the gamedesigninitiative at cornell university

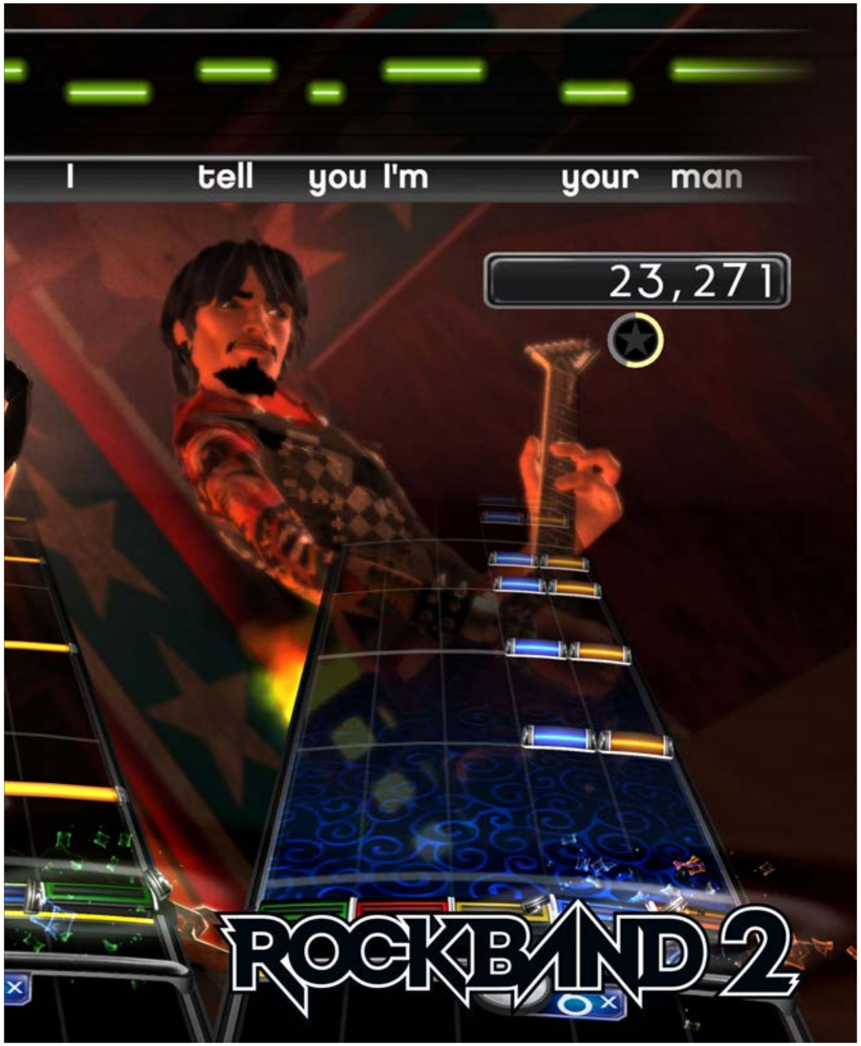
Lecture 16

Game Audio

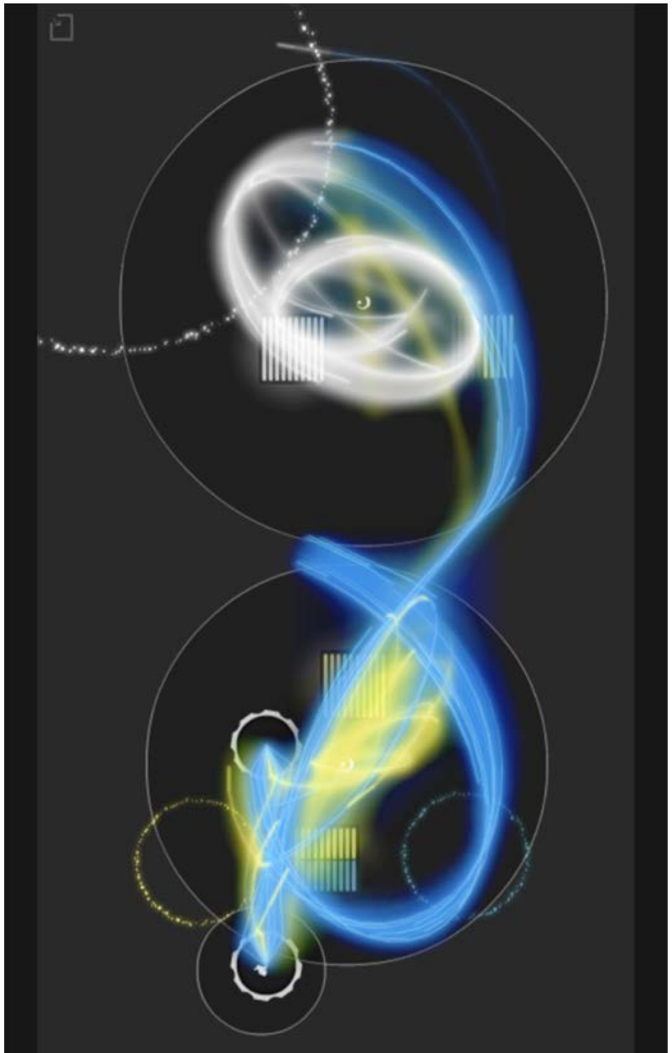
The Role of Audio in Games

Engagement

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



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
- Other reasons?

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

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

History of Sound in Games

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- 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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- Starts w/ MIDI
- 5th generation (Playstation)
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- Sample selection
- Volume
- Pitch
- Stereo pan

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

The Technical Challenges

- Sound **formats** are not (really) cross-platform
  - It is not as easy as choosing MP3
  - Android, iOS favor different formats
- Sound playback **APIs** are not standardized
  - SDL (& CUGL) is a layer 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
  - .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

Data Formats and Platforms

Format	Description	iOS	Android
MP3	You know what this is	Yes	Yes
(HE-)AAC	A lossy codec, Apple's MP3 alternative	Yes	Yes
Linear PCM	Completely uncompressed sound	Yes	Yes
MIDI	**NOT SOUND**; Data for an instrument	Yes	Yes
Vorbis	Xiph.org's alternative to MP3	**No**	Yes
ALAC	Apple's lossless codec (but compressed)	Yes	**No**
FLAC	Xiph.org's alternative lossless codec	**No**	Yes
iLBC	Internet low bit-rate codec (VOIP)	Yes	**No**
IMA4	Super compression for 16 bit audio	Yes	**No**
μ-law	Like PCM, but optimized for speech	Yes	**No**

The Associated File Formats

Format	File Types
MP3	.mp3
(HE-)AAC	.aac, .mp4, .m4a
Linear PCM	.wav
MIDI	.mid

- Any other file format is **not cross-platform**
- Apple/iOS is pushing the .caf file
  - Stands for Core Audio Format
  - Supports MP3, (HE-)AAC, PCM, ALAC, etc…
  - But not cross-platform

The Associated File Formats

Format	File Types
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Limited support due to patent issues

- Any other file format is **not cross-platform**
- Apple/iOS is pushing the .caf file
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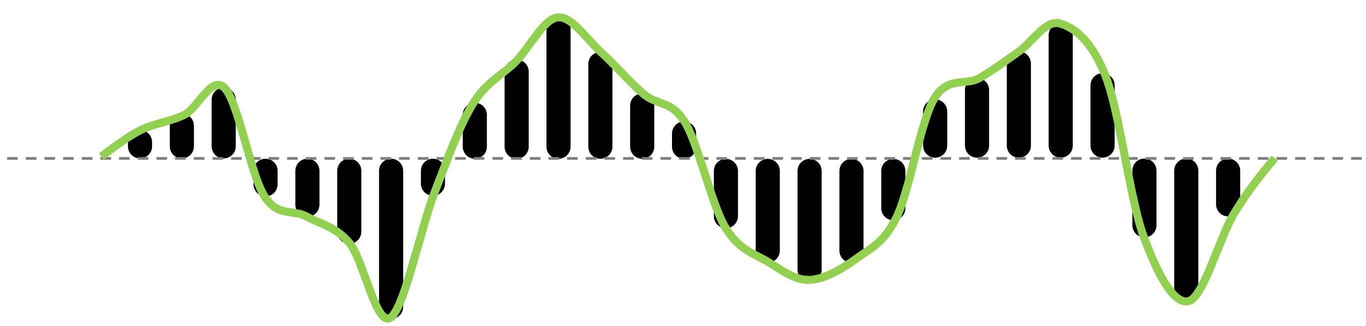
Linear PCM Format

- 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

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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

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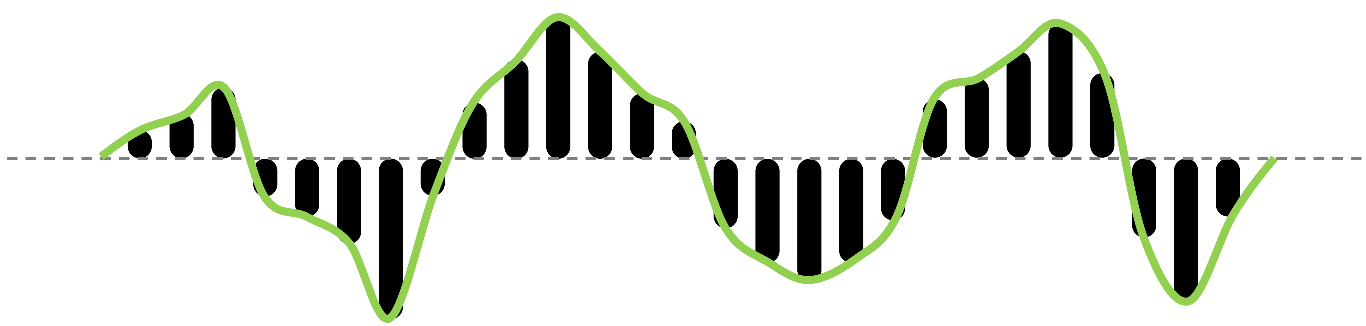
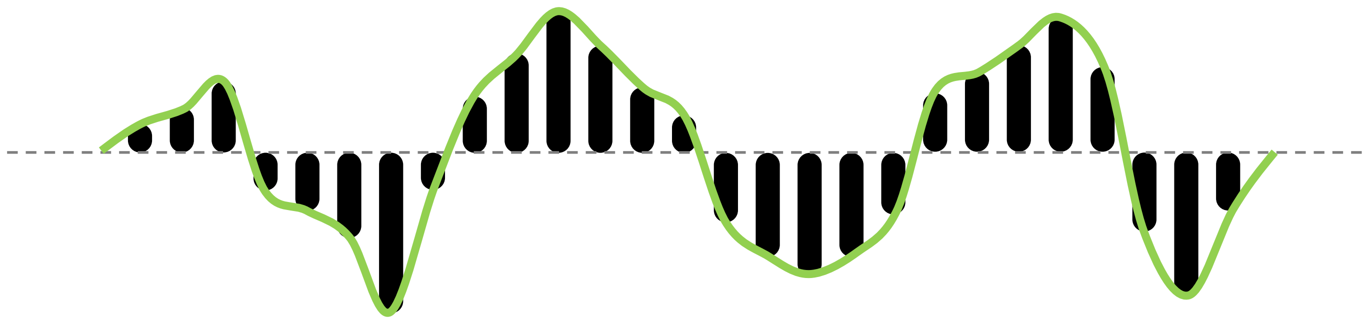


- Sometimes encoded as shorts or bytes MIN to MAX

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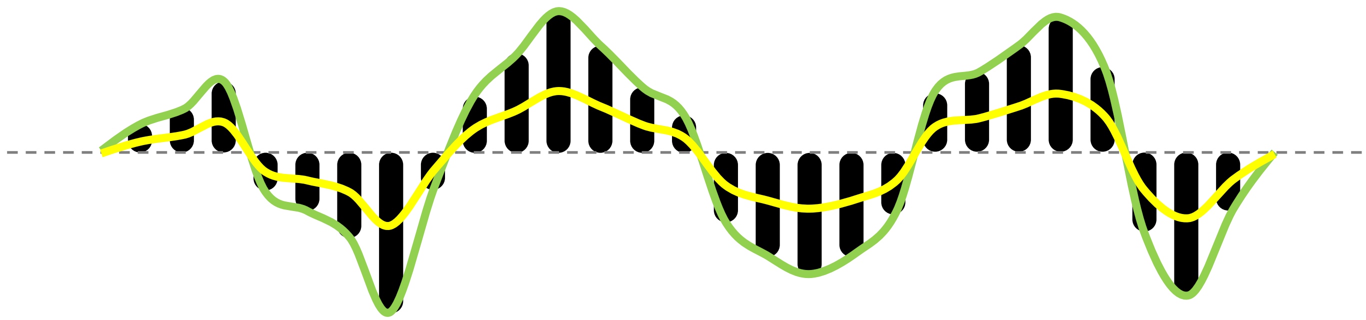
0.5	0.2	-0.1	0.3	-0.5	0.0	-0.2	-0.2	0.0	-0.6	0.2	-0.3	0.4	0.0



- Magnitude of the amplitude is the volume
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  - Multiply by number 0 to 1 to change global volume

