Network and Cybersecurity

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Contents

1 Computer networks

A system connected to the internet is called a **host / end system**. This is done through a **communication link** and **packet switch**. By this it has a **transmission rate** which describe the speed measured in bits/second.
When data is send it is done through **packets** which consist of a data header and including data. A packet is send first to the packet switch often taking form of a router or link-layer-switch. Which then sends it further to the communication link.
ISP then interconnect the communication links and connects to a higher up ISP or between higher up IPS. The highest tier is tier 1 of which other ISP connect with, even multiple such a **multi-home** setup is done in case of failure. An ISP may also connect to a **PoP** which simply are a group of routers from another ISP.
ISPS may also peer between each other to reduce exchange with higher tier ISP and reduce cost. Most often are peering free and third-party companies may even create a IXP where multiple ISP can connect and peer to each other.
Big companies like Google also have their own private network infrastructure, which spans the globe between their datacenters. They are therefore peering between same tier ISPs and are also connected to 1 tier ISP but with reduced exchange.
Edge network refers to the connections to all the end host.

1.1 Different types of connections

Home access can be done in different ways:

- DSL - Digital subscriber line is done through a telephone lines in higher frequencies such phone and network can coincide. Current max limit of downstream is 1 Gbps, but the whole system requires small distance to the telco provider 8 - 16 km.

- Cable internet access - Internet done through the existing television coax cable or hybrid fiber coax (HFC), a modem is then used to translate the analog signal to digital signal. This allows up to 1.2 Gbps downstream.

- FTTH - Fiber to the home is a direct connection from the cable office to the home. Here are two types of active optical network (AON) and passive optical network (PON), PON works such a network terminator at the home goes through a splitter between up to 100 homes into the

terminator at the telco.

- 5G fixed wireless - This is a wireless solution which are beginning to gather popularity in cities.

At the home a Local Access Network is then created, and most often extended to a Wireless Local Access Network using a router using the Wi-Fi protocol.

1.2 Mediums of signals

When a signal is sent through a physical media it is called guided media whereas over a wireless media it is called unguided.
Example of mediums are:

- Twisted cobber wire - used for telephone connections a still in use, for ethernet and more. A communication link done with this is called unshielded twisted pair (UTP). Modern cables of category 6a can transfer up to 10 Gbps over hundreds of meters.
- Coaxial cable - much like the twisted cobber cable, but the two copper conductors work concentric rather than parallel.
- Fiber Optics - Works by sending light instead of electricity, can have speeds in up to hundreds of Gbps over distances up to 100km due to no electromagnetic interference.
- Terrestrial Radio Channels - radio signals which characteristics depend a lot on environmental factors. Can be found in three categories, short distance up to 2 meters, medium distance up to a few hundred meters and long-distance spanning tens of kilometers.
- Satellite Radio Channels - Works as a transmitter receiver between two points on earth. Either in a geostationary orbit or low-earth orbit. geostationary eliminate the need of always finding the optimal satellite but at consequence of 280 ms delay.

1.3 Packet switching

Most packet switches use **store-and-forward transmission**, such before forwarding a transmission all packages have to be obtained and processed before transmission.
This therefore extend the transmission time according to package number

and speed.
When the router has processed all packages, they can be send as one unit.
This therefore can be described with:

$$d_{end-to-end} = N\frac{L}{R}$$

Where N is number of links, L is number of bits in a package and R is the speed.

In case of a package is already being sent the new package is sent to the **output buffer**, and in case of a full buffer **packet loss** will occur.
A packet switch most often has multiple queues for different forwarding packages. The packages are placed based on a forwarding map, using the packages IP address.

1.4 Circuit switching

A circuit switch networks works on reservations. Instead, if queues, a reservation is made to the receiver in both bandwidth and that a port is open for the package.
This also has the advantage that the number of links can be ignored, in transmission time.
In this way a package can be guaranteed to come at a specific time and no loss of any package.
To route more connections either **frequency-division multiplexing** or **time-division multiplexing** can be used.
FDM splits the available frequencies into smaller bandwidth of which data can be sent through.
TDM splits up the full bandwidth into frames which are reserved with a number a number of slots for data. Therefore, the speed is dependent on frame rate multiplied by number of bits in a slot.
The downside of circuit switching compared to package switching, is the need for allocated space for every user, even in inactivity whereas package switching can work with changing demand.

