Principles of Communications

Supervision Notes

These notes are for the use of supervisors. They may be shared with students, later

Where these cover problems, they are notes, not complete solutions. They are intended to give you guidance to help with marking students' work.

Oct 2018

1 Context

The slides and schedule for 2018-2019 are available online: `http://www.cl.cam.ac.uk/teaching/1819/PrincComm/materials.html` plus I blog progress, problems, feedback etc `http://clogspotclog.blogspot.co.uk/` each week.

In networking, there is a collection of related courses, which we have done some alignment chats to check dovetail, which I think really only lacks enough time for practicals which could be done IF we were allowed to do ticks in part II and get resources to have supervisors for lab sessions...my part II course could hen be half evaluated through that (which would build on Andrew Moore's 1b and the other part II and ACS/Part III networking.

In 1B, there key pre-requisite is Andrew Moore's Computer Networking course. Clearly, it is last year's material you want to look at: `http://www.cl.cam.ac.uk/teaching/1718/CompNet/`

Key things a supervisor might want to cover in first 1-3 supervisions would be to revise *Routing* (Distance Vector and Link State); *TCP* (fairness, flow and congestion control); and *Ethernet* (CSMA/CD), for which this course relies on knowledge. Revisiting the supervisors materials from Andrew's course should work (`http://www.cl.cam.ac.uk/teaching/1718/CompNet/supervisors/`). [1]

This also compliments the Concurrent and Distributed Systems IB course by Robert Watson: `http://www.cl.cam.ac.uk/teaching/1718/ConcDisSys/materials.html` which is useful for context (again look at last year's notes especially, as that is what this year's students should know).

There are several closely related part II courses in communications and systems, and I recommend a well rounded systems person should be interested in all of these (in US terms *audit*):

Security II `http://www.cl.cam.ac.uk/teaching/1819/SecurityII/`

Computer Systems Modelling `http://www.cl.cam.ac.uk/teaching/1718/CompSysMod/`,

Information Theory `http://www.cl.cam.ac.uk/teaching/1718/InfoTheory/`,

Digital Systems Processing `http://www.cl.cam.ac.uk/teaching/1718/DSP/`

2 Reading/Book

There isn't really a book that covers the material of this course, which reaches from underlying maths (graphs, control, optimisation) to real world experiences (BGP, TCP, Data Centers).

The maths is covered in Keshav's book on the Mathematical Foundations of Networking, `http://www.amazon.co.uk/Mathematical-Foundations-Networking-Addison-Wesley-Professiona dp/0321792106`, very nicely - indeed, it also covers material in other courses mentioned above, so for example

Chapter 1/Chapter 2 Discrete math/stats reminder:)

Chapter 5 - Chapter 8 Control (See also ch3, for 8.0)

Chapter 3/Chapter 4 Optimisation

[1]Delta to previous years this course were given in: No longer cover graph theory; compact routing; TCP with random linear codes; wireless ad hoc and mesh networks with network coding, or switches.

Chapter 6 Perhaps useful for the Computer Systems Modeling material

Chapter 9 Perhaps useful background for John Daugman's Information Theory course.

3 Systems and Layering

1. A communication system is generally made up of the following elements. Describe the function of each element and how it interacts with the system as a whole *(10 Marks)*:

   (a) Sender
   sender is origin of message. encodes it and transmits it

   (b) Receiver
   receiver is destination of message. receives and decodes it

   (c) Message
   The actual information to be conveyed

   (d) Channel
   The medium over which the message is sent. e.g. air, cable, fibre

   (e) External Constraints
   noise, errors, etc

2. System Design is the art and science of combining a number of different elements/resources into a complete system that functions as efficiently as possible. Typically resources can be quantified by a combination of Time, Space, Computation, Money and Labour. For each of the following design approaches explain the trade-offs that are being made and give a practical example of where the approach might be used *(6 Marks)*:

   (a) Multiplexing
   trades time and space for money. Might be used to distribute live TV over Internet or for shared memory

   (b) Batching
   Groups like tasks to reduce space and increase efficiency. Example might be ??????

