Protocol Design and The Real World

Protocol Design – S-38.3157

Living below the Internet:

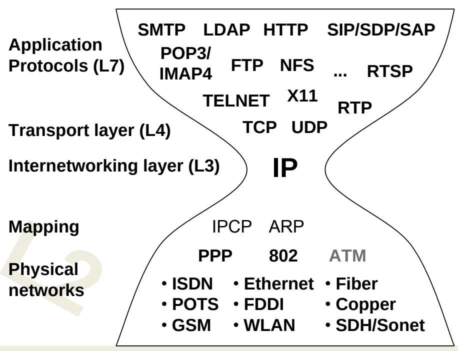
Advice for Internet Subnetwork Designers

RFC 3819, July 2004

Internet: The Network of Networks

- Networking technologies come and go
  - Modem, ISDN, DSL, Cable modem, Sat modem, Fiber to the home, Hybrid Fiber Coax, Powerline, Wireless Local Loop/WiMax, WiFi, …
- Each of the technologies has some characteristic technical parameters:
  - Bitrate/data rate/throughput
  - Transmission latency (light speed!)
  - Range/coverage/availability
  - Cost!
- Moore's Law keeps shifting the design tradeoffs
  - More transistors allow more processing
  - And new technologies are invented every day
- IP must be able to interface to all of the network technologies
- But each of the subnetwork technologies can help or hurt with this

Internet Protocol "Suite"



What is a "subnetwork"?

- IP packets are carried by "links", "link layer", "L2"
- RFC 2460 defines "Link" as:
  - a communication facility or medium over which nodes can communicate at the link layer, i.e., the layer immediately below IPv6.
    Examples are Ethernets (simple or bridged); PPP links; X.25, Frame Relay, or ATM networks; and internet (or higher) layer "tunnels", such as tunnels over IPv4 or IPv6 itself.
- A "Link" can be highly structured
  - Ethernets are connected by switches (= bridges) and formerly repeaters
  - Some "Links" are multi-layer networks, e.g. the serial line emulation defined by GSM runs its own mobility protocol
- IP generally does not care too much
  - But its performance can be helped or hurt

Optimizing subnetwork performance

- Provide functionality sufficient for carrying IP
  - Move IP packets back and forth
  - Provide some form of L3 → L2 address mapping
- Eliminate unnecessary functions that increase cost or complexity
  - IP does not need perfect retransmission persistence
  - Traditionally, subnetwork designers have erred on the side of **too much** functionality (remember the end-to-end arguments)
  - Waist-expanders (multicast, QoS) do benefit from L2 support
- Choose subnetwork parameters that maximize the performance of the Internet protocols
  - E.g., losses should be predominantly congestion losses

MTUs, fragmentation, segmentation (1)

- IPv4 has been designed to “work” with MTUs of 68 bytes
- Minimum **reassembly unit** was 576 bytes originally in IPv4
- Dominance of Ethernet has caused the expected MTU to be 1500 Bytes
  - Often with some bytes taken away for tunneling, PPPoE etc.
  - IPv6 formalizes this to a **minimum MTU** of 1280 bytes
- IP packets
  - Carry their own length (unless header compression is used)
  - Allow fragmentation at the router (IPv4) or at the sender (IPv6)
    - Typically avoided by “Path MTU discovery”, so MTU should be stable
    - Internet fog may cause ICMP “packet too big” messages to be lost, though
  - Have only 16 bits (IPv4) or 32 bits (IPv6) for fragment IDs

MTUs, fragmentation, segmentation (2)

- IPv6 links must, IPv4 links should attain 1280..1500 byte MTU
  - May need adaptation layer for segmentation/reassembly
  - Much more efficient to do on the link layer
- Larger MTUs (9000+) become increasingly desirable at high speeds
  - Sometimes called “jumbograms” (these are really packets > 64KB)
- Slow network **may** benefit from smaller MTUs
  - Serialization delay (1.25 s @ 9600 bit/s!) should not exceed 100..200 ms
  - When large packets block high-priority ones:
    Suspend-resume schemes (e.g., RFC 2687) or brute-force segmentation with multiple reassembly queues (e.g., RFC 2686, ATM) can help