Linear PCM Format

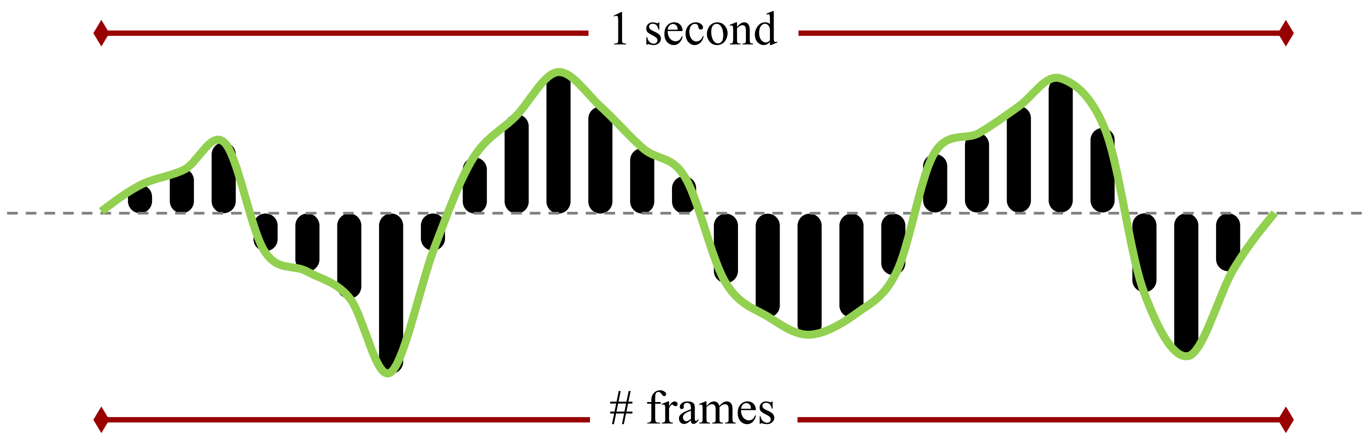
- Samples are organized into (interleaved) **channels**

$$\boxed{0.5}\,\boxed{0.2}\,\boxed{-0.1}\,\boxed{0.3}\,\underbrace{\boxed{-0.5}\,\boxed{0.0}}_{\text{frame}}\,\boxed{-0.2}\,\boxed{-0.2}\,\boxed{0.0}\,\boxed{-0.6}\,\boxed{0.2}\,\boxed{-0.3}\,\boxed{0.4}\,\boxed{0.0}$$

- 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

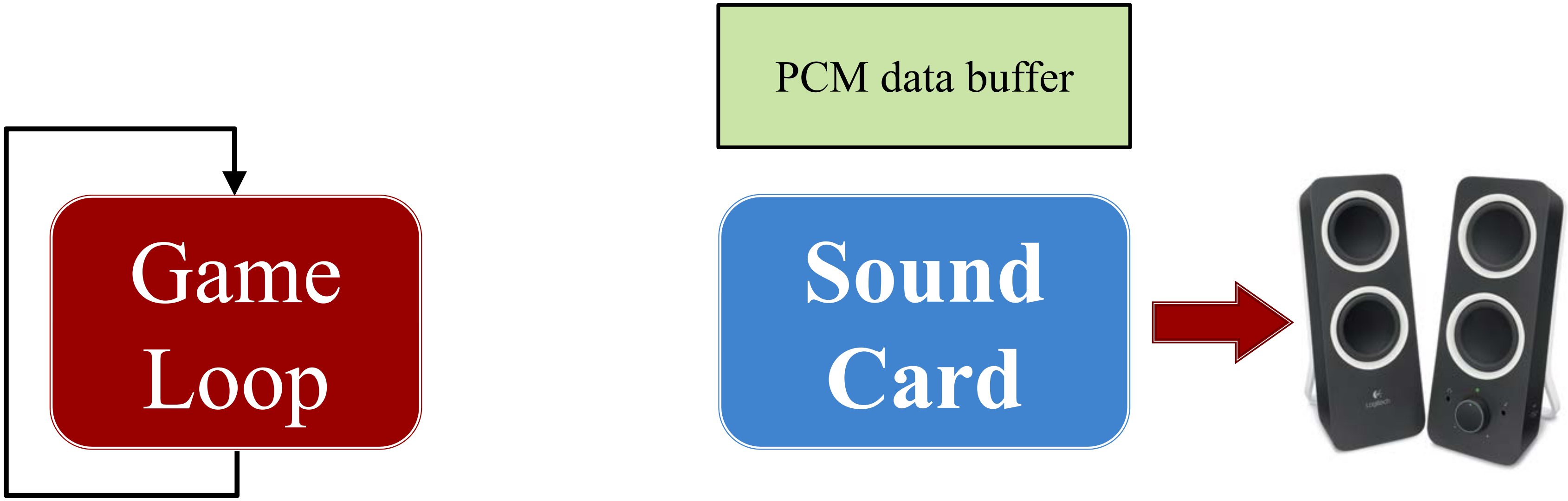
Linear PCM Format

- The sample rate is frames per second

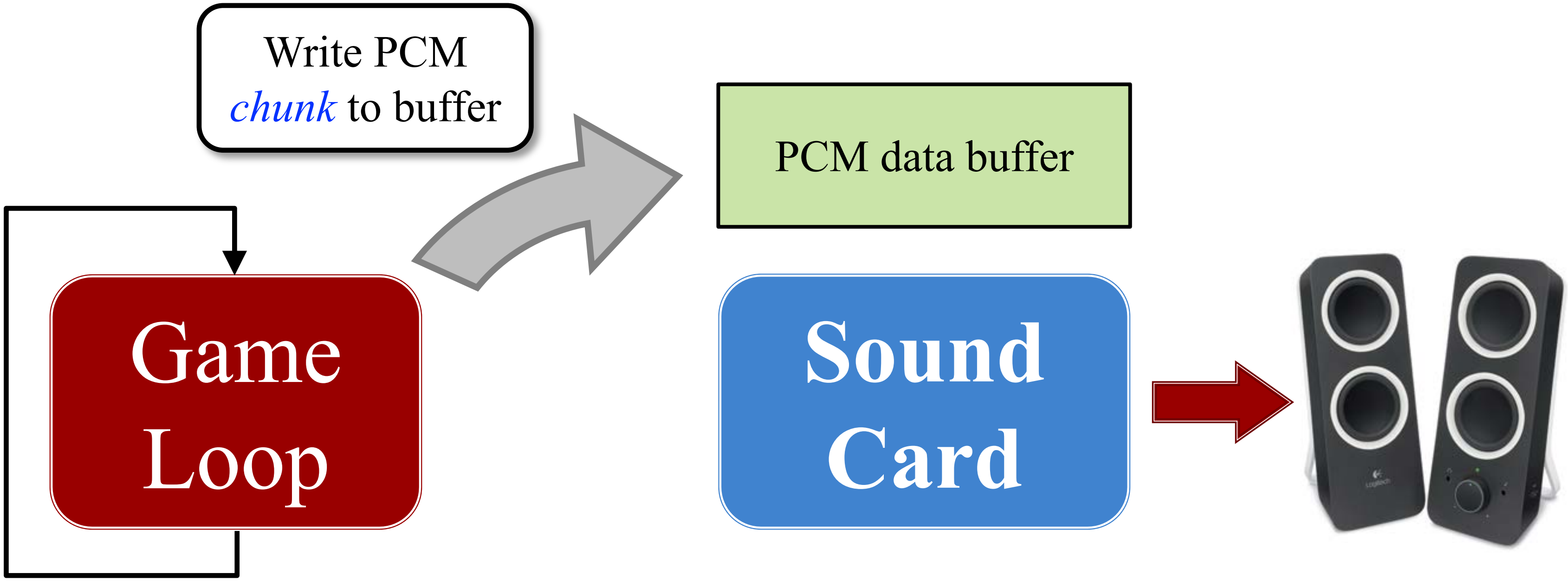


- **Example**: 0.5 seconds of stereo at 44.1 kHZ
  - 0.5 s * 44100 f/s = 22050 frames
  - 2 samples/frame * 22050 frames = 44100 samples
  - 4 bytes/sample * 44100 samples = 176.4 kBytes
- 1 minute of stereo CD sound is 21 MB!

Playing Sound Directly

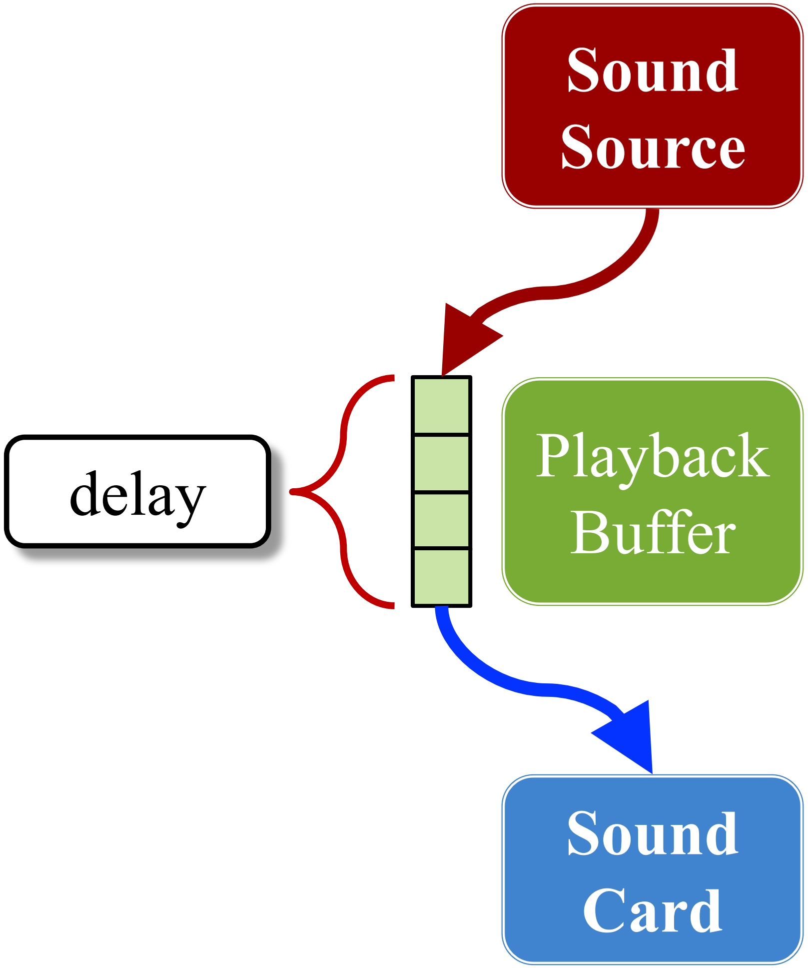


Playing Sound Directly

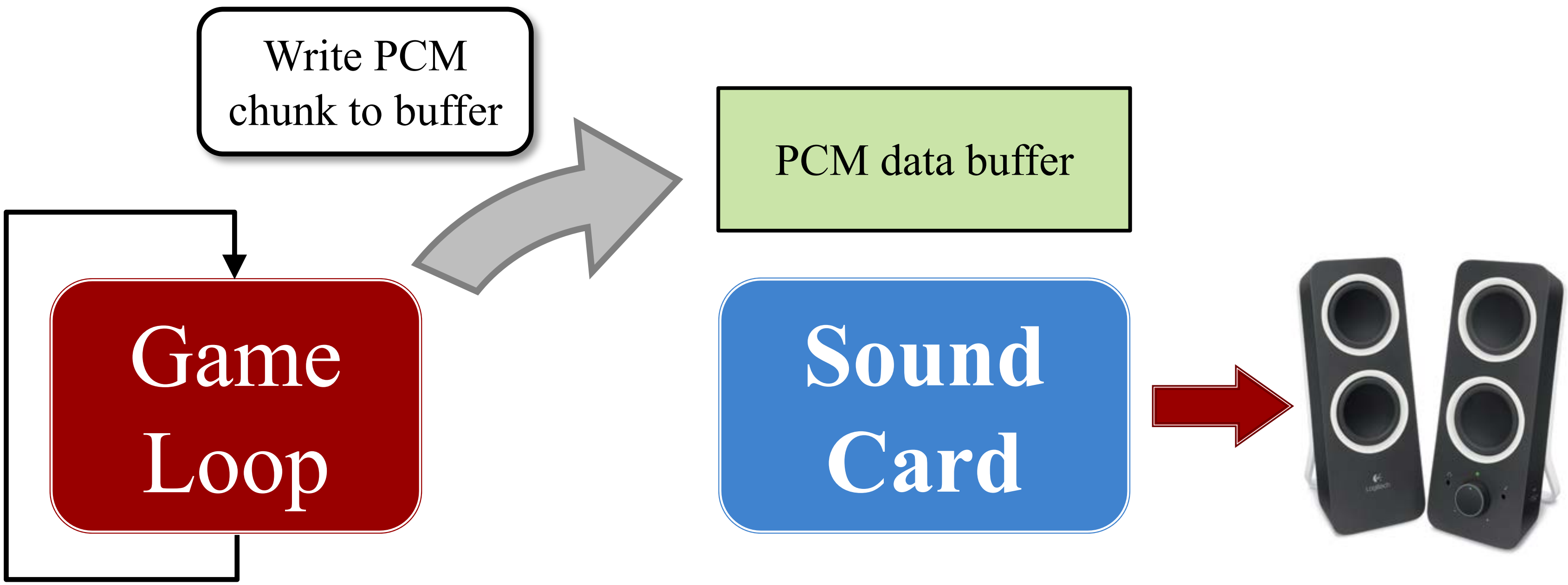


The Latency Problem

- 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*



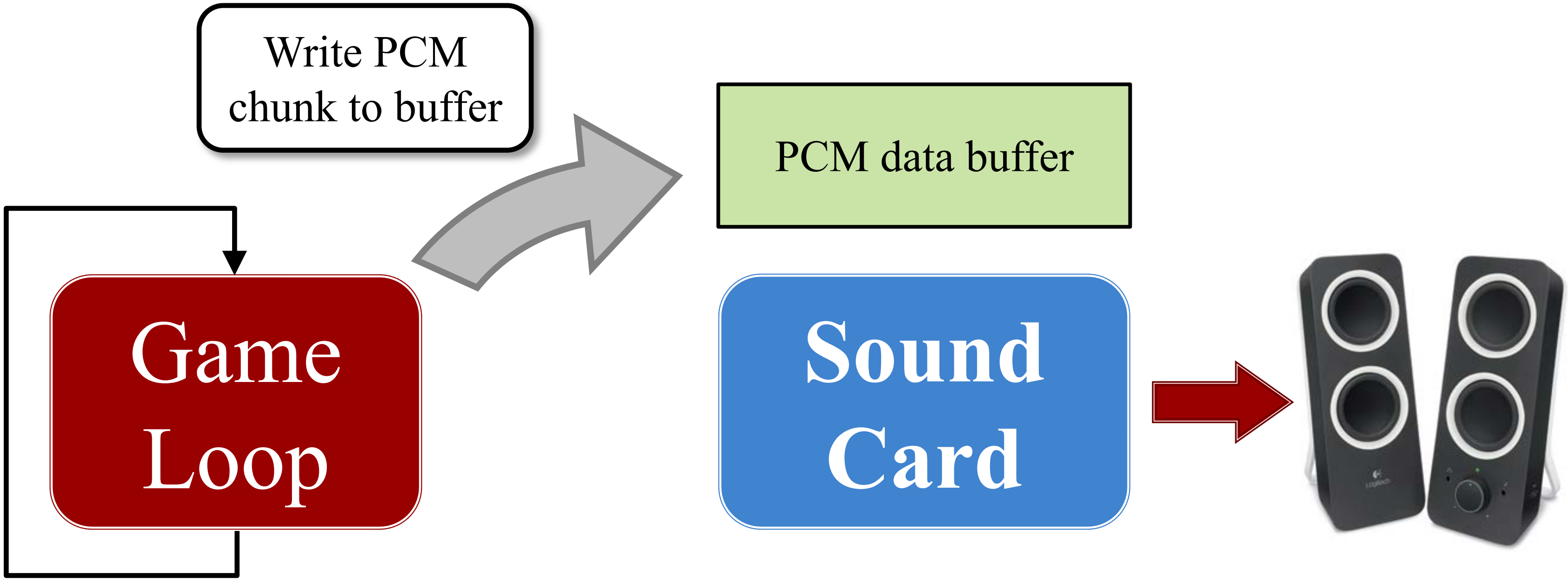
Playing Sound Directly



Choice of buffer size is important!

