

Operating Systems

Lecture 25: The OS Network Stack





Context for today's lecture

- A quick overview of the OS network stack
 - Has evolved over decades
 - Many different components
 - Many different protocols
 - Today is just a brief overview (details in 4450)
- What shall we focus on?
 - Interaction with other components that we have studied in 4410

Recap: what do computer networks do?

A computer network delivers data between the end points

- One and only one task: Delivering the data
- This delivery is done by:
 - Chopping the data into **packets**
 - Sending individual packets across the network
 - Reconstructing the data at the end points
- That is all!

Recap: Data delivery as a fundamental goal

- Support the logical equivalence of <u>Interprocess Communication (IPC)</u>
 - Mechanism for "processes on the same host" to exchange messages
- Computer networks allow "processes on two different hosts" to exchange messages
- Clean separation of concerns
 - Computer networks deliver data
 - Applications running on end hosts decide what to do with the data
- Keeps networks simple, general and application-agnostic

End-to-end story

Four fundamental problems!

- Naming, addressing: Locating the destination, receiver app
- Routing: Finding a path to the destination host
- Forwarding: Sending data from the sender app to the receiver app
- Reliability: Handling failures, packet drops, etc.

Fundamental problem #1: Naming and Addressing

- Network Address: where host is located
 - Requires an address for the destination host
- Host Name: which host it is
 - Consider when you access a website
 - URL is **user-level name** (eg, <u>www.cornell.edu</u>)
 - Network needs address (eg, where is <u>www.cornell.edu</u>)?
- Must map names to addresses
 - Just like we use an address book to map human names to addresses

Must be done at the end-host;

The source knows the name—

Maps that name to an address using DNS!

(Done only once, when establishing a connection—low overhead)

Routing packets through network elements (eg, routers) to destination

- Given destination address (and name), how does each switch/router know where to send the packet so that the packet reaches its destination
- When a packet arrives at a router
 - a routing table determines which outgoing link the packet is sent on
 - Computed using **routing protocols**

Mostly done within the network switches/routers; Has little to do with the OS

Queueing and Forwarding of packets at <u>switches/routers</u>

- Queueing: When a packet arrives, store it in "input queues"
 - When a packet arrives:
 - Look up its destination's address (how?)
 - Find the link on which the packet will be forwarded (how?)
- Forwarding: When the outgoing link free
 - Pick a packet from the corresponding virtual output queue
 - forward the packet!

Done at switches/routers; Has little to do with the OS

Queueing and Forwarding of packets at <u>the host</u>

- When a process wants access to the network, it opens a socket, which is associated with a port
- Socket: an OS mechanism that connects processes to the network stack
- **Port:** number that identifies that particular socket
 - used by the OS to direct incoming packets
- There is a sender-side socket/port, and a receiver-side socket/port

Done at the host OS; Lot of interesting tradeoffs

What must packets carry to enable forwarding?

- Packets must describe where it should be sent
 - Requires an address for the destination host
 - A port number (socket identifier) for the destination application
- Packets must describe where its coming from
 - For handling failures, etc.
 - Requires an address for the source host
 - A port number (socket identifier) for the source application
- Packets must carry data
 - can be bits in a file, image, whatever

Must be done at the host; Network just delivers the packet

How do you deliver packets reliable?

- Packets can be dropped along the way
 - Buffers in router can overflow
 - Routers can crash while buffering packets
 - Links can garble packets
- How do you make sure packets arrive safely on an unreliable network?
 - Or, at least, know if they are delivered?
 - Want no false positives, and high change of success

Mostly implemented at the host (end-to-end principle) Using a protocol called TCP

There is also an unreliable transmission mechanism: UDP

The end-to-end story

- Application opens a **socket** that allows it to connect to the **network stack**
- Maps name of the web site to its address using DNS
- The network stack at the source embeds the address and port for both the source and the destination in packet header
- Each router constructs a routing table using a distributed algorithm
- Each router uses destination address in the packet header to look up the outgoing link in the routing table
 - And when the link is free, forwards the packet
- When a packet arrives the destination:
 - The network stack at the destination uses the port to forward the packet to the right application

Four fundamental problems!—what does the OS do?

- Naming, addressing: Locating the destination
 - Setting up connection (name resolution, etc.)—low overhead
- Routing: Finding a path to the destination
 - Little or nothing
- Forwarding: Sending data to the destination
 - Create/insert packet headers—high overhead
 - Move data around based on sockets/ports—high overhead
 - Enable applications to read/write data—very high overheads
- Reliability: Handling failures, packet drops, etc.
 - Protocol-level processing—high overhead

Questions?

End-host network stack: Questions to ask

- Where is the sender-side socket located?
 - Depends on where the application is currently scheduled (CPU scheduler)
- How does the OS move packets from the sender-side socket to the network hardware?
 - Data is in some memory location specified by the socket (virtual memory)
 - The OS performs sender-side processing (create packets, headers, TCP processing, ..)
 - Data moved to the network hardware using DMA (studied in IO and devices)
- Where is the receiver-side socket located?
 - Depends on where the application is currently scheduled (CPU scheduler)

• How does the OS move packets from the network hardware to the receiver-side socket?