   (c) Pipelining
   trades labour, space and computation to reduce time and money. Examples are pipelines in processors and assembly lines in factories

3. (a) Explain what is meant by the terms Abstraction, Hierarchy and Virtualisation. For each give an example of their use within a communications system *(6 Marks)*.
   Abstraction describes the process of defining a universal model for a system that is able to ignore the specific implementation. A good example is layering
   Hierarchy desscribes how different functions within the system relate to one another. The TCP/IP stack is strongly hierarchical
   Virtualisation describes the process of defining an abstract model for something ???????

   (b) Describe the difference between soft and hard state *(2 Marks)*.
   soft state uses implicit information you know about a system to determine the state. For instance TCP uses implicit congestion information to determine the state of the path. hard state requires an explicit message to determine the state. For instance a TCP connection requires a FIN to be closed.

(c) In the context of the Internet, give an example of each of explicit and implicit state notifications *(2 Marks)*.
explicit e.g. TCP handshake, ECN. Implicit, e.g. packet drop,

4. (a) List the 7 layers of the OSI protocol model. For each layer include a description of the functions and give an example protocol *(10 marks)*
From bottom up: 1: Physical layer, e.g. optical fibre, twisted pair. Carries the physical signal. protocols might be physical part of etehrnet (100-baseT), ATM. 2: Datalink layer. Frames the data into units bound for a given physical destination. examples are ethernet, PPP. 3: Network layer. Routes data to the correct destination and adds logical addressing over the physical addressing. protocols include IP. 4: transport layer. responsible for e2e connectivity, flow control and reliability. protocols include TCP and UDP. 5: Session layer. inter-host communication - connects the local and remote hosts. protocols include RTCP. 6: presentation layer. Deals with issues such as endianness, encryption. Presents the data to the application in the required form. protocols include telnet. 7: applicaiotn layer. This is the final layer that is actually going to use the data. protocols inlcude HTTP, NNTP, NTP, etc

(b) TCP/IP is often described as a 4-layer model. Give the 4 layers and show how this fits into the 7-layer model *(4 Marks)*.
1: Physical layer is equivalent to layers 1 and 2 of OSI model. 2: Internet layer is same as network layer in OSI. In TCP/IP this is always IP (IP over everything, everything over IP). 3: Transport layer is the same as OSI layer 4, but some argue it also includes some sessioning functionality. 4: Applicaiotn layer, This covers layers 5, 6 & 7 of OSI.

5. (a) Explain why layering is such a powerful design approach for the Internet *(4 Marks)*.
abstraction, allows you to consider all other layers as black boxes, etc

(b) What is meant by layer-violation? Give an example of where you might want to do this *(3 Marks)*.
layer violation = 1 layer changing something in another, or signalling between the layers. ECN is a good example

(c) Describe how the end-to-end principle relates to the concept of layering *(3 Marks)*.
e2e principle states that you do complex stuff at end hosts and allow middle to get on with just sending bits. The layering model allows one to cleanly separate end-host functions (layers 4-7) from netowrk functions (layers 1-3)

Exam Questions

2012 Paper 7 Q8(a)
2008 Paper 7 Q3
2014 Paper 9 Q10 (but only for 1b reminder)

4 Routing

1. Describe the dynamic alternative routing system *(5 Marks)*.
DAR is simple system used in telephone network. If a route is full, re-route a share of teh traffic via an alternative route. Choose alternative according to pre-determined

alternative (fixed tandem) or (better) via a sticky random tandem where you randomly choose an alternative but stick with it. Trunk reservation is a refinement that tries to reserve some bandwidth for calls where the link is the shortest path.