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

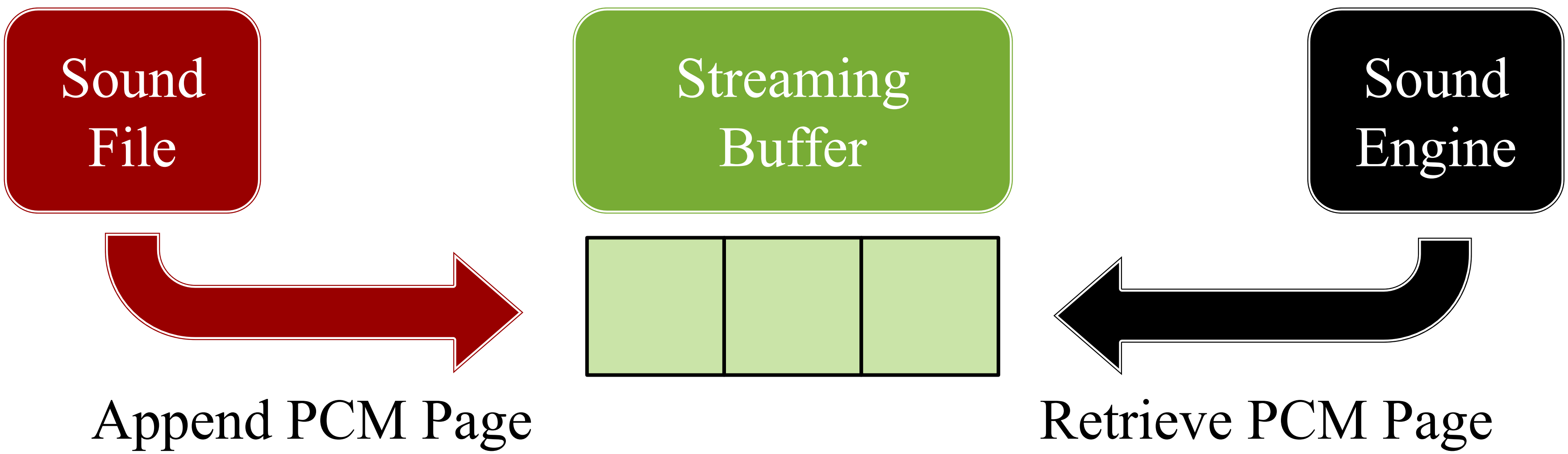
Playing Sound Directly



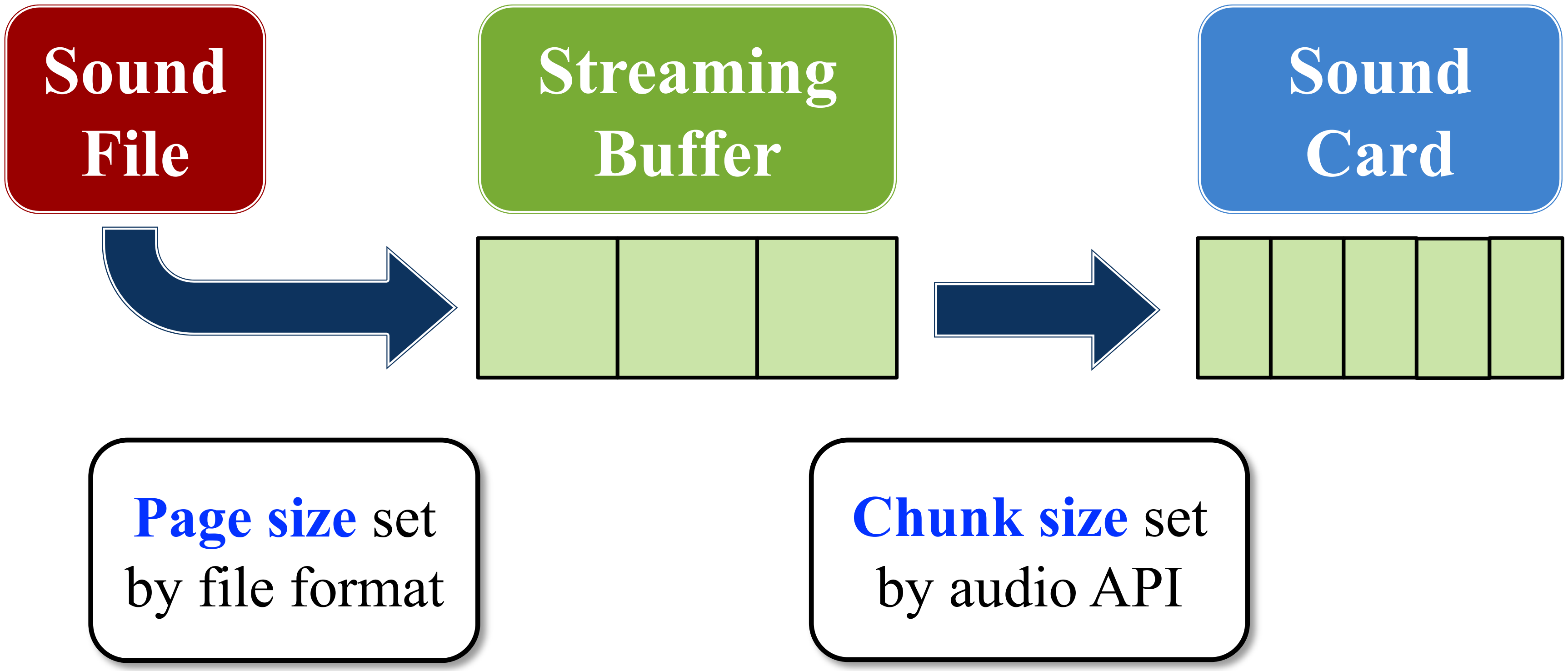
- Windows: 528 bytes (even if you ask for larger)
- MacOS, iOS: 512-1024 bytes (hardware varies)
- Android: 2048-4096 bytes (hardware varies)

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
- This is how OGG support was added to CUGL

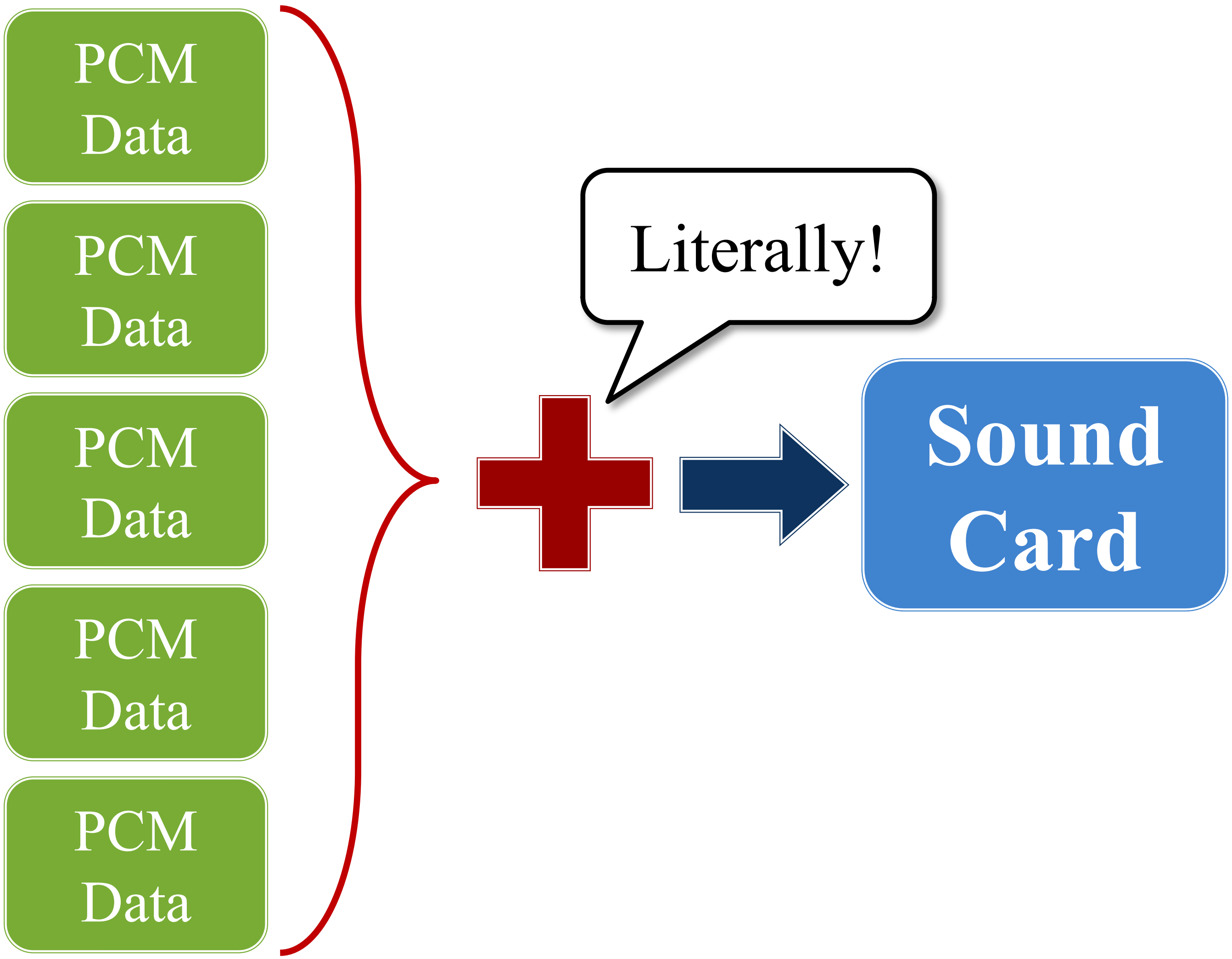


How Streaming Works

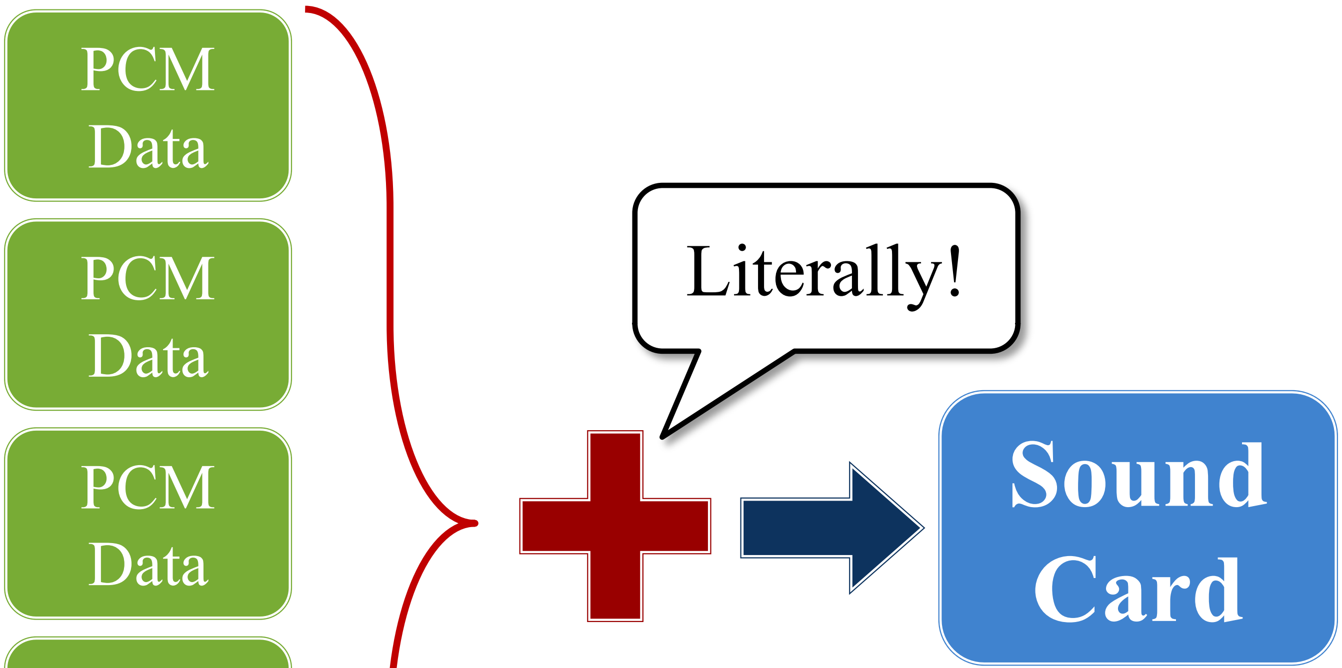


- **Sound**: Sound asset that is *preloaded* as full PCM
- **Music**: Sound asset that is *streamed* as PCM pages

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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  - Fu…
  - …nnels)
  - …pause, resume, or loop sound

