Networking: Other Transports, NAT

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Transports in wide use:

- UDP: thin wrapper over IP, unreliable, unsequenced
- TCP: complete transport service, offers reliable, sequenced delivery with guarantee of either success or a positive failure indication.
- Together majority of Internet traffic

RTP

- Real-time Transport Protocol
- Used to transport voice (telephony) and video (streaming) in some applications.
- Doesn't do anything you can't do yourself with UDP.

Problems for voice and video

- Consistent timing
- Choice between dropping and catching up
- Trade off with buffering

For telephony…

- Usual claim is anything over 35ms latency is problematic for conversation (“toll quality”)
  - Figure has no experimental basis
  - Partly about echo cancellation, partly about difficulty in maintaining conversation
- 35ms is easy to achieve in traditional telephone networks (roughly 10k km speed of light) but is difficult to achieve reliably in IP based networks with slow/ congested local links.

Reality is more generous

- Latency over networks with complex compression (“codecs”) is higher, GSM for example.
    - Although GSM has no “side tone”, which is why people shout in mobile phones.
- Increasingly, people will tolerate GSM-quality voice (~3kbps) rather than “toll quality” voice (~56kbps).
- Counter example is difficulty people have with geo-stationary satellite communications (ie 1960s/70s phone calls to Australia), but there latency approaches 500ms with heavy echo cancellation.

RTP

bit offset	0-1	2	3	4-7	8	9-15	16-31
0	Version	P	X	CC	M	PT	Sequence Number
32	Timestamp						
64	SSRC identifier						
96	CSRC identifiers ...						
96+32×CC	Profile-specific extension header ID						Extension header length
128+32×CC	Extension header ...						

RTP

- Each packet contains a sequence number, which can be used to spot gaps and re-order packets.
- But each packet also contains a time-stamp (resolution decided when the stream is set up)
  - Say, 8KHz for voice, as voice is most commonly 8KHz sampling rate, 4KHz bandwidth
  - Or frame-rate for video

Difference with TCP

- No acknowledgements.
- Receiver knows when packet was sent, and how many were sent.
- Receiver can therefore discard packets in order to stay “current”, or can pause replay to wait for arrival of missing packets, or some other strategy.
- Duplicates are detected.

RTP Setup

- RTCP (“real time control protocol”) used to set up video replay and similar
- SIP (“session initiation protocol” used to set up Voice over IP telephony.
- Co-ordination of RTCP/SIP session with RTP stream is difficult for firewalls: in voice-land, “Session Border Controllers” combine SIP and firewalling, while emptying your wallet.
- Most video streaming now uses traditional TCP with sufficient buffering to deal with variation in latency, plus heavy compression with MPEG/etc.

SCTP

- Stream Control Transport Protocol
- Attempt to tunnel traditional voice signalling (“SS7”) over internet.
- Again, UDP with a few extra facilities
- Largely moribund