1.5 Delay

There are different kind of delays the most important are:

- Nodal processing delay - delays in the router such as determine package header or bit error correction (μs)

- Queuing delay - delays in case of heavy traffic and non-empty buffers (μs - ms)
- Transmission delay - delays that occur when waiting for a whole package and transmission rate is low (μs - ms) - $d_{trans} = \frac{L}{R}$
- Propagation delay - delays from the actual bit transportation limit by medium (ms) - $d_{prop} = \frac{m}{s}$

These in total are the total nodal delay.

1.6 Protocols

A protocol defines the format and the order of messages exchanged between two or more communicating entities, as well as the actions taken on the transmission and/or receipt of a message or other event.

Protocols are therefore used everywhere; such everything works as expected in between the layers of the network.
When different layers of protocols work together it is called a protocol stack, this may be for the internet:

- Application - These are protocols which are sent pr application such as HTTP, SMTP, DNS and such, referred as message
- Transport - These protocols are for sending application messages, here TCP and UDP, referred as segment
- Network - Protocols for distributing the segments using IP protocols or other routing protocols therefore referred as IP layer
- Link - Protocols here are used for the IP layer to transmit to until reaching the physical layer, this is protocols such as Ethernet and Wi-Fi, referred as frames
- Physical - Protocols for transmitting frames, here protocols depend on medium such as for fiber

When data goes through each layer, a new header is added, and previous layer data is used as payload.

1.7 Network attacks

Some of the different kind of internet attacks include:
Denial of service attacks (DoS) which prevent access to network host or infrastructure.
It can work by 3 ways:

- Vulnerability attack - Few messages which use a exploit such the host either stops or crashes
- Bandwidth flooding - A large number of packages is sent to the host resulting in overfilled buffers
- Connection flooding - A large number of open TCP connections are made such every entry is occupied

For flooding attacks, they may also be a distributed Dos (DDoS) attack through multiple devices often in a botnet.

A packet sniffer is a type of passive attack of which a device will listen for packages and possibly find sensitive information in packages sent through it.
IP Spoofing is a package which source address is spoofed, and therefore faked to look like a different source.

2 Protocol layers

For an application communicates to network layer, a socket is setup, which identifies by a port. An app-layer protocol defines

- Messages exchanges - request, response,...
- Message syntax - what fields are in a message
- Message semantics - Meaning of fields
- Rules for send and responds

Transport service requirements - The different requirements an application may set for the transport of data
The different types of requirements an application may set

- Data loss - May be strictly no loss or just some percentage
- Throughput - May need a constant throughput or can be elastic

- Time sensitive - May need a minimum latency for data
- Security - Some data may be more sensitive than other

2.1 Client server architecture

The server side is on an always on host, with permanent ip, which handles backend stuff.
Ex. are hosting API and database.
The client in this architecture communicates via HTTP protocols are some alike.
The clients are the dynamic part, may not be on same ip, or connect with each other

2.2 P2P

No host involved but clients communicating directly between each other.
Are scaled based on number of users.

2.3 TCP Service

Reliable transport with flow control to not overwhelm receiver
Able to throttle sender in an overloaded network.
Does not provide timing, min. throughput guarantees or security
Need a setup between client and server established before use.

2.4 UDP

Unreliable data transfer.
Does not provide reliability, flow control, congestion control, timing, throughput guarantee, security
Smarter for data which can handler larger loss.
TCP may get stuck on a packet whereas UDP would simply skip the package and go forward for the next package.
UDP is also faster and connectionless therefore making it possible to handle more connections faster.
The header includes 4 16 bits data: source port, destination port, length of data, checksum
The checksum works by the data is split up into 16-bit words, then they are summed, and overflow is wrapped around. Then the sum is converted to 1s

compliment (everything is flipped). This will then be the checksum.
If the checksum fails, the data is either discarded or delivered with an error.

