apple supported, but HARD deprecated in iOS 9
- FMOD/WWISE
  - Industry standard for game development
  - Mobile support is possible but not easy
  - Not free; but no cost for low-volume sales

Proprietary Sound Engines

- Apple AVFoundation
  - API to support modern sound processing
  - Mainly designed for music/audio creation apps
  - But very useful for games and playback apps
- OpenSL ES
  - Directed by Khronos Group (OpenGL)
  - Substantially less advanced than other APIs
  - Really only has support in Android space
  - Google is deprecating in 2022

Proprietary Sound Engines

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And many competing 3rd party solutions

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  - Directed by Khronos Group (OpenGL)
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What Does LibGDX Use?

- LibGDX support is actually OS specific
  - Recall the core/desktop package distinction
  - Because LibGDX supports mobile and computer
- Different platforms have different backends
  - All desktop platforms are built on **OpenAL**
  - The android backend uses `android.media`
- Needs an **abstraction** bringing all together
  - This is done with the `Audio` interface

The LibGDX Audio Interface

- LibGDX provides an audio **singleton**
  - One global object referencing audio device
  - Access via `GDX.audio` (static field of `GDX`)
  - Same principle as `System.out`
- Singleton implements the `Audio` interface
  - Use it to access `AudioDevice` for direct sound
  - Use it to allocate new `Sound`, `Music` instances
  - But do not use it for much sound manipulation

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  - But do not use it for much sound manipulation

Essentially a **factory** for other classes

The LibGDX Sound Classes

Sound

- Primary method is play()
  - Returns a long integer
  - Represents sound *instance*
  - `loop()` is a separate method
- Has no public constructor
  - Use `Audio.newSound(f)`
  - `Audio` can cache/preload
- Must dispose when done

Music

- Primary method is play()
  - This is a `void` method
  - Only allows **one instance**
  - `loop` is an attribute of music
- Has no public constructor
  - Use `Audio.newMusic(f)`
  - `Audio` can cache the file
- Must dispose when done

Playing a Sound

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Playing a Sound

- Playback may include **multiple sounds**
  - Sounds may play simultaneously (offset)
  - Simultaneous sounds may be same asset
  - [redacted]

Requires an understanding of OpenAL

- Pla[redacted]
  - It may span multiple animation frames
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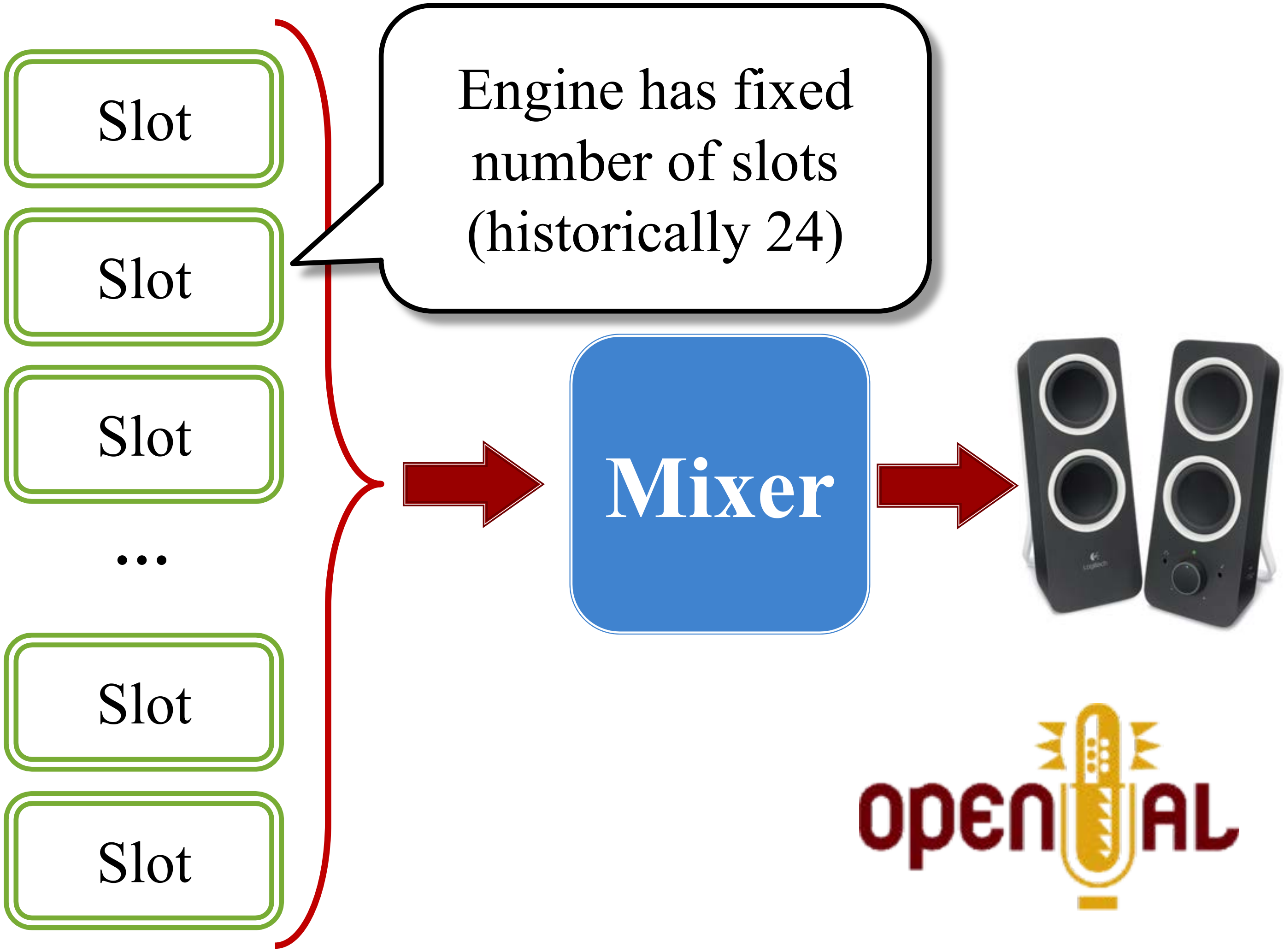
Classic Model: Playback Slots