2. Compare and contrast intra-domain routing with inter-domain routing. Explain why distance vector routing is used for inter-domain routing and link state routing is used for intra-domain. Why can't they be used the other way round? *(10 Marks).*
Intra-domain routing should be fast, should scale to thousands of nodes, should be shortest path with no loops. Link state achieves all of these. Inter-domain routing needs to handle policy - sometimes want to route round a neighbour, avoid another downstream network, etc. This is not really feasible with link state because it is designed to ensure that each node has the information to find the shortest path to any other node. Distance vector allows you to specify the cost to reach any other node and hence allows you to favour some routes over others even if they aren't the shortest path. The problem with DV is that it takes a long time to converge.

3. (a) Describe how reverse path forwarding works in multicast. Explain how it ensures you find the shortest path *(2 Marks).*
Reverse path forwarding: Forward a packet FROM S iff it arrived on a link that is (one of) the shortest paths TO S

   (b) Explain what is meant by pruning *(2 Marks).*
Pruning refers to the act of cutting out unnecessary links by explicitly telling a parent node to stop sending you packets (e.g. you see the same packet twice from 2 parents, tell one of them to stop sending it)

   (c) Briefly describe 3 more multicast routing protocols *(6 Marks).*
At least 4 to choose from, described in notes:
***Multicast routing protocol** builds on "tunnels" that avoid routers that are not multicast capable, floods along shortest path including tunnels.*
***Distance vector multicast routing protocol (DVMRP)** - basically RIP for multicast uses hop count to find shortest path, uses reverse path forward, prune, rejoin and explicit join messages.*
***Multicast OSPF (MOSPF)** basically what it says on tin!*
***Core based Trees (CBT)** similar to DVMPRP but uses a core router - all join messages go via this which cuts out loops, etc.*
***Protocol Independent Multicast (PIM)** combines best bits of CBT and DVMRP - uses flood and prune for dense groups, core-based for sparse.*

4. (a) Explain the goals and at least two examples of Compact Routing Schemes. What are the disadvantages for compact routing in practice, and in theory? *Reduce routing table size from n ln(n) to something - depends on approarc - interval, landmark, or name independant. Downsides are stretch, and practical obstacle is multi-homing, and last, but not least, the high cost of updates. There's a really good introduction to this in the CCR paper* `http://ccr.sigcomm.org/online/files/p43-krioukovA.pdf`

5. (a) Describe the deployment scenario, goals and approach to Centralize routing based on fibbing. What could possibly go wrong? *See notes - deployment is as an add on to a link-state, intra-domain routing system; main trick is to introduce a fib/virtual node for every node on a path you want to influence. Downsides are increased convergence time if the controler goes down, and increased RIB size, and FIB computation times.*

6. (a) BGP Problems see revision slides `http://www.cl.cam.ac.uk/teaching/1516/PrincComm/revision/interdomain-remedial.ppt` Explain the operation of the Border Gateway Protocol, including some of the well known scaling problems. How are the two key relationships (peering, and customer/provider) supported? Explain the Stable Paths problem. What is a BGP Wedgie? *See notes - for BGP - see attributes like ASpath, community etc; SPP is really about convergence - BGP has a collection of rules for route selection (see notes) but a set of dynamics (annouce/withdraw) and topologies that can interact badly. The Wedgie is an extreme livelock example of this.*

Exam Questions

2018 Paper 7 Q 13
2017 Paper 8 Q 9(b)
2016 Paper 7 Q13
2014 Paper 7 Q12 2013 Paper 8 Q10
2011 Paper 7 Q11
2010 Paper 9 Q6

5 Flow Control and Control Theory

1. An ISP (Comcast perhaps :) decides to limit the rate of P2P traffic to a maximum of $r\%$ in any given link. If this is exceeded the ISP sends TCP reset packets to the P2P connection endpoints.[2]

   (a) Explain how this reduces the rate of P2P traffic. *(4 Marks)*
   TCP RST forces a TCP connection to close instantly without the 4 way close. Effectively this instantly kills a connection so if you can identify a TCP flow that is carrying P2P traffic and send a RST to the source the flow will die.

