Mac OS X IOAudio

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Who am I

- I'm Godfrey van der Linden
- Past member of Apple's Mac OS X CoreOS kernel group
- One of the two architects for the object oriented input/output infrastructure, IOKit
- I designed and implemented much of IOKit's synchronisation (threading model) and its user/kernel/IOMMU memory description objects
- Involved in many family designs, though I personally owned the serial family only

Apple Architect, Kernel I/O Infrastructure (IOKit), 1997-2006
Now a Ph.D. student researching complete system power modelling
A family in IOKit encapsulated a lot of knowledge in a particular hardware class. PCI, USB, Firewire have bus families, Block Storage and Serial are example of leaf hardware class designs.

Audio case study

- This presentation is a case study of Mac OS X's IOAudio family
  - First hour will describe the design process
  - Second hour will discuss subtle threading issues with the design
  - Last hour will discuss the latest IOAudio's evolution
- Questions are welcome and in the last hour I'd prefer to lead a discussion rather than lecture
- Slides for the last hour will be posted soon after this lecture

A family in IOKit terminology is a group of related C++ classes that encapsulate the OS's knowledge about an I/O class of devices.

Mac OS X

- Mach 'micro'-kernel encourages light-weight thread use
- Also provides extensive, flexible and slow virtual-memory APIs
- Dispatching is priority sorted, with round robin within 127 priority levels
- 'Time constrained' - threads are considered soft real-time
  - There is no guaranteed scheduling
- High priority does not increase CPU allocation but rather speeds thread re-schedule
- Only primary interrupts have a higher priority than time-constrained threads

100% per CPU is as fast as you can go.

Thread priority bands

Primary interrupts
Time constrained band
thread_call timers
Page-out thread
IOWorkLoops & kernel_threads
Top user band & Window Manager events
Carbon Async threads
user time-share threads
Idle threads (one per CPU)

Broadly the priority levels are grouped into similar functionality
Carbon threads are becoming less relevant
Idle threads basically put CPUs into progressively lower power states

Why is audio interesting

- In 1998 Apple was dying
  - Still held onto its multi-media customers, just!
- They needed a modern operating system and must not loose their remaining loyal customers
- Audio is very hard to do
  - Especially on an established platform
- IOAudio is the IOKit solution to this problem
  - It required extensive ground up implementation from interrupt path through to user land libraries
    - Kernel: Interrupt latency, thread priority protocols
    - User-land: Source code compatibility with Classic

In 1998 nobody had a good protected memory space solution to audio. The dominant meme was “unix can not do glitch free audio”.
The main issue is that primary interrupts tend to run unboundedly
As a study of AOS techniques audio touches most parts of a system, from in kernel design to client process libraries, through to hardware and clock interrupt delivery and services. Shared memory and multi-process synchronisation.

Goals

- Must be capable of delivering glitch free audio
- Minimise system load
- Mixing audio from multiple processes and threads
- Shortest possible input to output audio latency
- Highly reactive to audio controls
- Reasonably bounded 'jitter', i.e. variation from scheduled code start time
- Clients must live entirely in user-space
- Kernel has its own address space, typically 4/4KVM
    - This implies expensive user/kernel transitions

Ability to play audio WITHOUT ever glitching
Minimise audio system load
Allow multiple processes to play audio simultaneously
Jitter - Variation in routine start time
4/4 KVM, 4Gb Kernel and user virtual memory on 32bit machines
User input such as volume, balance etc

Classic audio drivers

- Audio drivers typically used the double buffer paradigm
  - After buffer completion, next buffer is scheduled...
  - ...and all clients are then called to fill the alternate
- Problems
  - Single address space
  - Low latency required very small buffers, which forced high interrupt rates. Usually not required
  - Totally dependant on other interrupt handlers
  - Clients often have an optimal buffer size for their algorithm, usually requires yet more double buffers and complexity

Double buffer
Single address space
usually single process

Classic duty cycle

1. Wait for a buffer done interrupt
2. On interrupt call out to sound mixing library to fill buffer
3. On return, wrap buffer in DMA request
4. Schedule DMA

