

# Linux, Locking and Lots of Processors

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## A little bit of history



- Multix in the '60s
- Ken Thompson and Dennis Ritchie in 1967–70
- USG and BSD
- John Lions 1976–95
- Andrew Tanenbaum 1987
- Linus Torvalds 1991

The history of UNIX-like operating systems is a history of people being dissatisfied with what they have and wanting to do something better. It started when Ken Thompson got bored with MULTICS and wanted to write a computer game (Space Travel). He found a disused PDP-7, and wrote an interactive operating system to run his game. The main contribution at this point was the simple file-system abstraction. And the key ingredient there was that the OS did not interpret file contents — an ordinary file is just an array of bytes. Semantics are imposed by the user of the file.

Other people found it interesting enough to want to port it to other systems, which led to the first major rewrite — from assembly to C. In some ways UNIX was the first successfully portable OS.

After Ritchie & Thompson (1974) was published, AT&T became aware of a growing market for UNIX. They wanted to discourage it: it was common for AT&T salesmen to say, 'Here's what you get: A whole lot of tapes, and an invoice for \$10 000'. Fortunately educational licences were (almost) free, and universities around the world took up UNIX as the basis for teaching and research.

The University of California at Berkeley was one of those universities. In

1977, Bill Joy (a postgrad) put together and released the first Berkeley Software Distribution — in this instance, the main additions were a pascal compiler and Bill Joy's `ex` editor. Later BSDs contained contributed code from other universities, including UNSW. The BSD tapes were freely shared between source licensees of AT&T's UNIX.

John Lions and Ken Robinson read Ritchie & Thompson (1974), and decided to try to use UNIX as a teaching tool here. Ken sent off for the tapes, the department put them on a PDP-11, and started exploring. The license that came with the tapes allowed disclosure of the source code for 'Education and Research' — so John started his famous OS course, which involved reading and commenting on the Edition 6 source code.

In 1979, AT&T changed their source licence (it's conjectured, in response to the popularity of the Lions book), and future AT&T licensees were not able to use the book legally any more. UNSW obtained an exemption of some sort; but the upshot was that the Lions book was copied and copied and studied around the world, *samizdat*. However, the licence change also meant that an alternative was needed for OS courses.

Many universities stopped teaching OS at any depth. One standout was Andy Tanenbaum's group in the Netherlands. He and his students wrote an OS called 'Minix' which was (almost) system call compatible with Edition 7 UNIX, and ran on readily available PC hardware. Minix gained popularity not only as a teaching tool but as a hobbyist almost 'open source' OS.

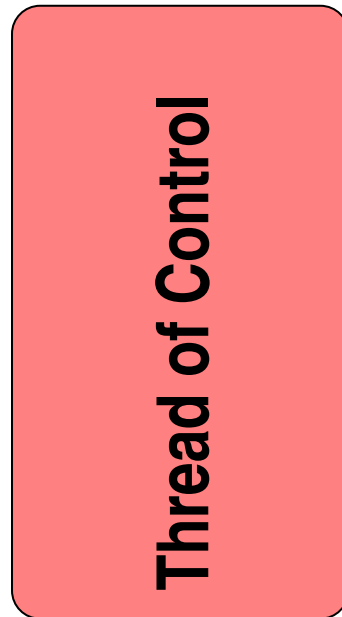
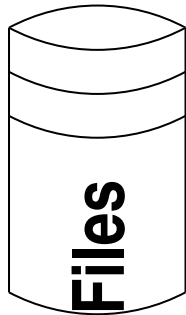
In 1991, Linus Torvalds decided to write his own OS — after all, how hard could it be? — to fix what he saw as some of the shortcomings of Minix. The rest is history.

- Basic concepts well established
  - Process model
  - File system model
  - IPC
- Additions:
  - Paged virtual memory (3BSD, 1979)
  - TCP/IP Networking (BSD 4.1, 1983)
  - Multiprocessing (Vendor Unices such as Sequent's 'Balance', 1984)

The UNIX core concepts have remained more-or-less the same since Ritchie and Thompson published their CACM paper. The process model and the file system model have remained the same. The IPC model (inherited from MERT, a different real-time OS being developed in Bell Labs in the 70s) also is the same. However there have been some significant additions.

The most important of these were Paged Virtual Memory (introduced when UNIX was ported to the VAX), which also introduced the idea of Memory-mapped files; TCP/IP networking, Graphical terminals, and multiprocessing, in all variants, master-slave, SMP and NUMA. Most of these improvements were from outside Bell Labs, and fed into AT&T's product via open-source-like patch-sharing.

In the late 80s the core interfaces were standardised by the IEEE, in the so-called POSIX standards.



Linux Kernel

As in any POSIX operating system, the basic idea is to abstract away physical memory, processors and I/O devices (all of which can be arranged in arbitrarily complex topologies in a modern system), and provide threads, which are gathered into processes (a process is a group of threads sharing an address space and a few other resources), that access files (a file is something that can be read from or written to. Thus the file abstraction incorporates most devices). There are some other features provided: the OS tries to allocate resources according to some system-defined policies. It enforces security (processes in general cannot see each others' address spaces, and files have owners). Unlike in a microkernel, some default policy is embedded in the kernel; but the general principal is to provide tools and mechanisms for an arbitrary range of policies.

- Root process (`init`)
- `fork()` creates (almost) exact copy
  - Much is shared with parent — Copy-On-Write avoids overmuch copying
- `exec()` overwrites memory image from a file
- Allows a process to control what is shared

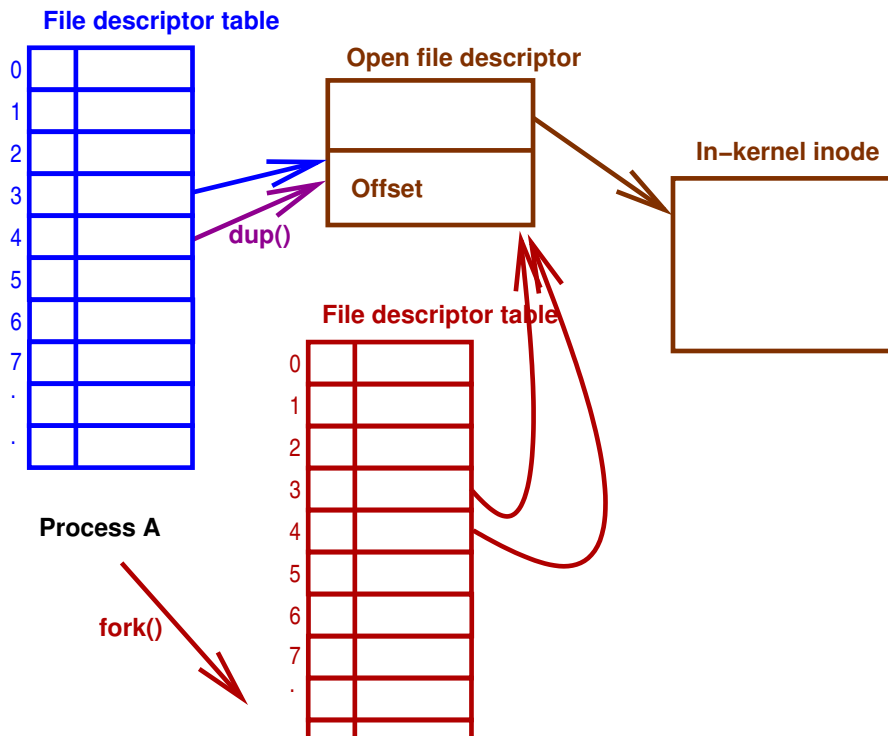
The POSIX process model works by inheritance. At boot time, an initial process (process 1) is hand-crafted and set running. It then sets up the rest of the system in userspace.

- A process can clone itself by calling `fork ()`.
- Most attributes *copied*:
  - Address space (actually shared, marked copy-on-write)
  - current directory, current root
  - File descriptors
  - permissions, etc.
- Some attributes *shared*:
  - Memory segments marked `MAP_SHARED`
  - Open files

First I want to review the UNIX process model. Processes clone themselves by calling `fork ()`. The only difference between the child and parent process after a `fork ()` is the return value from `fork ()` — it is zero in the child, and the value of the child's process ID in the parent. Most properties of the child are logical *copies* of the parent's; but open files and shared memory segments are *shared* between the child and the parent.

In particular, seek operations by either parent or child will affect and be seen by the other process.

## Files and Processes:



Each process has a file descriptor table. Logically this is an array indexed by a small integer. Each entry in the array contains a flag (the `close-on-exec` flag and a pointer to an entry in an *open file table*. (The actual data structures used are more complex than this, for performance and SMP locking).

When a process calls `open ()`, the file descriptor table is scanned from 0, and the index of the next available entry is returned. The pointer is instantiated to point to an *open file descriptor* which in turn points to an in-kernel representation of an index node — an *inode* — which describes where on disc the bits of the file can be found, and where in the buffer cache can in memory bits be found. (Remember, this is only a logical view; the implementation is a lot more complex.)

A process can *duplicate* a file descriptor by calling `dup ()` or `dup2 ()`. All `dup` does is find the lowest-numbered empty slot in the file descriptor table, and copy its target into it. All file descriptors that are dups share the open file table entry, and so share the current position in the file for read and write.

When a process `fork ()`s, its file descriptor table is copied. Thus it too shares its open file table entry with its parent.



```
switch (kidpid = fork()) {
case 0: /* child */
    close(0); close(1); close(2);
    dup(infd); dup(outfd); dup(outfd);
    execve("path/to/prog", argv, envp);
    _exit(EXIT_FAILURE);
case -1:
    /* handle error */
default:
    waitpid(kidpid, &status, 0);
}
```

So a typical chunk of code to start a process looks something like this. `fork ()` returns 0 in the child, and the process id of the child in the parent. The child process closes the three lowest-numbered file descriptors, then calls `dup ()` to populate them again from the file descriptors for input and output. It then invokes `execve ()`, one of a family of exec functions, to run *prog*. One could alternatively use `dup2 ()`, which says which target file descriptor to use, and closes it if it's in use. Be careful of the calls to `close` and `dup` as order is significant!

Some of the exec family functions do not pass the environment explicitly (`envp`); these cause the child to inherit a copy of the parent's environment. Any file descriptors marked *close on exec* will be closed in the child after the `exec`; any others will be shared.

- 0 Standard Input
- 1 Standard Output
- 2 Standard Error
- Inherited from parent
- On login, all are set to *controlling tty*

There are three file descriptors with conventional meanings. File descriptor 0 is the standard input file descriptor. Many command line utilities expect their input on file descriptor 0.

File descriptor 1 is the standard output. Almost all command line utilities output to file descriptor 1.

File descriptor 2 is the standard error output. Error messages are output on this descriptor so that they don't get mixed into the output stream. Almost all command line utilities, and many graphical utilities, write error messages to file descriptor 2.

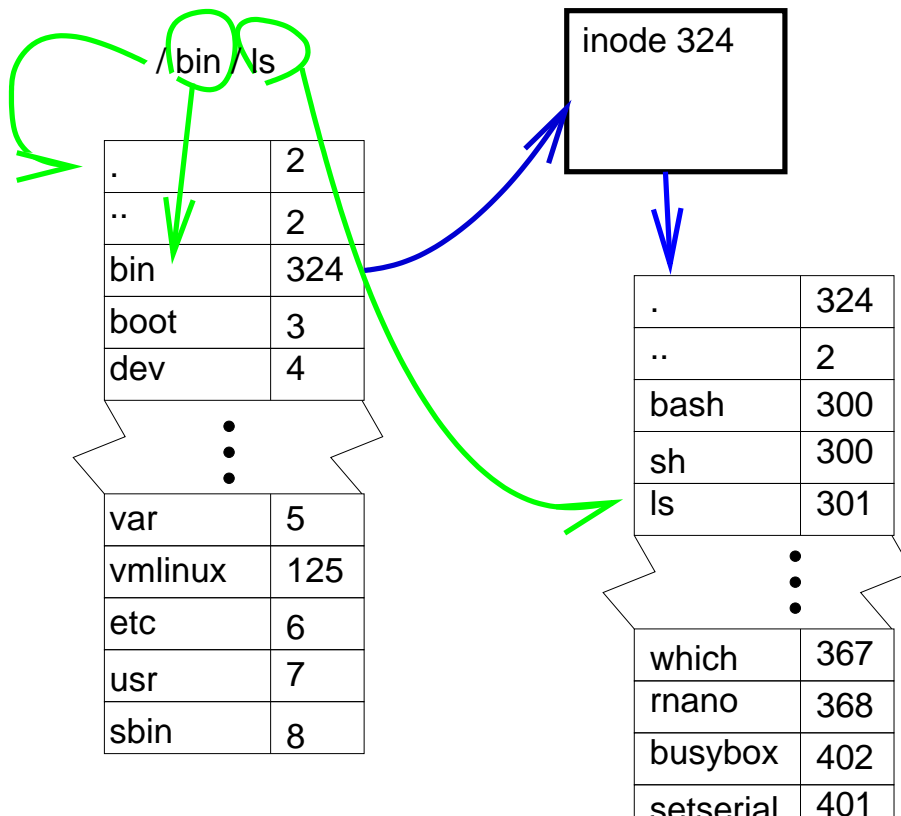
As with all other file descriptors, these are inherited from the parent.

When you first log in, or when you start an X terminal, all three are set to point to the *controlling terminal* for the login shell.

- Separation of names from content.
- ‘regular’ files ‘just bytes’ → structure/meaning supplied by userspace
- Devices represented by files.
- Directories map names to index node indices (`inums`)
- Simple permissions model

The file model is very simple. In operating systems before UNIX, the OS was expected to understand the structure of all kinds of files: typically files were organised as fixed (or variable) length records with one or more indices into them. One very common organisation was essentially an image of a card deck! By contrast, UNIX regular files are just a stream of bytes.

Originally in UNIX directories were also just files, albeit with a structure understood by the kernel. To give more flexibility, they are now opaque to userspace, and managed by each individual filesystem. The added flexibility makes directory operations more expensive, but allows Linux to deal with over thirty different filesystems, with varying naming models and on-disk structures.



The diagram shows how the kernel finds a file.

If it gets a file name that starts with a slash (/), it starts at the root of the directory hierarchy for the current process (otherwise it starts at the current process's current directory). The first link in the pathname is extracted ("**bin**") by calling into the filesystem code, and searched for in that root directory.

That yields an inode number, that can be used to find the contents of the directory. The next pathname component is then extracted from the name and looked up. In this case, that's the end, and inode 301 contains the metadata for "**/bin/ls**".

- translate name → inode
- abstracted per filesystem in VFS layer
- Can be slow: extensive use of caches to speed it up *dentry cache* — becomes SMP bottleneck
- hide filesystem and device boundaries
- walks pathname, translating symbolic links

Linux has many different filesystem types. Each has its own directory layout. Pathname lookup is abstracted in the Virtual FileSystem (VFS) layer. Traditionally, looking up the name to inode (**namei**) mapping has been slow; Linux currently uses a cache to speed up lookup.

At any point in the hierarchy a new filesystem can be grafted in using **mount**; **namei** () hides these boundaries from the rest of the system.