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

- Want ability to **mix** sounds together
  - Functions to add together sound data quickly
  - Background process for dynamic volume adjustment

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

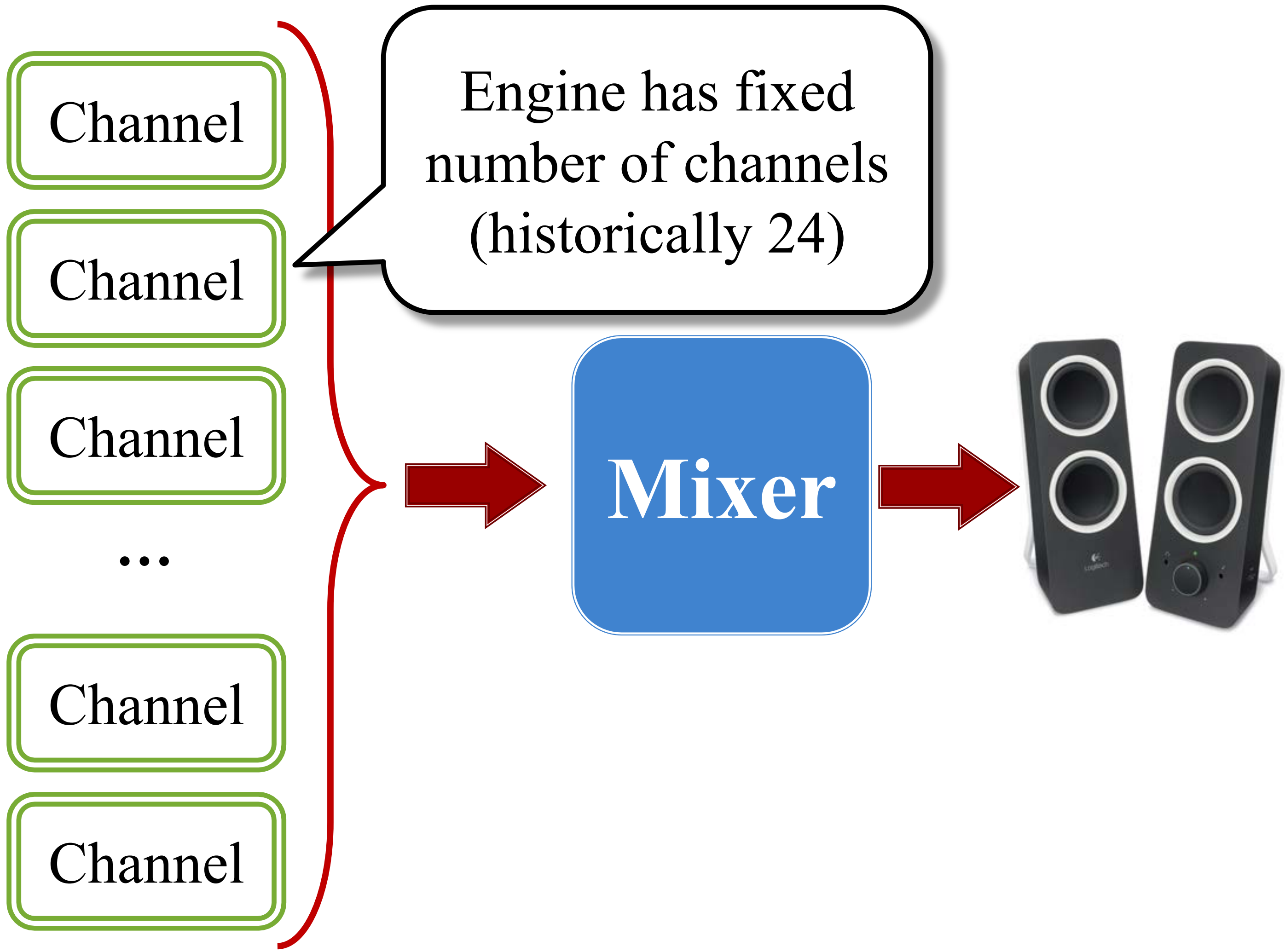
What about SDL?

- CUGL is on top of SDL
  - SDL has its own audio API
  - Works on all platforms
- But it is a **extremely** low-level API
  - Fill the buffer with linear PCM data
  - Either pull (callback) or push (queue)
  - No support for non-WAV audio formats
  - No support for mixing, pausing, or anything

Solution: CUGL Audio Classes

- **AudioChannels:** Simple audio interface
  - Essentially uses the OpenAL model
  - Very easy to use and understand
  - Limited to pre-recorded sound files
- **AudioManager:** Advanced audio interface
  - Direct access to the *audio filter graph*
  - Requires a lot of audio knowledge to use
  - Can support complex audio assets (patches)

Classic Model: **Channels**



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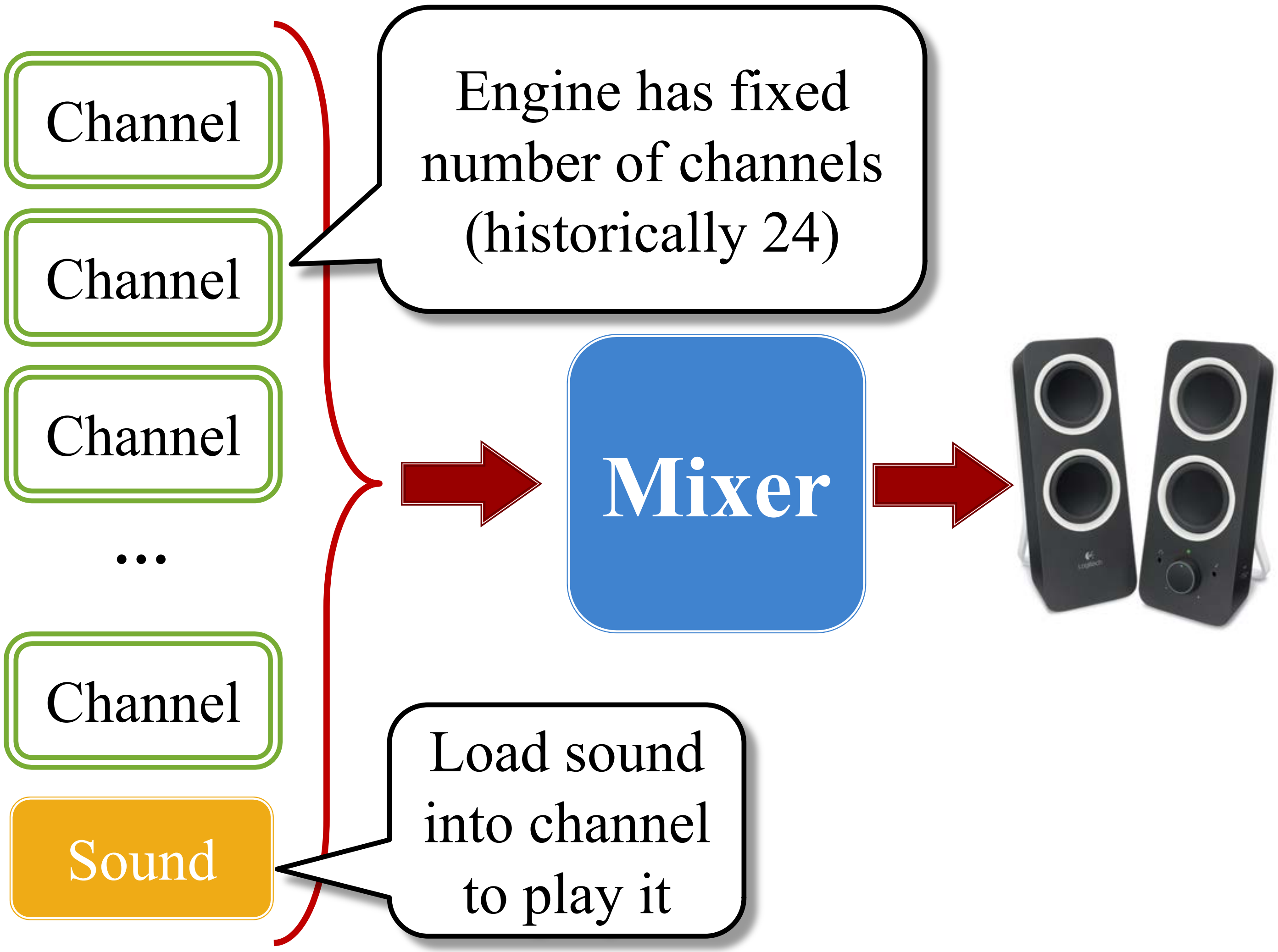

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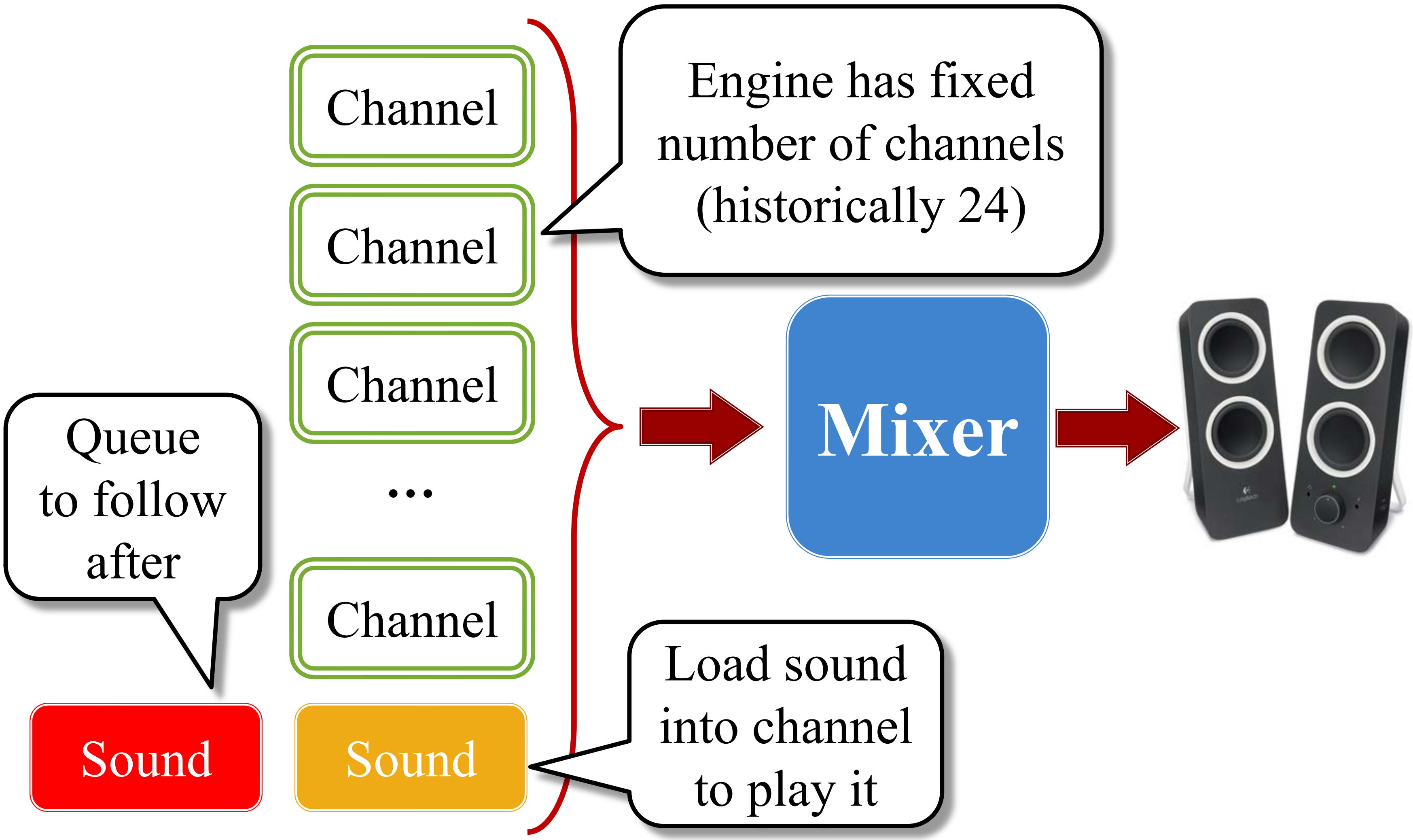


Classic Model: **Channels**

Channel

Channel

Channel

…

Channel

Engine has fixed number of channels (historically 24)