- Step 2 can take an arbitrarily long time, if it doesn't return promptly the audio engine will idle
- If buffers are too big then audio isn't reactive to user input
- If buffers are too small then too many interrupts are delivered
- If sound clients live in a different address space then switching costs soon mount

Introducing karaoke

- It seems trivial use, but internet streaming background and mixing microphone input is one of the hardest applications to support
  - It requires microphone to speaker latency < ~4ms
  - Sound track is streamed across the internet, i.e. long delays
  - Video is expensive in CPU resource
  - May need to signal process input for echo removal, needs actually output audio
- Ideal implementation would be to combine a public address app and a streaming media player, that's karaoke

Full video stream from the internet
Full music stream from the internet
Microphone plugged into audio input

Why is karaoke hard

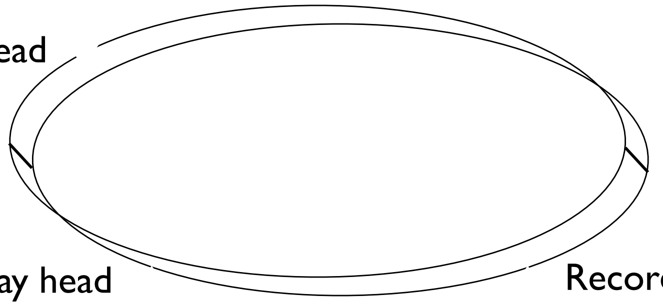
- The PA app requires very low latency but is otherwise lightweight
- The background stream needs large buffers to download into and is usually heavyweight
- Conflicting requirements, the traditional solution is to run with very small audio buffers, calling every client per buffer
  - Very expensive, on virtual memory systems
  - Background must also double buffer. From natural decode buffers to small low latency buffers

Human sensitivity to echoes
You only have a few of milliseconds
Remember that sound travels about 33cm/s (1 foot/second)
So need high-speed mix of input audio into audio stream
However given internet latencies should buffer sound for 30 seconds!
Ideally the decode of audio is done in large buffers, but input->output is vital

Observations

- Observe that audio plays at a known and constant rate
- Also most modern audio hardware has advanced DMA
- Hardware reads data when it needs it, not when DMA is scheduled
- Audio hardware often allows for free-running DMA program, that is the engine automatically loops back to the first DMA
- Since the audio rate is known it is possible to determine which sample is playing
    - Any buffer size can be simulated using timer interrupts instead of primary audio interrupts and each client can use any buffer size that suits their algorithm
- Notice that the DMA runs all the time, with or without data

Note that buffers take a fixed amount of time to play
This means that time is isomorphic to sample
However modern processors have sub-microsecond timers
Divorce anecdote, QA engineer who set his system up at home and left the house and systems paniced.



Tape loop metaphor

This metaphor isn't entirely correct, in fact the tape stays in one location and the heads move around the tape.

The audio ‘framebuffer’

- Audio has some similarities with video
  - A piece of hardware automatically reads memory
  - ... and outputs the data through a set of DACs
  - Compositing layers graphics into video memory
- We can consider mixing of an audio stream to be a compositing operation
- Instead of using xy-coordinates, an audio mixbuffer is addressed with time-coordinates
- A client plays audio by
  - presenting a buffer to the driver with a start time
  - driver translates time to an address and accumlates onto the specified location

The implies an interesting solution, don’t double buffer, use a free running DMA program.
Every program can choose its own natural audio rate
Kernel does regular mix/clip and sample to float conversions
Need locks to serialise mix-clip, needs semphores, 2 kernel round trips

In kernel

- You may have noticed a hole in this design
  - Unlike video audio compositing is inherently additive
  - The buffers must be zeroed after they have been played
  - Some already played data can be convenient for echo cancelation
- The kernel driver's task is to:-
  - Build a time/sample mapping, Interrupt timestamps
  - Accumulate each client stream into the mixbuffer
  - Convert the mixbuffer format into hardware samples
  - At some point after samples are played, zero the mixbuffer out
  - In case of a kernel panic, zero the sample buffers

Since userland solution needed two kernel roundtrips for locks, it was easier and more efficient to take a single kernel trip to provide buffers and sleep thread waiting for next input buffer

Userland audio?