Symbolic links haven't been mentioned yet. A symbolic link is a special file that holds the name of another file. When the kernel encounters one in a search, it replaces the name it's parsing with the contents of the symbolic link. Some filesystems encode the symbolic name into the directory entry, rather than having a separate file.

Also, because of changes in the way that pathname lookups happen, there is no longer a function called `namei()`; however the files containing the path lookup are still called `namei.[ch]`.

## KISS:

- Simplest possible algorithm used at first
  - Easy to show correctness
  - Fast to implement
- As drawbacks and bottlenecks are found, replace with faster/more scalable alternatives

This leads to a general principle: start with KISS. Many of the utilities that are common on Linux started out as much simpler programs wrapped in shell scripts; as people elaborated the scripts to provide more functionality, they became less maintainable, and eventually were refactored into a compiled language.

- Extra keywords:
  - Section IDs: `__init`, `__exit`, `__percpu` etc
  - Info Taint annotation `__user`, `__rcu`, `__kernel`, `__iomem`
  - Locking annotations `__acquires(X)`, `__releases(x)`
  - extra typechecking (endian portability) `__bitwise`

The kernel is written in C, but with a few extras. Code and data marked `__init` is used only during initialisation, either at boot time, or at module insertion time. After it has finished, it can be (and is) freed.

Code and data marked `__exit` is used only at module removal time. If it's for a built-in section, it can be discarded at link time. The build system checks for cross-section pointers and warns about them.

`__percpu` data is either unique to each processor, or replicated.

The kernel build system can do some fairly rudimentary static analysis to ensure that pointers passed from userspace are always checked before use, and that pointers into kernel space are not passed to user space. This relies on such pointers being declared with `__user` or `__kernel`. It can also check that variables that are intended as fixed shape bitwise entities are always used that way—useful for bi-endian architectures like ARM.

- Extra iterators
  - `type_name_foreach()`
- Extra O-O accessors
  - `container_of()`
- Macros to register Object initialisers

Object-oriented techniques are used throughout the kernel, but implemented in C.

Almost every aggregate data structure, from lists through trees to page tables has a defined type-safe iterator.

And there's a new built-in, `container_of` that, given a type and a member, returns a typed pointer to its enclosing object.

In addition there is a family of macros to register initialisation functions. These are ordered (early, console, devices, then general), and will run in parallel across all available processors within each class.



- Massive use of inline functions
- Quite a big use of CPP macros
- Little `#ifdef` use in code: rely on optimizer to elide dead code.

The kernel is written in a style that does not use `#ifdef` in C files. Instead, feature test constants are defined that evaluate to zero if the feature is not desired; the GCC optimiser will then eliminate any resulting dead code. Because the kernel is huge, but not all files are included in every build, there has to be a way to register initialisation functions for the various components. The Linux kernel is quite object-oriented internally; but because it runs on the bare metal functions that would usually be provided by language support have to be provided by the OS, or open coded. The `container_of()` macro is a way to access inheritance; and the `xxx_initcall()` macros are a way to handle initialisation. Obviously, initialisation has to be ordered carefully; but after interrupts are set up, all the processors are on line, and the system has a console, the remaining device initialisers are run; then all the general initialisers.

## Goals:

- $O(1)$  in number of runnable processes, number of processors
  - good uniprocessor performance
- ‘fair’
- Good interactive response
- topology-aware

Because Linux runs on machines with up to 4096 processors, any scheduler must be scalable, and preferably  $O(1)$  in the number of runnable processes. It should also be ‘fair’ — by which I mean that processes with similar priority should get similar amounts of time, and no process should be starved. In addition, it should not load excessively a low-powered system with only a single processor (for example, in your wireless access point); and, at a higher level, applications should not be able to get more CPU by spawning more threads/processes.

Because Linux is used by many for desktop/laptop use, it should give good interactivity, and respond ‘snappily’ to mouse/keyboard even if that compromises absolute throughput.

And finally, the scheduler should be aware of the caching, packaging and memory topology of the system, so it when it migrates tasks, it can keep them close to the memory they use, and also attempt to save power by keeping whole packages idle where possible.

## Implementation:

- Changes from time to time.
- Currently 'CFS' by Ingo Molnar.

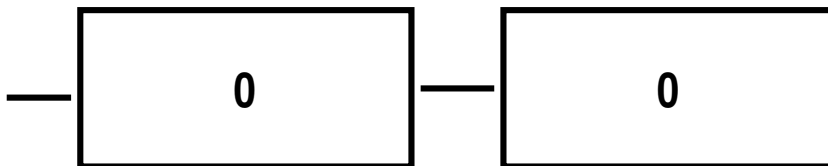
Linux has had several different schedulers since it was first released. The first was a very simple scheduler similar to the MINIX scheduler. As Linux was deployed to larger, shared, systems it was found to have poor fairness, so a very simple dual-entitlement scheduler was created.

## Dual Entitlement Scheduler

Running



Expired



The idea here was that there were two queues: a deserving queue, and an undeserving queue. New and freshly woken processes were given a timeslice based on their 'nice' value. When a process's timeslice was all used up, it was moved to the 'undeserving' queue. When the 'deserving' queue was empty, a new timeslice was given to each runnable process, and the queues were swapped. (A very similar scheduler, but using a weight tree to distribute time slice, was used in Irix 6)

The main problem with this approach was that it was  $O(n)$  in the number of runnable and running processes—and on the big iron with 1024 processors, that was too slow. So it was replaced in the early 2.6 kernels with an  $O(1)$  scheduler, that was replaced in turn (when it gave poor interactive performance on small machines) with the current 'Completely Fair Scheduler'

CFS:

1. Keep tasks ordered by effective CPU runtime weighted by nice in red-black tree
2. Always run left-most task.

Devil's in the details:

- Avoiding overflow
- Keeping recent history
- multiprocessor locality
- handling too-many threads
- Sleeping tasks
- Group hierarchy

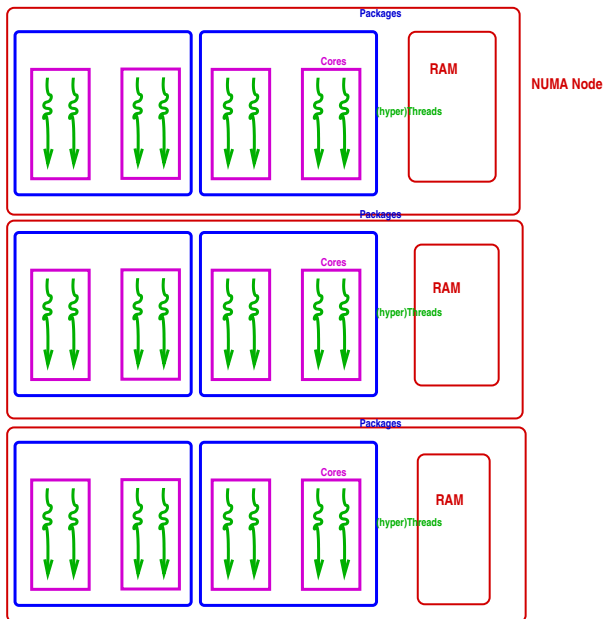
The scheduler works by keeping track of run time for each task. Assuming all tasks are cpu bound and have equal priority, then all should run at the same rate. On a sufficiently parallel machine, they would always have equal runtime.

The scheduler keeps a period during which all runnable tasks should get a go on the processor — this period is by default 6ms scaled by the log of the number of available processors. Within a period, each task gets a time quantum weighted by its *nice*. However there is a minimum quantum; if the machine is overloaded, the period is stretched so that the minimum quantum is 0.75ms.

To avoid overflow, the scheduler tracks 'virtual runtime' instead of actual; virtual runtime is normalised to the number of running tasks. It is also adjusted regularly to avoid overflow.

Tasks are kept in vruntime order in a red-black tree. The leftmost node then has the least vruntime so far; newly activated entities also go towards the left — short sleeps (less than one period) don't affect vruntime; but after awaking from a long sleep, the vruntime is set to the current minimum vruntime if that

is greater than the task's current vruntime. Depending on how the scheduler has been configured, the new task will be scheduled either very soon, or at the end of the current period.



Your typical system has hardware threads as its bottom layer. These share functional units, and all cache levels. Hardware threads share a *core*, and there can be more than one core in a *package* or *socket*. Depending on the architecture, cores within a socket may share memory directly, or may be connected via separate memory buses to different regions of physical memory. Typically, separate sockets will connect to different regions of memory.

## Locality Issues:

- Best to reschedule on same processor (don't move cache footprint, keep memory close)
  - Otherwise schedule on a 'nearby' processor
- Try to keep whole sockets idle
- Somehow identify cooperating threads, co-schedule on same package?

The rest of the complications in the scheduler are for hierarchical group-scheduling, and for coping with non-uniform processor topology.

I'm not going to go into group scheduling here (even though it's pretty neat), but its aim is to allow schedulable entities (at the lowest level, tasks or threads) to be gathered together into higher level entities according to credentials, or job, or whatever, and then schedule those entities against each other.

Locality, however, is really important. You'll recall that in a NUMA system, physical memory is spread so that some is local to any particular processor, and other memory is a long way off. To get good performance, you want as much as possible of a process's working set in local memory. Similarly, even in an SMP situation, if a process's working set is still (partly) in-cache it should be run on a processor that shares that cache.

Linux currently uses a 'first touch' policy: the first processor to write to a page causes the frame for the page to be allocated from that processor's nearest memory. On `fork()`, the new process's memory is allocated from the same node as its parent, and it runs on the same node (although not necessarily on the same core). `exec()` doesn't change this (although there is an API



to allow a process to migrate before calling `exec()`. So how do processors other than the boot processor ever get to run anything?  
The answer is in runqueue balancing.

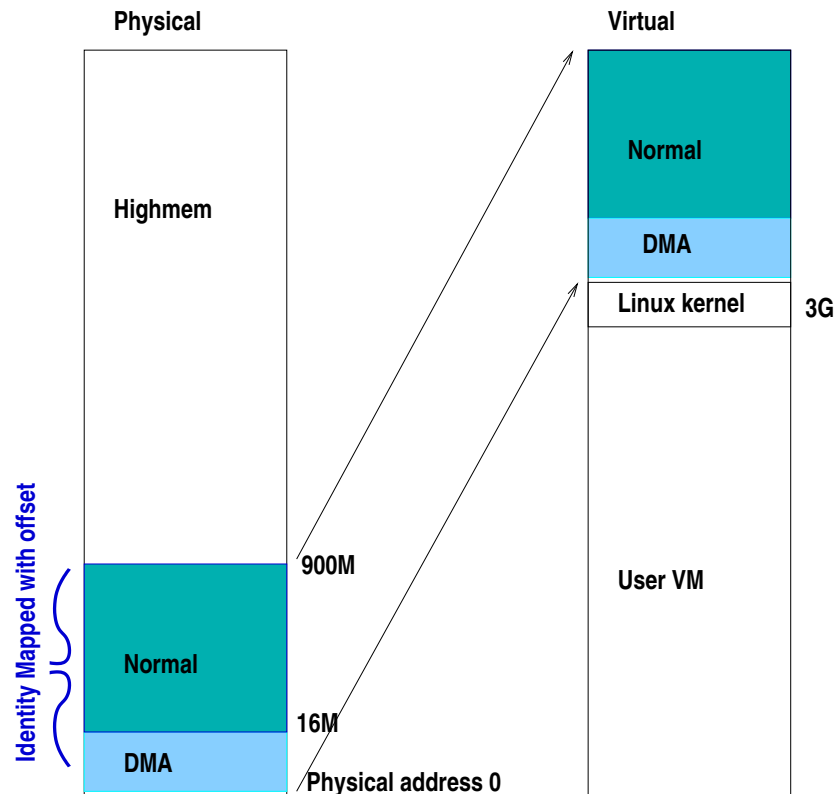
- One queue per processor (or hyperthread)
- Processors in hierarchical ‘domains’
- Load balancing per-domain, bottom up
- Aims to keep whole domains idle if possible (power savings)

There is one runqueue for each lowest schedulable entity (hyperthread or processor). These are grouped into ‘domains’. Each domain has its ‘load’ updated at regular intervals (where load is essentially sum of vruntime/number of processors).

One of the idle processors is nominated the ‘idle load balancer’. When a processor notices that rebalancing is needed (for example, because it is overloaded), it kicks the idle load balancer. The idle load balancer finds the busiest domains, and tries to move tasks around to fill up idle processors near the busiest domain. It needs more imbalance to move a task to a completely idle node than to a partly idle node.

Solving this problem perfectly is NP-hard — it’s equivalent to the bin-packing problem — but the heuristic approach seems to work well enough most of the time.

## Memory in zones



Some of Linux's memory handling is to account for peculiarities in the PC architecture. To make things simple, as much memory as possible is mapped at a fixed offset, at least on X86-derived processors. Because of legacy devices that could only do DMA to the lowest 16M or memory, the lowest 16M are handled specially as **ZONE\_DMA** — drivers for devices that need memory in that range can request it. (Some architectures have no physical memory in that range; either they have IOMMUs or they do not support such devices). The Linux kernel maps itself in, and has access to all of user virtual memory. In addition, as much physical memory as possible is mapped in with a simple offset. This allows easy access for in-kernel use of physical memory (e.g., for page tables or DMA buffers).

Any physical memory that cannot be mapped (e.g., because there is more than 4G of RAM on a 32-bit machine) is termed 'Highmem' and is mapped in on an ad-hoc basis. It is possible to compile the kernel with no 'Normal' memory, to allow all of the 4G 32-bit virtual address space to be allocated to userspace, but this comes with a performance hit.

The boundary between user and kernel can be set at configuration time; for

64-bit systems it's at  $2^{63}$  – i.e., all addresses with the highest bit set are for the kernel.

- Direct mapped pages become *logical addresses*
  - `--pa()` and `--va()` convert physical to virtual for these
- small memory systems have all memory as logical
- More memory  $\rightarrow$   $\Delta$  kernel refer to memory by `struct page`

Direct mapped pages can be referred to by *logical addresses*; there are a simple pair of macros for converting between physical and logical addresses for these. Anything not mapped must be referred to by a `struct page` and an offset within the page. There is a `struct page` for every physical page (and for some things that aren't memory, such as MMIO regions). A `struct page` is less than 10 words (where a word is 64 bits on 64-bit architectures, and 32 bits on 32-bit architectures).