2.5 Domain name system

DNS binds name/string to an ip address.
This also includes alias names for mail server or sub domains.
Useful for more dynamic ip setup and readability
Implemented via a distributed database.
The databases stores resource records (RR) in the format (name,value,type,ttl)
The DNS is implemented in a tree structure.
First are the root DNS which contains addresses of top-level domain severs like .com, .org, .dk
They then contain then the DNS servers for domain such as google.com, which itself contains addresses for subdomain.
So, when a request is made first, the cache is checked, then a request to a preinstalled root such as 1.1.1.1 is made.
This then returns a top-level address for a DNS server, which a request is made to.
This is repeated until an ip matching the request is returned.
This is the iterative method.
The recursive query finds the ip via the DNS servers which contacts each other putting the burden on them
Local DNS servers does not belong to hierarchy but is hosted by an ISP.
TTL (time-to-live) is a numerical value representing the amount of server hops a packet can make before being outdated.
The local DNS handles the requesting and caching for an IP to the hierarchy DNS servers.

2.5.1 Types of resource records

To do a lookup the dig tool can be used

- Type=A - name: hostname, value: ip-address
- Type=NS - name: domain, value: nameserver
- Type=CNAME - name: alias, value: hostname
- Type=MX: name: '@', value: mail server

2.5.2 DNS protocol

Query and reply are in the form Message, header
The header consists of 12 bytes dedicated to

- 16-bit identification
- 16-bit flags
  - Query (0) or reply (1)
  - 4-bit opcode: standard query (0), domain name form ip (1), status request of server (2), (3) is reserved for status an not used
  - AA: The server is authoritative (1), non-authoritative/cache (2)
  - TC: the message exceeds 512 bytes and are truncated (1)
  - RD: Recursion desired (1)
  - RA: Recursion available from server (1)
  - Zero: 3 bits of zeros reserved
  - rCode: Response code, no error (0), format error (1), server failure (2), did not find name (3), request is supported (4), policy denies execution of query (5)
- 16-bit Number of questions in body
- 16-bit Number of answers RRs and is 0 from client and set by server
- 16-bit Number of authority RRs and is likewise 0 from client
- 16-bit Number of additional RRs

The body then consist of

- Questions - query from client
- Answers RRs- Response to query from non-authority
- Authority RRs - Response to query from authority
- Additional information

2.5.3 Security

A person could bombard the DNS servers with traffic and deny other traffic in form of DDoS.
Not successful to date on root server but TLD (top-level domain) has succeeded
NXDOMAIN attack is requesting non existing domains and spending the DNS server resources to find non valid addresses.
Random subdomain attack is like NXDOMAIN attack but with subdomains to target the namespace rather than root or TLD.
Phantom domain attack is setting up DNS servers which does not respond or very slow responses, such in a recursive lookup the TLD will have to wait for response.
TCP SYN is the attack of which a bunch of TCP request are opened but never used.
DNS domains lock-up is an extended TCP SYN attack where after a connection is established random packages is sent to the server, and the server will wait for a correct response.
DNS rebinding attack is used to get past browsers same-origin policy, this is done by first the user lands on a shady website, the website then makes a request to itself, but the dns record is updated to point at a new site which the script now can be run upon.
DNS cache poisoning is where an attacker imposes as a nameserver, and then creates a request for the nameserver and answers before the real nameserver and thereby creating a fake lookup in the cache.

3 Web and HTTP

3.1 HTTP

A HTTP request is sent by the client, and server sends using HTTP protocol an object in response
The request is sent at port 80 using a TCP request.
Non-persistent HTTP send a single object and then closes
Persistent HTTP can send multiple files between client and server
Response time RTT is the time for a small packet to travel from client to server and back.
Persistent has longer open connection but every referenced object can be sent at as little as one RTT
Non-persistent requires 2 RTT at least pr referenced object and loses a lot of time to OS overhead for each established connection.