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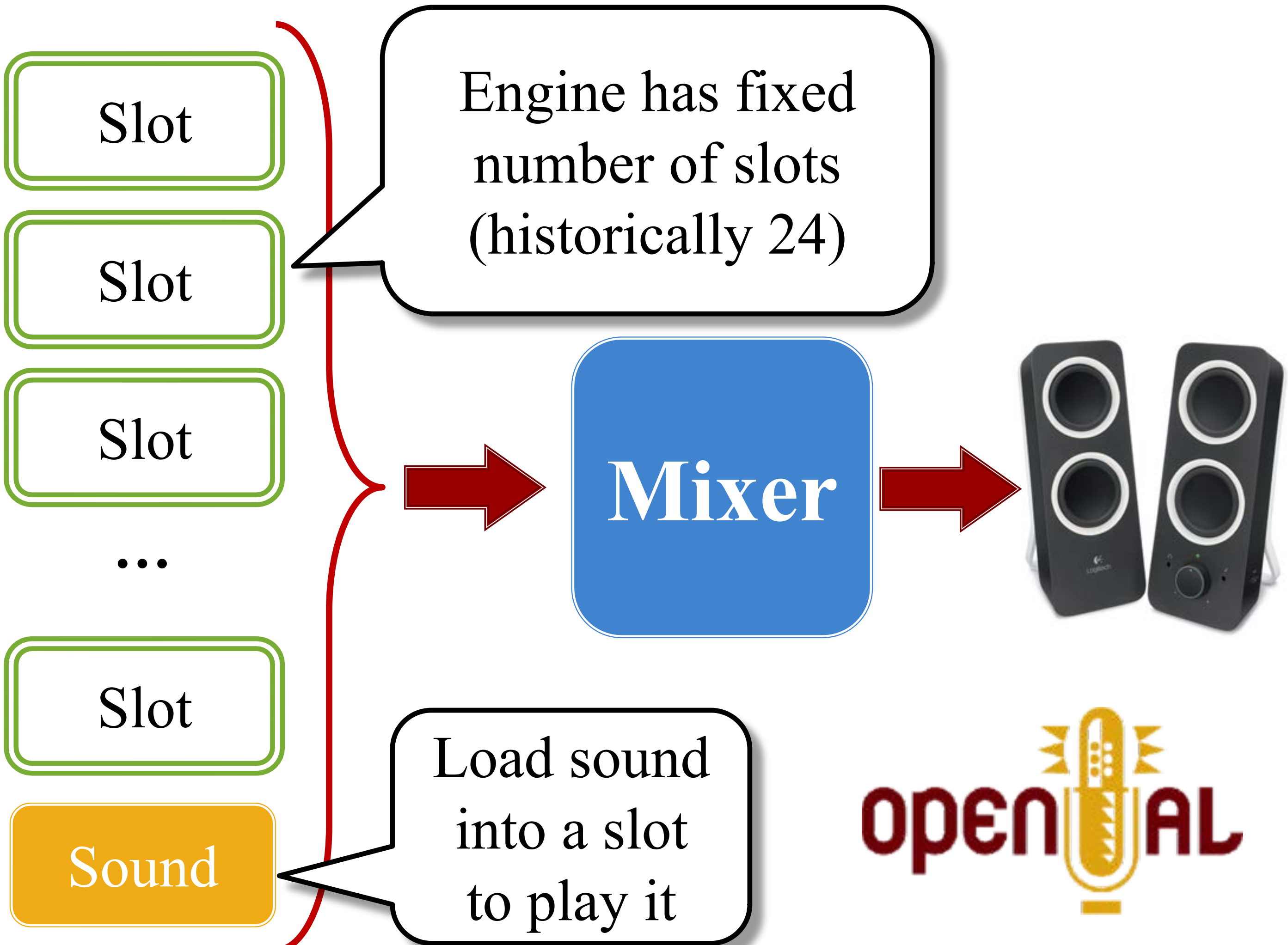