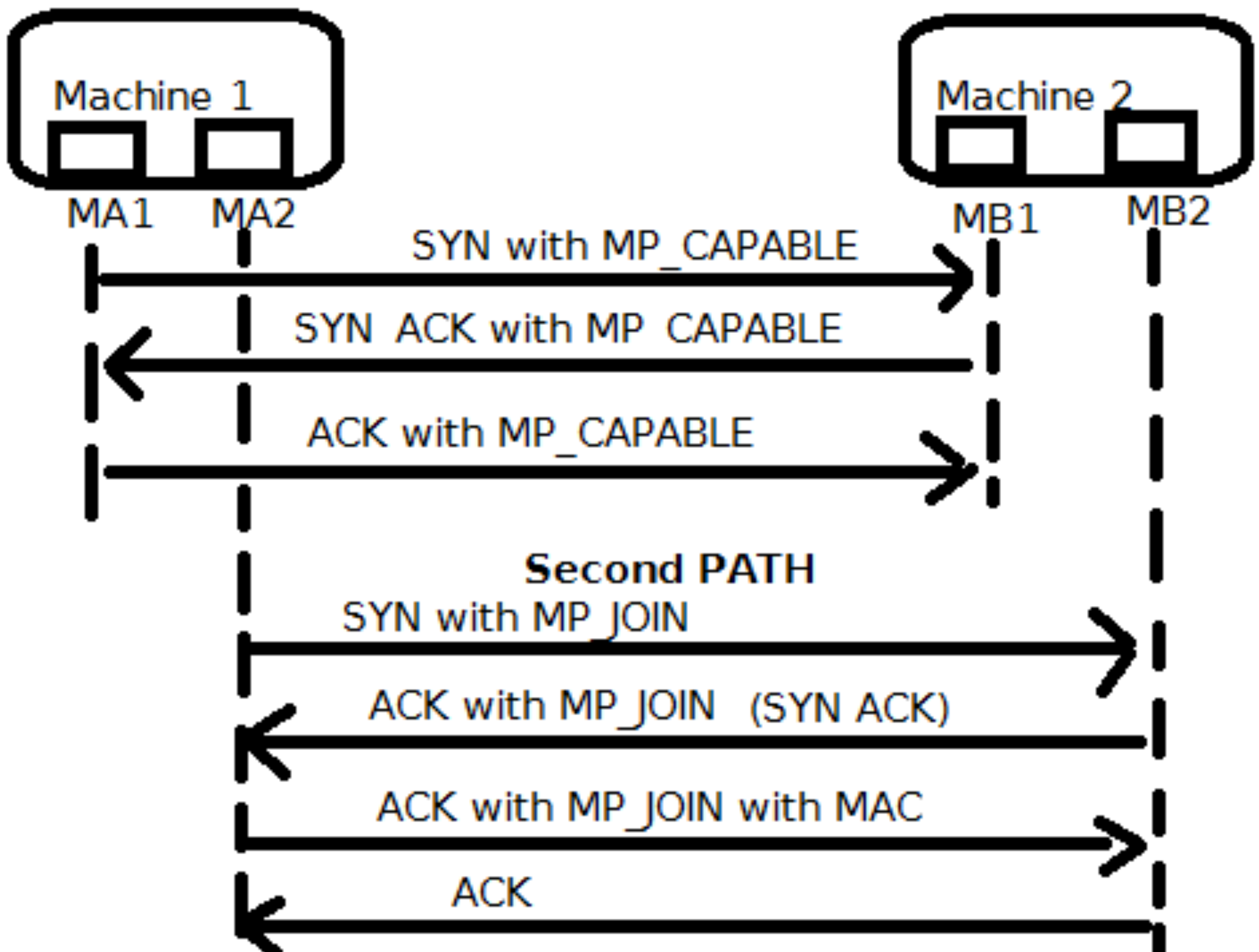
DCCP

- Datagram Congestion Control Protocol
- Another UDP plus frills, again for time-sensitive delivery.
- Again, moribund
- General lesson: “UDP plus a bit” is too complicated if it is general, insufficiently attractive to implementors if it is too specific.

Multipath TCP

- Now something more exciting!
- RFC6824 is well worth reading
- Allows multiple paths to be used by one TCP connection
    - For example, Wifi **and** 4G **simultaneously**

Multipath TCP



Not only performance

- By having a link multiplexed over WiFi and 4G, failure of one path appears as just some packet loss, and the link rapidly reconfigures.
  - This is very hard otherwise, as you will have different IP numbers in each realm
- Also makes effective use of multiple network cards, particularly in networks with a lot of resilience / redundancy.

New, but growing

- Implemented in iOS 7 et seq
- Reference implementation in Linux (much of the data centre world)
- Coming soon in Solaris (rest of the data centre world)
- Doesn’t require significant application changes, most applications work unmodified (may require recompilation)
- Looks promising

Address Translation

- Mechanism to extend scarce IP numbers
- Incidentally provides some security, although this was not a design goal and should be treated with care
- Breaks “end to end principle”
- Causes some people (such as me) to start shouting uncontrollably