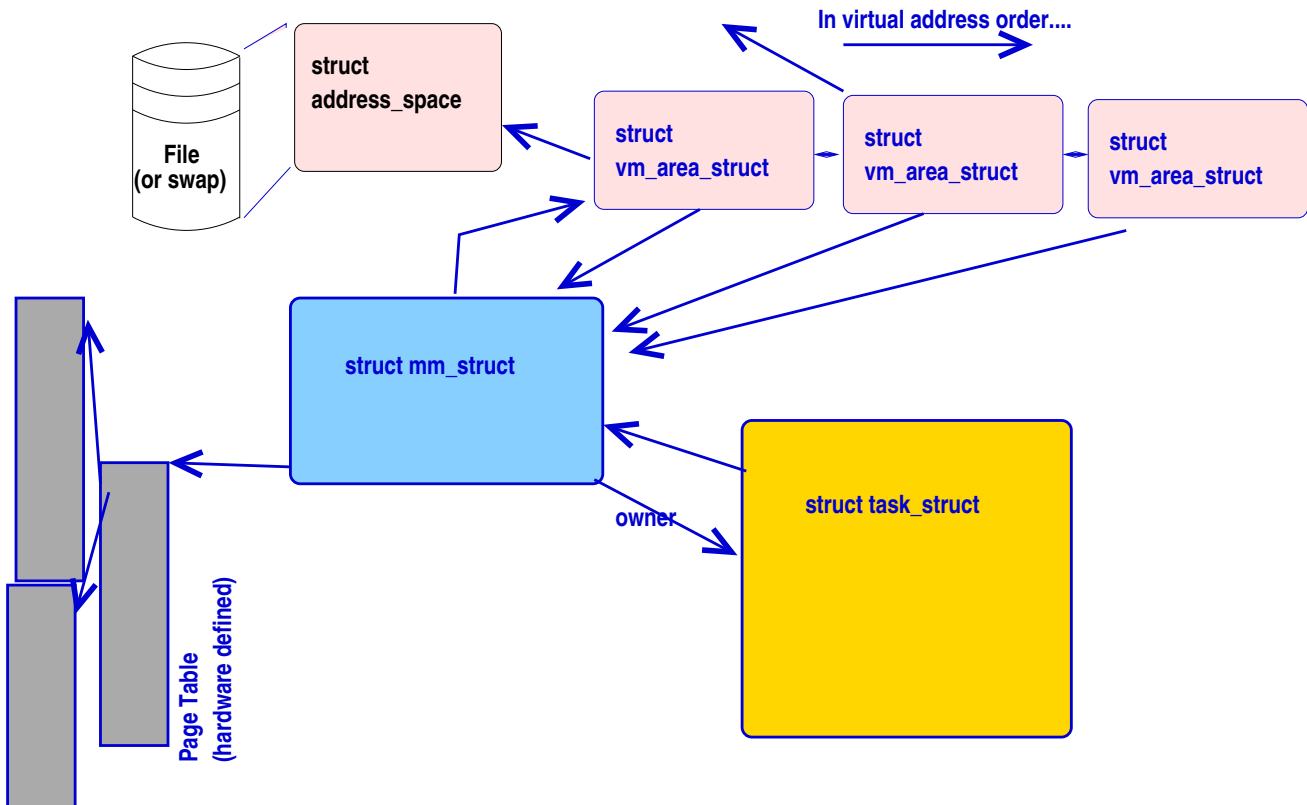
`struct page:`

- Every frame has a `struct page` (up to 10 words)
- Track:
  - flags
  - backing address space
  - offset within mapping *or* freelist pointer
  - Reference counts
  - Kernel virtual address (if mapped)

A **struct page** lives on one of several lists, and is in an array from which the physical address of the frame can be calculated.

Because there has to be a **struct page** for every frame, there's considerable effort put into keeping them small. Without debugging options, for most architectures they will be 6 words long; with 4k pages and 64bit words that's a little over 1% of physical memory in this table.

A frame can be on a free list. If it is not, it will be in an active list, which is meant to give an approximation to LRU for the frames. The same pointers are overloaded for keeping track of compound frames (for SuperPages). Free lists are organised per memory domain on NUMA machines, using a buddy algorithm to merge pages into SuperPages as necessary.



Some of the structures for managing memory are shown in the slide. What's not visible here are the structure for managing swapping out, NUMA locality and SuperPages.

There is one `task_struct` for each thread of control. Each points to an `mm_struct` that describes the address space the thread runs in. Processes can be multi-threaded; one, the first to have been created, is the *thread group leader*, and is pointed to by the `mm_struct`. The `struct mm_struct` also has a pointer to the page table for this process (the shape of which is carefully abstracted out so that access to it is almost architecture-independent, but it always has to be a tree), a set of mappings held both in a red-black tree (for rapid access to the mapping for any address) and in a double linked list (for traversing the space).

Each *VMA* (*virtual memory area*, or `struct vm_area_struct`) describes a contiguous mapped area of virtual memory, where each page within that area is backed (again contiguously) by the same object, and has the same permissions and flags. You could think of each `mmap()` call creating a new VMA. Any `munmap()` calls that split a mapping, or `mprotect()` calls that

change part of a mapping can also create new VMAs.



## Address Space:

- Misnamed: means collection of pages mapped from the same object
- Tracks inode mapped from, radix tree of pages in mapping
- Has ops (from file system or swap manager) to:
  - dirty** mark a page as dirty
  - readpages** populate frames from backing store
  - writepages** Clean pages — make backing store the same as in-memory copy
  - migratepage** Move pages between NUMA nodes

Each VMA points into a **struct address\_space** which represents a map-pable object. An **address\_space** also tracks which pages in the page cache belong to this object.

Most pages will either be backed by a file, or will be anonymous memory. Anonymous memory is either unbacked, or is backed by one of a number of swap areas.

- Special case in-kernel faults
- Find the VMA for the address
  - segfault if not found (unmapped area)
- If it's a stack, extend it.
- Otherwise:
  1. Check permissions, SIG\_SEGV if bad
  2. Call `handle_mm_fault()`:
    - walk page table to find entry (populate higher levels if nec. until leaf found)
    - call `handle_pte_fault()`

When a fault happens, the kernel has to work out whether this is a normal fault (where the page table entry just isn't instantiated yet) or is a userspace problem. Kernel faults are rare: they should occur only in a few special cases, and when accessing user virtual memory. They are handled specially.

It does this by first looking up the VMA in the red-black tree. If there's no VMA, then this is an unmapped area, and should generate a segmentation violation. If it's next to a stack segment, and the faulting address is at or near the current stack pointer, then the stack needs to be extended.

If it finds the VMA, then it checks that the attempted operation is allowed — for example, writes to a read-only operation will cause a Segmentation Violation at this stage. If everything's OK, the code invokes `handle_mm_fault()` which walks the page table in an architecture-agnostic way, populating 'middle' directories on the way to the leaf. Transparent SuperPages are also handled on the way down.

Finally `handle_pte_fault()` is called to handle the fault, now it's established that there really is a fault to handle.

`handle_pte_fault ()` : Depending on PTE status, can

- provide an anonymous page
- do copy-on-write processing
- reinstantiate PTE from page cache
- initiate a read from backing store.

and if necessary flushes the TLB.

There are a number of different states the pte can be in. Each PTE holds flags that describe the state.

The simplest case is if the PTE is zero — it has only just been instantiated. In that case if the VMA has a fault handler, it is called via `do_linear_fault ()` to instantiate the PTE. Otherwise an anonymous page is assigned to the PTE. If this is an attempted write to a frame marked copy-on-write, a new anonymous page is allocated and copied to.

If the page is already present in the page cache, the PTE can just be reinstantiated — a ‘minor’ fault. Otherwise the VMA-specific fault handler reads the page first — a ‘major’ fault.

If this is the first write to an otherwise clean page, it’s corresponding `struct page` is marked dirty, and a call is made into the writeback system — Linux tries to have no dirty page older than 30 seconds (tunable) in the cache.

Three kinds of device:

1. Platform device
2. enumerable-bus device
3. Non-enumerable-bus device

There are essentially three kinds of devices that can be attached to a computer system:

1. *platform devices* exist at known locations in the system's IO and memory address space, with well known interrupts. An example are the COM1 and COM2 ports on a PC.
2. Devices on a bus such as PCI or USB have unique identifiers that can be used at run-time to hook up a driver to the device. It is possible to enumerate all devices on the bus, and find out what's attached.
3. Devices on a bus such as *i<sup>2</sup>c* or ISA have no standard way to query what they are.

## Enumerable buses:

```
static DEFINE_PCI_DEVICE_TABLE(cp_pci_tbl) = {
{ PCI_DEVICE(PCI_VENDOR_ID_REALTEK, PCI_DEVICE_ID_REALTEK_813)
{ PCI_DEVICE(PCI_VENDOR_ID_TTTECH, PCI_DEVICE_ID_TTTECH_MC322)
{ } ,
};
MODULE_DEVICE_TABLE(pci, cp_pci_tbl);
```

Each driver for a bus that identifies devices by some kind of ID declares a table of IDs of devices it can driver. You can also specify device IDs to bind against as a module parameter.

Driver interface:

**init** called to register driver

**exit** called to deregister driver, at module unload time

**probe ()** called when bus-id matches; returns 0 if driver claims device

**open, close, etc** as necessary for driver class

All drivers have an initialisation function, that, even if it does nothing else, calls a `bus_register_driver()` function to tell the bus subsystem which devices this driver can manage, and to provide a vector of functions.

Most drivers also have an `exit ()` function, that deregisters the driver.

When the bus is scanned (either at boot time, or in response to a hot-plug event), these tables are looked up, and the 'probe' routine for each driver that has registered interest is called.

The first whose probe is successful is bound to the device. You can see the bindings in `/sys`

## Platform Devices:

```
static struct platform_device nslu2_uart = {  
    .name = "serial8250",  
    .id = PLAT8250_DEV_PLATFORM,  
    .dev.platform_data = nslu2_uart_data,  
    .num_resources = 2,  
    .resource = nslu2_uart_resources,  
};
```

Platform devices are made to look like bus devices. Because there is no unique ID, the platform-specific initialisation code registers platform devices in a large table.

Here's an example, from the SLUG. Each platform device is described by a **struct platform\_device** that contains at the least a name for the device, the number of 'resources' (IO or MMIO regions) and an array of those resources. The initialisation code calls **platform\_device\_register()** on each platform device. This registers against a dummy 'platform bus' using the name and ID.

The 8250 driver eventually calls **serial8250\_probe()** which scans the platform bus claiming anything with the name 'serial8250'.

non-enumerable buses: Treat like platform devices

At present, devices on non-enumerable buses are treated a bit like platform devices: at system initialisation time a table of the addresses where devices are expected to be is created; when the driver for the adapter for the bus is initialised, the bus addresses are probed.



- Describe board+peripherals
  - replaces ACPI on embedded systems

Recent kernels are moving away from putting platform devices into C code, in favour of using a *flattened device tree*, which describes the topology of buses, devices, clocks and regulators, so a single kernel can run on more than one board.

- I've told you status today
  - Next week it may be different
- I've simplified a lot. There are many hairy details

I'm assuming:

- You've already looked at ext[234]-like filesystems
- You've some awareness of issues around on-disk locality and I/O performance
- You understand issues around avoiding on-disk corruption by carefully ordering events, and/or by the use of a Journal.

If you don't understand any of this – stop me now for a quick refresher course!

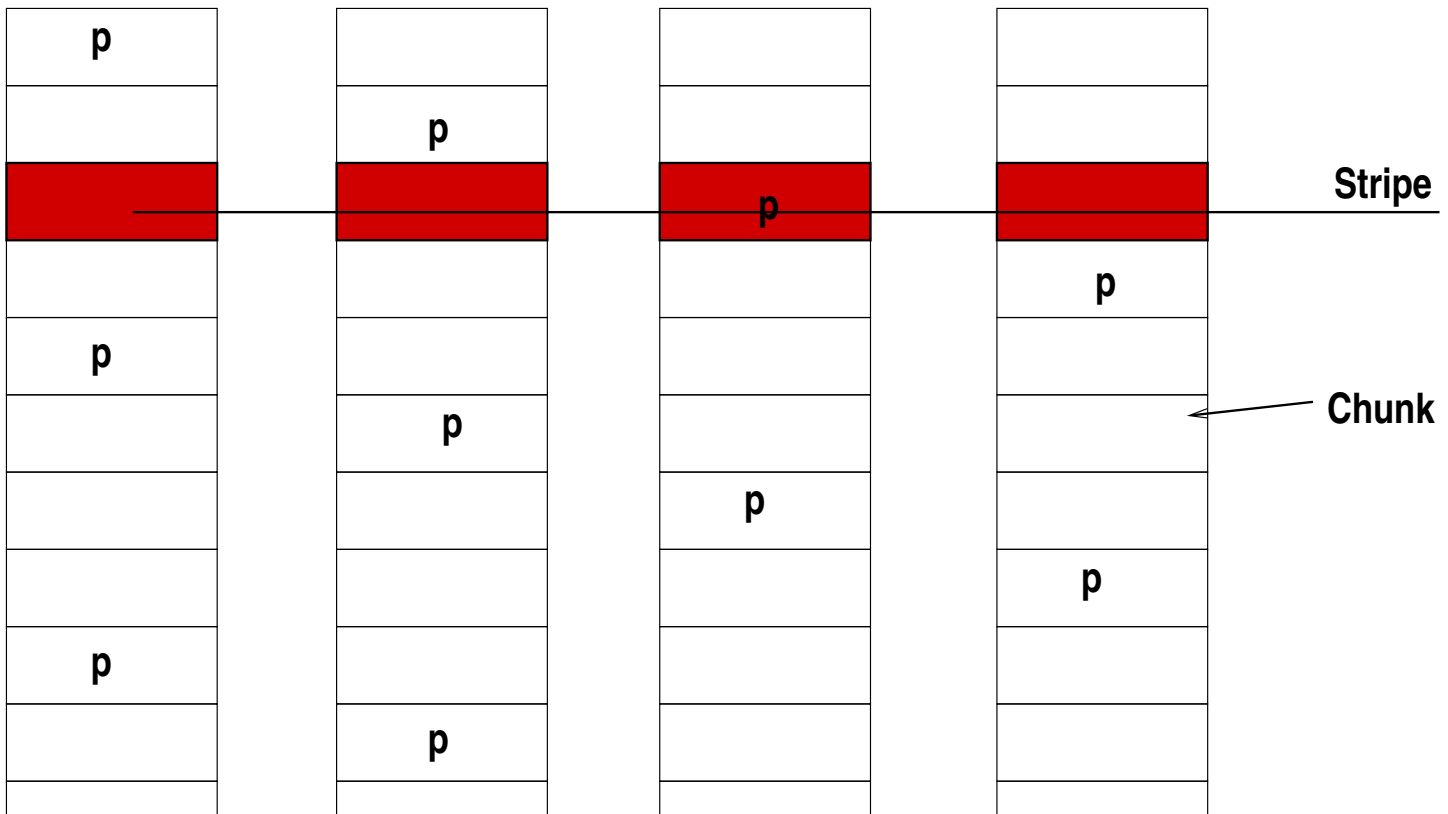
- Optimised for use on spinning disk
- RAID optimised (especially XFS)
- Journals, snapshots, transactions...

Most 'traditional' file systems optimise for spinning disk. They arrange the disk into regions, and try to put directory entries, inodes and disk blocks into the same region.

Some also take RAID into account, and try to localise files so that a reasonably-sized single read can be satisfied from one spindle; and writes affect only two spindles (data plus parity).

Many use a journal for filesystem consistency. The journal is either in a fixed part of the disk (ext[34], XFS, etc) or can wander over the disk (reiserFS); in most cases only metadata is journalled.

And more advanced file systems (XFS, VxFS, etc) arrange related I/O operations into transactions and use a three-phase commit internally to provide improved throughput in a multiprocessor system.