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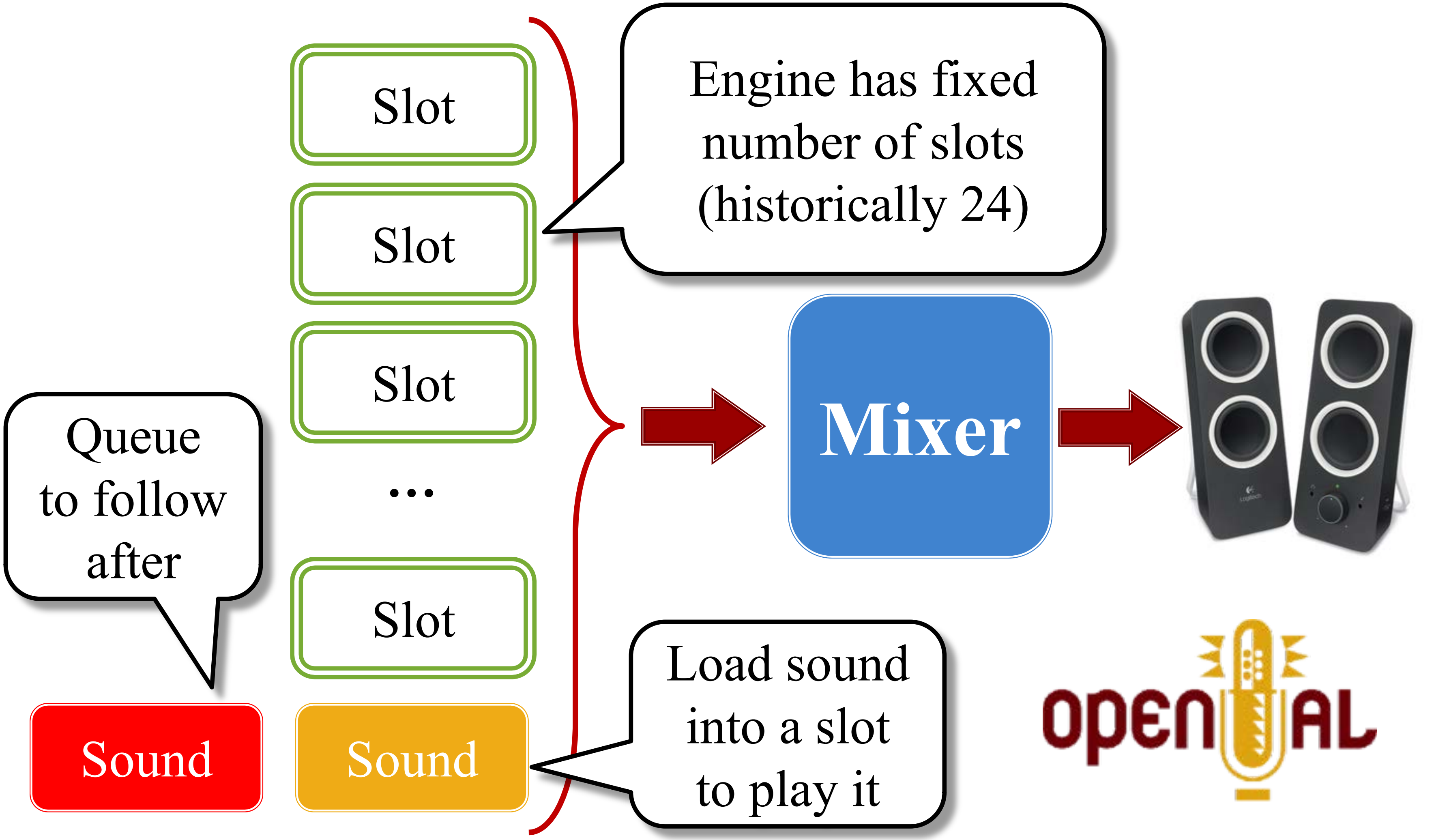
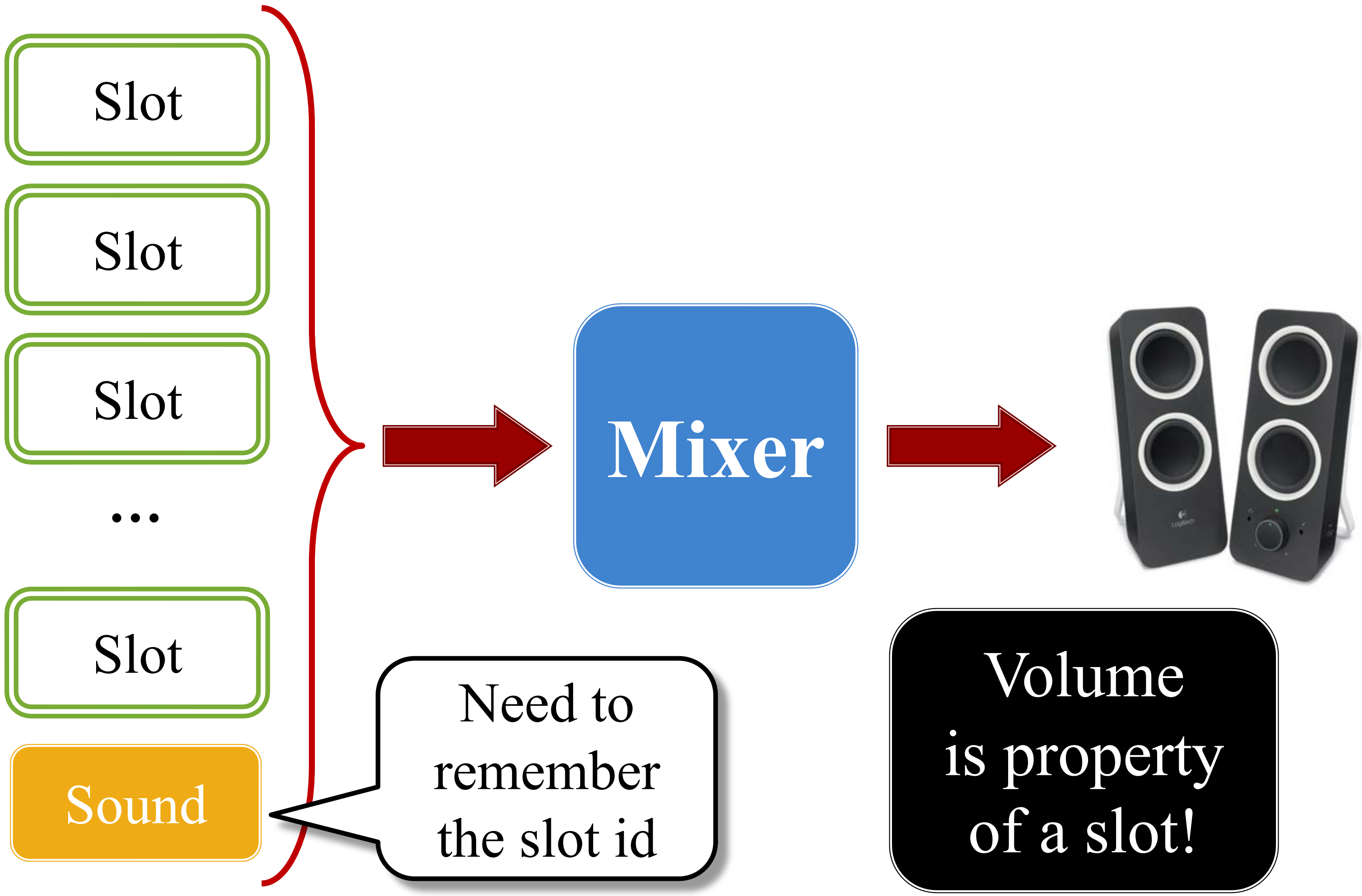


Playing a Sound with Slots

- **Request** a playback slot for your asset
  - If none is available, sound fails to play
  - Otherwise, it gives you an id for the slot
- **Load** asset into the slot (but might stream)
- **Play** the playback slot
  - Playing is a property of the slot, not asset
  - Playback slot has other properties, like volume
- **Release** the slot when the sound is done
  - This is usually done automatically

Application Design



The Sound API

- ```
  /**
    * @return channel id for sound playback
    *
    * If no channel is available, returns -1
    * @param volume  The sound volume
    * @param pitch   The pitch multiplier (>1 faster, <1 slower)
    * @param pan     The speaker pan (-1 full left, 1 full right)
    */
  public long play(float volume, float pitch, float pan);
  ```
- `public void stop(long audioID);`
- `public void resume(long audioID);`
- `public void setLooping(long audioID, boolean loop);`
- `Public void setVolume(long audioID, float volume);`
  ```

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    */
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  ```

Returns available channel id

- `public void stop(long audioID);`
- `public void resume(long audioID);`
- `public void setLooping(long audioID, boolean loop);`
- `Public void setVolume(long audioID, float volume);`

Need to remember channel id
  ```

Why This is Undesirable

- Tightly couples architecture to sound engine
  - All controllers need to know this playback slot id
  - Playback must communicate id to all controllers
- Instances usually have a *semantic meaning*
  - **Example**: Torpedo #3, Ship/crate collision
  - Meaning is independent of the slot assigned
  - Would prefer to represent them by this meaning
- **Solution**: Refer to instances by *keys*

Idea: SoundController Class

- A `SoundController` is essentially a hashmap
  - Map strings (keys) to integers (slot ids)
  - Only stores a key when instance is playing
- This class needs to be a **singleton**
  - So we can access this anywhere at all time
  - **Demo:** See the class provided with this lecture
- To work, the map must be **up-to-date** at all times
  - We use this controller to play the sounds
  - And it must be notified when a sound is done