The general request consists of: method, url, protocol, headers
The general response consists of: protocol, status code, status phrase, headers, data
There are 4 method types for HTTP/1.1

- GET - get resource
- POST - send resource
- HEAD - meta data to check for updates
- PUT - Uploads file in entity body to url field
- DELETE - delete file in the url field

The most common response status codes

- 200 OK
- 301 Moved Permanently - object is moved to new location given in message
- 400 Bad Request - not understood by server
- 404 Not Found - Fil not found on server
- 505 HTTP version not supported
- 418 I'm a teapot - When a teapot is requested to brew coffee

3.2 Cookies

Cookies are the solution for http being stateless.
Cookies allows to store files in the browser of the user.
This can be used for saved storage like shopping cart or authority like a session cookie.

3.3 HTTP/2.0

Problem in 1.0 was request was treated in order, 1.1 made it a little better using pipelining which allowed for multiple sequential request.
2.0 introduced streams, where request is numbered in odds and responses are given in even numbers.

3.4 Electronic mail

Electronic mail consists of 3 major components: User agents, Mail servers and Simple mail transfer protocol (SMTP)
User agents are essentially mail clients which allows for creation and reading of emails.
Mail servers have a mailbox for incoming messages and a message queue for outgoing mails
SMTP uses TCP on port 25 to transfer emails.
The protocol work on command response, where command is in ASCII 7 bit and response are status codes.
The mail message consists of header (To, from, subject), blank line and body (Only ASCII)
The user agents access the mail server via the IMAP protocol or HTTP

3.5 Video streaming and content delivery networks

To reduce the amount of data, coding is used on the data, spatial (groups pixels together) and temporal (only send difference of video frames)
CBR - constant bit rate
VBR - variable bit rate
DASH - Dynamic, Adaptive Streaming over HTTP
DASH divides video files into chunks, with each having different rates.
All chunks are managed in the manifest file which provides URL to each chunk.
The client then handles, when to get chunks, at what encoding rate and which server to request chunks.
CDN networks work by distributing the content to multiple servers around the globe.
To find the best server in a CDN period tests are made in the network of which speeds are in between each server.

3.6 P2P

No need for an always on server
End systems directly communicate.
Is more scalable than a server setup.
For n clients on a server the sever time will be $D_{Client-Server} > max(NF/u_s, F/d_{min})$, it will therefore scale linearly.

Whereas in P2P the time will be $D_{P2P} > max(F/u_s, F/d_{min}, NF/(U_s + \sum u_i))$
Where U_s is the central server speed, u_i is user upload speed, d_i user download speed and F is file size.
A torrent is a group of peers exchanging files divided into chunks of 256Kb. The peers are then managed in a tracker, which also participate in the torrent.
A torrent client then uses those peers, request which chunks they have, and missing chunks are downloaded from fastest connection.

4 Network security

The properties of secure communication

- Confidentiality - Only sender and receiver should be able to read the message using encryption
- Message integrity - The message should not be altered by intent or accident which is checked using check sum
- End-point authentication - Insuring sender and receiver are able to confirm their identity
- Operational security - A firewall or deep packet inspection may be setup between local network and public network to counter malicious attacks

4.1 Principles of cryptography

A text is first plain text and after using the encryption algorithm it becomes a ciphertext.
The most used method used today are based on keys between sender and receiver.
A symmetric key system both sender and receiver have to same key to decrypt an encrypt.
In a public key system sender and receiver has 2 keys a public K^+ and private K^-. The sender then encrypts using the public receiver key which then can be decrypted by the receiver using the private key.

4.2 RSA

RSA is an algorithm to implement public key system.
To generate a public and private key the following is done

- Two large primes are chosen to larger to higher encryption though longer de,- and encryption times, but the product should be in the order of 1024 bits
- Computer $n = pq$ and $z = (-1)(q-1)$
- A number e is chosen which is $e < n$ and $gcd(e, n) = 1$
- Find a number d such $ed - 1 \mod z = 1$
- $K^+ = (n, e)$ and $K^- = (n, d)$