   (b) Using control theory identify the plant, the command, the control input, the disturbance and the output. *(6 Marks)*
   Plant is congested link, command is maximum percentage of P2P traffic, input is number of P2P connections that are reset, disturbance is the fluctuation in the number of active P2P connections, Output is the percentage of connections that need to be closed

   (c) Give a state space representation of this system. Assume that the control rule is to reduce the number of P2P connections by u connections over a time period T whenever the P2P fractions exceeds $r\%$. *(5 Marks)*
   See Keshav for the solution. Basically the natural state variable is the fraction of P2P traffic at a given time. If this exceeds threshold r, then the system reduces the number of connections by u. In other words given x is current fraction of P2P traffic and $\dot{x}$ is the level after T seconds, then

$$\dot{x} = \left\{ \begin{array}{ll} -\frac{u}{T} + w & if x > r \\ w & otherwise \end{array} \right\}$$

   (d) Find the transfer function for the system when the congested link has more than $r\%$ P2P traffic. *(5 Marks)*

[2] Taken from Keshav Chapter 8 Exercises 1, 3 & 4

They'll hate this! Basically it is taking the Laplace transform of the equation above for when x ¿ r. This simplifies to

$$\frac{X(s)}{U(S)} = G(s) = \frac{-1}{T}$$

where X(s), U(s) etc are the Laplace transforms

2. Network coding can be used instead of ARQ to cover for missing packets. Describe how linear combinations of packets can be transmitted. How are combined packets decoded? How can acknowledgements for a TCP like protocol be modified to indicate to senders what information has been received? *This is straight out of the mitzenmacher work, but also gaussian elimation which is covered in David Greaves Numerical Methods lectures in the last 3 years (see online materials for his course!). in* `http://www.cl.cam.ac.uk/teaching/1314/NumMethods/nummeths13slides.pdf`. *For TCP and Network coding, see* `http://www.rle.mit.edu/ncrc/wp-content/uploads/2013/12/2011_Network_CodingMeets.pdf`

Exam Question

2017 Paper 7 Q13
2014 Paper 8 Q10
2013, Paper 8 Q10
2011 Paper 9 Q9

6 Optimisation and Scheduling

1. You are the head of operations for a hot air balloon company. Flights are losing money and your job is to make them profitable again. The number of flights is fixed but you can control which launch location to use and the duration of the flight (there is a minimum duration of 15 minutes). The cost of a flight depends on the duration (to cover fuel, pilot's wages and the cost of the chase vehicle), the launch location and how far the landing site is from a road (as you have to compensate any farmer whose fields you have to cross). You have 1 pilot and up to 9 passengers and are free to set the charge to whatever you want but the number of passengers decreases as the cost goes up.

   Construct this problem as an optimisation (e.g. construct plausible transfer and objective functions). What are the fixed parameters? What are the inputs and outputs? What are the control variables? How would you empirically estimate the transfer function[3]? *(10 Marks)*
   This is quite an open-ended question. Fixed parameters are no. of flights, the list of take-off locations costs (of pilot, chase vehicle, fuel for hot air balloon, etc), possible launch locations, the actual road network. Inputs are things that determine where a given flight goes (e.g. wind speed and direction). Outputs are the distance the balloon ends from the road, the number of passengers (this is a tricksy one - this is not a control parameter because it is influenced by the cost of the ticket which is a control parameter...), the cost. Control variables are where the flight starts, how long it lasts and the cost of a ticket. The objective function is to maximise the difference between the income from selling tickets and the cost of the price (e.g. maximise the profit or

[3]from Keshav Ch.4 Ex.1

minimise the loss). Empirical estimation is the easy one - given wind speed/direction estimate the landing spots from each launch spot. Use market studies to estimate the relationship between cost of tickets and number of tickets sold. Over time determine the cost of average flight.

2. Optimisation-based congestion control aims to steer the network towards its optimal operating point, avoiding bottlenecks at any point, while maximising what economists call social welfare (the greatest number of end-users getting the best service they can). A new protocol called multi-path TCP (MPTCP) allows a multi-homed host to use two or more physical connections for a single TCP flow.