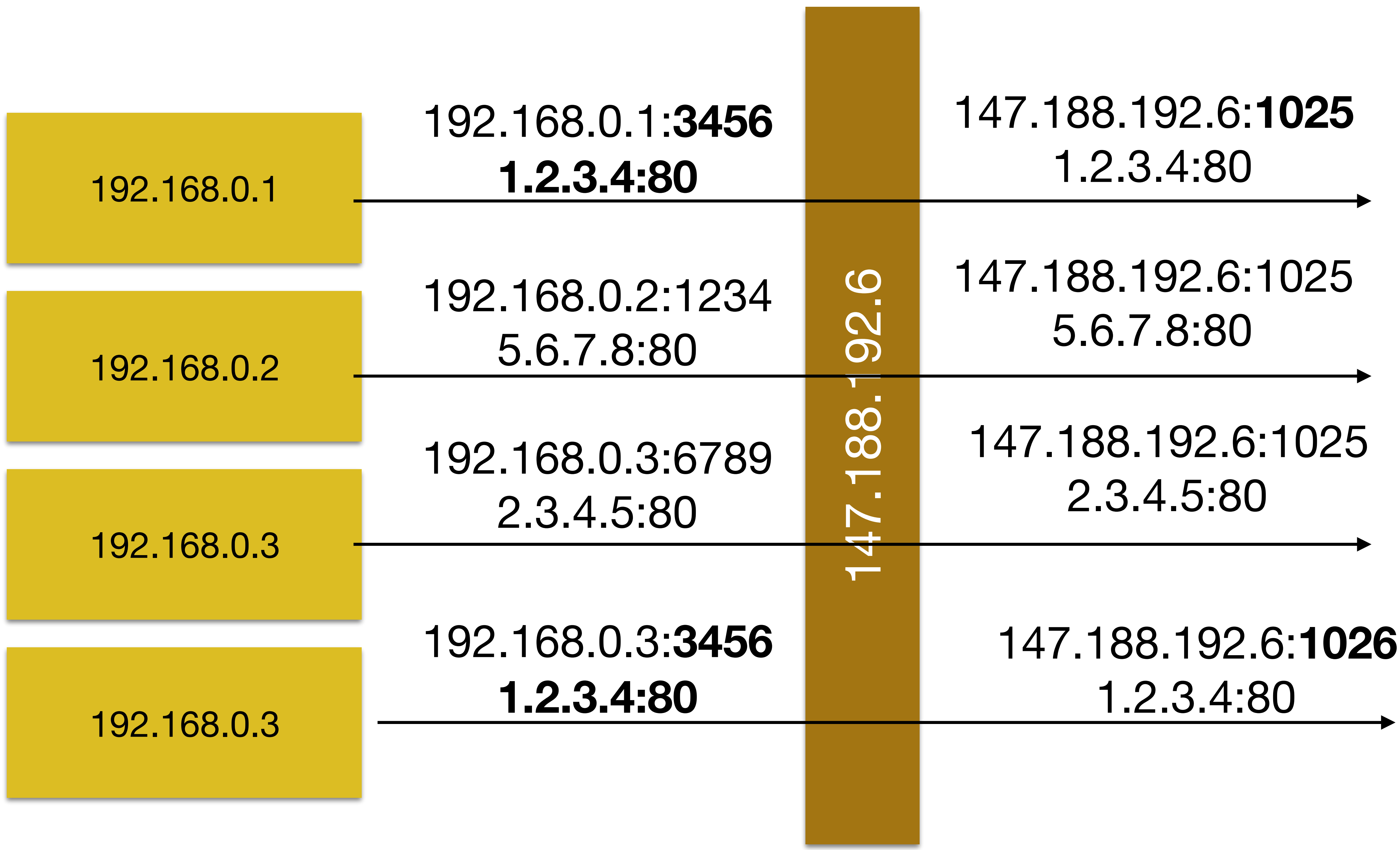
Basic Principles

- Outbound NAT:
  - Connection is modified so that connections from multiple source IP addresses are encoded into port number space of a smaller number of addresses
- Inbound NAT
  - Connection is modified so that connections to multiple ports on a small number of IP addresses are expanded out to a large number of addresses

Recall:

- TCP connection identified by source IP, source port, destination IP, destination port.
- So long as one element in the quad is different, it’s a different (and distinguishable) connection
- Destination IP and port identify called service
- But the source can be changed