Mixer

Queue to follow after

Sound

Sound

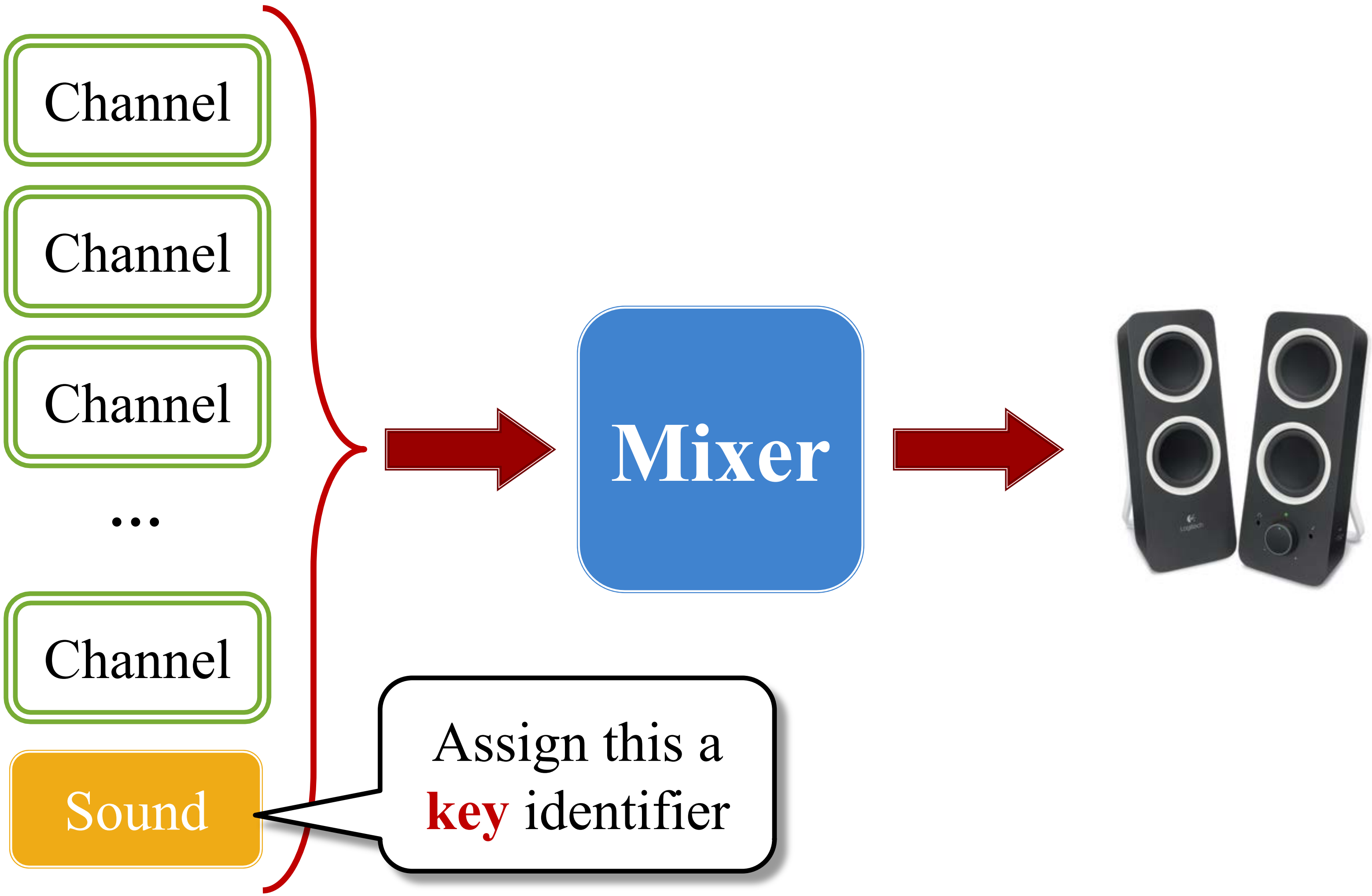
Load sound into channel to play it

openAL

Playing a Sound with Channels

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

Application Design



Stopping Sounds

- Would like to know when a sound is finished
  - To free up the channel (if not automatic)
  - To stop any associated animation
  - To start a follow-up sound
- Two main approaches
  - **Polling**: Call an `isPlaying()` method/function
  - **Callback**: Pass a listener to the engine
- `AudioChannels` only allows both approaches

The AudioChannels API

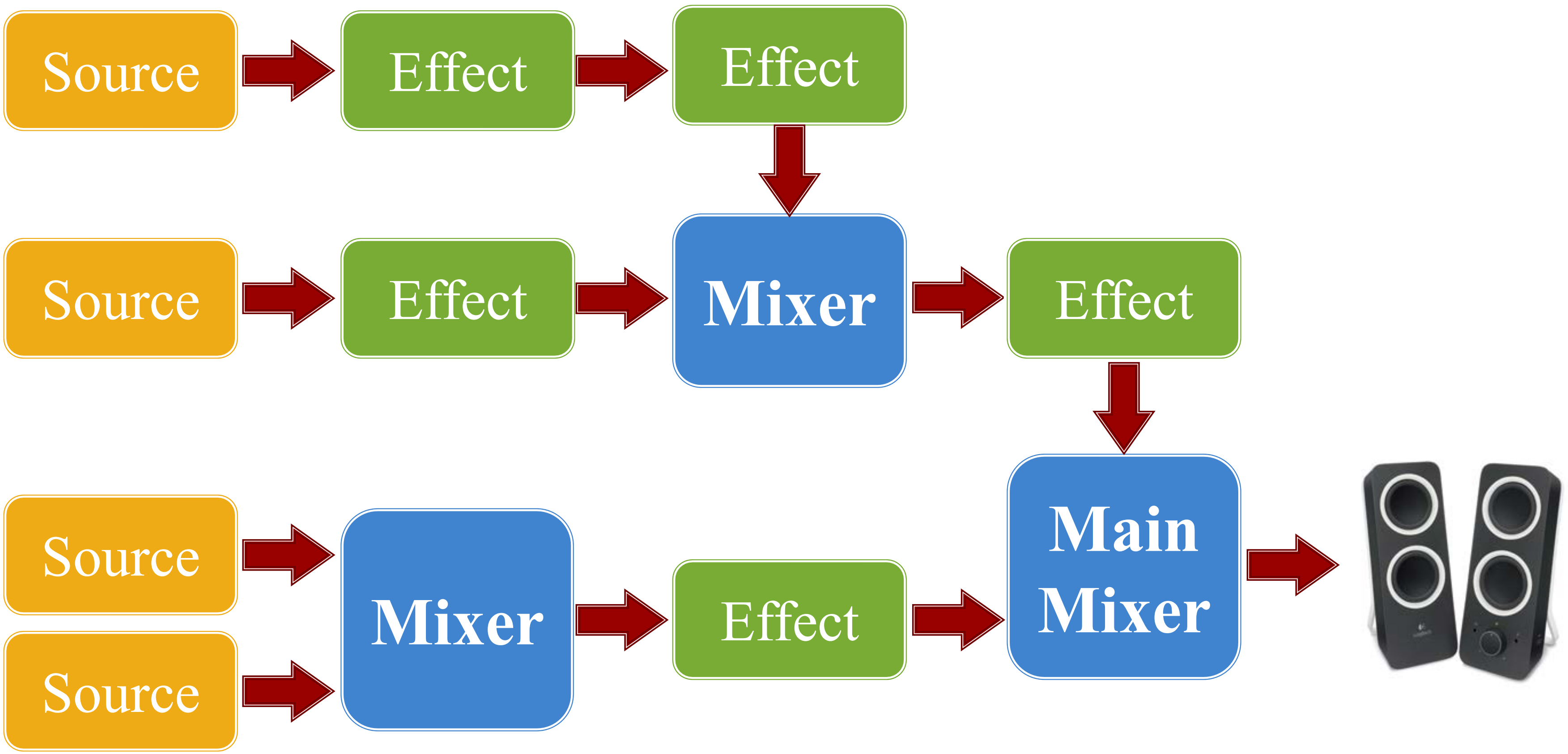
- ```
  /**
   * Plays given sound as a sound effect (paging out as necessary)
   *
   * @param  key      the reference key for the sound effect
   * @param  sound    the sound effect file to play
   * @param loop      Whether to loop indefinitely
   * @param volume    The sound volume
   */
  void playEffect(string key, const std::shared_ptr<Sound>& sound);
  ```
- `void stopEffect(string key);`
- `void setEffectVolume(string key, float volume);`
- `void getEffectState(string key);`