Stopping Sounds

- Would like to know when a sound is finished
  - To free up the slot (if not automatic)
  - To stop any associated animation
  - To start a follow-up sound
- Two main approaches
  - **Polling**: Call an isPlaying() method
  - **Callback**: Pass a function when play
- Default LibGDX cannot do *either* of these

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Cannot do in android.media

- Default LibGDX cannot do *either* of these

Solution: AudioEngine

- You are all making **desktop games**
  - This means you are always using OpenAL
  - Just need a way to expose OpenAL features
  - This is the purpose of GDIAC audio backend
- Basic interface is `AudioEngine`
  - Upcast `GDX.audio` to this interface
  - Now have access to SoundBuffer, MusicBuffer
  - These classes give extra features you need
- **Note:** `AssetDirectory` handles this automatically

The **GDIAC** Sound Classes

SoundBuffer	MusicBuffer

SoundBuffer

- Works just like `Sound`
  - Primary method is play()
  - Returns a long integer
- But has **playback control**
  - Can poll if still playing
  - Can add listener to monitor
- Exposes **OpenAL features**
  - Elapsed playback time
  - Panning between speakers
  - Sound pitch control

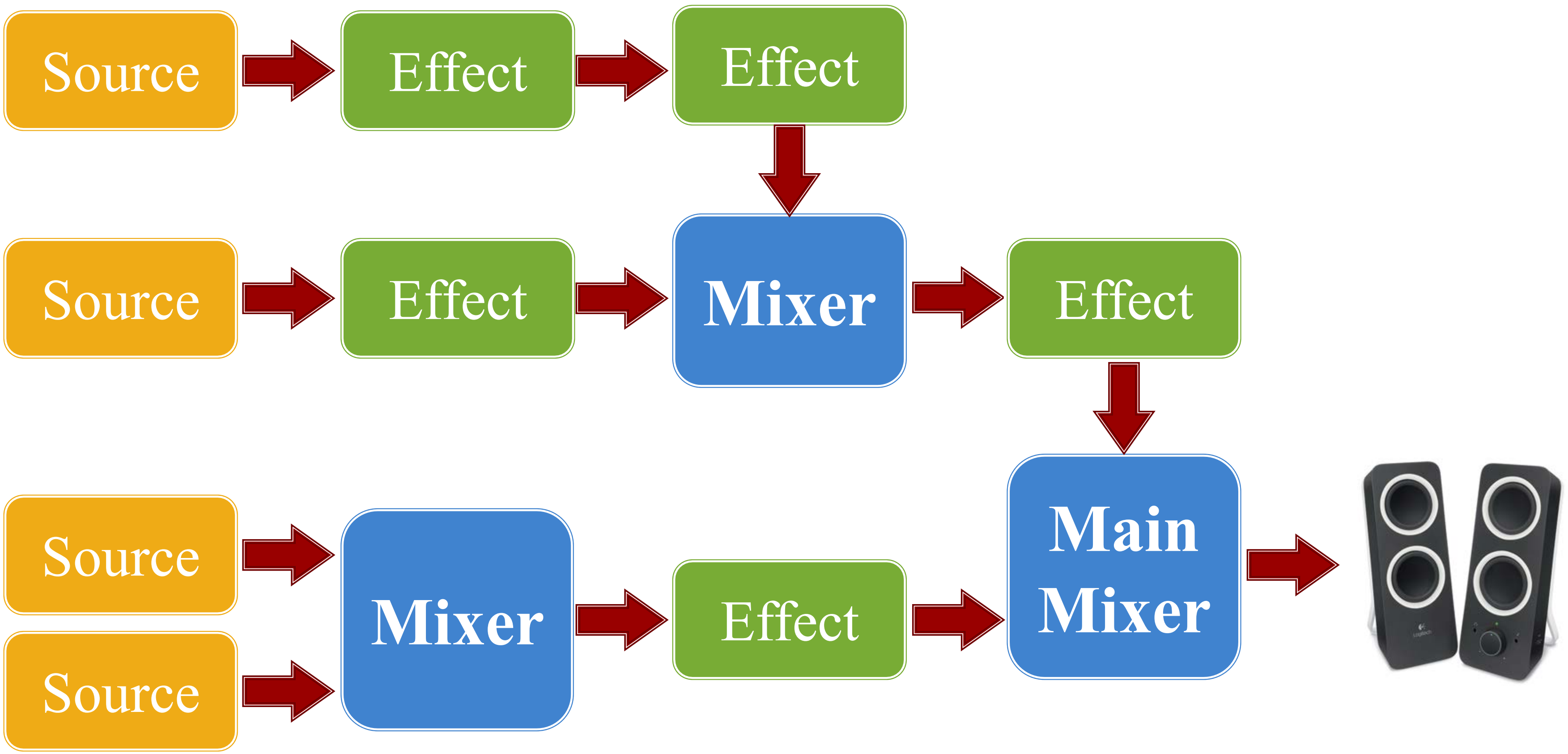
MusicBuffer

- Works just like `Music`
  - Primary method is play()
  - This is a `void` method
- But has a **playback queue**
  - Can add AudioSource to it
  - Provides gapless playback
- Methods **manage the queue**
  - Add or remove music
  - Swap out music at position
  - Skip over current music