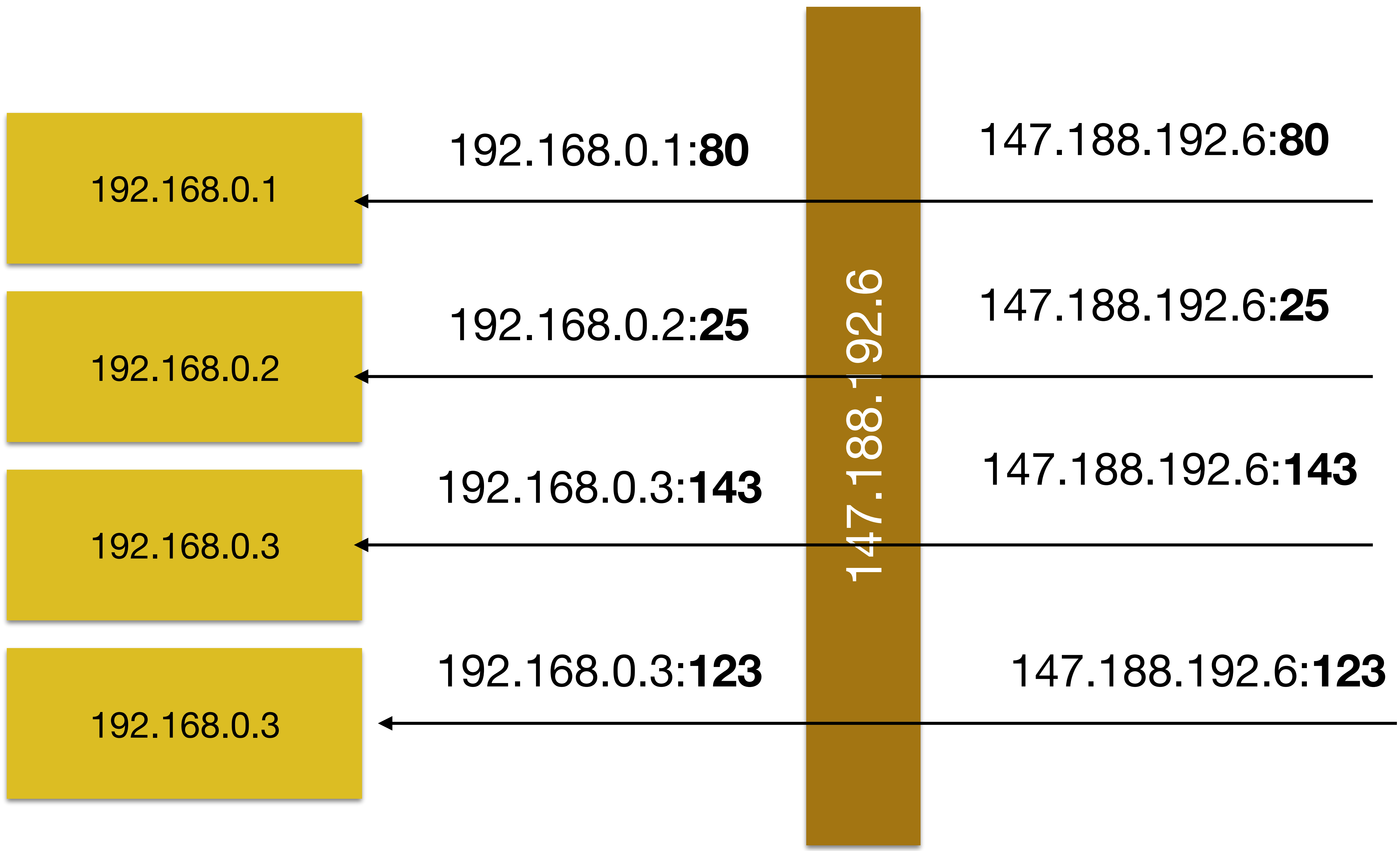
Outbound (Source) NAT



In reality…

- Often not necessary to overload port numbers as shown: each connection gets distinct source port number
  - Gives 65535 connections per IP number
- Large installations use multiple IP numbers at NAT point