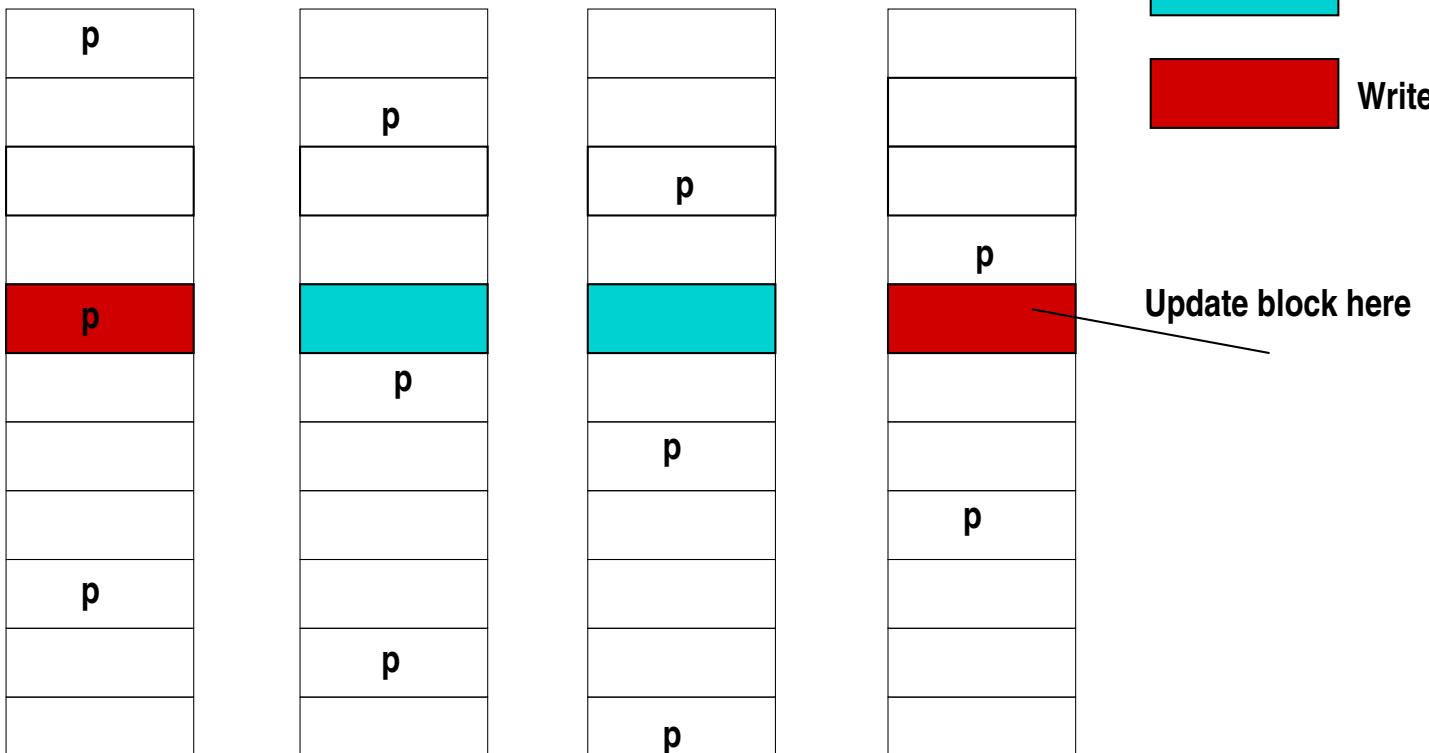


RAID (Redundant array of inexpensive discs) is a way to get more reliability and performance out of cheap disks. It's a truism that you can pick any two from reliable, fast, large, and inexpensive.

The idea is to use multiple discs together. There are obviously a variety of arrangements for that; either simply concatenating the discs (raid 0); simply mirroring some discs (each has the same content, raid 1), or more complex parity or parity plus mirroring (raid 5 or 6; raid 6 has two parity blocks per stripe, raid 5 has one; raid 50 mirrors two raid 5 arrays).

For random read I/O, it's likely that the seek latency will be reduced, because adjacent reads fall on different spindles. And reliability is enhanced (providing the driver can detect errors) because there is a parity block in each stripe (for raid 5 or 6) so data can be recovered even if a drive fails.

# RAID performance



But consider writes. If you update a single block, the parity block also has to be updated. That means reading every other block in the stripe as well.

So it's best to write an entire stripe at once. Filesystems that understand RAID attempt to do this, if the stripe isn't too big.

So there is a tension between small chunk size (say, 64k) which means that large reads have to span spindles, but that allow better write and small-file performance, and larger chunk sizes (up to a couple of Mb) that allow good read performance for large contiguous chunks, but are expensive to write.

Also consider reliability. The error rate for commodity SATA and SAS drives is around  $10^{-15}$  — so after moving only 0.88PB of data you expect to see an error. Rebuilding the array after a failure will hit this after only a few hundred rebuilds — see <http://www.enterprisestorageforum.com/technology/feat> for details.

- NOR Flash
- NAND Flash
  - MTD — Memory Technology Device
  - eMMC, SDHC etc — A JEDEC standard
  - SSD, USB — and other disk-like interfaces

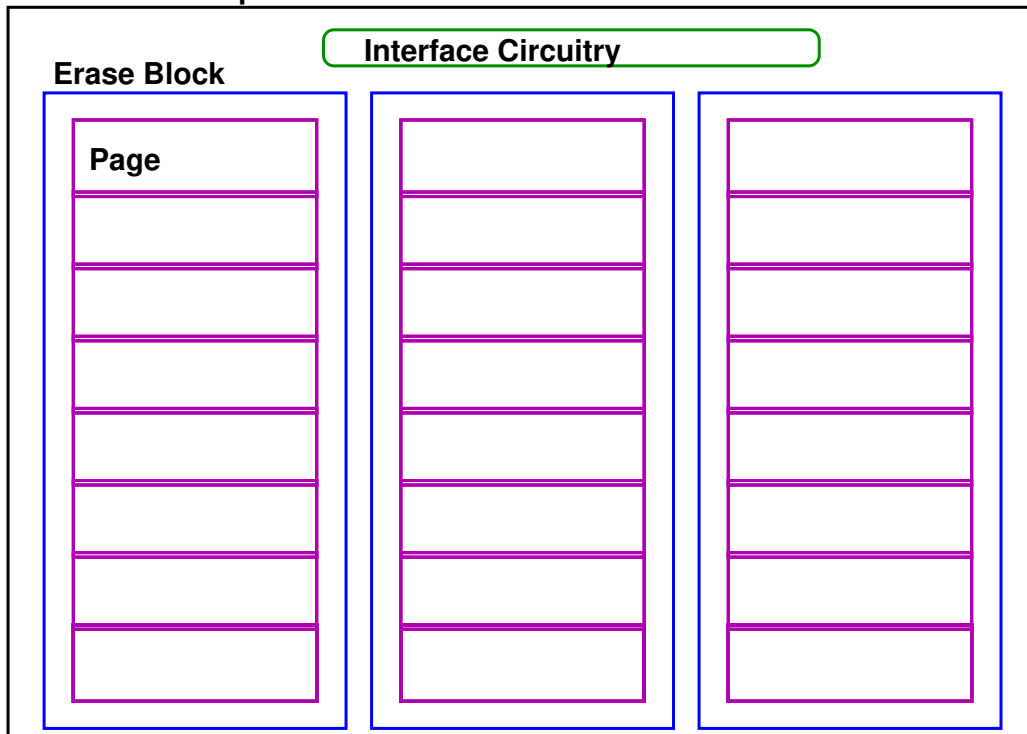
But more and more storage nowadays is no longer on spinning rust. Flash-memory-based storage has very different characteristics.

There are basically two kinds of Flash memory: NOR flash, and NAND flash. Check the Wikipedia article if you want a more detailed view of how they differ; from a systems perspective, NOR flash is byte oriented so you can treat it like (slow) RAM; NAND flash is block oriented, so you have to read/write it a block at a time, like a disk.

I'm not going to talk any more about NOR flash; NAND flash memory has much higher density, and is the common cheap(ish) flash used on-board in embedded systems, SD cards, USB sticks and SSDs.

NAND flash comes with a variety of system interfaces. The most common are the Memory Technology Device interface (MTD), the MMC (Multi-Media Card) interface (a JEDEC standard, used for eMMC on-board flash and for SD cards) and a standard disk interface (such as used in USB sticks and in SSDs).

NAND Flash Chip



NAND flash is accessed as *pages*, typically 512 bytes. (To avoid confusion with operating system pages, I'm going to refer to them as blocks). Writes write only zeroes, so data has to be erased before it is written.

Erasure happens in larger units — an erase block can be two or four megabytes, or even more. What's more, as each NAND cell can hold one (Single-level cell, SLC) or two or three (multi-level-cell, MLC) values, erase blocks can be power-of-three as well as power-of-two blocks big. This means that one cannot update a block in-place.

You'll have noticed that the available capacity on an SD card is significantly less than the rated capacity. Although the flash inside an SD card or USB stick is always a power-of-two size, it's quoted in thousands of megabytes, not 1024s of megabytes. The 'spare' capacity is used for holding metadata, and as erased blocks for use on writes. The controller cannot usually tell which blocks are unused by the filesystem, and has to assume that all blocks are always in use.

There are some other gotchas.

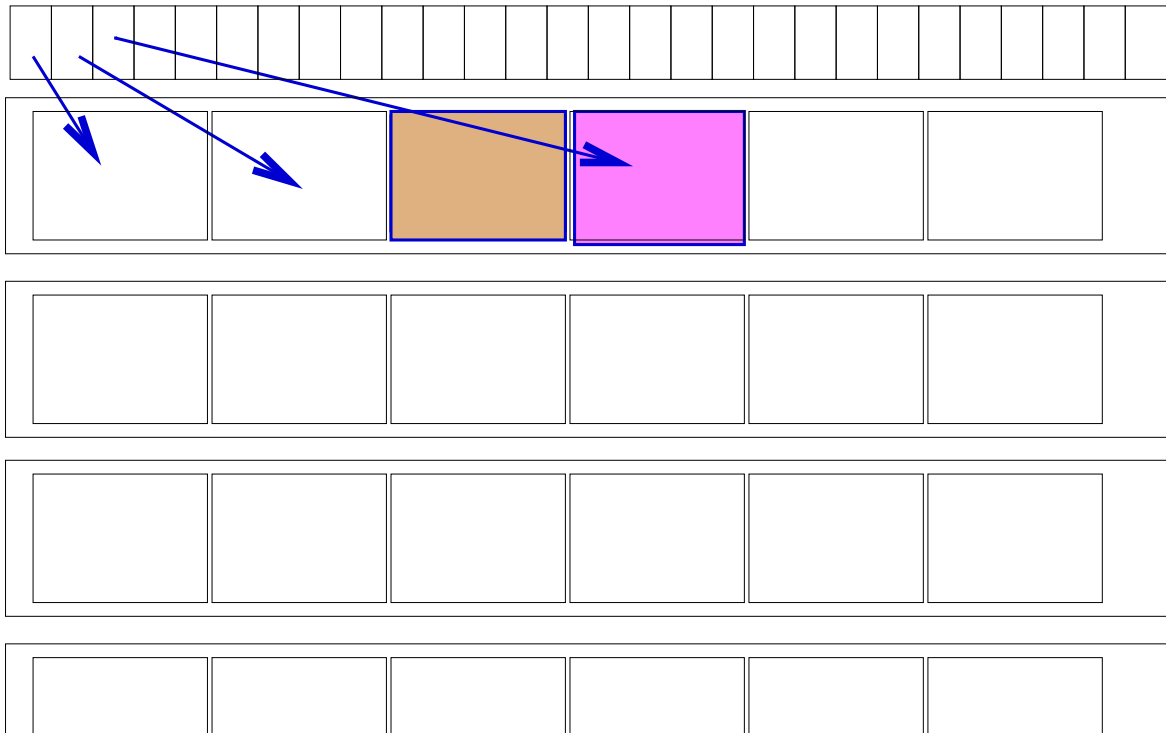
Blocks wear out. Currently available SLC NAND chips have around 100 000



erase cycles before they wear out (MLC chips are around half this); there is some research into adding heaters onto the chip to anneal and restore the cells, which would give three orders of magnitude better lifetime, but such NAND flash is not yet commercially available.

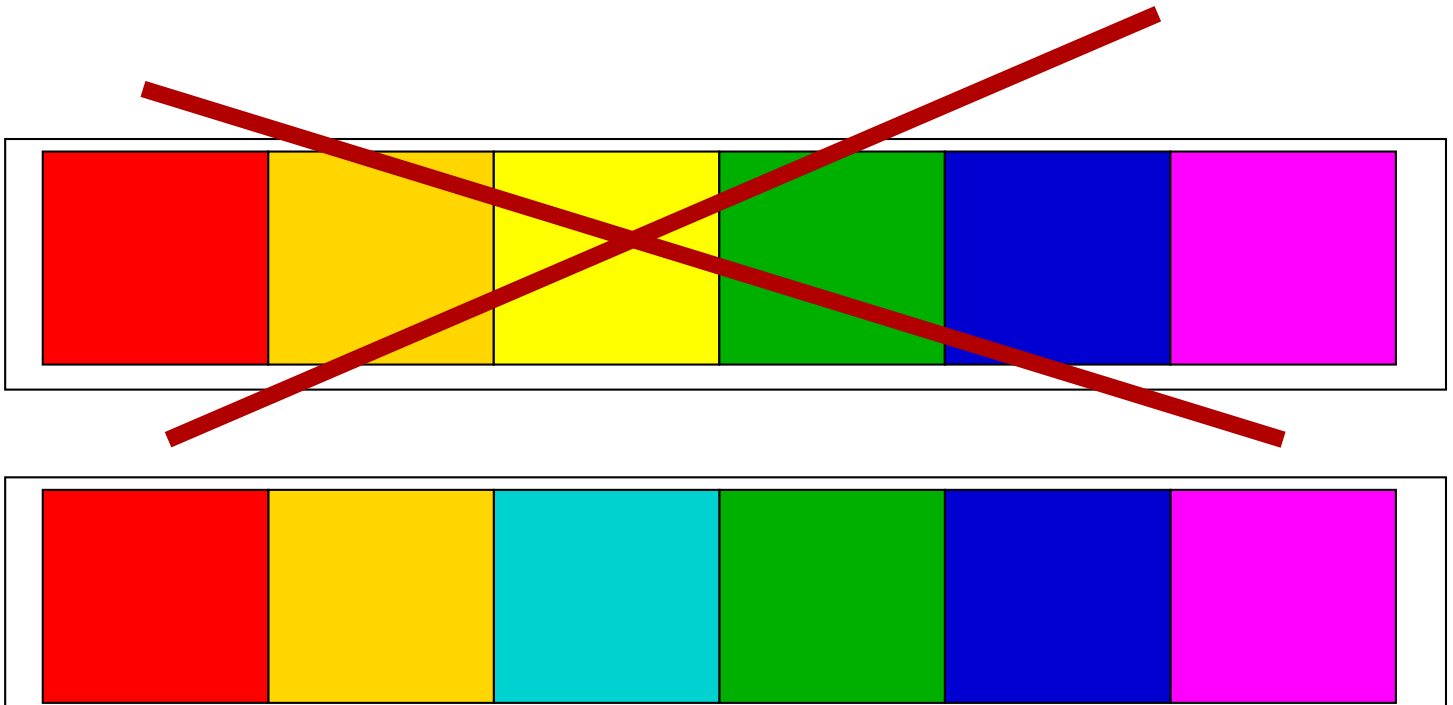
In addition, reading can disturb adjacent cells. Read-Disturb operates over a few hundred thousand read operations, and will cause errors, not in the cells being read, but in adjacent cells. Erasure fixes this problem.