- OS gives hardware some memory addresses to write the data (virtual memory)
- Network hardware copies data using DMA (studied in IO and devices)
- The OS knows where the data is copied (virtual memory)
- The OS performs receiver-side processing (reliability, strip off headers, etc...)
- Tells the application where to read the data from (virtual memory)



Write system call

- Initiates data copy
 - From the application buffers (address space) to kernel buffers
- High CPU overheads
 - Just moving data around (read from one buffer, write to another buffer)
 - All kinds of caching and page replacement issues come up
- Packets are constructed at this point
 - Push data to socket's write queue until the queue is full
 - Block until queue is empty



TCP/IP processing

- All reliability-specific operations
- If protocol says okay to send data
 - Pop packets from socket's write queue and push to the next layer
 - Must not delete packets yet, in case the packet gets lost in the network
- Delete packets once ack-ed by the receiver
 - A lot of book keeping (could be complicated)



NetFilter

- Performs "filtering" of packets
 - e.g., firewall
- Network address/port translation
 - E.g., when one wants to hide sender port/addresses from other servers
- In Linux, iptable and nftable commands are used for filtering
 - Lightweight



XPS

- Network Hardware (NIC) has multiple queues
 - Just like other storage hardware that we have discussed
- To which queue should one forward packets from a particular socket?
 - How should the mapping work?
 - All sockets forward to one queue?
 - Each socket is assigned its own queue?
 - If many-to-many mapping, how to map sockets to queues?
- Linux XPS layer is used to define/perform this mapping
 - Usually maps all sockets running on the same core to the same NIC queue
 - But can define any mapping



Queueing Discipline

- Performs "traffic shaping" and packet scheduling
 - Shaping: how much bandwidth to give to each socket
 - Scheduling: among sockets mapped to a queue, which packet to choose next?
 - Performed on a per-queue basis
- Each transmit queue has its own queueing discipline (qdisc) in the OS
 - In Linux, tc command is used for managing qdisc





Segmentation

- Traditionally, data processed and transmitted at 1500byte granularity
 - But, if the application has a lot of data to send (on the same socket)
 - Many of the previous processing steps will be similar for all packets
 - Individual processing unnecessarily wastes CPU cycles
 - High packet processing overheads
- General Segmentation Offload (GRO)
 - Software-based solution to batch packet processing (e.g., 64KB granularity)
 - But packets transmitted at 1500byte granularity
 - Thus, once processed by the OS, we must "segment" packets before transmission
- GSO saves cycles for packet processing using batches of packets (~64KB)
 - But has overheads (implemented in software, after all): perform segmentation
- TCP Segmentation offload (TSO)
 - Perform packet processing in batches in the OS
 - Offload segmentation of packet batches to the hardware
 - Most NICs support TSO



Driver Tx

• Manage "shared memory" between the NIC and the OS

- Share memory region: a ring (circular) buffer
- Each element in the buffer referred to as a "packet descriptor"
 - Memory address where data in a particular packet will be copied
- Operations:
 - Write data into one of the descriptors
 - Signal to the NIC that data is ready to be transmitted (ring doorbell)
 - Ring doorbell in per-descriptor basis has high CPU overheads
 - NIC then fetches packets from DRAM pointed by the packet descriptor
 - Descriptors reinserted into the ring buffer as data in a descriptor is transmitted





Driver Rx

- NICs maintain multiple Rx ring buffers
- For each buffer, OS does the following operations:
 - Prepare new descriptors for the NIC to do DMA
 - Push new descriptors containing empty pages to the ring buffer
 - Once NIC finishes DMA, unmap DMA mappings





IRQ Handling and NAPI

- Packets are DMA-ed to Rx ring buffer in shared (kernel) memory
- NIC triggers an interrupt to wake up OS for handling packets
 - Downside: per-packet interrupt has very high overheads
- NAPI (new API): disable the interrupt and start a poll loop for handling packets
 - Reduce #interrupts
 - Only the first packet triggers an interrupt





Generic Receiver Offload (GRO)

- Receiver-side optimization similar to GSO/TSO
- Aggregate packets of the same flow before TCP/IP processing
 - software-based
 - Extra CPU overheads (similar to GSO)
- LRO: offload GRO to the hardware (NIC)
 - Downside: NIC has limited memory to store packets





Packet and flow steering

- Which core should NIC forward packets to?
- RSS/RPS: choose core based on the hash of the packet header
 - RSS: hardware-based; RPS: software-based
 - Scale packet processing with bandwidth
 - Downside: NUMA, cache unawareness
- aRFS/RFS: choose core based on where the application is running
 - aRFS: hardware-based; RFS: software-based
 - Benefits: Read/write data from local cache/memory
 - Downside: poor scalability when #apps in the same core increases





TCP/IP and read system call

- Push packets to socket read queue
- Wake up application thread for copying data to the application buffers
 - Downside: extra scheduling overhead/delay
- Send ACK packets
 - Sender can clear out packets that have been delivered