- You may have noticed that this design would work as well in user land, by mapping the mixbuffer into every address space
- Unfortunately mixing, which is accumulative (+=), must be serialised
- We realised that taking and dropping an interprocess mutex required 2 kernel roundtrips plus one more for the data transfer and finally a call to sleep the thread
- But doing the serialisation, mixing, sleeping & input data copying in the kernel only had a single kernel round trip
- What a shame, basic rule for everybody else is to stay out of my kernel!
- Have you seen what we missed, at the time?

Missing only a few samples can cause audible clicks
Audio has the highest priority in the system, higher than drivers!

Human sensitivity

- Early in the design process it was apparent that humans are very sensitive to missed samples
- The user land audio library team convinced us that they must have the highest priority in the system to provide glitch free sound
- We warned them that this is "just soooooo" dangerous
  - Infinite loops would not be debuggable
  - You must give up the processor occasionally to allow the system to run
- We caved and the audio threads in OS X are the highest priority in the system, only primary interrupts run at a higher priority

Mac OS X primary interrupts usually, disable the interrupt line in the PIC then schedule a 'work loop' to clear the interrupting condition. The work-loop is lower priority to the real-time band.
They convinced us that they could deal with the responsibility

Audio duty cycle

- Register a pair of input and output buffers with the kernel
  1. Fill output buffer
  2. Call kernel driver with output buffer and a deadline
  3. Compute starting sample of buffer
  4. Mix data into floating point buffer and clip into sample buffer if required
  5. Block thread until deadline passes expires
  6. Convert input samples to floating point
  7. Return input data to user land
  8. Repeat

Every client can use what ever buffer size most suits them. Also that the data can be presented to the system at any time, independent of the DMA engine.

Kernel responsibility

- Track number of registered client processes
- Accumulate each client's data into the mixbuffer
- Track buffer start times precisely using CPU counters
    - On interrupt a sample counter is read from the hardware
- Convert from CPU time to sample clock and back
- When last client buffer is presented for a time period clip the floating point data into hardware samples
- Setup a timer interrupt to erase the outgoing data
- We chose to use 3 x 1/8s buffers for most drivers

We could push this design down to .5 ms or 22sample buffers and degrade well provided enough time was given to provider I/O threads to complete.

Implementation

- Mac OS X was a brand new system. It was not necessary to maintain backwards compatibility with classic mac's 68K driver model
- This design was only practical by keeping very tight control over primary interrupts
- But we were free to define an I/O model that uses threads to prioritise interrupts
- During the primary interrupt the interrupt line is disabled until the workloop calls the interrupt handler
- Most Apple hardware does NOT share interrupts, this means that only add-on PCI drivers are *required* to implement primary interrupt filters

yes many drivers were still written in 68K even on PowerPCs
Why did I mention shared interrupts? Only code that must run at primary interrupt is the 'filter', which has a simple duty, to determine if the hardware raised an interrupt and return true if so.

Intermission

- Consider the implications of audio processing being at such a high priority compared to drivers and the window manager

Highest priority

- The first problem we had to fix was the infinite loop debugging
  - Scheduler enforces a maximum thread run-time and reclassifies a real-time thread to the time-share band
- What about the I/O threads?
  - No easy solution, the audio engine must regulate its system use and give up the processor regularly
- But even giving up the processor may not be sufficient, what about audio input from one of the serial buses?
- In this hour I will describe two completely different solutions to solve the same basic priority inversion

No surprise we had warned the audio library guys, however
Can’t debug infinite loops! Run time was about 2 seconds
And since the thread has been running for a long time its time share priority is automatically depressed.
Hold off disk and network drivers
What about those drivers that have audio protocols, such as: USB and Firewire buses

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Remember this slide?