These two characteristics together mean that flash has to be managed, either by a filesystem that understands FLASH characteristics (e.g., JFFS2, YAFFS), or by a wear-levelling translation layer and a garbage collector.

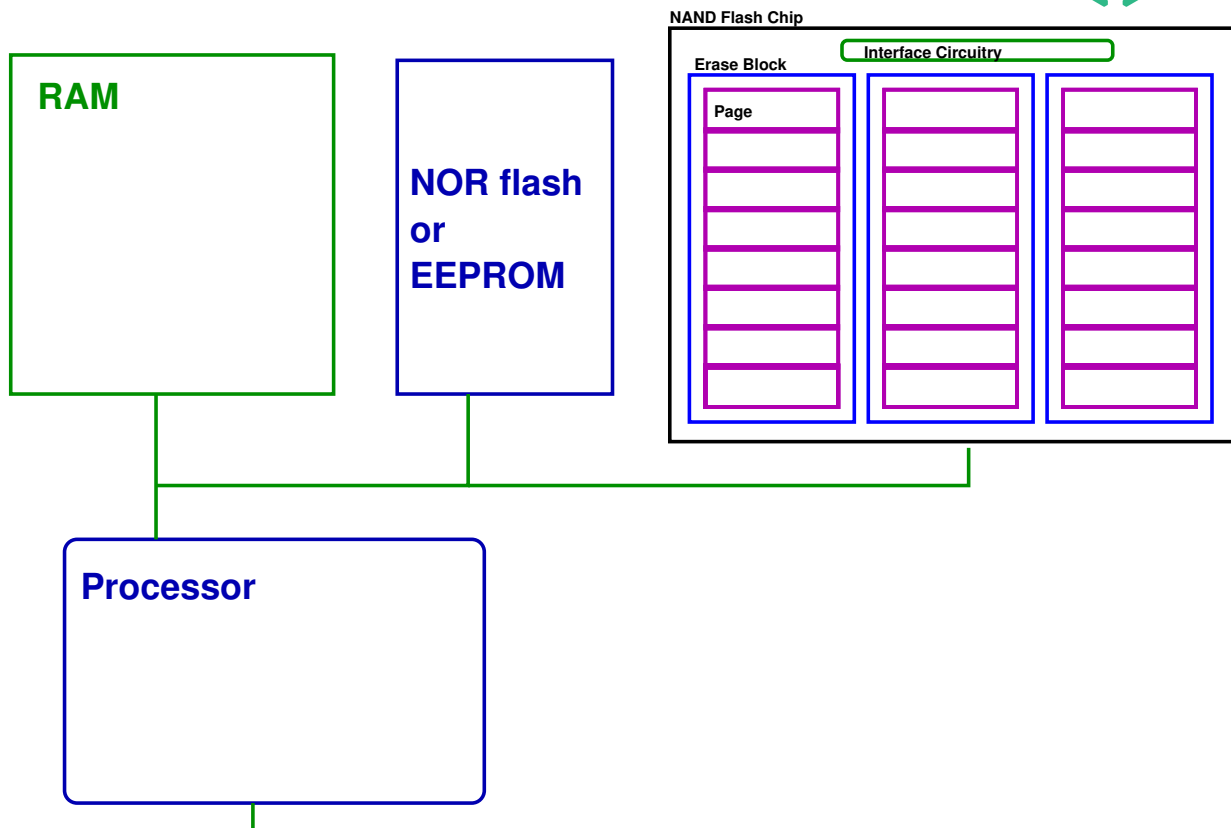


To update a block, the controller first finds an empty erased block, writes the update to it, updates the FTL bookkeeping pointer, and marks the old block as garbage to be collected later.

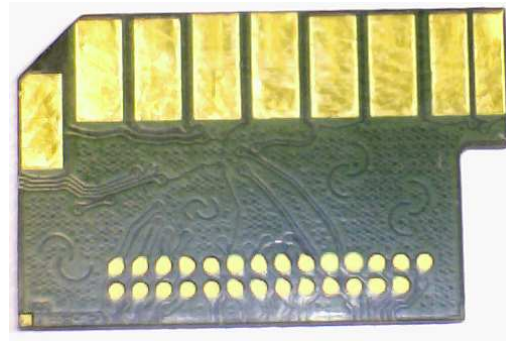
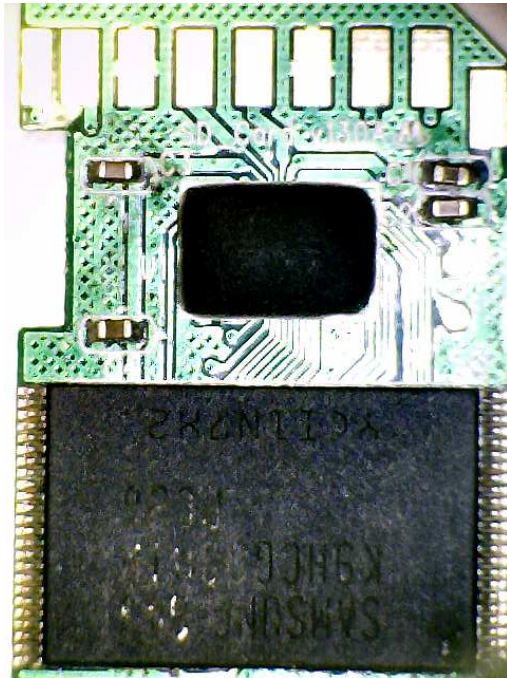
But doing translation on 512 byte chunks is really inefficient. It means you have to keep a *lot* of data around, and because most filesystems write in larger chunks, you have to do lots of book-keeping for every write. For example, FAT filesystems always write an entire cluster size (typically 64k for FAT32, but could be more). So every write involves 128 updates, and modifies 256 words of storage, if the controller maps individual blocks.



So instead of mapping per-block, most SD cards use what are variously called segments or allocation units. An allocation unit is a strict multiple of the erase-size, typically 4M or 8M.



An SD card or USB flash memory contains a controller in addition to the NAND flash chip. The controller is typically a custom-designed low-power microcontroller, together with a few words of non-volatile memory (perhaps NOR flash), some RAM, and some interface and power-management circuitry. It is often mounted chip-on-board. One of the big differences between cards is the controller plus its firmware, rather than the flash memory.



Here are some pictures. The one on the left is a fake Lenxs 16G SD card; you can see the 8Gb (Samsung) NAND flash chip, and the Chip-On-Board (COB) controller under the epoxy blob,

The one on the right is a SanDisk 1G card. Everything is encapsulated between the layers of the card, so there's not a lot to see.

I've been told that SanDisk design and fabricate their own flash and controllers; some other companies use third-party controllers and flash.

# The controller:



- Presents illusion of 'standard' block device
- Manages writes to prevent wearing out
- Manages reads to prevent read-disturb
- Performs garbage collection
- Performs bad-block management

Mostly documented in Korean patents referred to by US patents!

The controller has to do a number of things, at speed. Total power consumption is fairly small — up to 2.88 Watts for a UHS-1 card; much less for standard cards. So it tends to be a fairly limited mmu-less microcontroller, such as the MCS-51 (or more usually, one of its variants).

The main things the controller has to do in its firmware is to present the illusion of a standard block device, while managing (transparently) the fact that flash cannot be overwritten in-place.

It also has to be aware when an SD card or USB stick is wrenched from its socket. The power pins on the card are longer than the others; this gives a few milliseconds of power to finalise writes, and to update the controller's NVRAM with the block address of any metadata in the Flash.

Two ways:

- Remap blocks when they begin to fail (bad block remapping)
- Spread writes over all erase blocks (wear levelling)

In practice both are used.

Also:

- Count reads and schedule garbage collection after some threshold

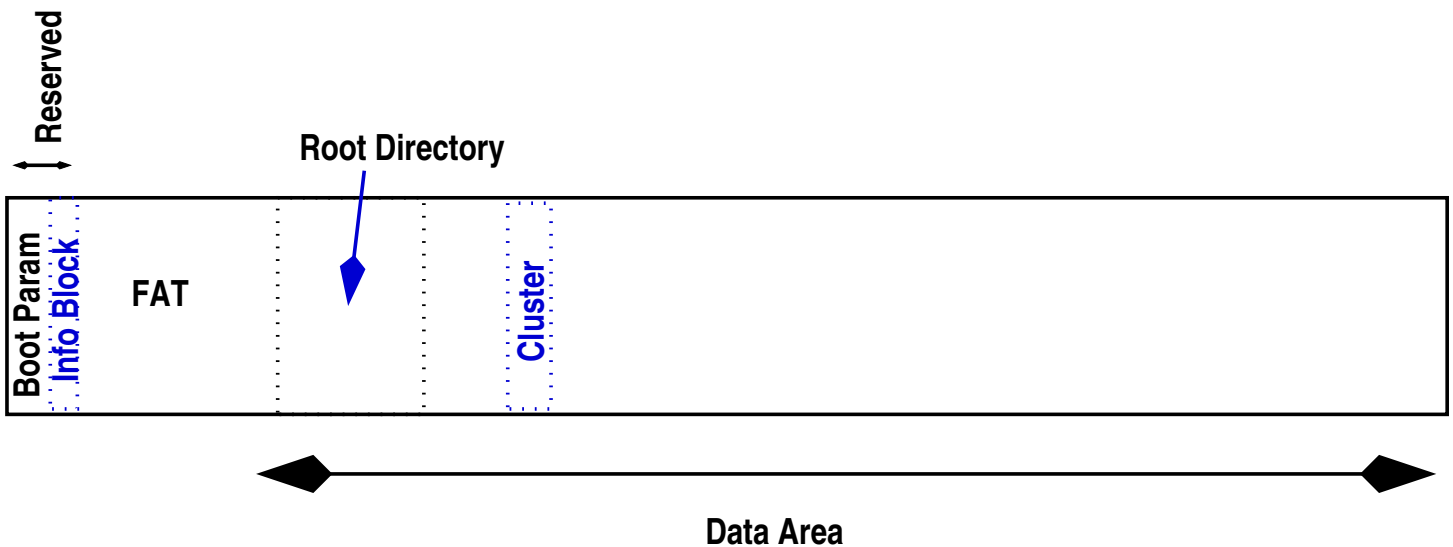
There are two ways to extend the limited write lifetime of a flash block. The first is to use ECC to detect cells going bad, and remap a block when errors start to appear. The second is to spread out writes over all blocks. By combining with garbage collection, such *wear levelling* can be achieved with low overhead.

- Typically use FAT32 (or exFAT for sdxc cards)
- Always do cluster-size I/O (64k)
- First partition segment-aligned

**Conjecture** Flash controller optimises for the preformatted FAT fs

Removable Flash devices almost always are preformatted with a FAT file system. Typically, the first partition starts on an erase boundary (thus being a good hint for the size of the erase block), and uses a cluster size that is a good compromise for the allocation unit size used by the controller. It's likely that the controller will optimise for the write patterns experienced when using a FAT fs.





The reserved area at the start of the filesystem contains a Boot Parameter Block (essentially the same as a superblock on a UNIX file system). Key parameters are the location, size, and number of FATs (file allocation tables), the location of the root directory, and the cluster size.

All disk allocation and I/O is done in cluster-size chunks.

We looked at a lot of different preformatted cards. All used a 64k cluster size, and had the FATs in the second erase block of the disk. The first erase block was used only for the MBR.

The Directory entry for a file contains the index of its first cluster. The first cluster index is then used to look up the next cluster in the FAT, as a chain. So extending a file involves writing the data into the data area, and cluster indices into the FAT, and finally updating the directory entry with the file size.

**Conjecture** The controller has some number of buffers it treats specially, to allow more than one write locus.

Given that the SD I/O speed is higher than the speed to write to Flash, the controller must have RAM to buffer the writes.

We timed writes at various offsets from each other to determine the size of the buffer (we expect that two adjacent writes within the same open buffer will be fast, but when the buffer finally is committed to flash, it'll be slower — which is what we found), and to discover how many write loci could be handled simultaneously.

# Testing SDHC cards



We ran fairly extensive benchmarks on four full-sized cards using a Sabre-Lite: a Kingston 32Gb class 10, a Toshiba 16Gb class 10, and two different SanDisk UHS-1 cards: an Extreme, and an Extreme Pro.

# SD Card Characteristics

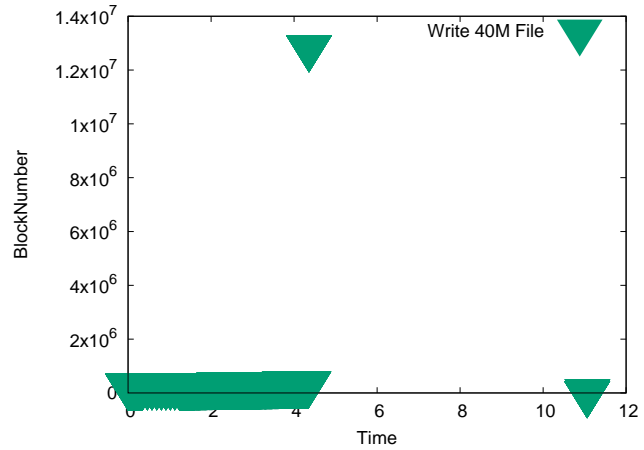
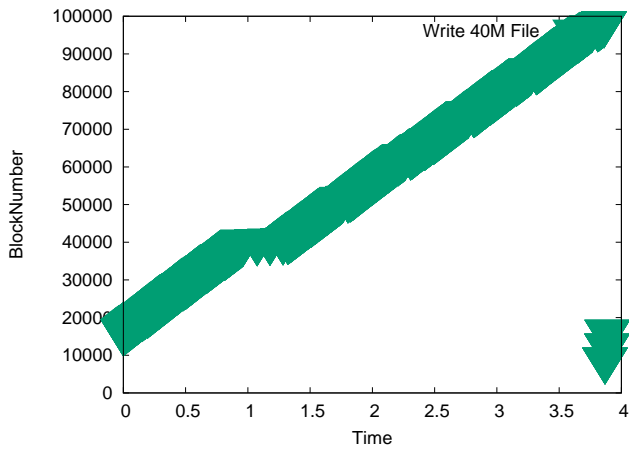


Card	Price/G	#AU	Page size	Erase Size
Kingston Class 10	\$0.80	2	128k	4M
Toshiba Class 10	\$1.20	2	64k	8M
SanDisk Extreme UHS-1	\$5.00	9	64k	8M
SanDisk Extreme Pro UHS-1	\$6.50	9	16k	4M

The Toshiba card we measured had two open allocation units, but didn't seem to treat the FAT area specially.