The encryption c is then done by

$$c = m^e \mod n$$

where $m < n$
The decryption is then done by

$$m = c^d \mod n$$

This does take a lot of data and time therefore often sessions keys are generated and shared using RSA.
This works by when a message is encrypted and decrypted it is expressed as

$$\begin{aligned} m &= (m^e \mod n)^d \mod n && (1)\\ &= m^{ed} \mod n && (2)\\ &= m^{ed \mod z} \mod n && (3)\\ &= m^1 \mod n && (4)\\ &= m && (5) \end{aligned}$$

(2) can be done as property of modulo. (3) is possible because be definition p and q are prime and $n = pq$ and $z = (p-1)(q-1)$ makes it such $x^y \mod n = x^{y \mod x}$.
(4) can be done due to z was chosen by definition to be $ed \mod z = 1$. (5) is done since $m < n$.
Since e and d are multiplied we would get the same result if the message were encrypted or decrypted first.

The safety of RSA relies on there is no known fast factoring of number algorithm

4.3 Symmetric key encryptions

- Caesar cipher - Every letter is shifted a number in the alphabet with wrap around
- Monoalphabetic cipher - Every letter is mapped to another letter
- Polyalphabetic - Using multiple monoalphabetic ciphers in a certain order
- Block cipher (ECB) - The message is split up into blocks of size k which then is mapped to an encrypted block in same size
- Advanced block cipher (OFB) - Split message up to smaller block of size k_1 and then again to size k_2, the k_2 is then scrambled in place. This is repeated n times
- Cipher block chaining (CFB) - First a Initialization vector is generated and send, then every block is XOR'ed with the last send message and encrypted using shared key.

To attack a simple encryption there are different methods

- Ciphertext-only attack - Can be attacked using statistical analysis
- Known-plaintext attack - If some of the plaintext is known and therefore can be matched to the encryption
- Chosen-plaintext attack - If the attacker has access to the plaintext and can therefore get information about the encryption

4.3.1 Feistel cipher

The cipher works by

1. Plaintext is spitted into 2 chunks left and right
2. A function is then performed on the right using key 1
3. The left chunk is then XOR'ed with the output of the function
4. The XOR output is now the right chunk, and the original right chunk is the left chunk
5. 2 - 3 is repeated n times with the last chunk as input and the key is iterated

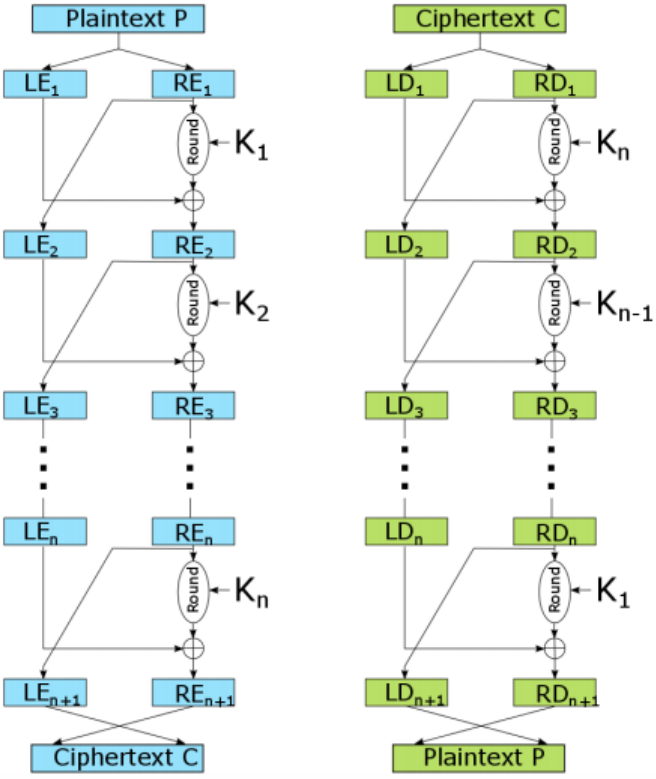


Figure 1: Feistel cipher illustration by terodee

6. Left and right are switched

To then decrypt the same algorithm is used with the reverse order of the keys.
This will work due to XOR being reversible

4.3.2 DES

A symmetric encryption system, such input and output are the same size
First the plaintext is divided into 64-bit chunks.
A 64-bit key is generated.
Then an initial permutation is done on the 64-bit text using a predetermined vector of value mappings.
A Feistel cipher is then done with 16 rounds, where the function is defined as:

1. The 32-bit input is expanded to 48 bit
2. The 48 bit is XOR'ed using the input key

3. The 48 bit is then substituted using the S-Box table to 32 bits. This is done by splitting the 48 bits into 6 bits chunks. Then bit 1 and 6 is the row number and 2,3,4 and 5 is the column number.