Framing

- L1 transports (groups of) bits, L2 builds frames
- Delimiters vs. counting
  - Delimiters: maintain data transparency by bit stuffing, byte stuffing, etc.
  - COBS (constant overhead byte stuffing) is good way of providing transparency
- Easiest case: 1:1 mapping of IP packets to L2 packets
- SAR (small fixed-size frames, as in ATM): avoid complexity
  - AAL5: SNDUs with IP packet, length, CRC are chopped up
  - Reassembly errors are caught in the CRC (and SNDU length)
- Where L2 already has (large) fixed-size frames: mix and match
  - RFC 4326 (ULE) defines one such mapping on MPEG-2 frames (188 bytes)
  - To avoid error propagation, resynchronization should be quick

L2 connection management

- L2 may need connections (e.g., POTS/ISDN!)
- Manual setup
  - Acoustic coupler, anyone? ☺
- Automatic setup:
  - Nailed-up (i.e., reconnect after each failure)
  - Dial-on-demand + idle timeout
    - Timeout value hard to choose
  - Bandwidth-on-demand (multiple connections “as needed”)
    - “Need” hard to find out from L2 as there is no L7 intention signaling
- Related: connection-less BoD
  - DAMA (Demand-Assignment Multiple Access)
  - 802.11 PCF

Multipoint networks

- Simplest case: PPP — address resolution is trivial
  - As is multicast
- Broadcast networks
  - IPv4 ARP requires broadcast (designed for Ethernet)
  - May have efficient multicast (IPv6 ND relies on this)
  - Infrastructure (e.g., Ethernet switches) may have to do the work
  - IGMP/MLD snooping (or explicit signalling protocol) to minimize exposure to unwanted multicast
- NBMA (non-broadcast multiple access)
  - Need additional support for discovery/address resolution
  - E.g., ATM had ATMARP, MARS

Error Control

- Ultimate responsibility: hosts (end-to-end argument)
  - Internet has license to drop, corrupt, duplicate, or reorder packets
- End-to-end repair is more expensive, though:
  - requires effort at multiple hops
  - Can only happen at path RTT timescales (as opposed to hop RTT)
  - Losses are interpreted as congestion by L4 and reduce throughput
- L2 may repair errors **to aid performance**
- Actually: some loss is OK (or even needed!)
  - Perfect persistence will be overtaken by TCP retransmission
- L2 reliability should be "lightweight”
  - it only has to be "good enough"

Assessing L2 error control

- Yardstick: TCP
  - Most traffic is TCP anyway
  - Other traffic is supposed to be TCP-friendly (and generally have similar performance characteristics)
- Secondary consideration: RTP
  - Looks different
  - Has different requirements
    - **consistently** low delay keeps the playout timer short
    - Every packet drop reduces quality (but a couple percent can be tolerated)
- Two approaches to add redundancy:
  - Always: Forward error correction (FEC), usually at L1
  - On demand: retransmission ("ARQ"), at L2

FEC

- From a total throughput perspective, worse than ARQ
  - But for ARQ you first have to get entire packets (frames) through
- Now universally used at L1 (Trellis coding etc.)
- Issue: FEC vs. fading
  - FEC requires interleaving to ride through deep fades
  - Interleaving adds **delay**
  - TCP performance inversely proportional to delay
- Modern thinking ("4G") : minimize delay
  - Hop-by-hop ARQ works quite well on a low-delay channel
  - Need to leave some spare capacity for retransmissions, though

ARQ

- RFC3366
- Hop-by-hop retransmission wins:
  - Can operate on link-layer friendly segments (e.g., < 100 Byte)
  - Involves only the resources of one hop
  - Operates at the time constants of one hop
- Wild delay variation introduced by ARQ loses:
  - TCP timers will fire ahead of time if ARQ takes too long
  - Leads to duplicate packets — possibly both in the same L2 queue…
- Limit retransmission persistency
  - Should be on the order of **path** delay
  - Somewhat hard to predict (LAN vs. country vs. continent vs. world)
  - If possible, distinguish TCP (higher persistency) and RTP (lower persistency)

Outages

- "Elevator events": system enters tunnel/metal cage/…
- TCP timers **will** fire
  - No sense transmitting all the duplicate packets from multiple retransmissions
  - High persistence not very useful
  - **Do not deliver all the stale packets** after the outage
- However: There is no way in IP to notify the end of the outage
  - TCP timers may have backed off into some high region
  - It may take a while until the next timer fires
  - Dead time after the end of the outage
- Trick: Keep **some** packets around at L2 during an outage
  - Delivery after outage will trigger L4 machinery