   (a) Discuss how a scheme like MPTCP might benefit the end-user. Consider throughput, reliability, etc. *(5 Marks)*
   MPTCP divides te flow into multiple subflows. Each subflow is managed as a stand-alone TCP connection, but (usually) their windows are linked so they remain TCP fair. Benefits are potential improved throughput (2 pipes = twice the throughput), improved reliability (same traffic on two pipes? Forward error correction info on one, data on other?), Allows you to use multiple providers - overcome fair use policies? Overcome provider-based destination blocking.

   (b) How might this improve the overall performance of the network, and how does this help achieve global optimisation? *(5 Marks)*
   This approach will spread the load more evenly across the network, effectively increasing the size of the resource pool available to each connection. This reduces congestion which benefits everyone, it also allows providers to make more efficient use of their resources (by helping balance the load away from bottleneck links). Ultimately could achieve a globally optimal solution that maximises social welfare and minimises costs.

   (c) MPTCP is recommended to use a coupled congestion controller [RFC6356] to ensure fairness when competing with other traffic. What is meant by fairness in this respect? Describe the operation of the coupled congestion controller. *(5 Marks)*
   TCP fairness means that at any given bottleneck your flow should perform no better than it would have had you been using single path. The coupled controller takes this a stage further and ensures that you don't get an unfair performance gain across all the subflows. Windows in each subflow grow independently (e.g. 1 packet per RTT). But if you hit congestion then the window on that flow is reduced in proportion to the total window size across all your flows.

3. The max-min fair share criterion for allocation m_n of resources to a set of N flows, with respective demands x_n for a resource of capacity C, can be computed using the following equations:

$$m_n = min(x_n, M_n), \quad 1 \leq n \leq N$$

$$M_N = \frac{C - \sum_{i=0}^{n-1}(m_i)}{(N - n + 1)}$$

   (a) Explain, perhaps with the use of an example, how this criterion operates to mitigate between over- and under-demands[4]. *(5 Marks)*
   Easiest is to use a basic example. If you have capacity C = 12, 4 flows wanting

[4] From 2010 Paper 7 q.9

2, 2.5, 3.5, 6. Divide the available bandwidth 4 ways (to get 3). All flows are now given 3. This leaves a remainder of 1.5 (because 2 flows wanted less than 3). The remaining 1.5 is divided into 2 again to give 0.75. Flow 3 only wanted an additional 0.5 so there is 0.25 left which can go to the final flow giving 2, 2.5, 3.5, 4 as th split. Basically in each "round" anyone who asks less than their share gets it, then the remainder is shared between the next set of flows, etc until all flows have a share.

(b) Contrast max-min fairness with proportional fairness. Try to do this with reference to real-world examples. *(5 Marks)*
Proportional fairness splits the available load between the demands in proportion to their demand. So flow 4 gets 6/14, flow 3 3.5/14, flow 2 2.5/14 and flow 1 gets 2/14. This is used in CDMA. Round robin is an example of max-min

Exam Questions

2014 Paper 8 Q9
2012 Paper 8, Q10

7 Contention and Traffic

1. Traffic sources can be described as elastic or inelastic

(a) Explain these two terms and give examples of each sort of traffic. *(4 Marks)*
Elastic traffic is defined as traffic that, whatever throughput it currently receives, will get a better utility from an increase in throughput. Bulk data transfer is the classic example. Inelastic traffic doesn't follow this, and indeed often has a minimum throughput below which it gets 0 utility. Best example is live streaming traffic.