4. The 32 bits are then permutated using the P-table.

After the Feistel cipher, a final permutation is done by with the inverse of the first permutation.

To find the subkeys for each of the 16 rounds the following is done to the key.
First the key is reduced to 56 using permuted choice 1 table, where two chunks of 28 is created is also divided.
Each chunk is then circular left shifted once on iteration 1,2,9 and 16 and shifted twice on every other iteration.
Then the second permutation table is used to generate the 48-bit sub key.
By this method a bit will be used in 14 out of the 16 keys.

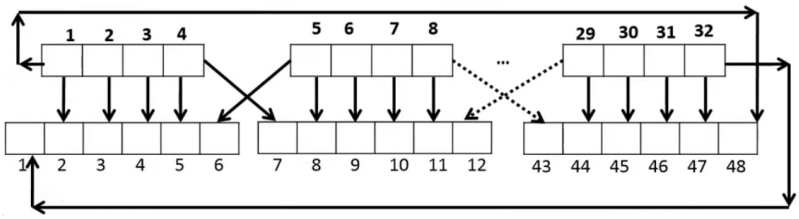


Figure 2: Des expansion visualization

Column Number

Row No.	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
0	14	4	13	1	2	15	11	8	3	10	6	12	5	9	0	7
1	0	15	7	4	14	2	13	1	10	6	12	11	9	5	3	8
2	4	1	14	8	13	6	2	11	15	12	9	7	3	10	5	0
3	15	12	8	2	4	9	1	7	5	11	3	14	10	0	6	13

P

16	7	20	21
29	12	28	17
1	15	23	26
5	18	31	10
2	8	24	14
32	27	3	9
19	13	30	6
22	11	4	25

Figure 3: Des S-Box lookup table to the left and P-box table to the right

4.3.3 Triple DES

To get a longer key 3 DES algorithms can be chained.
The encrypt it is done by the three keys $k1, k2, k3$ such ciphertext= $E_{k3}(D_{k2}(E_{k1}(\text{plaintext})))$
For the decryption the encrypt and decrypt is reversed.

PC-1

57	49	41	33	25	17	9
1	58	50	42	34	26	18
10	2	59	51	43	35	27
19	11	3	60	52	44	36
63	55	47	39	31	23	15
7	62	54	46	38	30	22
14	6	61	53	45	37	29
21	13	5	28	20	12	4

PC-2

14	17	11	24	1	5
3	28	15	6	21	10
23	19	12	4	26	8
16	7	27	20	13	2
41	52	31	37	47	55
30	40	51	45	33	48
44	49	39	56	34	53
46	42	50	36	29	32

Figure 4: Des permutation choice tables for creating sub keys

The chaining of encrypt decrypt encrypt, is such if a program only implements DES, then the triple DES will still work.
Triple is required over double due to double being vulnerable to a meet in the middle attack.
If an attacker has access to the input and output of encryption and decryption and knowns a pair lets say A→B
Then a DES bruteforce is done with A as input and all outputs are saved.
Then B is bruteforce decrypted and all output are saved.
Then a matching pair can be found. This therefore result in two bruteforce has to be done on DES therefore only making the combinations 2^{57} and not 2^{112}

4.3.4 AES

Plain text is split up into chunks of 128 bits.
There are 3 levels of encryption where

1. key: 128 bits, rounds: 10
2. key: 192 bits, rounds: 12
3. key: 256 bits, round 14

The algorithm works by representing the input as a 4x4 matrix of bytes where the matrix is filled in by columns such b2 is in row 1 column 0, in the following steps

1. Plaintext is XOR'ed with key 0
2. Sub-bytes - A S-box lookup table is used for substituting bytes
3. Shift rows - Each row is cyclic left shifted (in bytes) x times where x is equal to the row number

4. A mix column matrix is multiplied to each column of the matrix

5. The round key is XOR'ed to the matrix

Step 2-5 is repeated, until last round where step 4 is not performed.
To decrypt the keys order is reversed.