SanDisk and Samsung cards had between six and nine allocation areas, and didn't seem to treat the FAT area specially.

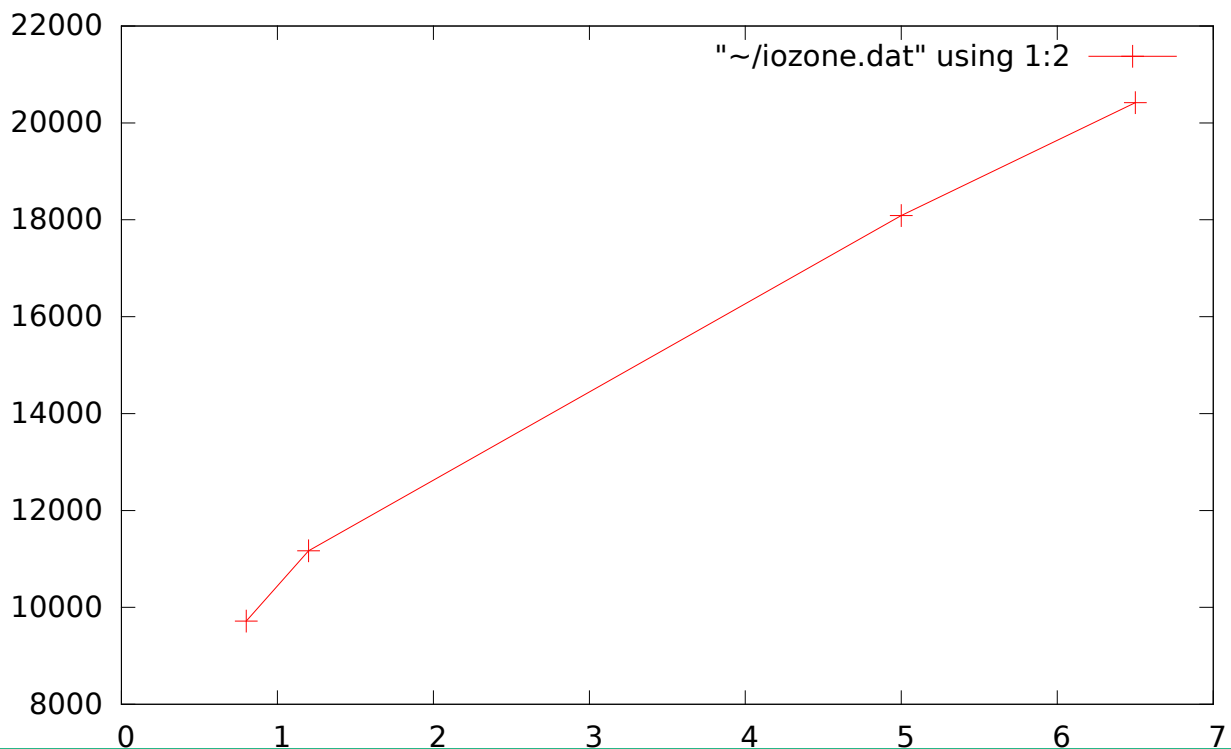
For the Kingston cards, which have only two open allocation units, one of them appears to be pinned to the FAT area. So you can do fast writes to a single open file, extending it in the FAT area, and in the data area. But multiple files are slow, and any filesystem that doesn't use the FAT area in the same way will be slow.



(On Toshiba Exceria card)

You can see from the scatter plots that creating a file on a FAT32 file system is almost ideal for a two open-allocation-unit system. All writes are in 64k chunks; the FAT remains open for extensions; and the buffer becomes available for the directory entry after closing off the last write to the file's data. However, ext4 and XFS don't behave like this at all. Ext4 on an unaged filesystem is pretty good at maintaining locality, but has some writes a while after the file is closed: one to update the free-block bitmap, one for the inode, and one for the journal.

# Write Patterns: File Create



Bonnie showed very little difference between the cards; large write performance was around proportional to the price per gigabyte, other benchmarks were around the same for all cards.

- By Samsung
- ‘Use on-card FTL, rather than work against it’
- Cooperate with garbage collection
- Use FAT32 optimisations

F2FS was designed by Samsung for SD cards and other higher-level Flash devices. It works well on USB sticks and SD cards.

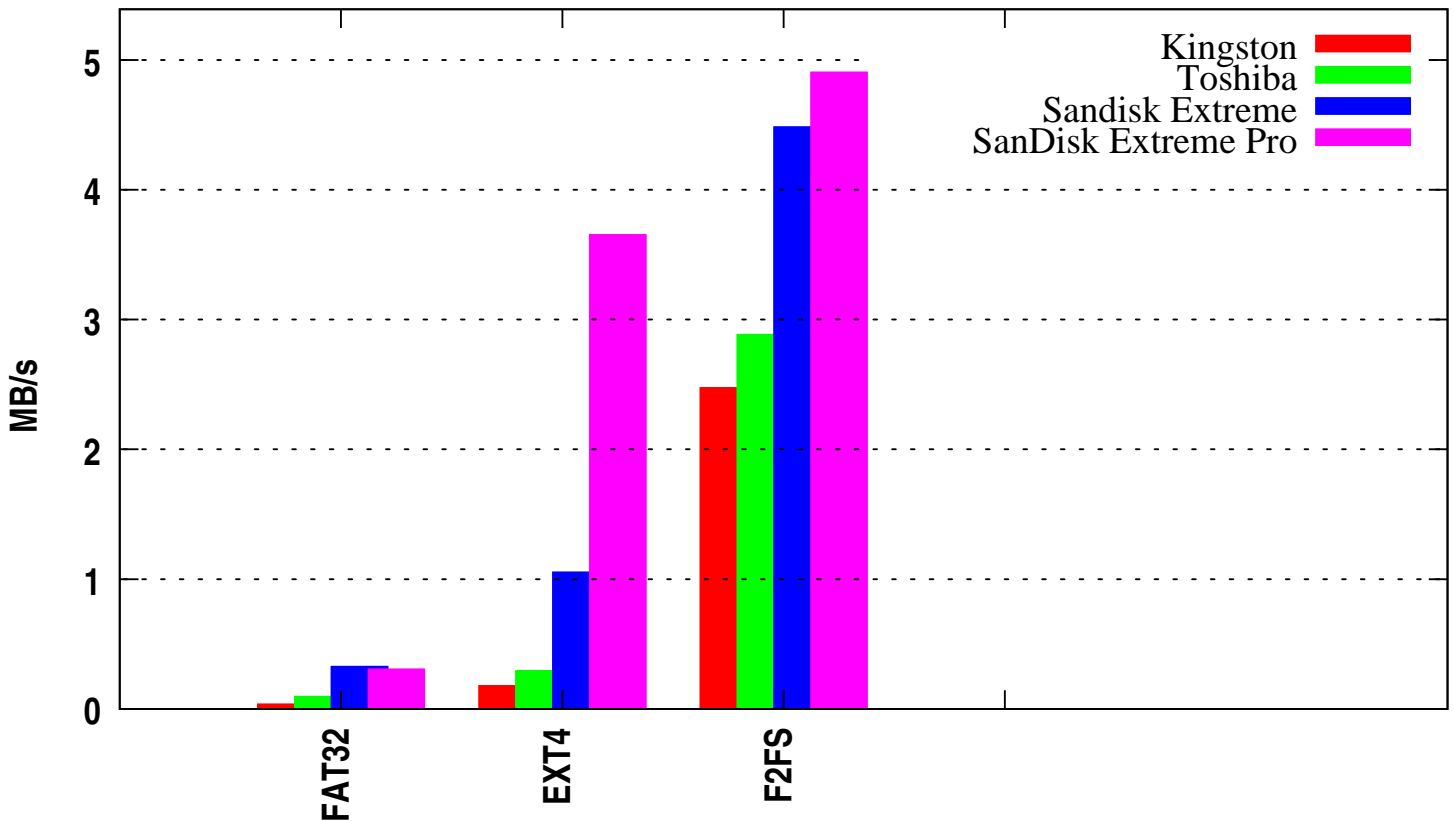
- 2M Segments written as whole chunks — always writes at log head  
— aligned with FLASH allocation units
- Log is the only data structure on-disk
- Metadata (e.g., head of log) written to FAT area in single-block writes
- Splits Hot and Cold data and Inodes.

It is designed to work *with* the flash translation layer. It understands the way that garbage collection might happen, and that full-segment writes are the most efficient. It uses log-based allocation to make the most of the FLASH characteristics.

It also divides files up based on their type. Files that are likely to be long lived, and written only once (e.g., photos, movies), are marked as 'cold' and stored in a different area of the file system from other files. This helps the garbage collector.



# Benchmarks: PostMark 32k Read



Postmark, which writes lots of small files, showed massive differences between the cards. As its read and write sizes are much less than the page size, it forces a program/erase or a garbage collection on every write — making it worse case for the cards. The file systems that hide this (F2FS) do much better, even on the cheapest card.

- Observation: XFS and ext4 already understand RAID
- RAID has multiple chunks, and a fixed stride, so...
- Configure FS as if for RAID

F2FS is newish, and maybe you don't trust it for a production machine. Can we get as good performance using a more mature filesystem?

A RAID system has a fixed number of spindles, and stripes data across all of them in fixed size chunks. If we configure a filesystem, telling it the allocation-unit-size is the chunk size, and the RAID stripe width is the number of open allocation units possible, would we get as good performance as F2FS gives?

Short answer: You see some performance improvement

But the effect isn't generally enough to bother with if you ha

## The Multiprocessor Effect:

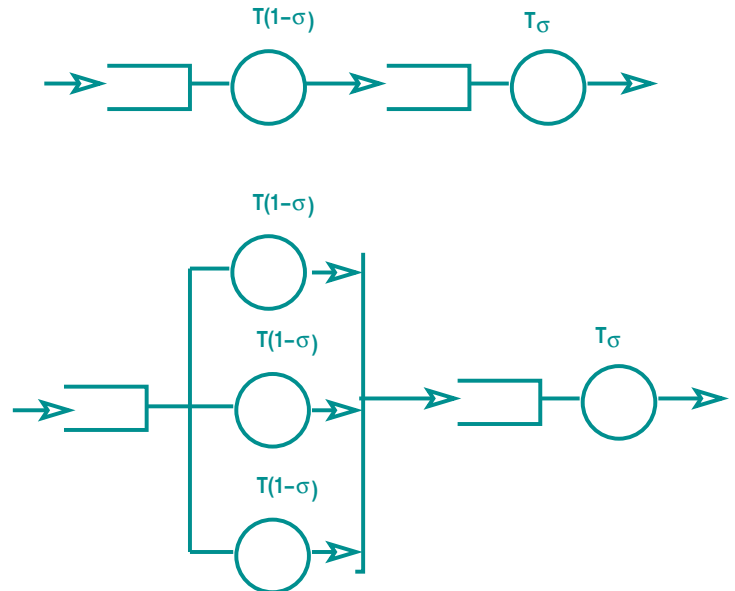
- Some fraction of the system's cycles are not available for application work:
  - Operating System Code Paths
  - Inter-Cache Coherency traffic
  - Memory Bus contention
  - Lock synchronisation
  - I/O serialisation

We've seen that because of locking and other issues, some portion of the multiprocessor's cycles are not available for useful work. In addition, some part of any workload is usually unavoidably serial.

Amdahl's law:

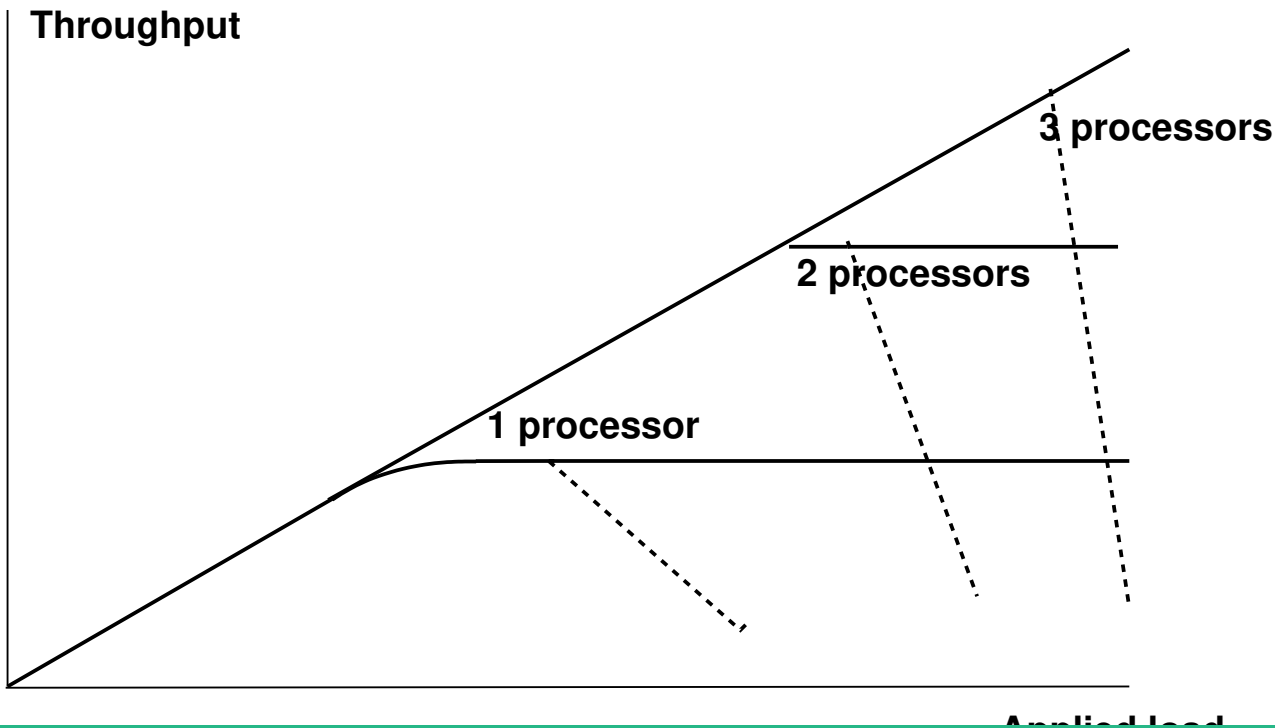
If a process can be split such that  $\sigma$  of the running time cannot be sped up, but the rest is sped up by running on  $p$  processors, then overall speedup is

$$\frac{p}{1 + \sigma(p - 1)}$$



It's fairly easy to derive Amdahl's law: perfect speedup for  $p$  processors would be  $p$  (running on two processors is twice as fast, takes half the time, than running on one processor).