Refer to instance logically
  ```

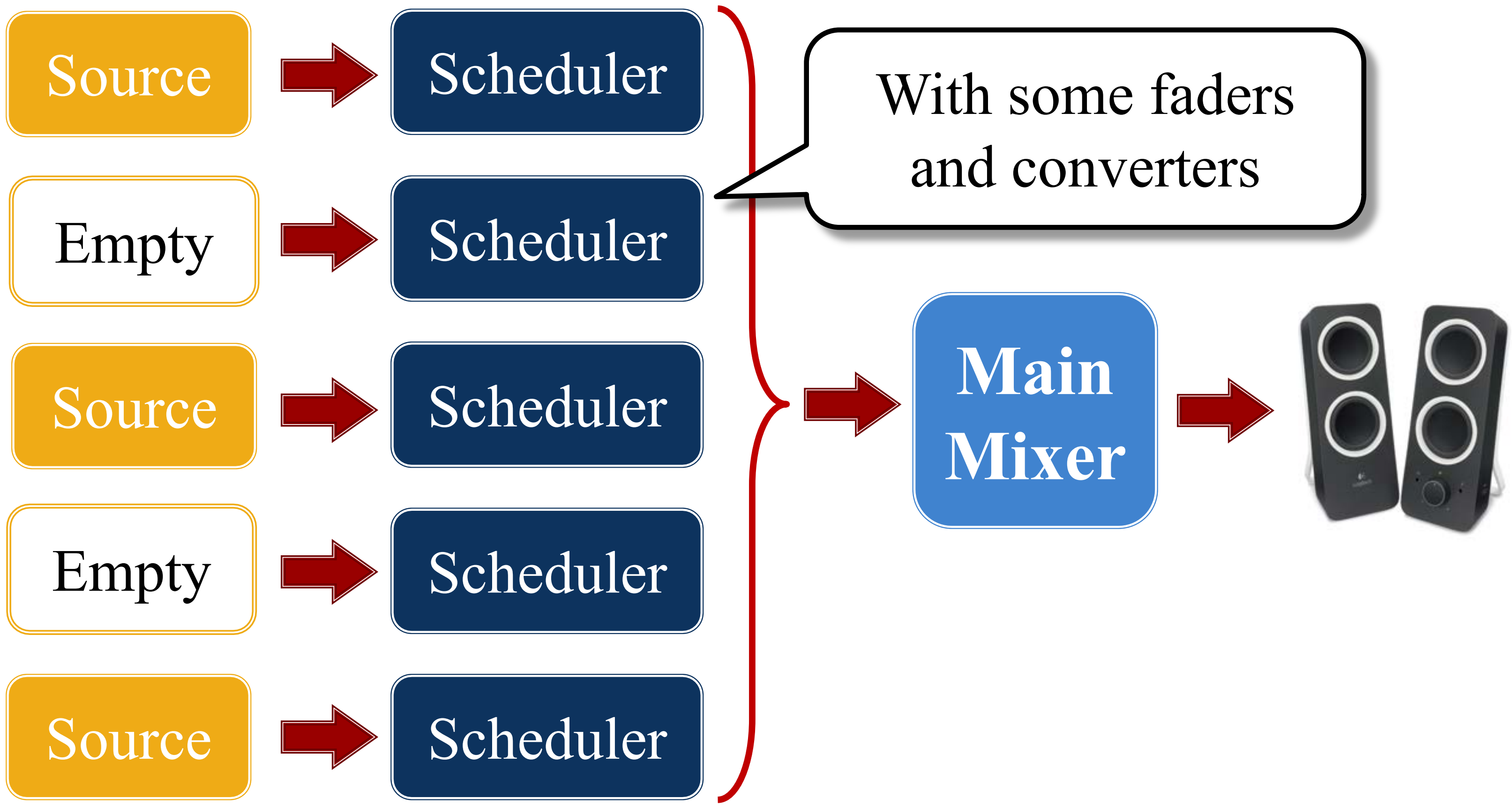
Problem with the Channel Model

- All controls are embedded in the channel
  - **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

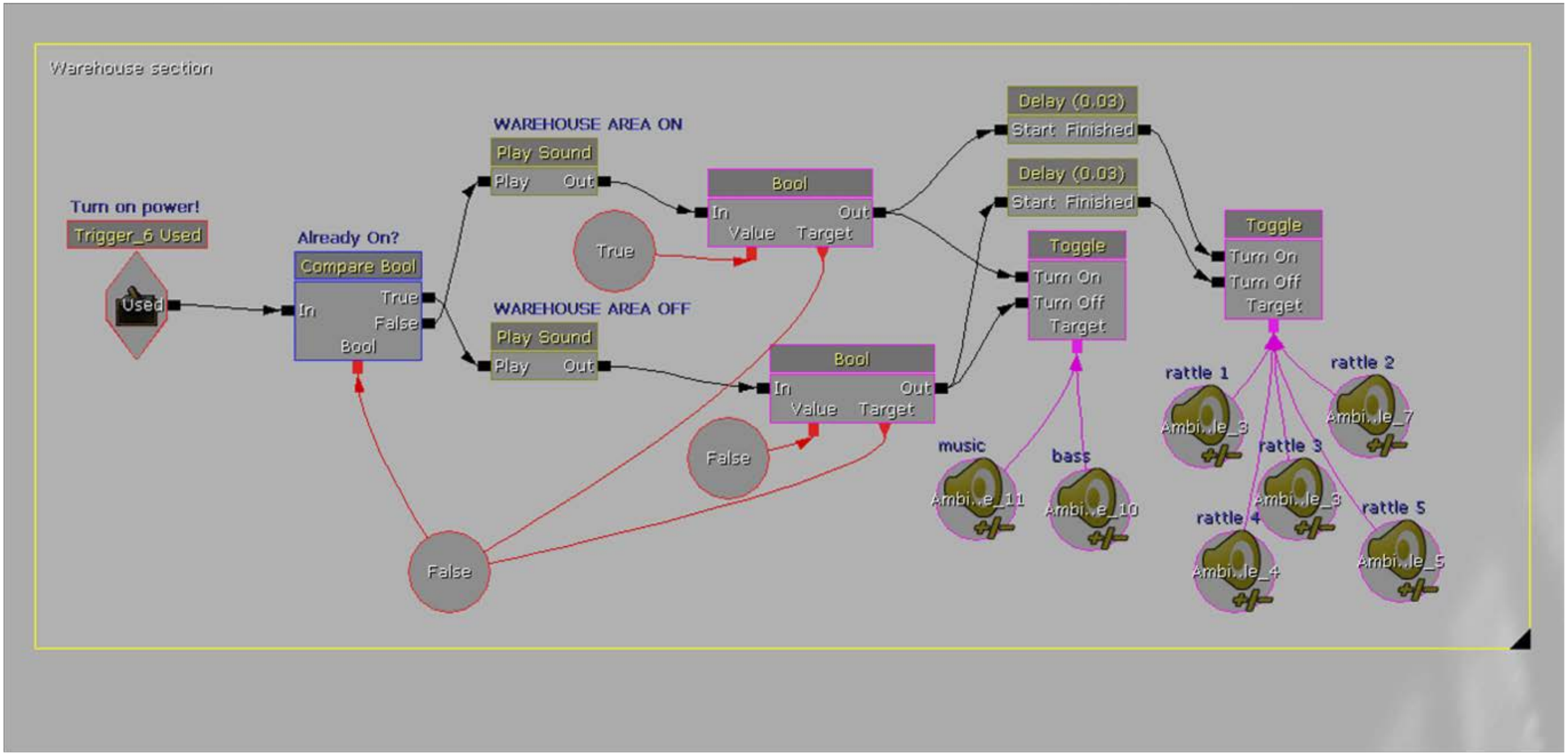


DSP Processing: The Mixer DAG

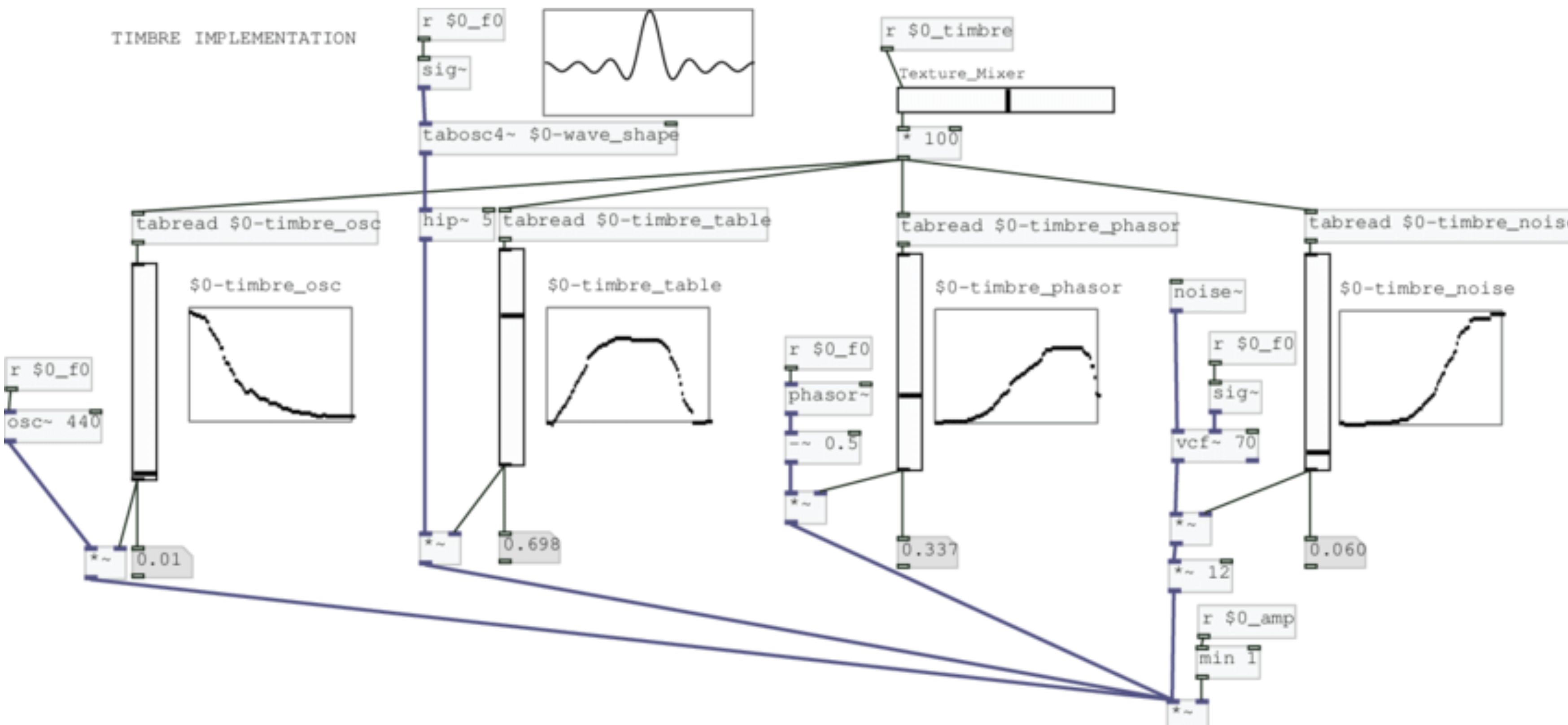


Channel model is a special case of this DAG

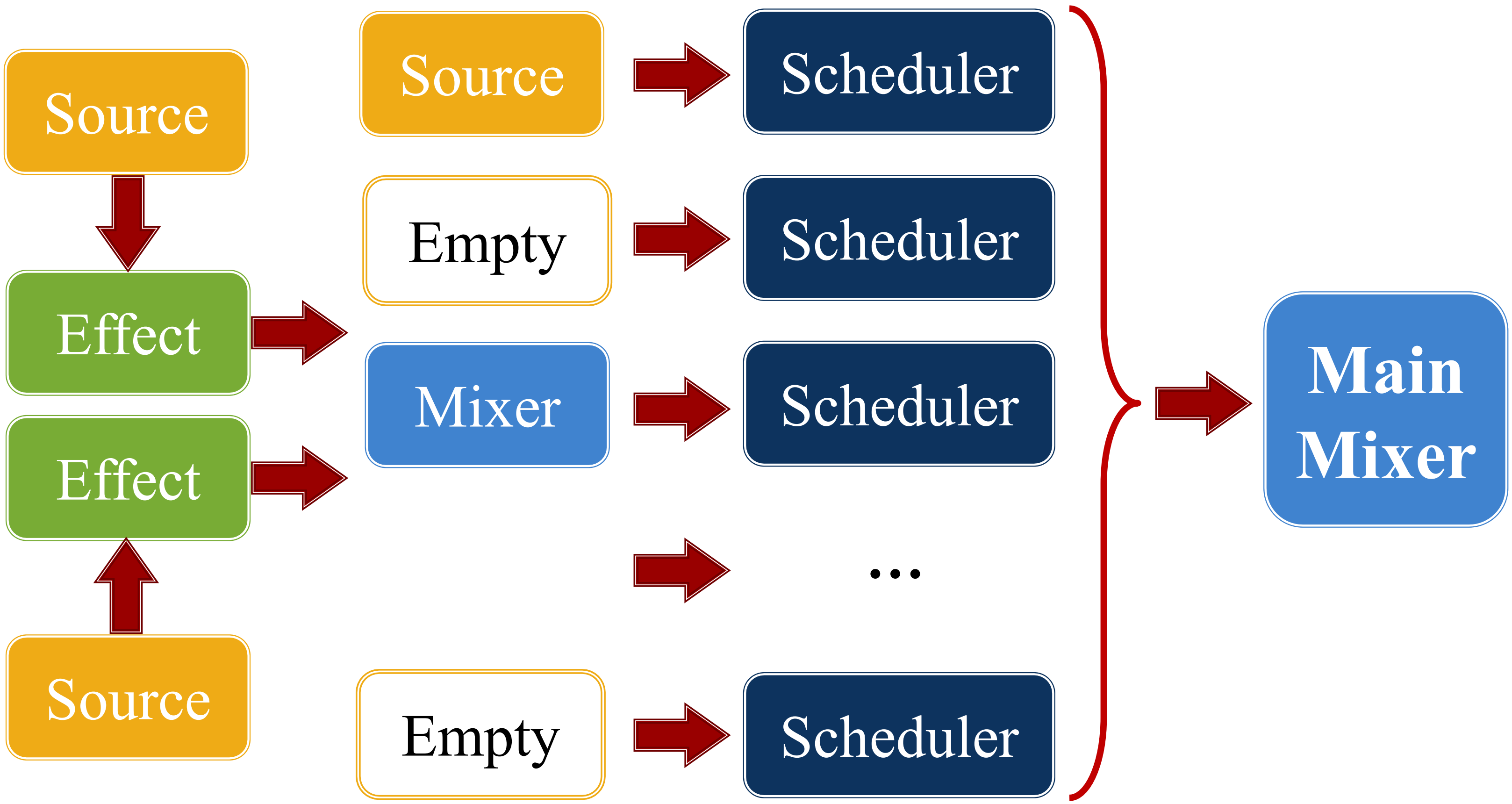
Example: UDK Kismet



Example: Pure Data



An Observation About "Channels"



Scheduler accepts any **audio subgraph**

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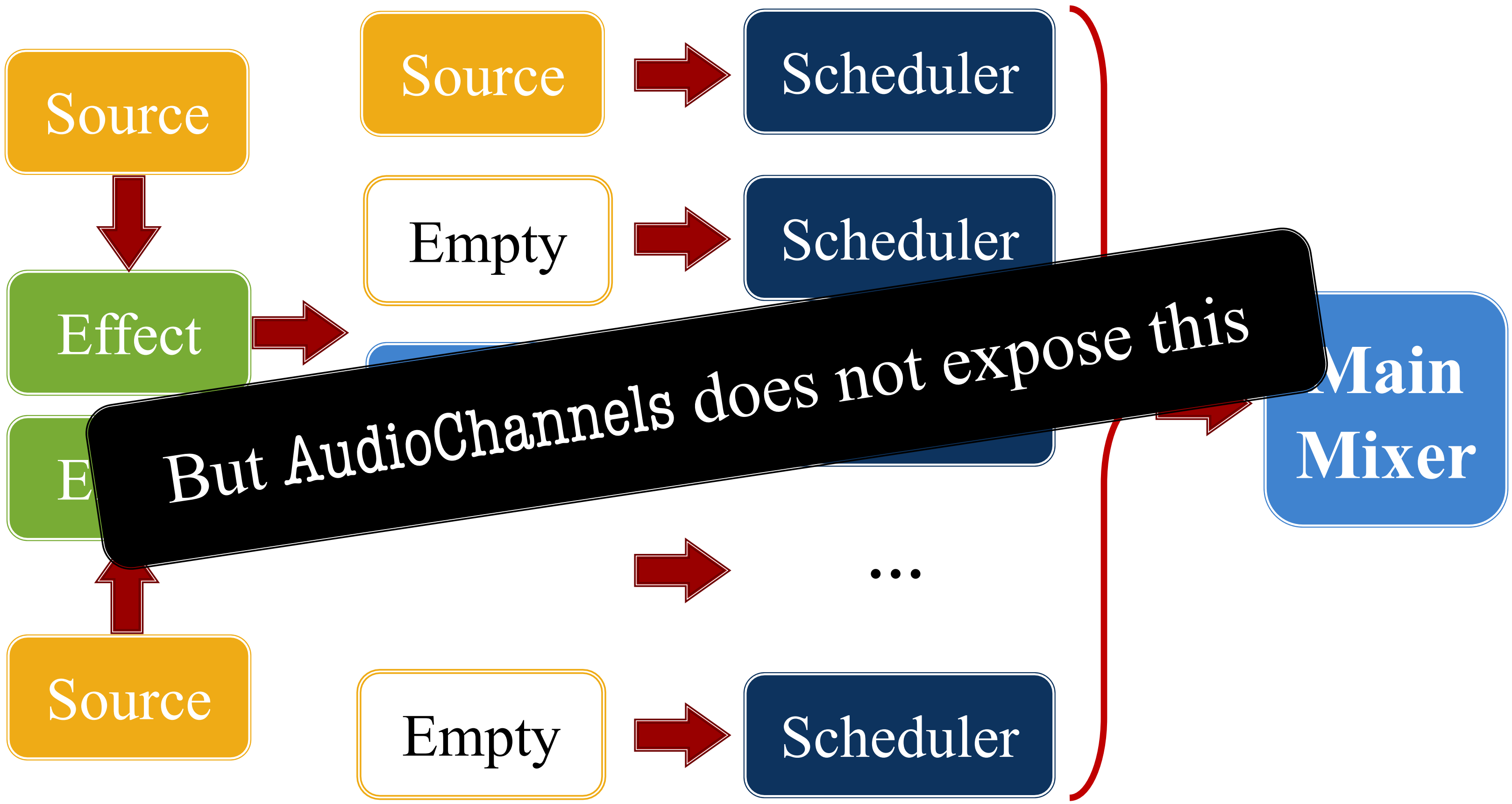


Scheduler accepts any **audio subgraph**

Creating Your Own Audio Graph

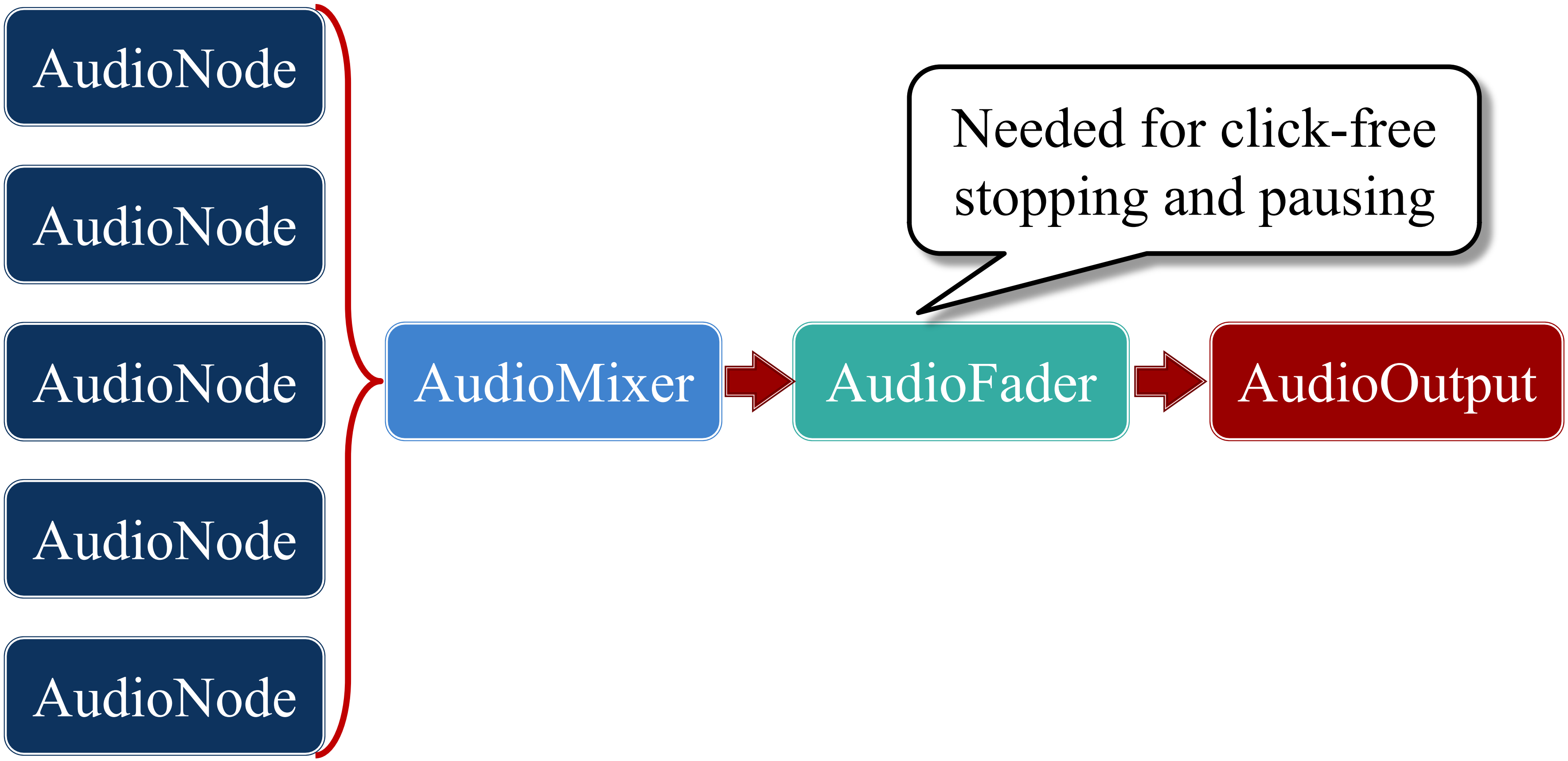
- Class AudioManager
  - Starts/stops audio system
  - Specifies the buffer size