Inbound (Destination) NAT



Inbound NAT

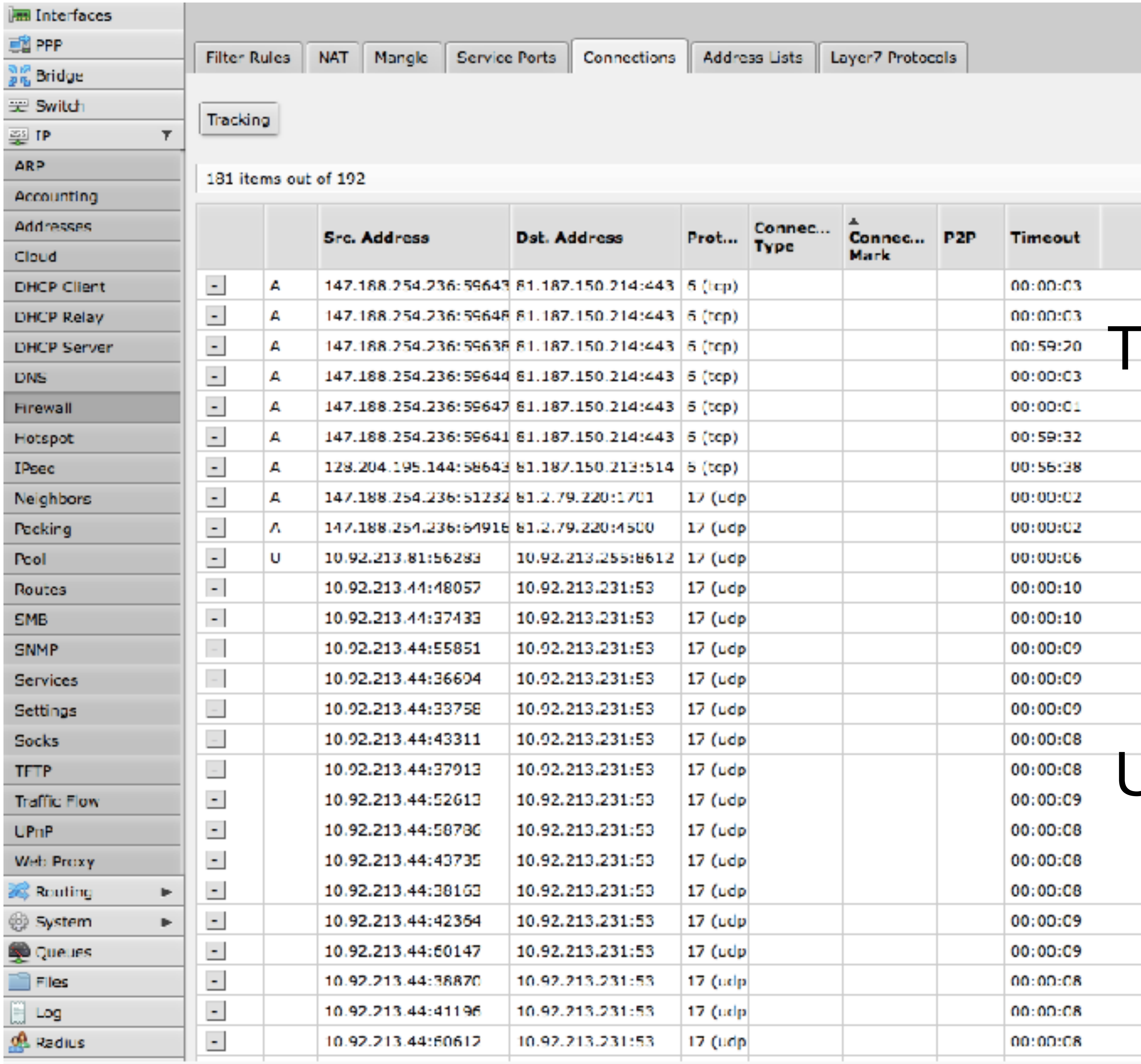
- Used to offer multiple services from single IP number (goes well with virtualisation to minimise attack surface)
- Also used in more complex situations to offer load balancing, failover, mobility, etc

NAT for TCP

- NAT device sees “SYN” packet and builds a mapping between inside and outside addresses.
- Modifies TCP packet (including IP header, as involves change to source address to be own), recomputes check sums, sends packet
- On receipt of packets, looks at source IP and port and destination port, performs reverse mapping and sends packet.
- Tracks TCP state, and deletes entry from translation table when FINs have all completed.

NAT for UDP

- No “state” as such.
- Rewrite outgoing UDP and then accept return packets until there is silence for 10s (typically).
- Can also impose limit on number of replies, as for example DNS.



		Src. Address	Dst. Address	Prot...	Connec... Type	Connec... Mark	P2P	Timeout
-	A	147.188.254.236:59643	81.187.150.214:443	6 (tcp)				00:00:03
-	A	147.188.254.236:59648	81.187.150.214:443	6 (tcp)				00:00:03
-	A	147.188.254.236:59638	81.187.150.214:443	6 (tcp)				00:59:20
-	A	147.188.254.236:59644	81.187.150.214:443	6 (tcp)				00:00:03
-	A	147.188.254.236:59647	81.187.150.214:443	6 (tcp)				00:00:01
-	A	147.188.254.236:59641	81.187.150.214:443	6 (tcp)				00:59:32
-	A	128.204.195.144:58643	81.187.150.213:514	6 (tcp)				00:56:38
-	A	147.188.254.236:51232	81.2.79.220:1701	17 (udp				00:00:02
-	A	147.188.254.236:64916	81.2.79.220:4500	17 (udp				00:00:02
-	U	10.92.213.81:56283	10.92.213.255:8612	17 (udp				00:00:06
-		10.92.213.44:48057	10.92.213.231:53	17 (udp				00:00:10
-		10.92.213.44:37433	10.92.213.231:53	17 (udp				00:00:10
-		10.92.213.44:55851	10.92.213.231:53	17 (udp				00:00:09
-		10.92.213.44:36604	10.92.213.231:53	17 (udp				00:00:09
-		10.92.213.44:33758	10.92.213.231:53	17 (udp				00:00:09
-		10.92.213.44:43311	10.92.213.231:53	17 (udp				00:00:08
-		10.92.213.44:37913	10.92.213.231:53	17 (udp				00:00:08
-		10.92.213.44:52613	10.92.213.231:53	17 (udp				00:00:09
-		10.92.213.44:58780	10.92.213.231:53	17 (udp				00:00:08
-		10.92.213.44:43735	10.92.213.231:53	17 (udp				00:00:08
-		10.92.213.44:38163	10.92.213.231:53	17 (udp				00:00:08
-		10.92.213.44:42364	10.92.213.231:53	17 (udp				00:00:09
-		10.92.213.44:60147	10.92.213.231:53	17 (udp				00:00:09
-		10.92.213.44:38870	10.92.213.231:53	17 (udp				00:00:08
-		10.92.213.44:41196	10.92.213.231:53	17 (udp				00:00:08
-		10.92.213.44:60612	10.92.213.231:53	17 (udp				00:00:08