System threads

- Audio threads are in the real-time band
- OS X has a simple dispatcher
  - The highest priority run-able thread is dispatched
  - Continues till end of time-slice
  - Round-robin within each priority
- What does this mean?
  - Audio threads can starve the rest of the system
  - OS X's scheduler is not 'fair'.
  - Real-time band have responsibilities
  - ..but they only operate with limited information

Interesting side effects, need to give enough time drivers to get raw audio data, such as mp3 steaming from disk, etc
I think that eventually Grand Central dispatch will address this issue

Serial bus input problem

- Input data from these buses are not laid down contiguously
  - 44100smpl/s uses 9 x 44 and 1 x 45 sample 1ms buffers
- Under heavy system load the firewire and usb bus work loops are held off
- When an audio DMA chain completes the buffers are converted to floating point and copied into the input buffer
- The data is on the system but can not be accessed until workloop is scheduled, usually too late for input processing

USB audio input

- All IOKit drivers use workloops to process interrupts
- The frame buffer design works fine for output
- Isoch DMA is not very time dependant, it reads data as necessary. Data need not be present at DMA program time
- Input is a different problem
  - With output the host can sprinkles the occasional larger DMA request on contiguous data as needed
  - For input however the DMA must provide 45 sample buffers for every millisecond, even though 90% of buffers are 44 samples
  - Input must be copied into a contiguous buffer, combined with conversion to floating point

The 125 millisecond chunks can be scheduled regularly, this should be possible for just about any synthesiser load.
The blocking issue is a result of 44100 samples per second, requires 9 44 sample blocks followed by one 45 sample block.
Not sure why the USB handler needs to run I think it is because they do not export the USB DMA engine directly and data needs to be copied into the DMA intermediate format

Input buffer size

- These holes are inconvenient
- Consider the public address use case, audio data needs to be copied from input to output every few milliseconds
- But the input DMA's don't complete until the end of the 125ms DMA chain
  - At which point the convert from input samples to floating is performed
- Even though the data is already on the system it needs to be copied into the input floating point buffer to be useful
- But this copy only occurs infrequently and is easily held off by higher priority threads

USB solution

- The USB family was enhanced with some high-performance isoch APIs
- The Audio driver can label an input stream as ‘high-performance’
- The USB bus driver adds interrupts after every 1ms I/O frame
- But limits itself to marking completed USB frames nothing further
- On the client thread the audio driver can walk its input DMA chain and copy and convert any available data

USB critique

- Great
  - Converting input data to floating point is done lazily
    - Either at 8Hz by driver workloop
    - ... or by high-priority client as needed
- Terrible
  - 1000 interrupts per second!
    - The primary interrupt is used for more than scheduling a client thread
    - All other interrupts are disabled while the USB controller does its thing
  - A lot of code needed to be written to support new API

Firewire

- Like USB that is input data layout is determined by the data source
- The DMA queues need to be processed and completed
- We needed a different solution, the USB solution was less than ideal
- The sneaky solution: Call Firewire driver interrupt handler directly while on the client thread
  - All drivers must be able to cope with ghost interrupts
  - Most kernel threads are preemptible
  - OS X implements priority hand-off on mutexes
  - But all completion routines are run, including the non-audio requests

Firewire implementation

- The change was easy for the Firewire family & IOKit
- IOKit:
  - modify IOWorkLoop to publish the interrupt handler loop as a function
- Firewire
  - An API that allows a client to call this function
- The firewire workloop can be in three states
  - Not running: new API simulates a ghost interrupt
  - Running and pre-empted: At workloop mutex promote thread to caller priority till mutex dropped
  - Running: Do not allow running thread to be halted till it drops the mutex

Diagram solution
At first they tried changing there workloops priority up to the real-time band, but this had other effects on the system.