  - Provides factor methods
  - Allocates input and output
- Class AudioOutput
  - Terminal node of the graph
  - Can be *named* or *default*
  - Defines the # of channels
  - Defines the sample rate

AudioOutput

AudioOutput

AudioOutput

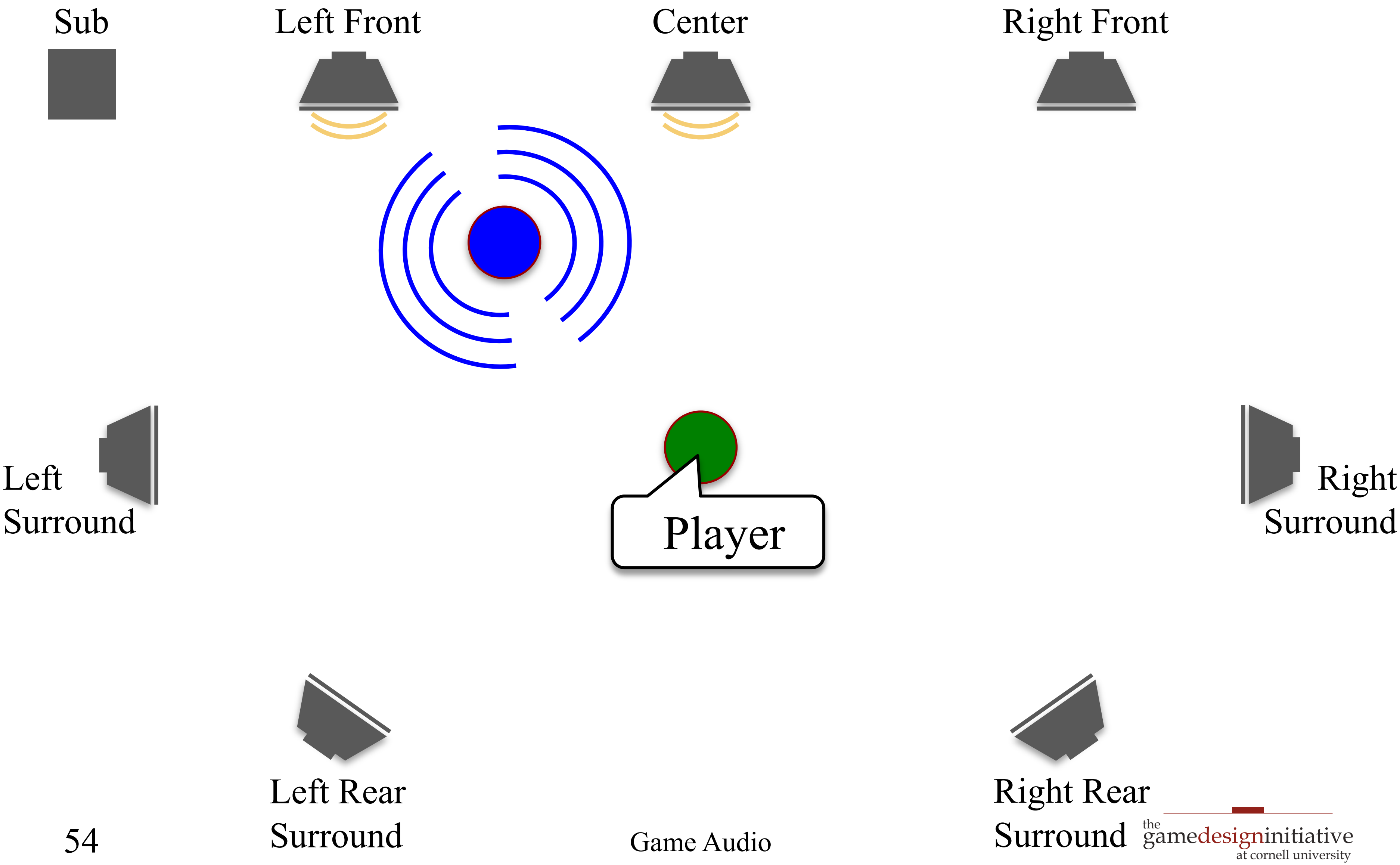
Creating Your Own Audio Graph



AudioNode Classes in CUGL

- **AudioPlayer**
  - Single playable instance for a sound asset
- **AudioFader**
  - Fade-in, fade-out and cross-fade effects
- **AudioPanner**
  - Simple stereo channel panning
- **AudioInput**
  - A recording node, for real-time playback

Advanced: Surround Sound



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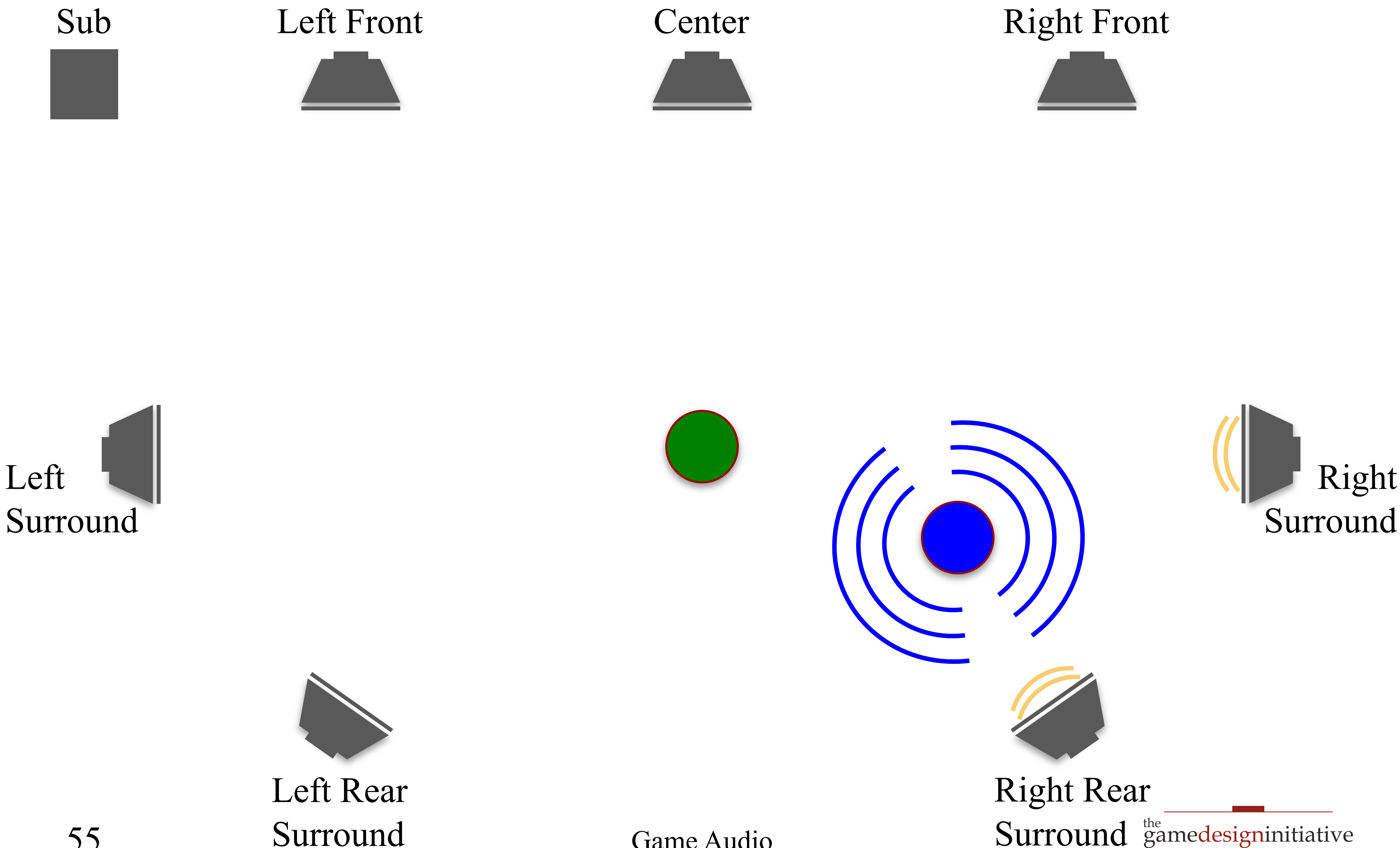

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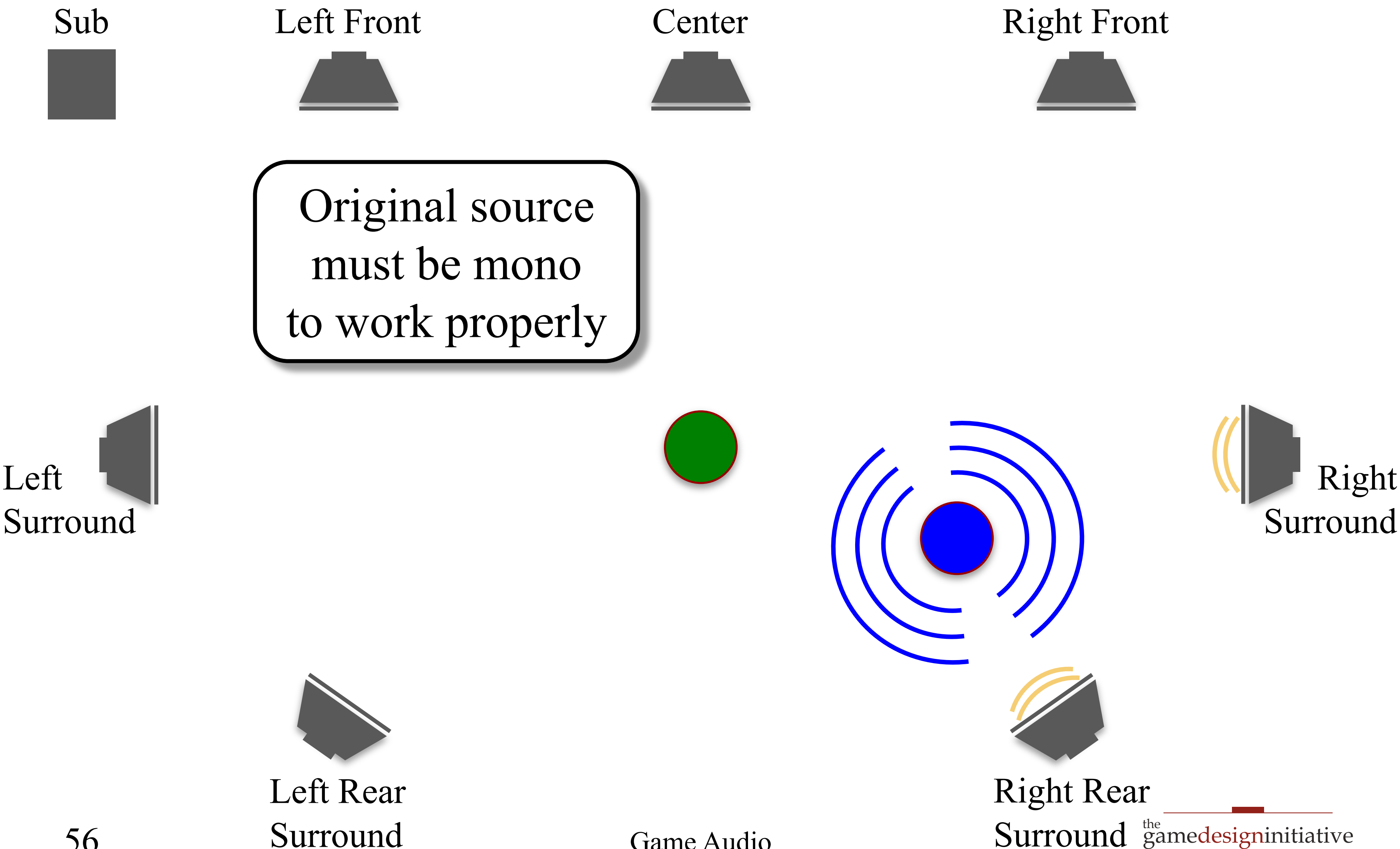
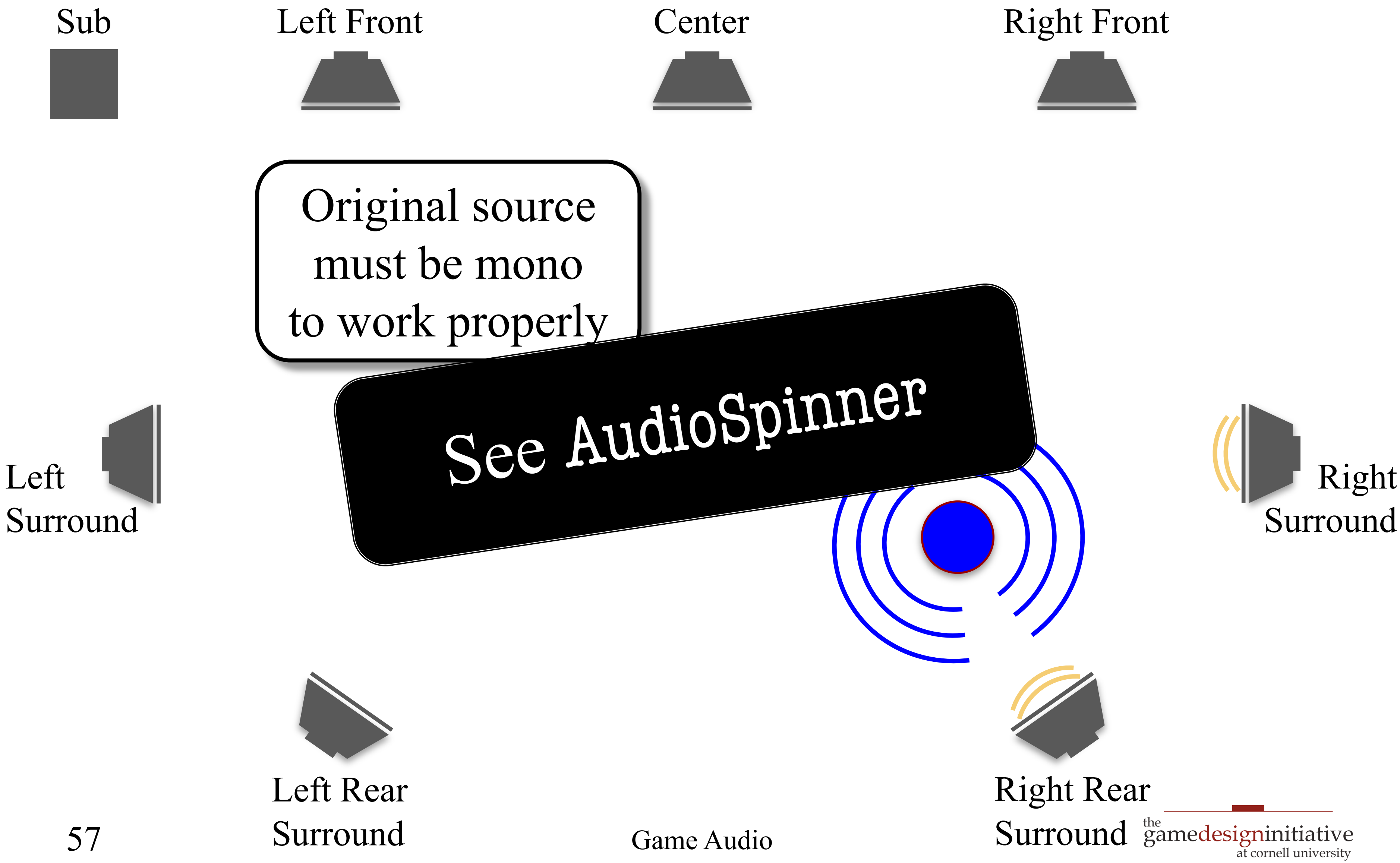