(b) These types of traffic require very different scheduling and resource guarantees. Discuss these differences in detail taking into account contention at queues and buffers, delay and jitter. Include suggestions for how network designers might optimise their network for both types of traffic. *(6 Marks)*
Elastic traffic can benefit from bursting into any available space in the network, but is also able to back off during busy periods. By contrast inelastic traffic needs a certain reserved bandwidth, but can't make use of any excess capacity. Usual solution is to effectively provide two networks - a high QoS network with reserved channels with specific characteristics (defined by e.g. DifServ) and a best effort network that can scavenge any unused space in the higher QoS classes. This may well require complex queueing, policing and shaping at network ingresses.

2. A major TelCo wants to redesign their telephone network to reduce the number of telephone exchanges they need. This will allow them to sell the old exchange buildings and make a healthy profit. Currently the TelCo uses Time Slot Interchange switches with each switch able to handle 10,000 calls simultaneously. They want to increase the size of the switch so it can handle 100,000 calls.

(a) Explain the operation of TSI. Explain the pros and cons of this approach. *(3 Marks)*
Basically re-shuffle the timeslots. So assume you have something in slot 1 wanting to go to slot 10, just move it to that slot at the output. Pros are it is REALLY

easy, cons are that you can't manage to cope with large numebrs of flows as you end up having to have ridiculously short time slots.

(b) Why might it not be possible to use TSI for the new larger switches? *(2 Marks)* *because if you are going to limit the switching delay you need to have slots that are at most a few microseconds long. This in turn means the data switching time (assuming standard 56kbps voice circuit) is of the order of a few fractions of a nanosecond (impossible currently)*

(c) Describe 2 alternative systems that the TelCo could use instead. *(5 Marks)* *Several alternatives spring to mind including space division interchange systems, and combined time and space or event time, space, time...*

3. Many broadband users in the UK are using DOCSIS (cable) to access the Internet. DOCSIS uses TDMA to provide access to the uplink capacity.

   (a) Why does this make the uplink a bottleneck? *(2 Marks)* *TDMA means that the uplink is the shared medium. In other words all users are conteding to access the available uplink BW (whereas in ADSL it is the downstream that has contention).*

   (b) Why is it perceived to be unfair when users are persistently uploading (e.g. P2P). *(3 Marks)* *The uplink is divided into time slots. Each time a slot comes up it is offered in turn to every user. If you have data waiting to send then you get the slot. If you are doing bulk upload then you will ALWAYS have data to send and hence will get a huge share of the slots. This is perceived as unfair (all users pay the same, a minority get a majority of the service)*

   (c) One suggested solution to this is called Fair Share [RFC6057]. Describe this solution in detail. *(5 Marks)* *Go read the RFC! Basically divide time into coarse intervals (10 mins perhaps). If in any time interval you notice that the uplink is very busy then look at which users were taking the greatest share. These users are then given a lower priority in the next interval - they are only offered uplink slots after all the normal priority users have been offered them.*

   (d) Prior to adopting Fair Share, Comcast was found to be using TCP RST messages to close certain TCP sessions. Describe in detail what the effect of this was and why this was seen as going against Net Neutrality. *(5 Marks)* *TCP RST has the effect of closing the connection instantly. This of course then means that traffic "goes away". This is anti net neutrality as they had to explicitly use DPI to identify certain applications, then target the resets at those apps. This is definitely not neutral traffic management! Would be nice if students discussed net neutrality and showed some understanding of the issues in the US and Europe (Which differ markedly)*

Exam Question

2018 Paper 8 Q10
2013 Paper 9 Q10
2012 Paper 7 Q11
2012 Paper 8 Q9
2009 Paper 8 Q5

8 Future

A student continuing to part III (or the MPhil) might be then interested in these courses:

Network Archteicture paper reading course was this: `http://www.cl.cam.ac.uk/teaching/1516/R02/materials.html`

Principles of Data Science `http://www.cl.cam.ac.uk/teaching/1516/L120/`

Data Centric Systems and Networking `http://www.cl.cam.ac.uk/teaching/1516/R212/`

Computer Security: Principles and Foundations `http://www.cl.cam.ac.uk/teaching/1516/R209/`

Building an Internet Router `http://www.cl.cam.ac.uk/teaching/1516/P33/`