Firewire driver

- We found that no completions ran long, as a result not required to preferentially handle isoch completions
  - Most of the other firewire clients do little work, they usually just wake up another thread
    - Network packets wake NetISR
    - Disk packets wake pager or client disk IO thread
- The change to the Firewire bus driver was small
- We didn't have to change the workloop threads priority!
- Firewire bus driver cleanly transitions to high-priority when a client requires it
- This few lines of change makes all data that has arrived available to the audio client

Firewire Audio changes

- Small mod to IOWorkLoop to export workloop function
- Small mod to IOFirewireDevice to re-export it
- On entry to input fetch function call interrupt handler loop
- On function completion, the audio driver knows exactly what data is available and can copy and convert it immediately
- Done

Re-export isn't necessary, just cleaner. Any client can determine what workloop it is running on and call workloop interrupt handler loop directly.

Clean solution

- High-priority threads are a scarce and expensive resource and should only be used when absolutely necessary
- Did you notice that the firewire workloop can continue running at its normal priority?
- Only when low-latency audio input is required does it ever run at a high priority
- Sad kernel person alert, I think this solution is really cool

Summary

- The basic audio mixbuffer design using 8Hz input processing is hopeless for input to output echoing
- Both USB and Firewire audio protocols require input copying and conversion to floating point contiguous data
- USB's Primary interrupt solution required extensive, dangerous and expensive development in bus driver
- Firewire's Priority handoff solution, required a 2 line function change to the bus driver and a slightly bigger change to IOKit
- Sometimes the sneaky solution is worth looking for!

Problems with Audio

- Too much code needs to run in kernel. How much is too much?
  - Any at all! Stay out of ***my*** kernel!
- Playing system sounds requires a client to spin up the entire audio infrastructure
- The audio thread's responsibility to give up some time for lower priority threads is unreliable and won't scale
- Have you seen a solution to audio design we missed?
- We'll look at audio v2 in the next hour.

Audio v2

- How would you address these problems?
- I’d like each team of you to discuss them and when we restart we will design a more elegant solution

Take a break and then start a 15 minute design discussion.
Present raw ideas

What must be in kernel

- Any resource that publishes via standard BSD APIs, such as the file system or network stacks
- Any driver that requires primary interrupts
- Any driver that talks directly to hardware registers
- Any resource that can be shared by many different processes
- The last step in performance enhancement, that is when too many context switches on data path

Note that the middle 3 reasons can easily be exported to userland!
Data path context switches can be wasteful, draw data path for a userland driver to a client process.

coreaudiod

- Requires client process to register double buffers with daemon
- Daemon runs in time-constrained band, probably at slightly higher priority for the floating point to sample clip
- Mix each client buffer when it must be valid, determined by 'play/record' head position and each buffer's deadline
- Daemon can play 'simple' audio streams
- In principle the daemon is nearly as efficient as a kernel driver
- Publish a userland plugin model for non-standard mix-clip and sample conversion

Draw kernel transition diagrams for in kernel and daemonised processes

Kernel support

- Any driver that could not be bothered to rewrite for new driver model
  - Over time Apple will port drivers to new model as hardware is released
- The output sample-buffer must always be mapped into kernel address space in case of a panic
  - Playing last 3/8ths of a second in a loop is unpleasant to say the least
- Interrupt time-stamping probably needs to remain in kernel
  - Driver could, in principle, wake a daemon interrupt thread

This is a bit political within Apple, they usually do not update already shipped hardware drivers. They almost always fork all of the drivers for every hardware release and rarely fix bugs in 'old' hardware

Further issues

- We have pulled a lot of complicated code out of the kernel
- Some drivers are more difficult to develop as they need to provide two different modules
  - mix-clip/conversion plugin to coreaudiod
  - driver that maps buffers and gets interrupts
    - new drivers are easier; only needs to support a single client, coreaudiod.
- Still has a problem with developer responsibility to give up some CPU, i.e. transitory priority inversions
- Very slightly higher CPU expenditure for one more address space transition

Fortunately Apple provides an extensive library of highly efficient mix-clip/conversion routines for many sample formats