Quality of error control

- TCP, UDP (as well as ICMP and IPv4 itself) use 16-bit two's complement checksum
  - Easy to compute (also by combining from partial checksums)
  - Not very strong — relies on good error detection at L2
  - Lots of undetected errors in practice [Stone/Partridge2000]
  - SCTP is the odd one out (CRC-32c, RFC3309)
- Higher layers can (and often do) use better error detection
  - E.g., cryptographic checksums in AH, ESP, TLS, SSH
- Still, some minimum quality from L2 is **expected**
  - Most L2 have at least 16-bit CRC
  - Make sure frame size and CRC are compatible
    - Long frames should use 32-bit CRC
  - Doing this at packet level is better than at segment level (cf. AAL5)

Unequal error protection

- Some applications can tolerate errors in some of their data
  - E.g., GSM speech codec can tolerate bit errors in excitation signal
- Need to protect header information, though
- Idea: error-protect initial part, but not all of the packet
- UDP-Lite (RFC3828): partial payload protection
  - Indicate which part of the UDP payload contributes to checksum
    - Reuses redundant UDP length field
- This separation is not visible at subnetwork layer
  - L2 error protection would need to make the same distinction
  - Could be divined by peeking at L4 header
- No L2 implementation yet

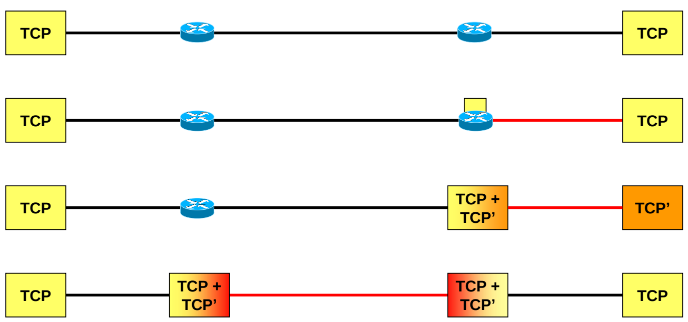
QoS

- Integrated Services have L2 mappings
(ISSLL — integrated services over specific link layers)
  - Controlled-Load may be easy to attain; Guaranteed is much harder
  - With a shared L2, also need to address admission control (reservation)
  - Tspec may be quite useful for planning resource usage (intention signal)
- Differentiated Services
  - PHB (per-hop behaviors) such as AF and EF again need to be mapped down to L2
  - AF has multiple priorities (as well as the backwards-compatible class selectors)
- Related issue: Buffering and Active Queue Management (AQM)
  - Provide adequate buffers
  - Start dropping some packets before latency gets really big (RED)
    - Hard to configure, though

Asymmetric Links

- RFC3449
- Some links have higher bitrates in one direction than in the other
  - ADSL
  - Satellites: downlink vs. return channels
  - Hybrid links built out of different technologies (Satellite + ISDN)
- Problem: When the ACKs don't fit into the return channel, forward channel is impaired
  - 1500/40 = 37.5 (usually less due to additional overheads)
- ACK compression etc. can help
- PEP (performance enhancing proxy) may be required
  - Can also assist TCP with other problems (high delay, high corruption error rate)

Summary: TCP in Extreme Networks



Compression

- Applications can compress their data
  - SSH
  - HTTP Content-Encoding
  - GIF, JPEG, PNG, video formats…
  - Very useful **before** encryption
- Many don't ➔ potential for performance increase at L2
  - Hard to do efficiently without sequencing/retransmission, though
  - Don't expand if L4 already compressed and/or encrypted
- Similar: Header compression
  - Most beneficial at small MTUs or for small-packet data (RTP voice)
  - Hop-by-hop can compress IP (and L4) headers, too
  - Needs to cope with packet losses, possibly reordering

Reordering

- IP allows for reordering of packets
- TCP, however, loses performance if that happens
  - May mis-diagnose a packet loss (three dup-acks)
- RTP, properly implemented, can be quite happy with reordering
  - As long as the timescales do not diverge too much