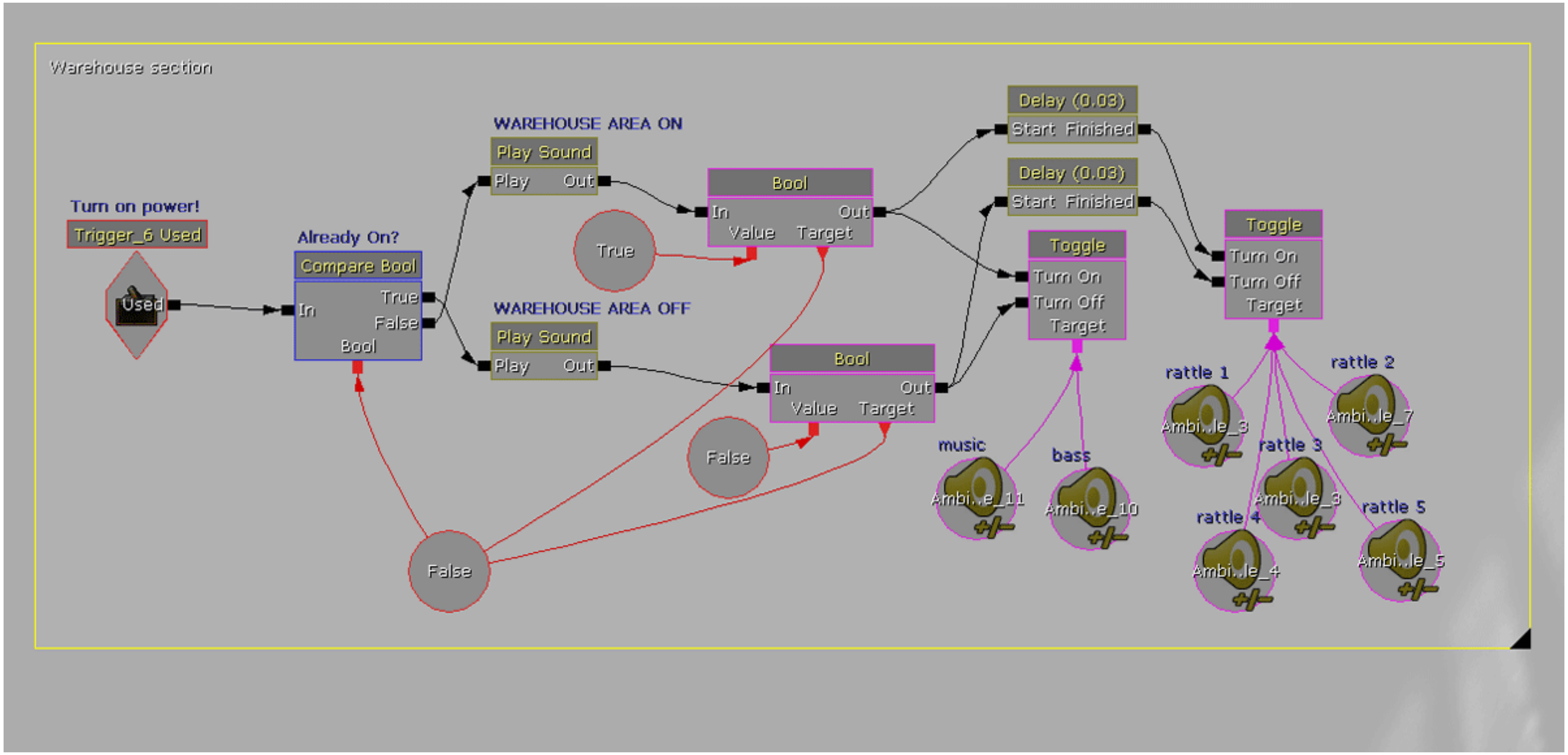
Problem with the Slots Model

- All controls are embedded in the slot
  - **Example**: Volume, looping, play position
  - Restricted to a *predetermined* set of controls
- Modern games want *custom sound-processing*
  - User defined sound filters (low pass, reverb)
  - Advanced equalizer support
  - Support for surround and 3D sound
  - Procedural sound generation

