Ethernet and Local Area Networking
Ethernet

- 1976, Metcalfe & Boggs at Xerox
  - Later at 3COM
- Based on the Aloha network in Hawaii
  - Named after the “luminiferous ether”
- Centered around a broadcast bus
- Can use different physical links
- Simple link-level protocol, scales well
- Simple algorithm for sharing the network well under load
Ethernet Goals

- Connect local area networks
  - Few buildings, short distances (<1 km)
- Inexpensively
  - Low infrastructure costs
- Without bottlenecks
  - No expensive routers, bridges, switches etc.
  - No state in the network, no store-and-forward

- Tremendously successful
- Simple conceptual model still in use
  - Despite two orders of magnitude increase in bandwidth
“CSMA/CD”

- Carrier sense
  - Listen before you speak
- Multiple access
  - Multiple hosts can access the network
- Collision detect
  - Detect and respond to cases where two hosts collide
Ethernet basics

An ethernet packet

- Destination Address
- Source Address
- Type
- ...Data...
- Checksum
Sending packets

- Carrier sense, broadcast if ether is available
ARP is used to discover physical addresses

- ARP = Address Resolution Protocol

“Addressing & ARP”

128.84.96.89

“I’m at 1a:34:2c:9a:de:cc”

128.84.96.90

“What is the physical address of the host named 128.84.96.89”

128.84.96.91
Addressing & DHCP

“I just got here. My physical address is 1a:34:2c:9a:de:cc. What's my IP?”

“Your IP is 128.84.96.89 for the next 24 hours”

◆ DHCP is used to discover network addresses
Collisions

What happens if two people decide to transmit simultaneously?
Collision Detection & Retransmission

- The hosts involved in the collision stop data transmission, sleep for a while, and attempt to retransmit.
- How long they sleep is determined by how many collisions have occurred before.
- They abort after 16 retries, hence no guarantee that a packet will get to its destination.

Advantages:
- Packet can be retransmitted at the link level immediately without high-level timeouts,
- Packets are truncated early to avoid wasting bandwidth
- Collision rates can be used to gauge net usage
Collisions

What happens if the packets are really short?
Odds & Ends

- Minimum packet size is 64 bytes, which is just right for the given length for all hosts to detect a collision.
- Truncated packets are filtered out of the network.
- CRC is used to detect malformed packets, e.g. electrical interference, noise.
Ethernet Features

- Completely distributed
  - No central arbiter

- Inexpensive
  - No state in the network
  - No arbiter
  - Cheap physical links (twisted pair of wires)
Ethernet Problems

- The endpoints are trusted to follow the collision-detect and retransmit protocol
  - Certification process tries to assure compliance
  - Not everyone always backs off exponentially
- Hosts are trusted to only listen to packets destined for them
  - But the data is available for all to see
  - Can place ethernet card in promiscuous mode and listen
Ethernet Lessons

- Best-effort delivery simplifies network design
- A simple, distributed protocol can tolerate failures and be easy to administer
- Networking infrastructure represents a large sunk cost
  - Best to keep it simple
  - Interoperable
  - Hard to upgrade means change occurs infrequently, when the gains are sizeable
Internet Structure & Protocols
Internetworking Origins

- Expensive supercomputers scattered throughout the US
- Researchers scattered differently throughout the US
- Need way to connect researchers to expensive machinery
- Point-to-point connections might have sufficed
Point to point connections
Internetworking Origins

- Department of Defense initiated studies on how to build a resilient global network
  - How do you coordinate a nuclear attack?
  - Especially, how do you tell people to stop firing missiles during a nuclear war?

- Interoperability and dynamic routing are a must
  - Along with a lot of other properties

- Result: Internet

- A complex system with simple components
Internet Overview

- Every host is assigned, and identified by, an IP address.
- Each packet contains a header that specifies the destination address.
- The network routes the packets from the source to the destination.

Question: What kinds of properties should the network provide?
Internet, The Big Picture

Endpoints

Routers
End-to-End Example

Should the network guarantee packet delivery?
- Think about a file transfer program
- Read file from disk, send it, the receiver reads packets and writes them to the disk

If the network guaranteed packet delivery, one might think that the applications would be simpler
- No need to worry about retransmits
- But still need to check that the file was written to the remote disk intact

A check is necessary if nodes can fail
- Consequently, applications need to be written to perform their own retransmits
- No need to burden the internals of the network with properties that can. and must. be implemented at the periphery
End-to-End Argument

- An Occam’s Razor for Internet architecture
- Application-specific properties are best provided by the applications, not the network
  - Guaranteed, or ordered, packet delivery, duplicate suppression, security, etc.
- The internet performs the simplest packet routing and delivery service it can
  - Packets are sent on a best-effort basis
  - Higher-level applications do the rest
Naming