- Try to avoid reordering
  - As long as it does not impair performance

- Many L2 protocols also □expect in-order delivery
  - PPP only works on order-preserving links
  - Existing header compression schemes: see RFC4224

L2 Security

- L2 security can
  - Protect the network (where its operation is expensive)
    - And protect against theft of service via that specific L2 network
  - Equalize security to other parts of the network
    - I.e., protection against casual snooping may be all a user wants
  - Thwart traffic analysis
- L2 security cannot really:
  - Protect the radio resources (jammers are easy to build)
  - Provide end-to-end security

From Specification to the Real World

Protocol Design – S-38.3157

You have designed a protocol – what now?

- Implement it
  - Good idea
    - Shows that you can implement it
    - And gives a clear idea how complex it really is
    - You will find errors, omissions, and ambiguities only when implementing
    - "Rough consensus and running code"
  - But requires a lot of effort
    - You may want to do partial validation with less effort early on
  - Errors in the spec: you may have to write parts over and over again
  - An implementation by itself does not tell you much
    - About the scalability of your protocol: what happens if many nodes run it?
    - About its reliability, robustness, and performance in the Internet
  - [An implementation alone is often insufficient for publications]
    - You need to "prove" your ideas right
    - You need to deliver some quantitative data ("plots") that show you are better in some way

Alternatives?

- Analysis: mathematical modeling and quantitative evaluation
  - Depends on your math skills and experience
  - Of course, you should always do the minimal math yourself
    - Basic thoughts on scalability, etc.
  - Anything coming close to the real world likely to get really complex
  - Not in our focus
- Simulation: test your algorithms in an artificial environment
  - Takes the place of real-world validation
  - Requires some "implementation" in a simulator
    - Most ideas never make it beyond this step
  - Often this is as close as you can get to real world experience
- Emulation: run your implementation in an artificial environment

Simulations

- There are many tools out there

  General purpose examples:
  - ns-2 [http://www.isi.edu/nsnam/ns/]
  - GloMoSim [http://pcl.cs.ucla.edu/projects/glomosim/]
  - OMNET++ [http://www.omnetpp.org/]
  - OPNET [http://www.opnet.com]
  - QualNet [http://www.scalable-networks.com/]
  - CSIM [http://www.atl.external.lmco.com/projects/csim/]
  - MIRAI-SF [http://mirai-sf.nict.go.jp/index_e.html]
  - MATLAB/Mathematica
  - (Spreadsheets…)

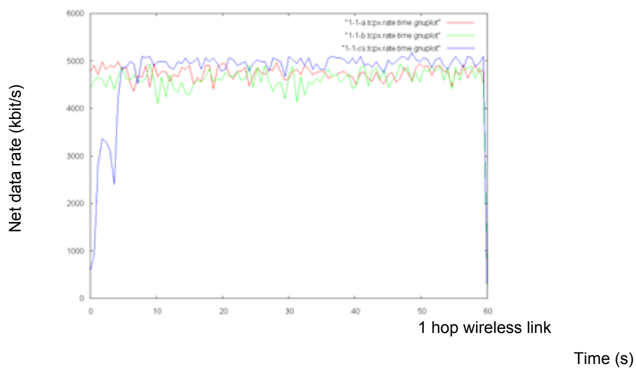
  Special purpose tools for specific simulation environments

  (and there are many community efforts and extensions available)

Issue #1 with Simulations

- Relation to Reality!
- Link layer: example wireless communication
  - Radio propagation has a gazillion dependencies
  - You cannot capture all
  - You cannot model all potential sources of interference
    - People, opening and closing doors, carried laptops and mobile phones (Bluetooth), etc.
    - Furniture, wall and window characteristics, water on windows, etc.
    - Vehicles (lorries with different loads and shapes vs. full buses vs. empty busses vs. different cars vs. motorcycles vs. bicycles) at different velocities, densities
    - Density of buildings, types of buildings, park areas, …
    - Non communication interference: micro wave ovens, …
- Mobile communications: reasonable mobility models
  - "Random waypoint considered harmful" — and indeed it is
  - General issue: how do humans, vehicles, etc. move?