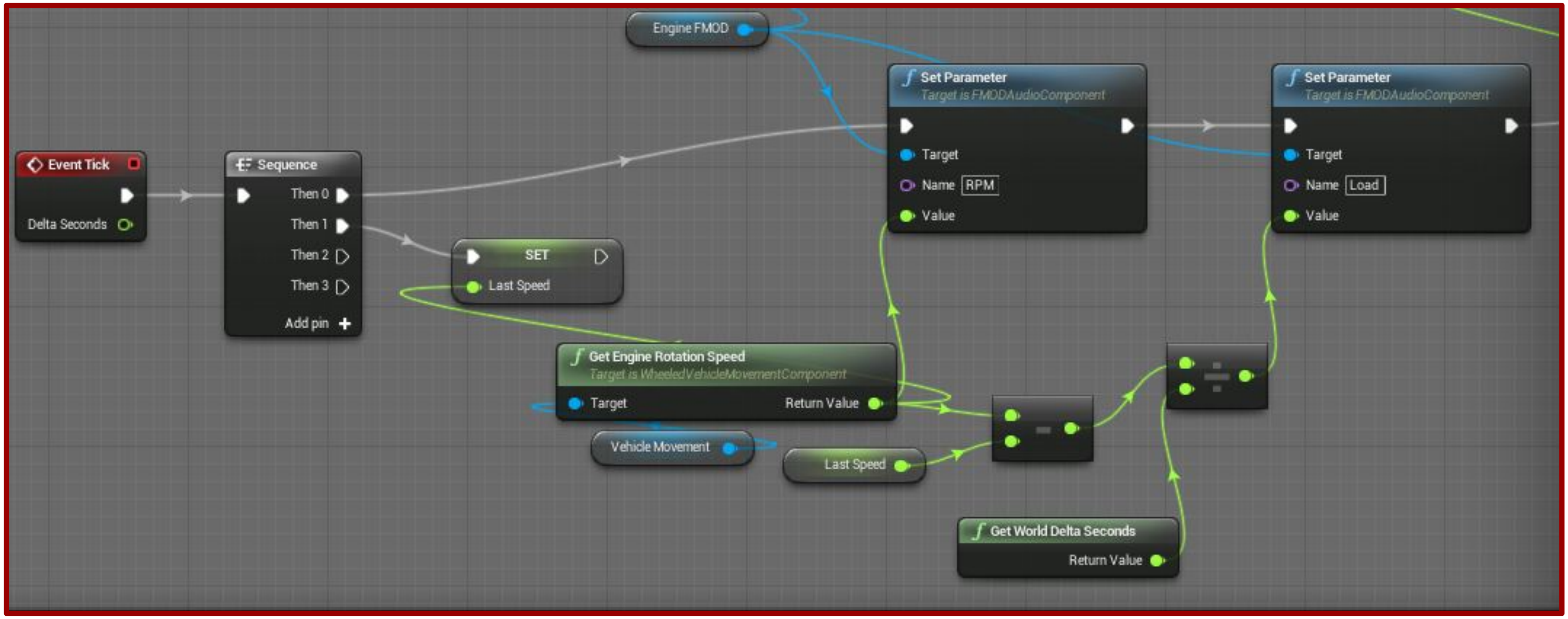
DSP Processing: The Mixer DAG



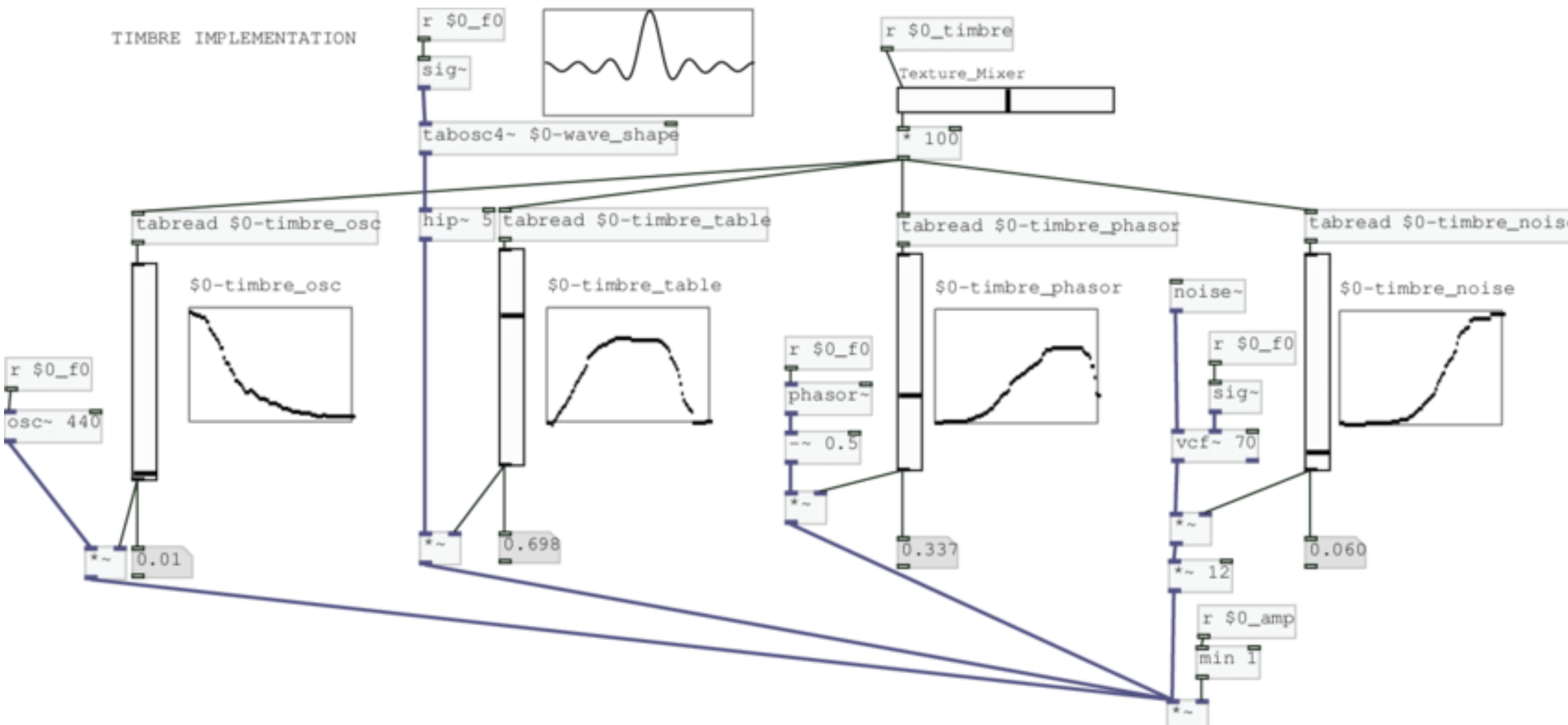
Example: UDK Kismet



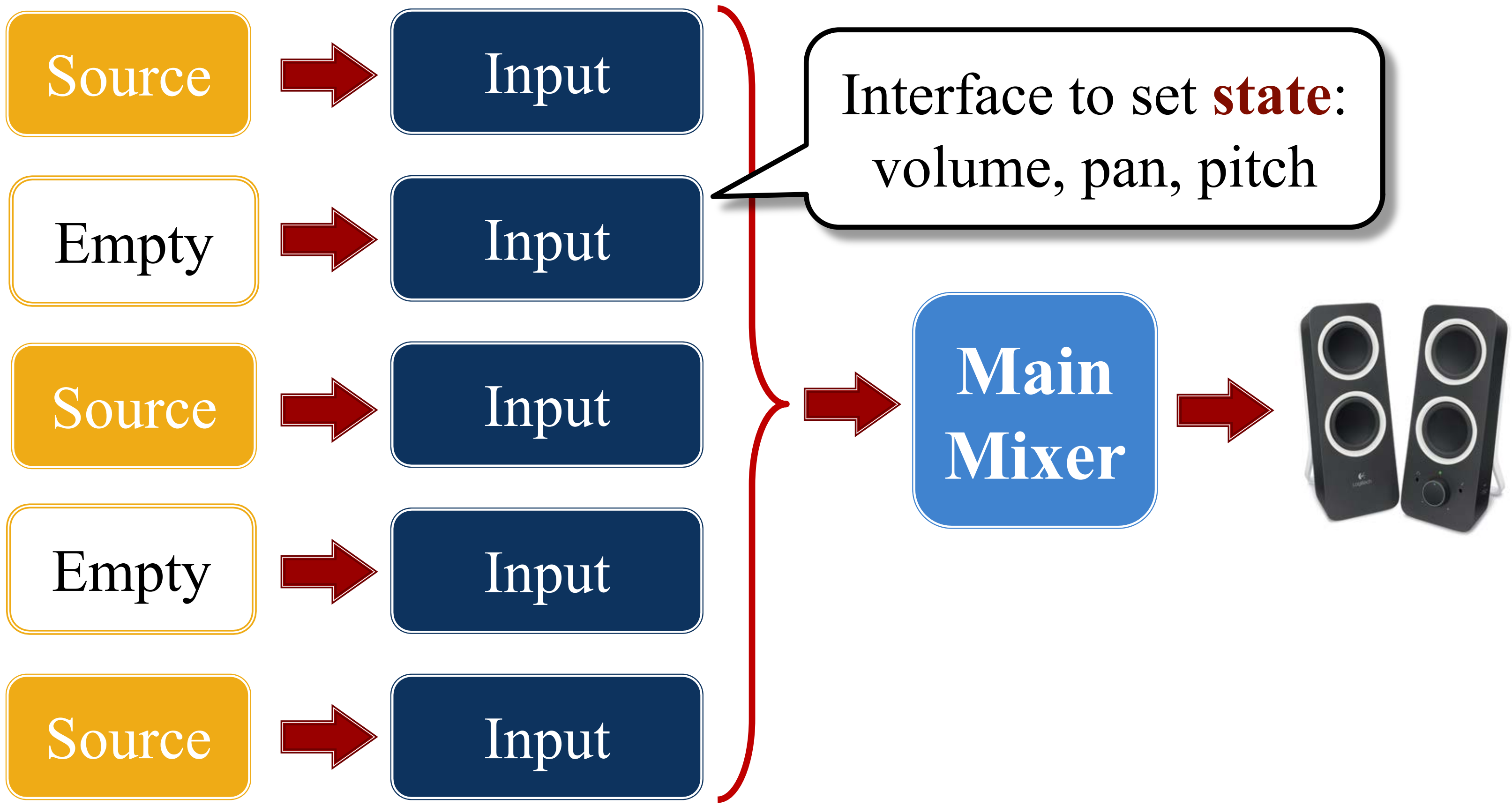
Example: FMOD



Example: Pure Data

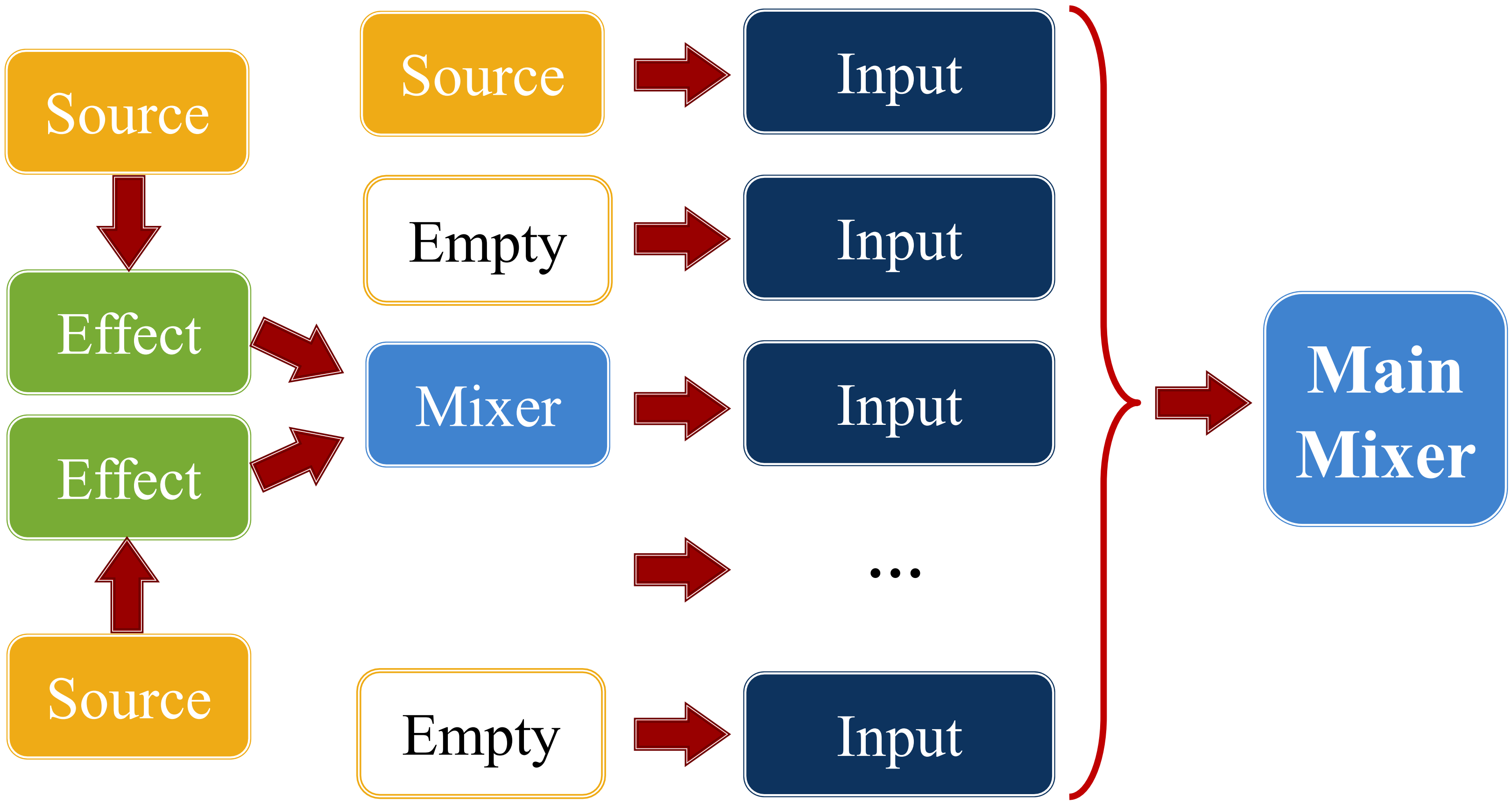


The Slot Model is a Special Case



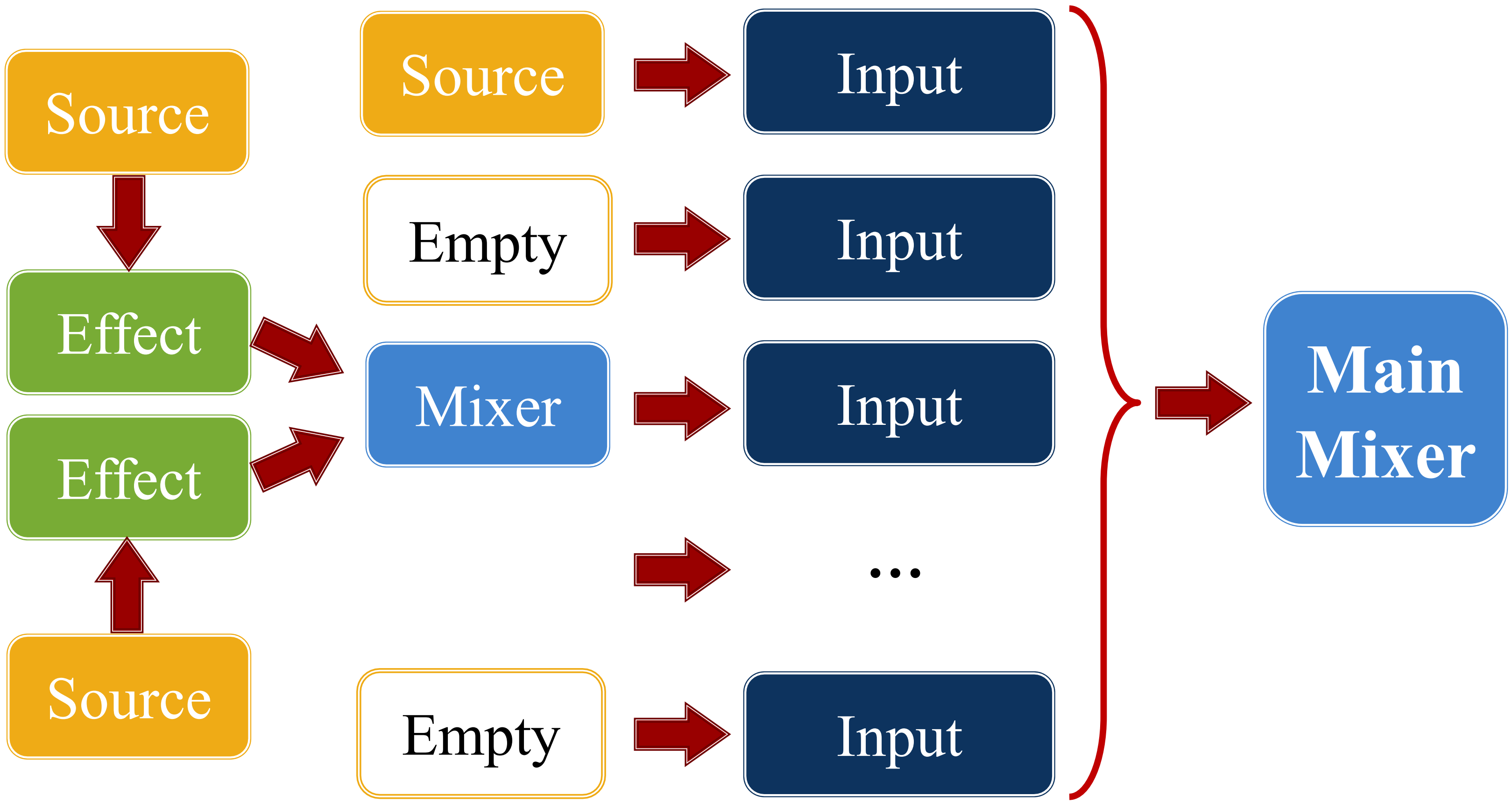