Every host on the Internet is identified by an IP address
- For now, 32-bit descriptor, like a phone number
- Plans underway to change the underlying protocols to use longer addresses

IP addresses are assigned to hosts by their internet service providers
- Not physical addresses: IP address does not identify a single node, can swap machines and reuse the same IP address
- Not entirely virtual: the IP address determines how packets get to you, and changes when you change your ISP

Need completely virtual names
DNS

- Protocol for converting textual names to IP addresses
  - www.cnn.com = 207.25.71.25
- Namespace is hierarchical, i.e. a tree.
- Names are separated by dots into components
- Components are looked up from the right to the left
DNS Tree

- All siblings must have unique names
- Root is owned by ICANN
- Lookup occurs from the top down
- DNS stores arbitrary tuples (resource records)
- The address field contains the IP address, other fields contain mail routing info, owner info, etc.
DNS Lookup

1. the client asks its local nameserver
2. the local nameserver asks one of the root nameservers
3. the root nameserver replies with the address of the authoritative nameserver
4. the server then queries that nameserver
5. repeat until host is reached, cache result.
DNS Lessons

- Simple, hierarchical namespace works well
  - Can name anything, can share names

- Scales OK
  - Caching
  - Even though it was meant to be hierarchical, people like short names, and use it like a flat namespace

- Arbitrary tuple database
  - Can delegate selected services to other hosts

- No security!

- Namespace = money
  - Innovations in this space are met with resistance from people who control name resolution
TCP/IP

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IP

- Internetworking protocol
  - Network layer

- Common packet format for the Internet
  - Specifies what packets look like
  - *Fragments* long packets into shorter packets
  - *Reassembles* fragments into original shape

- Some parts are fundamental, and some are arbitrary

  - IPv4 is what most people use
  - IPv6 clears up some of the messy parts, but is not yet in wide use
### IPv4 packet layout

<table>
<thead>
<tr>
<th>Version</th>
<th>IHL</th>
<th>TOS</th>
<th>Total Length</th>
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<tbody>
<tr>
<td></td>
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IP Fragmentation

- Networks have different maximum packet sizes
  - Big packets are sometimes desirable – less overhead
  - Huge packets are not desirable – reduced response time for others

- Higher level protocols (e.g. TCP or UDP) could figure out the max transfer unit and chop data into smaller packets
  - The endpoints do not necessarily know what the MTU is on the path
  - The route can change underneath

- Consequently, IP transparently fragments and reassembles packets
IP Fragmentation Mechanics

- IP divides a long datagram into N smaller datagrams
- Copies the header
- Assigns a Fragment ID to each part
- Sets the More Fragments bit
- Receiving end puts the fragments together based on the new IP headers
- Throws out fragments after a certain amount of time if they have not been reassembled
**IP Options**

- **Source Routing**: The source specifies the set of hosts that the packet should traverse.
- **Record Route**: If this option appears in a packet, every router along a path attaches its own IP address to the packet.
- **Timestamp**: Every router along the route attaches a timestamp to the packet.
- **Security**: Packets are marked with user info, and the security classification of the person on whose behalf they travel on the network.
  - Most of these options pose security holes and are generally not implemented.
**UDP**

- **Unreliable Datagram Protocol**
- **IP goes from host to host**
- **We need a way to get datagrams from one application to another**
- **How do we identify applications on the hosts?**
  - Assign *port numbers*
  - E.g. port 13 belongs to the time service
UDP Packet Layout

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<tr>
<td>Source Port</td>
<td>Destination Port</td>
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</tr>
<tr>
<td>Length</td>
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<td></td>
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- UDP adds Ports, Data Length and Data checksum
UDP

- **UDP is unreliable**
  - A UDP packet may get dropped at any time
  - It may get duplicated
  - A series of UDP packets may get reordered

- **Applications need to deal with reordering, duplicate suppression, reliable delivery**
  - Some apps can ignore these effects and still function

- **Unreliable datagrams are the bare-bones network service**
  - Good to build on, esp for multimedia applications
TCP

- Transmission Control Protocol
  - Reliable, ordered communication
- Enough applications demand reliable ordered delivery that they should not have to implement their own protocol
- A standard, adaptive protocol that delivers good-enough performance and deals well with congestion
- All web traffic travels over TCP/IP
# TCP/IP Packets

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<tr>
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</tr>
<tr>
<td>Sequence Number</td>
<td>Acknowledgement Number</td>
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<tr>
<td>Offset</td>
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TCP Packets

- Each packet carries a unique ID
  - The initial number is chosen randomly
  - The ID is incremented by the data length
- Each packet carries an acknowledgement
  - Can acknowledge a set of packets by ack’ing the latest one received
- Reliable transport is implemented using these identifiers
TCP Connections

- TCP is connection oriented
- A connection is initiated with a three-way handshake
- Three-way handshake ensures against duplicate SYN packets
- Takes 3 packets, 1.5 RTT
Typical TCP Usage

- Three round-trips to set up a connection, send a data packet, receive a response, tear down connection
- FINs work (mostly) like SYNs to tear down connection
  - Need to wait after a FIN for straggling packets
TCP keeps a copy of all sent, but unacknowledged packets.