To find the round keys first a key is generated equal to 128/192/256 bits.
For the 128 bits key These are then divided into 4 words of 32 bits denoted w_i
Therefore making $k_0 = [w_0, w_1, w_2, w_3]$
We can here denote $k_{0-1} = w_1$
To find round i's key the following is done

- $k_{(i-1)-3}$ is left cyclic shifted
- Then it is substituted using the same S-box for the algorithm
- Then the round constant is XOR'ed
- This is then XOR'ed with $k_{(i-1)-0}$ and is equal to k_{i-0}
- To find k_{i-1} $k_{(i-1)-1}$ XOR'ed with $k_{(i)-0}$
- To find k_{i-2} $k_{(i-1)-2}$ XOR'ed with $k_{(i)-1}$
- To find k_{i-3} $k_{(i-1)-3}$ XOR'ed with $k_{(i)-2}$

The round constants are as follows

4.4 Cryptographic hash functions

To ensure the integrity of a message a checksum is used.
The is done by the message being sent into a hash function which generates a unique value for the message.
To ensure authenticity of a message, not only is the messaged hashed but a shared key is added to the message. Called message authentication code (MAC)
Then when the message is received the message is also hashed and checked to be equal to the sent hash value.
This shared key could then be shared using RSA.

To get a digital signature the private key can be used to encrypt a document which then people can verify by decrypting using the public key.

This can then be combined with the MAC, such a method can both have integrity and verified author. After generating the hash, the senders private key is used to encrypt the hash, which then can be decrypted and verified.
A certification authority (CA) is used to hold public keys and verify to whom a public key belongs to.
An authentication protocol can work by first the sender sends an initiating message, the receiver response with a nonce (never used number), the sender encrypts using private key and receiver decrypts using public key. If they match an authentication is established.

4.5 Diffie Hellman key exchange

Allows two parties that have no prior knowledge of each other to jointly establish a shared secret key over an insecure channel.
The exchange is based on:

1. A common ground such as a number
2. Client and server adds their own number to the common ground
3. Client and server switch keys
4. Client and server again adds their own number
5. Client and server now has the same number

5 Transport layer

The transport layer is the messenger between the network layer and application layer.
Therefore, the transport layer is only at presence in the end systems of the network.
The transport layer provides protocols for UDP (User datagram protocol) and TCP (Transmission control protocol)
Transport layer packet is called a segment.
The IP is an unreliable service which does not guarantee segment delivery, in order and integrity
This is therefore encountered for in the transport layer.

5.1 Multiplexing and demultiplexing

Demultiplexing is when a segment redirects a data to the correct socket.
Multiplexing is creating segments with header information for the demultiplexing.
The segment header therefor has two fields of 16 bits the source port and destination port.
The well-known port numbers are from 0 - 1023 and are restricted, for things like HTTP and FTP
In case of a server setup with TCP, the server then uses the four parameters in the TCP request (src ip, dest ip, src port, dest port) to setup a demultiplexing on a new port.
Such two client may connect to the same port and unknowingly be switched to a new port without having to change TCP parameters.

5.2 Reliable data transfer

To verify (positive acknowledgement) or request repeat (negative acknowledgment) data transmission is known as ARQ (Automatic Repeat Request) protocols
An ARQ protocol requires error detection, receiver feedback and retransmission
This type of protocol will only send new data when the receiver acknowledges the package was received, therefore the name stop-and-wait protocols.
The problem is if the acknowledgment is corrupted, to counteract this the package is sent with a sequence number and in case of unidentifiable acknowledgement the packet can be sent again.
Likewise, the acknowledgment also gets the sequence number to counter act duplicate acknowledgments.
The sequence is simply a single bit, to know that the packet is in the right order relative to the last packet.
In case of a lost package or acknowledgement the sender has a time of which after it will resend the package
Choosing the time is hard and must be estimated to be fast enough to not wait too long but still not send too many duplicate packages.
Pipelining is the act of sending multiple packages at once and increasing the sequence numbering.