Calling play() assigns an input slot behind the scenes

The Slot Model is a Special Case



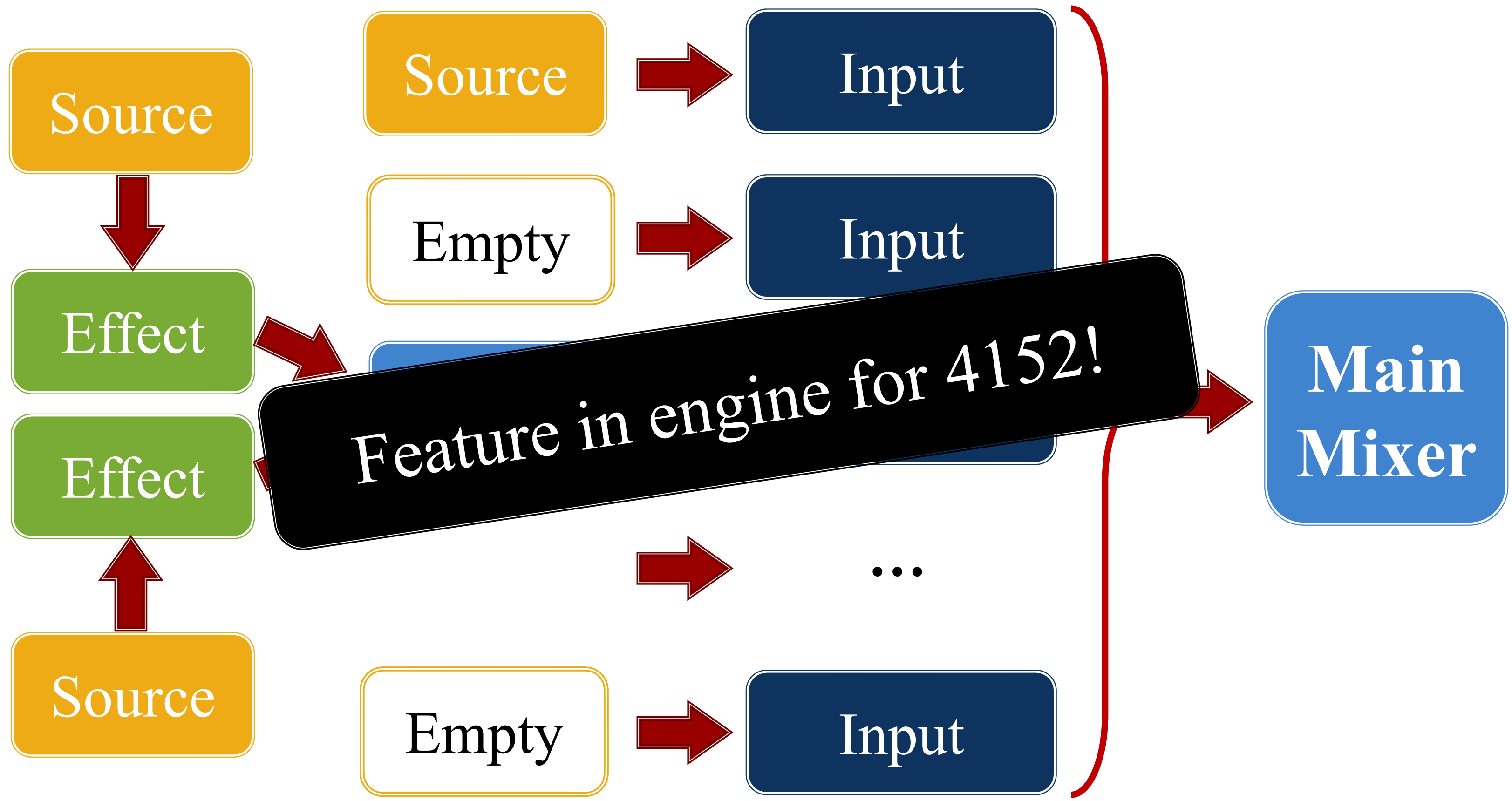
Theoretically input should accept any **audio subgraph**

The Slot Model is a Special Case



Even **OpenAL** cannot do this.

The Slot Model is a Special Case



Even **OpenAL** cannot do this.

Summary

- Audio design is about creating soundscapes
  - Music, sound effects, and dialogue
  - Combining sounds requires a sound engine
- Cross-platform support is a problem
  - Licensing issues prevent a cross-platform format
  - Very little standardization in sound APIs
- Best engines use digital signal processing (DSP)
  - Mixer graph is a DAG supporting sound effects
  - Unfortunately, we cannot do this in LibGDX