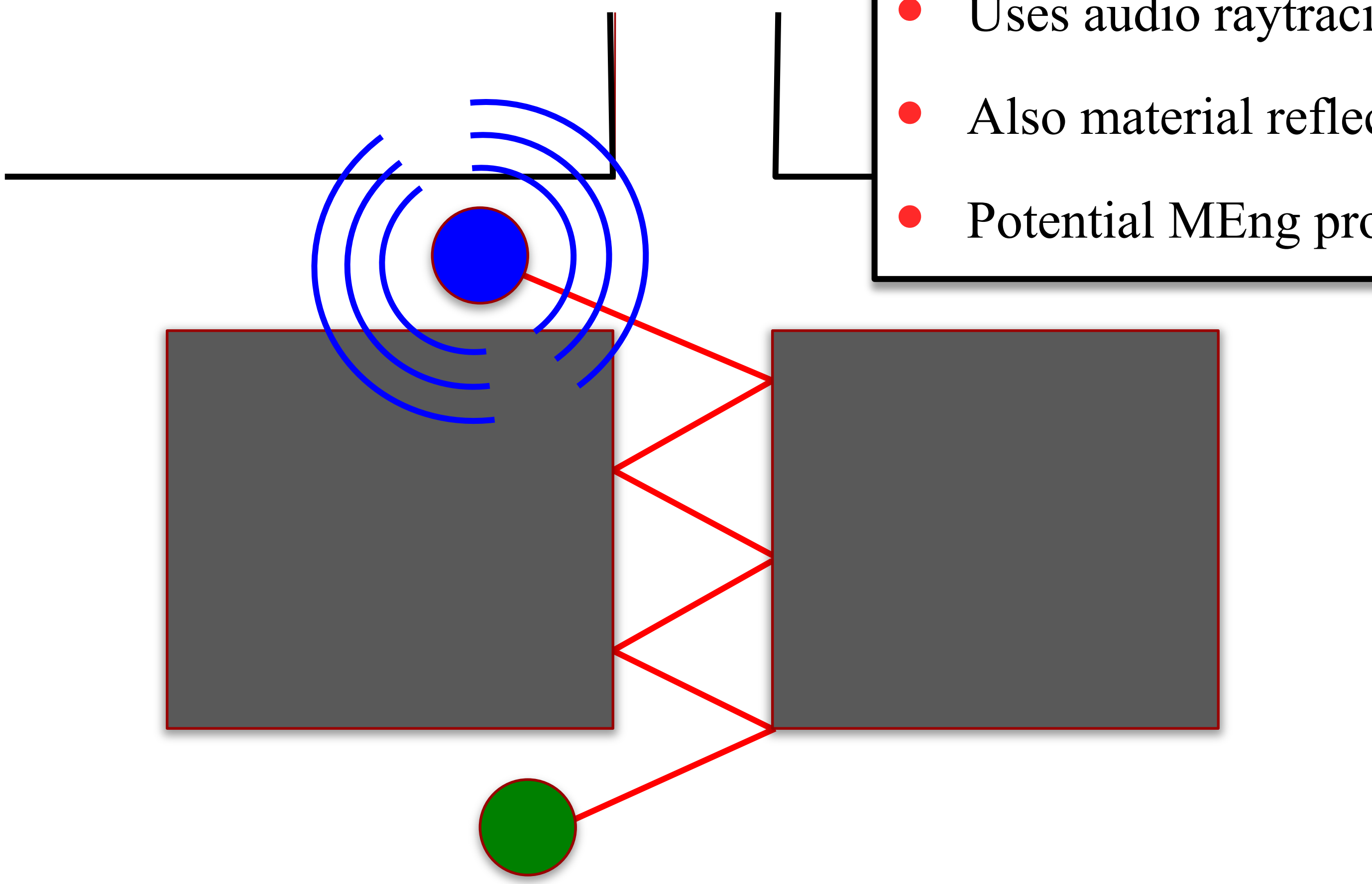


Original source must be mono to work properly

Advanced: Surround Sound



Advanced: Reverb Calculations



- Uses audio raytracing
- Also material reflection
- Potential MEng project

Advanced: Reverb Calculations

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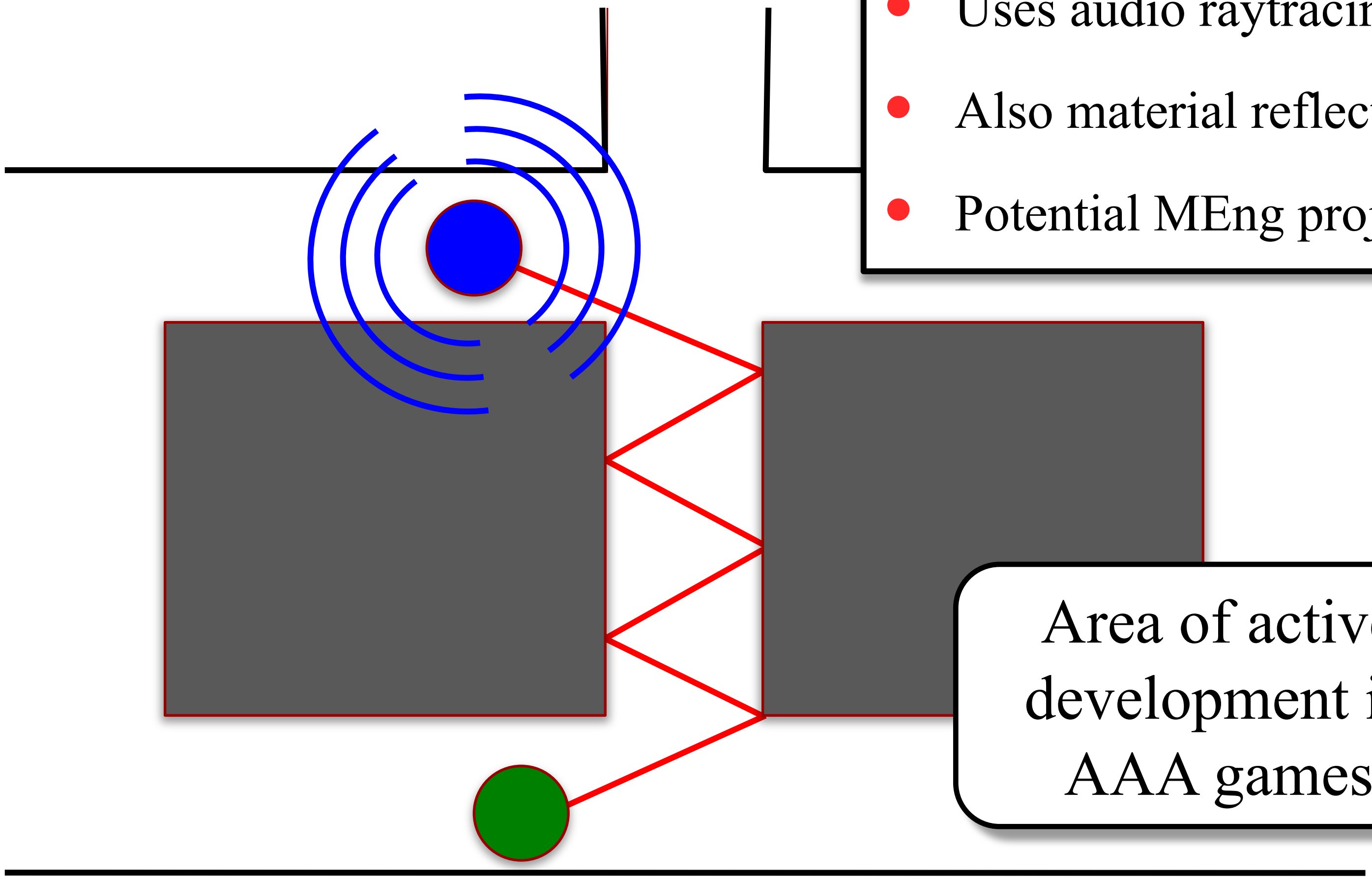
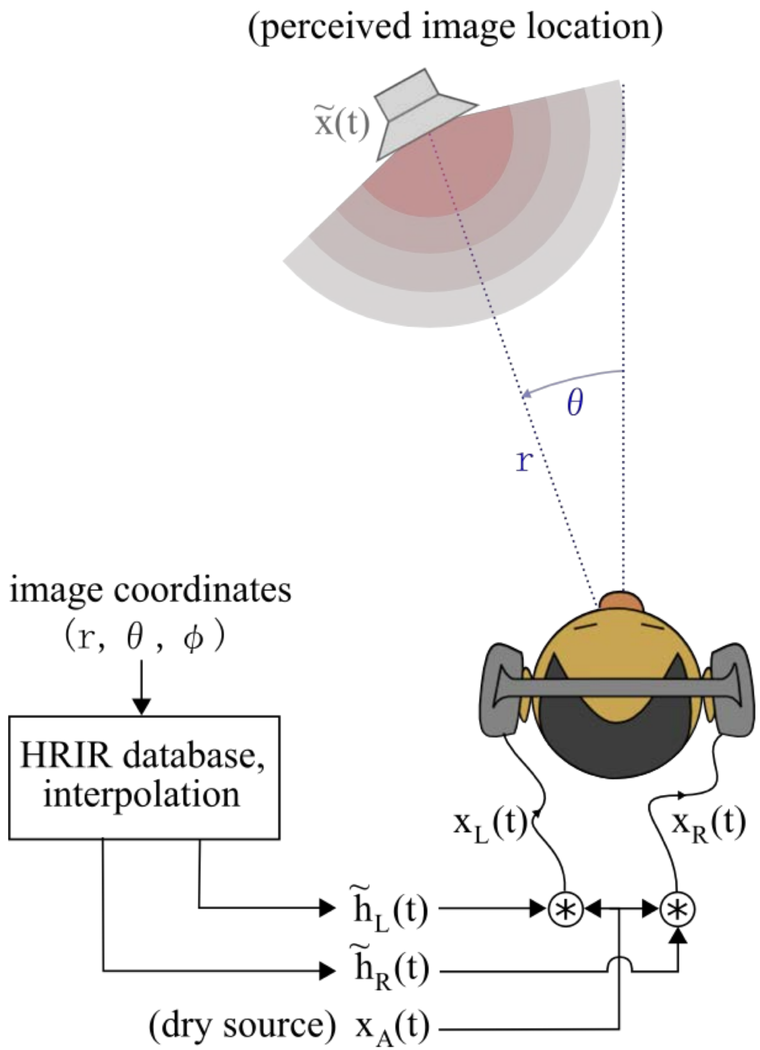


Area of active development in AAA games

Advanced: Binarual Synthesis

- Positional sound is fakey
  - Essentially volume control
  - Cannot pinpoint source
- **Goal**: realistic perception
  - Track the sound parallax
  - Account for shape of head
- Not (yet) in CUGL
  - In experimental branch
  - Will merge in summer



Example: Papa Sangre

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
  - CUGL has some early support for all this