If acknowledgement does not arrive within a "send timeout" period, packet is resent.

Send timeout adjusts to the round-trip delay.
TCP timeouts

What is a good timeout period?
- Want to improve throughput without unnecessary transmissions

NewAverageRTT = (1 - \(\alpha\)) OldAverageRTT + \(\alpha\) LatestRTT
NewAverageDev = (1 - \(\alpha\)) OldAverageDev + \(\alpha\) LatestDev

where LatestRTT = (ack_receive_time – send_time),
LatestDev = |LatestRTT – AverageRTT|,
\(\alpha\) = 1/8, typically.

Timeout = AverageRTT + 4*AverageDev

Timeout is thus a function of RTT and deviation
TCP Windows

Multiple outstanding packets can increase throughput
TCP Windows

- Can have more than one packet in transit
- Especially over fat pipes, e.g. satellite connection
- Need to keep track of all packets within the window
- Need to adjust window size
TCP Congestion Control

- TCP Increases its window size as long as no packets are dropped.
- It halves the window size when a packet drop occurs.
  - A packet drop is evident from the acknowledgements.
- Therefore, it will slowly build up to the max bandwidth, and hover around the max.
  - It doesn’t achieve the max possible though.
  - Instead, it shares the bandwidth well with other TCP connections.
- This linear-increase, exponential backoff in the face of congestion is termed *TCP-friendliness*.
TCP Window Size

- Linear increase
- Exponential backoff
- Assuming no other losses in the network except those due to bandwidth
TCP Fairness

Want to share the bottleneck link fairly between two flows
TCP Slow Start

Linear increase takes a long time to build up a window size that matches the link bandwidth*delay

Most file transactions are not long enough

Consequently, TCP can spend a lot of time with small windows, never getting the chance to reach a sufficiently large window size

Fix: Allow TCP to build up to a large window size initially by doubling the window size until first loss
TCP Slow Start

- Initial phase of exponential increase
- Assuming no other losses in the network except those due to bandwidth
TCP Summary

- **Reliable ordered message delivery**
  - Connection oriented, 3-way handshake

- **Transmission window for better throughput**
  - Timeouts based on link parameters

- **Congestion control**
  - Linear increase, exponential backoff

- **Fast adaptation**
  - Exponential increase in the initial phase
Routing

Emin Gun Sirer
Challenge

- Need to discover and maintain paths through the network between communicating endpoints

Metrics of importance

- Latency
- Bandwidth
- Packet Overhead ("Goodput")
- Jitter
- Memory space per node
- Computational overhead per node
Domains

- **Wired networks**
  - Stable, administered, lots of infrastructure
    - E.g. the Internet

- **Wireless networks**
  - Wireless, dynamic, self-organizing
  - Infrastructure-based wireless networks
    - A.k.a. cell-based, access-point-based
    - E.g. Cornell’s “rover”
  - Infrastructure-less wireless networks
    - A.k.a. ad hoc
    - E.g. the 414/415 ad hoc network
Classification

Route discovery and dissemination
- Proactive vs. reactive

Route selection and usage
- Single path vs. multipath
Model

- A graph $G(V,E)$, where vertices represent routers, edges represent available links
  - Assume a unity weight associated with each link
- Centralized algorithms for finding suitable routes are straightforward
  - All pairs shortest paths
  - Need distributed algorithms
Proactive Routing

- Routes are disseminated from each node to all others, periodically
  - Every host has routes available to every other host, regardless of need
  - Used on the internet, some wireless ad hoc networks
Bellman-Ford

- Each node keeps track of dist, shortest distance to node i, and nexthop

- Initially dist_self = 0, dist_i = infinity for i != self, nexthop = {}

- Announce the dist vector to all neighbors

- Update dist s.t. dist is set to min(all neighbor dist announcements) + 1, nexthop=neighbor with minimum dist

- After n-1 rounds, dist values converge
Proactive Routing

**Pros**
- Route discovery latency is very low

**Cons**
- $O(N)$ state in every router
- Constant background communication
Reactive Routing

- Routes are discovered on the fly, as needed
  - Route discovery often involves a network-wide query
  - Used on many wireless ad hoc networks

Example
- Proactive Bellman-Ford
DSR
- Source routing

AODV
- Ad hoc on demand distance vector
Reactive Routing

**Pros**
- State proportional to number of used routes
- Communication proportional to number of used routes and failure rate