The time taken for the workload to run on  $p$  processors if it took 1 unit of time on 1 processor is  $\sigma + (1 - \sigma)/p$ . Speedup is then  $1/(\sigma + (1 - \sigma)/p)$  which, multiplying by  $p/p$  gives  $p/(p\sigma + 1 - \sigma)$ , or  $p/(1 + \sigma(p - 1))$

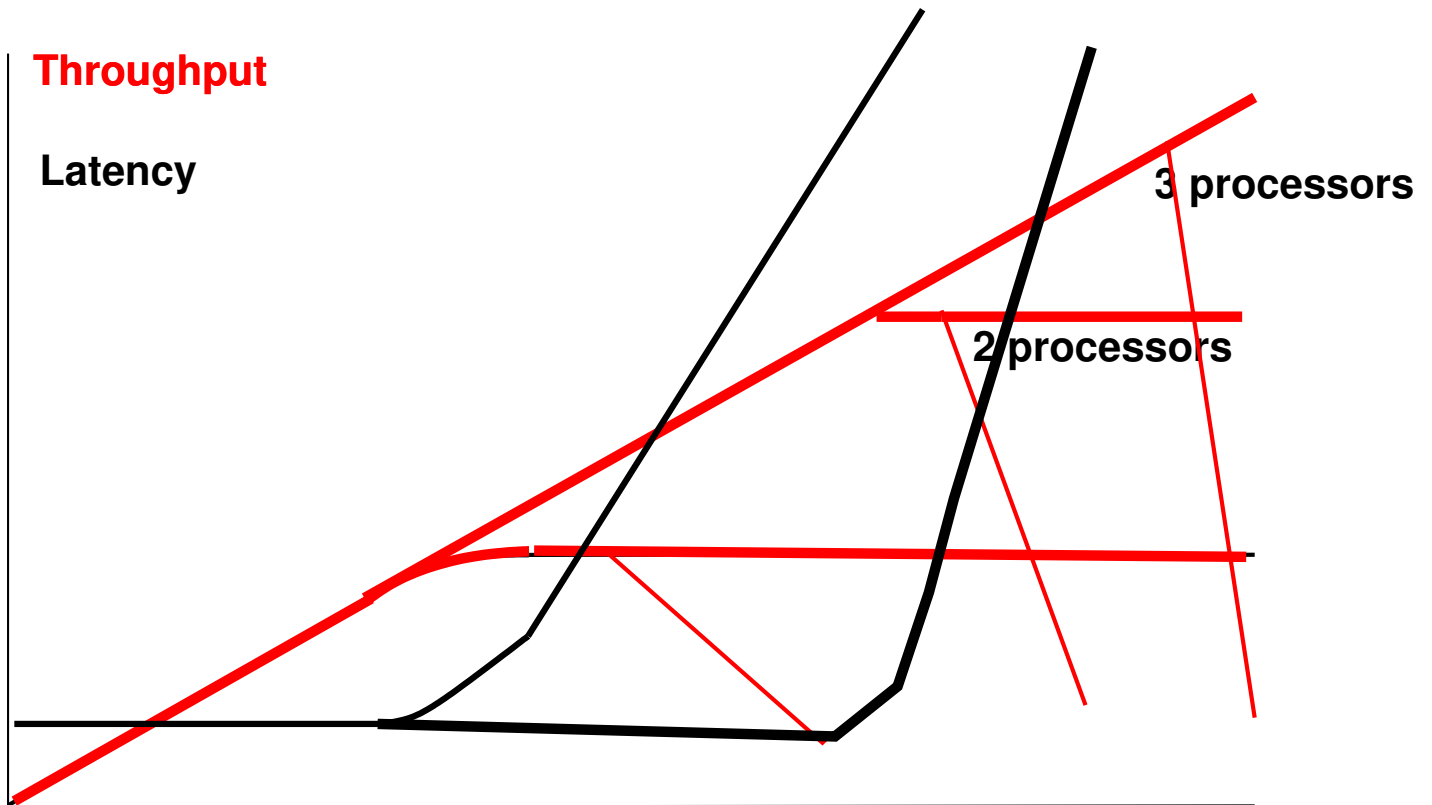


The general scalability curve looks something like the one in this slide. The Y-axis is throughput, the X-axis, applied load. Under low loads, where there is no bottleneck, throughput is determined solely by the load—each job is processed as it arrives, and the server is idle for some of the time. Latency for each job is the time to do the job.

As the load increases, the line starts to curve. At this point, some jobs are arriving before the previous one is finished: there is queueing in the system. Latency for each job is the time spent queued, plus the time to do the job.

When the system becomes overloaded, the curve flattens out. At this point, throughput is determined by the capacity of the system; average latency becomes infinite (because jobs cannot be processed as fast as they arrive, so the queue grows longer and longer), and the bottleneck resource is 100% utilised.

When you add more resources, you want the throughput to go up. Unfortunately, because of various effects we'll talk about later that doesn't always happen...



This graph shows the latency 'hockey-stick' curve. Latency is determined by service time in the left-hand flat part of the curve, and by service+queueing time in the upward sloping right-hand side. When the system is totally overloaded, the average latency is infinite.

Gunther's law:

$$C(N) = \frac{N}{1 + \alpha(N - 1) + \beta N(N - 1)}$$

where:

$N$  is demand

$\alpha$  is the amount of serialisation: represents Amdahl's law

$\beta$  is the coherency delay in the system.

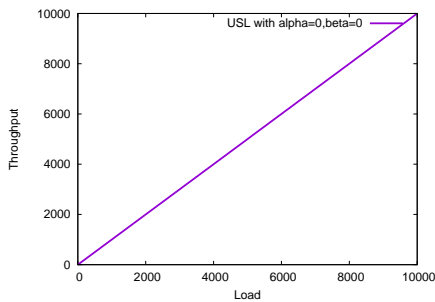
$C$  is Capacity or Throughput

Neil Gunther (2002) captured this in his 'Universal Scalability Law', which is a closed-form solution to the machine-shop-repairman queueing problem.

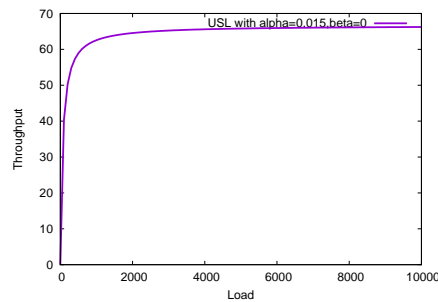
It has two parameters,  $\alpha$  which is the amount of non-scalable work, and  $\beta$  which is to account for the degradation often seen in system-performance graphs, because of cross-system communication ('coherency' or 'contention', depending on the system).

The independent variable  $N$  can represent applied load, or number of logic-units (if the work per logic-unit is kept constant).

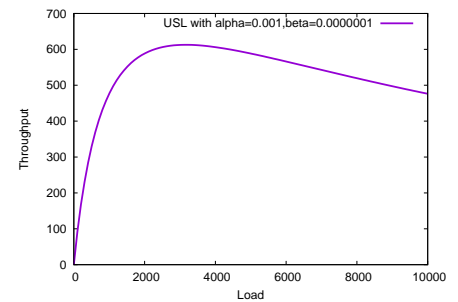




$$\alpha = 0, \beta = 0$$



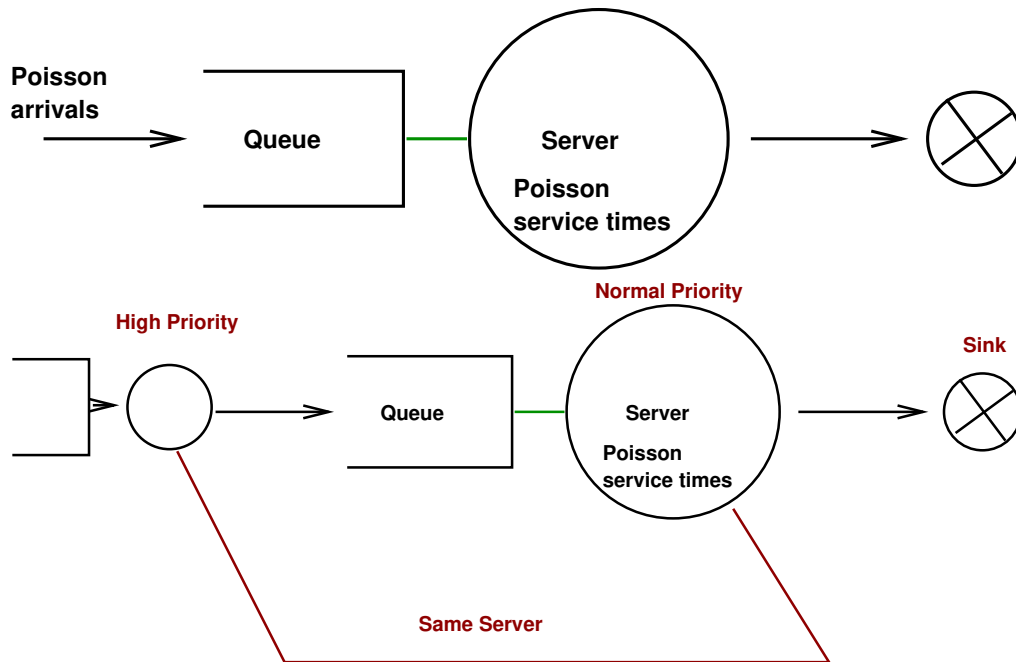
$$\alpha > 0, \beta = 0$$



$$\alpha > 0, \beta > 0$$

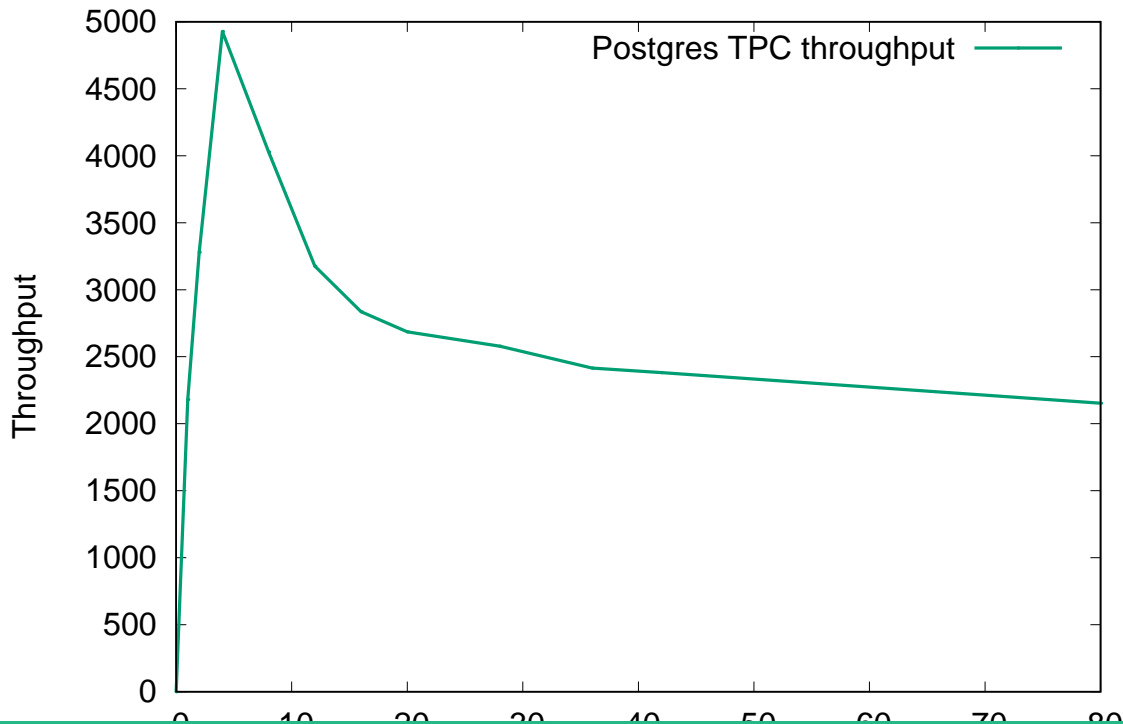
Here are some examples. If  $\alpha$  and  $\beta$  are both zero, the system scales perfectly—throughput is proportional to load (or to processors in the system). If  $\alpha$  is slightly positive it indicates that part of the workload is not scalable. Hence the curve plateaus to the right. Another way of thinking about this is that some (shared) resource is approaching 100% utilisation. If in addition  $\beta$  is slightly positive, it implies that some resource is contended: for example, preliminary processing of new jobs steals time from the main task that finishes the jobs.

## Queueing Models:

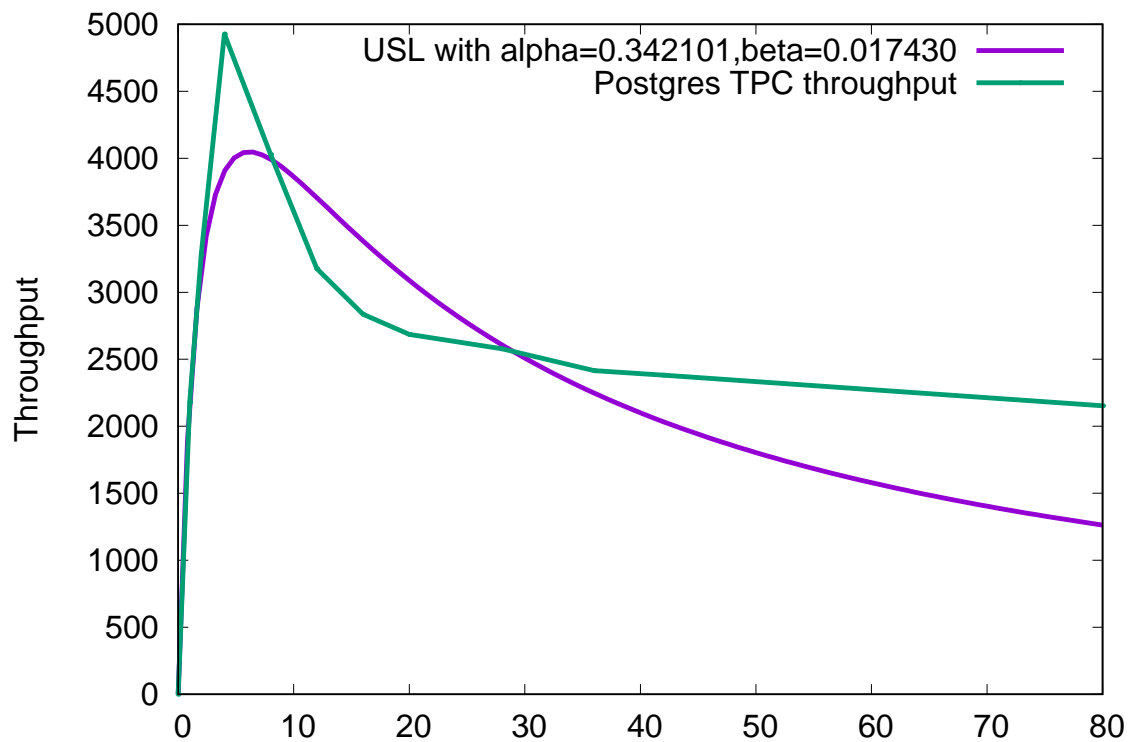


You can think of the system as in these diagrams. The second diagram has an additional input queue; the same servers service both queues, so time spent serving the input queue is stolen from time servicing the main queue.