Variation in Wireless Links (1)



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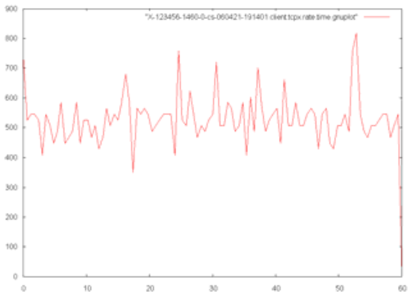
Variation in Wireless Links (2)

Net data rate (kbit/s)



6 hops wireless links

Time (s)

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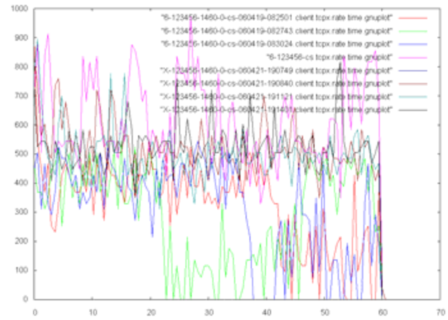
Variation in Wireless Links (3)

Net data rate (kbit/s)



Time (s)

6 hops wireless links

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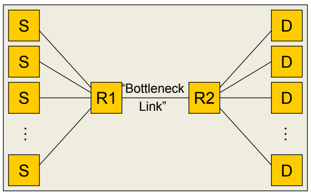
Issue #1 with Simulations (2)

Network layer

- Internet complexity
  - Interconnection topology
    - Networks, links, hosts
  - Virtually impossible to model even parts
- Internet diversity
  - Link data rates
  - Routers
    - Queue sizes, queuing disciplines
    - General behavior
  - Routing protocols
- "Background traffic"



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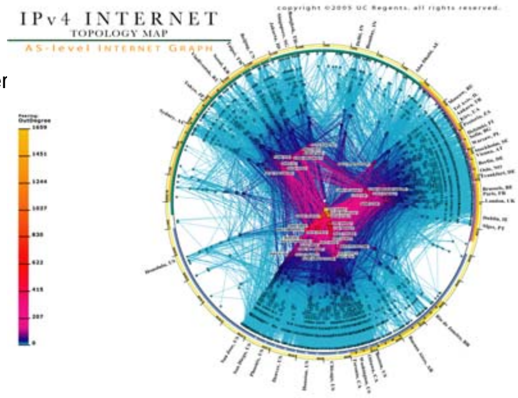
Issue #1

- Network layer
  - Internet:



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Issue #1 with Simulations (3)

- Transport layer
  - What mix of TCPs will you really find
  - How much SCTP?
  - How much UDP and similar traffic
- Applications
  - Which application are run?
  - Where?
    - Might be able to define this for web servers. But what about the others?
  - What is the usage pattern?
    - Ratio between applications?
    - Behavior of an individual user or a group of users?
    - Variation over time?
    - New applications?
    - What is the resulting traffic?
  - How well can large numbers help here?

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Further Pitfalls with Simulations

- Simulating itself is tricky
  - Find the right simulator, topology, traffic model, …
    - Difficult at all layers (virtually impossible for L1!)
  - Need to implement your protocol in the simulator
    - Different constraints from the real-world
    - Does it match your real-world implementation?
  - Choose the right simulation parameters
- Document everything
  - Recommended reading: "MANET Simulation Studies: The Incredibles"
- After all: simulations are like statistics
  - Don't trust any statistics you did not fake yourself!
  - For others' results: be critical
  - For your own simulations, this is like testing
    - Choose environments that are "real" and meaningful (rather than a perfect fit for what you want to prove)
    - Also choose environments that are "hostile"

Emulations

- Run the real code in a virtual environment
  - Allows testing the real thing
    - Instead of some imitation for a simulator
- Few simple examples
  - Dummynet [http://info.iet.unipi.it/~luigi/ip_dummynet/]
  - NIST Net [http://snad.ncsl.nist.gov/nistnet/]
  - Linux TCNG [http://tcng.sourceforge.net/] [http://lartc.org/]
  - Link layer packet bridges
  - Simple traffic shaping tools (such as udppipe)
- Virtual network environments
  - Virtualization of hosts (including kernel, interfaces, applications, etc.)
  - May use real and/or virtual links
  - May create complex artificial setups (similar to simulators)
  - But run real code
- Obviously, some issues similar to simulations apply