**Cons**
- Route discovery latency is high
- Jitter (variance of packet interarrival times) is high
Hybrid Routing

- Proactive & Reactive routing have some drawbacks
  - They work best under different network conditions
  - There are many parameters to pick to get optimal performance
- Perform hybrid routing
- Some routes are disseminated proactively, others discovered reactively
  - Can outperform both reactive and proactive across a wide range of scenarios
Remote Procedure Call
Clients and Servers

- A common model for structuring distributed computation is via the client/server paradigm.
- A server is a program (or collection of programs) that provides some service, e.g., file service, name service, ...
- The server may exist on one or more nodes.
- A client is a program that uses the service.
- A client first binds to the server, i.e., locates it in the network and establishes a connection.
- The client then sends requests to perform actions; this is done by sending messages that indicate which service is desired, along with params. The server returns a response.
The Problem with Messages

While messages provide very flexible communication, they also have certain problems:

- requires that programmer worry about message formats
- messages must be packed and unpacked
- messages have to be decoded by server to figure out what is requested
- messages are often asynchronous
- they may require special error handling functions

Basically, messages are not a natural programming model for most programmers.
A more natural way to communicate is through procedure call:
- every language supports it
- semantics are well defined and understood
- natural for programmers to use

Basic idea: let’s just define a server as a module that exports a set of procedures that can be called by client programs.

To use the server, the client just does a procedure call, as if it were linked with the server.
(Remote) Procedure Call

So, we would like to use procedure call as a model for distributed communication.

Lots of issues:
- how do we make this invisible to the programmer?
- what are the semantics of parameter passing?
- how is binding done (locating the server)?
- how do we support heterogeneity (OS, arch., language)
- etc.
Remote Procedure Call

- The basic model for Remote Procedure Call (RPC) was described by Birrell and Nelson in 1980, based on work done at Xerox PARC.
- Goals was to make RPC look as much like local PC as possible.
- Used computer/language support.
- There are 3 components on each side:
  - a user program (client or server)
  - a set of *stub* procedures
  - RPC runtime support
RPC

Basic process for building a server:

- Server program defines the server’s interface using an *interface definition language* (IDL)
- The IDL specifies the names, parameters, and types for all client-callable server procedures
- A *stub compiler* reads the IDL and produces two stub procedures for each server procedure: a client-side stub and a server-side stub
- The server writer writes the server and links it with the server-side stubs; the client writes her program and links it with the client-side stubs.
- The stubs are responsible for managing all details of the remote communication between client and server.
RPC Stubs

- Basically, a client-side stub is a procedure that looks to the client as if it were a callable server procedure.

- A server-side stub looks to the server as if it’s a calling client.

- The client program thinks it is calling the server; in fact, it’s calling the client stub.

- The server program thinks it’s called by the client; in fact, it’s called by the server stub.

- The stubs send messages to each other to make the RPC happen.
RPC Call Structure

```
call foo(x,y)
```

```
proc foo(a,b)
begin foo...
end foo
```

Client program makes local call to stub proc.

Server is called by its stub.

Client stub builds msg packet, inserts params.

Runtime sends msg to remote node.

Runtime unpacks params and makes call.

Call

```
10/20/2010
```

Message received

```
msg received
```

Server stub

RPC runtime

```
msg received
```

RPC runtime
RPC Return Structure

- Client program calls `foo(x,y)`
- Server program executes `proc foo(a,b)`
- Client continues
- Server returns
- Stub builds
- Return
- Client stub receives message, unpacks message, returns to caller
- Server stub sends message, builds result message with output arguments
- Server returns
- Message received by runtime
- Runtime receives message, calls stub
- Runtime, responds to original message
- Send message
- Return

10/20/2010
RPC Binding

- Binding is the process of connecting the client and server.
- The server, when it starts up, *exports* its interface, identifying itself to a network name server and telling the local runtime its dispatcher address.
- The client, before issuing any calls, *imports* the server, which causes the RPC runtime to lookup the server through the name service and contact the requested server to setup a connection.
- The *import* and *export* are explicit calls in the code.
RPC Marshalling

- Marshalling is the packing of procedure parameters into a message packet.
- The RPC stubs call type-specific procedures to marshall (or unmarshall) all of the parameters to the call.
- On the client side, the client stub marshalls the parameters into the call packet; on the server side the server stub unmarshalls the parameters in order to call the server’s procedure.
- On the return, the server stub marshalls return parameters into the return packet; the client stub unmarshalls return parameters and returns to the client.
RPC is the most common model now for communications in distributed applications.

RPC is essentially language support for distributed programming.

RPC relies on a stub compiler to automatically produce client/server stubs from the IDL server description.

RPC is commonly used, even on a single node, for communication between applications running in different address spaces.