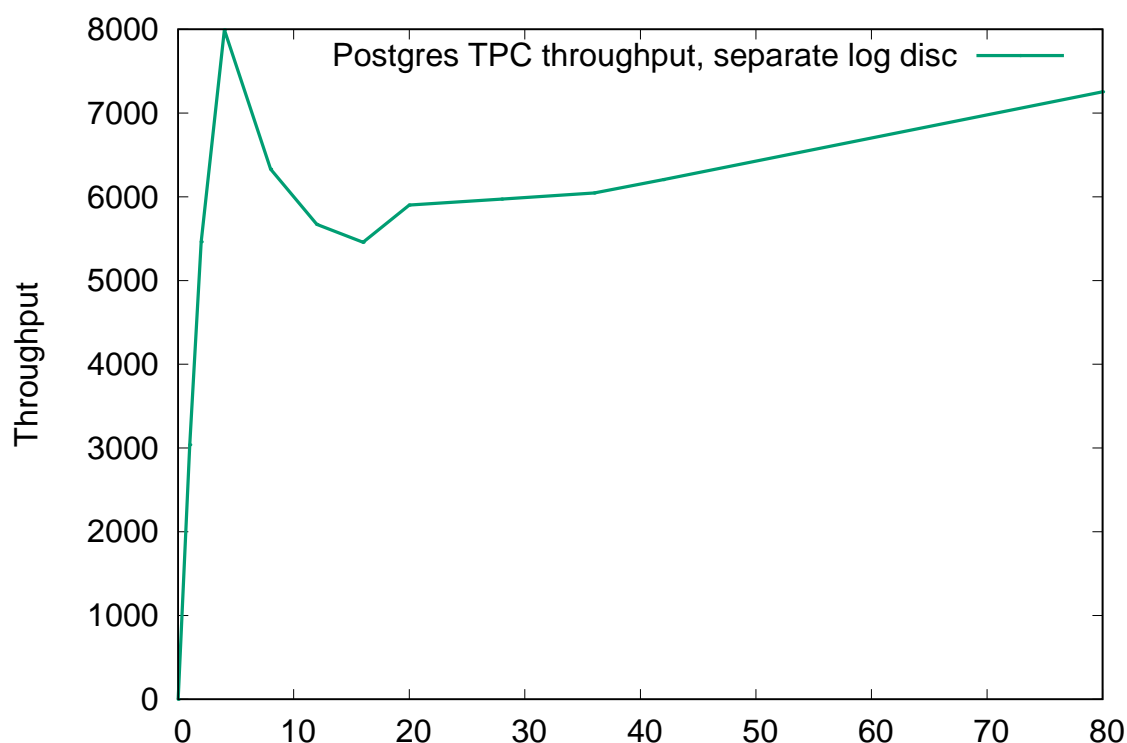
Real examples:



These graphs are courtesy of Etienne, Adrian and the RapiLog team. This is a throughput graph for TPC-C on an 8-way multiprocessor using the ext3 filesystem with a single disk spindle. As you can see,  $\beta > 0$ , indicating coherency delay as a major performance issue.

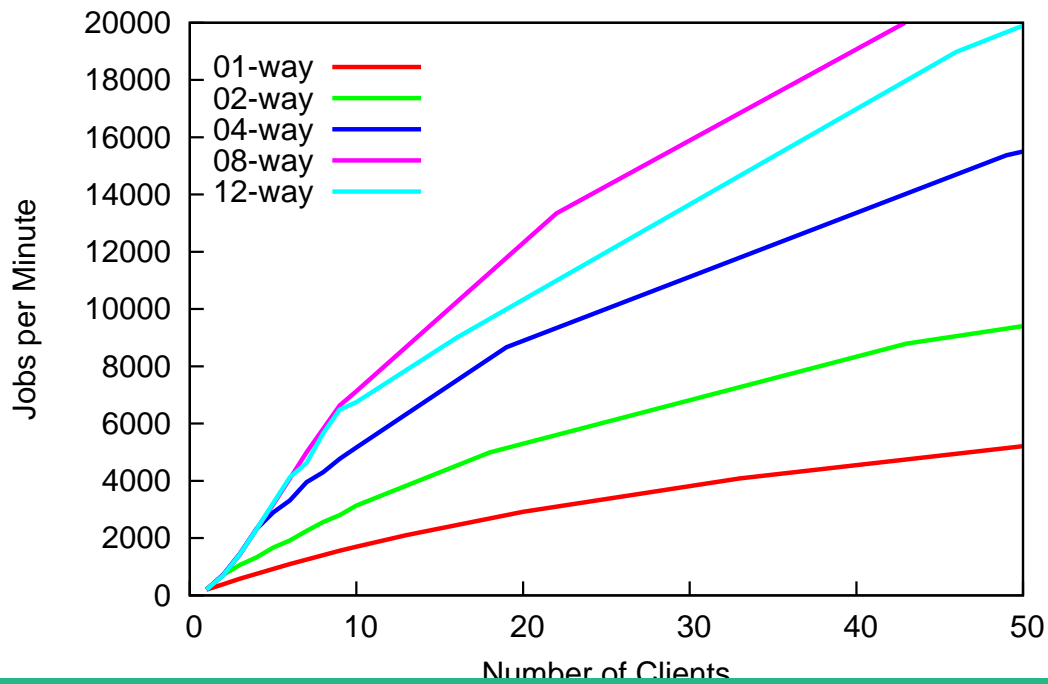


Using R to fit the scalability curve, we get  $\beta = 0.017$ ,  $\alpha = 0.342$  — you can see the fit isn't perfect, so fixing the obvious coherency issue isn't going to fix the scalability entirely.



Moving the database log to a separate filesystem shows a much higher peak, but still shows a  $\beta > 0$ . There is still coherency delay in the system, probably the file-system log. From other work I've done, I know that ext3's log becomes a serialisation bottleneck on a busy filesystem with more than a few cores — switching to XFS (which scales better) or ext2 (which has no log) would be the next step to try.

Another example:



This shows the reaim-7 benchmark running on various numbers of cores on an HP 12-way Itanium system. As you can see, the 12-way line falls below the 8-way line —  $\alpha$  must be greater than zero. So we need to look for contention in the system somewhere.

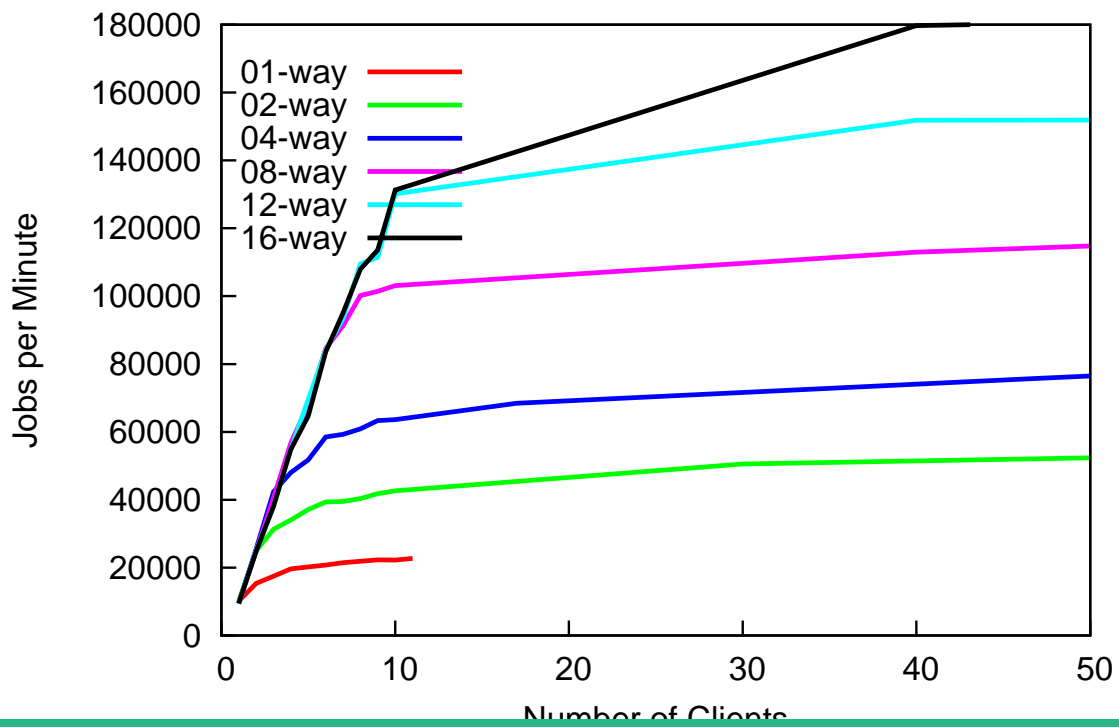
SPINLOCKS		HOLD		WAIT					
UTIL	CON	MEAN ( MAX )	MEAN ( MAX ) (% CPU)	TOTAL	NOWAIT	SPIN	RJECT	NAME	
72.3%	13.1%	0.5us (9.5us)	29us ( 20ms) (42.5%)	50542055	86.9%	13.1%	0%	find_lock_page+0x30	
0.01%	85.3%	1.7us (6.2us)	46us (4016us) (0.01%)	1113	14.7%	85.3%	0%	find_lock_page+0x130	

Lockmetering shows that a single spinlock in find\_lock\_page() is the problem:

```
struct page *find_lock_page(struct address_space *mapping,
                            unsigned long offset)
{
    struct page *page;
    spin_lock_irq(&mapping->tree_lock);
repeat:
    page = radix_tree_lookup(&mapping->page_tree, offset);
    if (page) {
        page_cache_get(page);
        if (TestSetPageLocked(page)) {
            spin_unlock_irq(&mapping->tree_lock);
            lock_page(page);
            spin_lock_irq(&mapping->tree_lock);
        }
    }
}
```

So replace the spinlock with a rwlock, and bingo:





The scalability is much much better.

- Find the bottleneck
- fix or work around it
- check performance doesn't suffer too much on the low end.
- Experiment with different algorithms, parameters

Fixing a performance problem for your system can break someone else's system. In particular, algorithms that have good worst-case performance on large systems may have poorer performance on small systems that algorithms that do not scale. The holy grail is to find ways that work well for two processor and two thousand processor systems.



- Each solved problem uncovers another
- Fixing performance for one workload can worsen another
- Performance problems can make you cry

Performance and scalability work is like peeling an onion. Solving one bottleneck just moves the overall problem to another bottleneck. Sometimes, the new bottleneck can be *worse* than the one fixed. Just like an onion, performance problems can make you cry.

## Avoiding Serialisation:

- *Lock-free* algorithms
- Allow safe concurrent access *without excessive serialisation*
- Many techniques. We cover:
  - Sequence locks
  - Read-Copy-Update (RCU)

If you can reduce serialisation you can generally improve performance on multiprocessors. Two locking techniques are presented here.

## Sequence locks:

- Readers don't lock
- Writers serialised.

If you have a data structure that is read-mostly, then a sequence lock may be of advantage. These are less cache-friendly than some other forms of locks.

Reader:

```
volatile seq;
do {
    do {
        lastseq = seq;
    } while (lastseq & 1);
    rmb();
    ....
} while (lastseq != seq);
```

Writer:

```
spinlock(&lck);
seq++; wmb();
...
wmb(); seq++;
spinunlock(&lck);
```

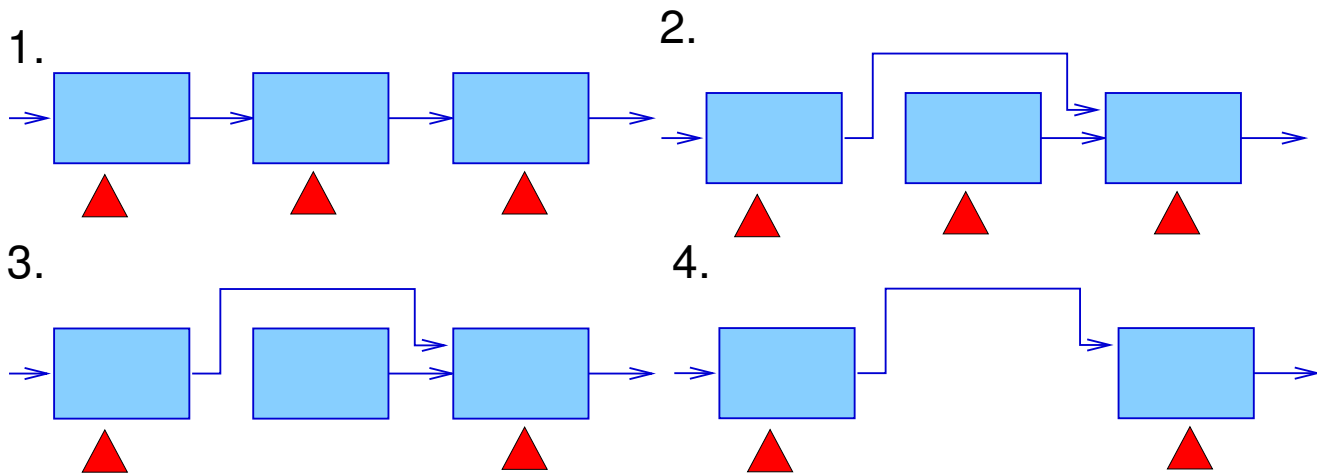
The idea is to keep a sequence number that is updated (twice) every time a set of variables is updated, once at the start, and once after the variables are consistent again. While a writer is active (and the data may be inconsistent) the sequence number is odd; while the data is consistent the sequence is even.

The reader grabs a copy of the sequence at the start of its section, spinning if the result is odd. At the end of the section, it rereads the sequence, if it is different from the first read value, the section is repeated.

This is in effect an optimistic multi-reader lock. Writers need to protect against each other, but if there is a single writer (which is often the case) then the spinlocks can be omitted. A writer can delay a reader; readers do not delay writers – there's no need as in a standard multi-reader lock for writers to delay until all readers are finished.

This is used amongst other places in Linux for protecting the variables containing the current time-of-day.

RCU: McKenney (2004), McKenney et al. (2002)



Another way is so called *read-copy-update*. The idea here is that if you have a data structure (such as a linked list), that is very very busy with concurrent readers, and you want to remove an item in the middle, you can do it by updating the previous item's *next* pointer, but you cannot then free the item just unlinked until you're sure that there is no thread accessing it.

If you prevent preemption while walking the list, then a sufficient condition is that every processor is either in user-space or has done a context switch. At this point, there will be no threads accessing the unlinked item(s), and they can be freed.

Inserting an item without locking is left as an exercise for the reader.

Updating an item then becomes an unlink, copy, update, and insert the copy; leaving the old unlinked item to be freed at the next quiescent point.

## References

McKenney, P. E. (2004), Exploiting Deferred Destruction: An Analysis of Read-Copy-Update Techniques in Operating System Kernels, PhD thesis, OGI School of Science and Engineering at Oregon Health and Sciences University.

**URL:**

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McKenney, P. E., Sarma, D., Arcangelli, A., Kleen, A., Krieger, O. & Russell, R. (2002), Read copy update, *in* 'Ottawa Linux Symp.'

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# Background Reading

*AT&T Bell Laboratories Technical Journal* **63**(8), 1577–1593.

**URL:** <ftp://cm.bell-labs.com/who/dmr/hist.html>

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