TCP, 60m timeout

UDP, 10s timeout

Problems with NAT

- It’s evil :-)
- Makes it very difficult to authenticate and log users
- NAT logging is part of “carrier grade NAT”, but requires time alignment of log on remote server and at the NAT point

Timing Problems

- my.popular.dom.ain server 1.2.3.4 has abusive connection from 147.188.192.6:1234 at 10:25:40
- 147.188.192.6 logging (if available) shows 1234 used for connections to 1.2.3.4 by 192.168.0.1 at 10:25:10 and 192.168.0.2 at 10:25:50.
- NAT logs won’t include URL, just IP number
- Who called my.popular.dom.ain? Requires **retrospective** knowledge of clock offsets.

Logging Problems

- Most web logging does not record source ports. It can, but usually doesn’t.
- So very difficult to request logs from NAT point, as there will be multiple connections to the same popular service, distinguished only by source port
- Claimed by law enforcement to be a serious problem.

Delays the IoT

- Internet of Things implies universal connectivity
- NAT delays universal connectivity, by making RFC1918 IP numbers usable for client devices.
- “Carrier Grade NAT” can even use RFC1918 for customer lines, NAT’d once at customer border and again at ISP border.

IPv6 has no NAT

- IPv6 does not require NAT, as plenty of addresses for everyone.
- IPv6 implementations don’t support NAT
- There are already proposals for IPv6 NAT, because of (bogus) security concerns.

NAT “Security”

- NAT is conceptually a stateful firewall: each TCP connection is being tracked for state, each UDP “connection” is being at least monitored for volume and duration
- Tendency to regard this as an actual firewall, cf. PCI-DSS requirement for NAT on low-end companies.
- NAT products not certified or designed for security
- To complicate matters, often common code (Linux NAT functionality is in iptables firewall).

Inbound NAT

- This is particularly confusing for inbound NAT
- Inbound permits connection to port 80 on outside of NAT to appear as connection to port 80 on internal machine.
- There is **no security** in this at all: even if the NAT point is regarded as a firewall, this is a complete pass-through.
- Yet inbound NAT is still used as a “security” feature.

Complications for NAT

- Protocols which embed IP numbers in control streams break under NAT, because the IP numbers are wrong.
- FTP is the worst offender, and requires custom NAT modules to re-write the contents of the control stream.
- Modified FTP (“Passive Mode”, “PASV”) is better solution, or just don’t use FTP (please, just don’t use FTP).

Complications for NAT

- IP-address based authentication schemes lose resolution, because all of a site appears as one address.
- Such schemes were arguably broken anyway, but are popular in academic publishing. Solutions involve complex proxying, but real solution is better authentication strategies.

Extra NAT protocols

- UPnP (“Universal Plug ’n’ Play” — who, one has to ask, names these protocols?)
- Allows “inside” devices to communicate with a NAT point and request inbound NAT, effectively automating a bypass of any firewall.
- Used heavily in residential products like Web Cams and “personal cloud” type products, as well as VoIP.
- UPnP is a dream for malware, as it makes opening a connection to a command and control server particularly easy.

Summary

- Quite a few alternatives to TCP and UDP, mostly used only for voice.
- Multipath TCP looks very promising.
- NAT is a necessary evil, but please, IPv6.