5.2.1 Go-Back-N

GBN is a protocol which allows the sender to send n packages at once.
To protocol states that the base is the sequence number which is next to be acknowledged
The nextseqnum is the next package to be sequenced and sent.
The protocol can also be called sliding-window protocol because it can be seen as a windows switch slides over the packages to send.
GBN only use a single timer for all package referring to the last not acknowledged package. If the time runs out all non-acknowledged packages are resend.
If the receiver gets an out of order package the package is discarded and a negative acknowledgement is sent, where the sender will resend all non-acknowledged packages.
Selective repeat combats this by only sending negative acknowledged or time-out packages from which the receiver must keep a buffer of correct and always acknowledge good packages.

5.3 TCP

TCP is duplex thus allowing data in both directions at once.
TCP is a hybrid if selective repeat and Go-Back-N
TCP is point-to-point at therefore only two host can be part of a TCP connection
First a three-way handshake is made with no payloads.
Then a send buffer is initialized, and the application fills it up.
Then the TCP chooses when the create a segment with buffer data.
The maximum segment size (MSS) is determined maximum transmission unit (MTU) (1500 bytes in ethernet and PPP) minus header data from IP and TCP (typically 40 bytes)
The segment header includes the same as UDPs (src, destination port number, checksum)

- 32-bit sequence field
- 32-bit sequence acknowledgement field
- Receive window indicating number of bytes receiver is willing to accept
- 4-bit header length field representing number of 32-bit header lengths
- Optional field include timestamp, agreed MMS, windows scaling factor

- 6-bit flag field:
  - ACK bit for acknowledgement
  - RST, SYN, and FIN for connection setup and teardown
  - CWR and ECE for congestion notification
  - PSH pass data to upperlayer immediately
  - URG for urgent segments

5.3.1 Sequence numbering

The TCP sequence numbering is on every byte and the segments numbering is the first bytes number.
Acknowledge numbering works by the response is the next expected acknowledge number.
Say the host receives a packet segment 0, length 447 bytes, then the host responds with acknowledge number 448
TCP is therefore said to provide cumulative acknowledgments.
In case of out of order packages the individual implementation decide to either buffer it or discard it.

5.3.2 Timeout time

To find segment timeout the first the round-trip time (RTT) is calculated using the formula

$$\text{EstimatedRTT} = (1 - \alpha) \cdot \text{EstimatedRTT} + \alpha \cdot \text{SampleRTT}$$

Where $\alpha = 1/8$ and it can be noted that the new sample is more weighted.
The variation is also important and is calculated using

$$\text{DevRTT} = (1 - \beta) \cdot \text{DevRTT} + \beta \cdot |\text{SampleRTT} - \text{EstimatedRTT}|$$

Where the recommended value is $\beta = 0.25$
The timeout is then determined as

$$\text{TimeoutInterval} = \text{EstimatedRTT} + 4 \cdot \text{DevRTT}$$

For the first package it is set to 1 second and when a timeout occurs the value is doubled. When measuring only original packages are measured such a false time is not measured from a newly sent and old acknowledgement

Fast retransmit can occur if a triple duplicate knowledge is gotten. Three is chosen since a double could occur in the event of out of order, but three will most likely be a missing segment.
Then the timer is ignored and the second is sent as soon as possible.

5.3.3 Flow control

Flow control is used in case the application is reading data slower than TCP is receiving the therefore overflowing the receiver buffer.
The receiver buffer spare room called receive window will always be equal to

$$\text{rwnd} = \text{RcvBuffer} - \text{LastByteRcvd} - \text{LastByteRead}$$

Where LastByteRead is the byte numbering read by the application.
Therefore, the sender will not send more bytes than LastSendByte-LastAckedByte
When the rwnd is 0 the sender has to send bytes until acknowledgment and a non-zero rwnd is sent by the receiver

5.3.4 Connection management

A TCP connection is established by

1. A segment without application data is sent with SYN flag set to 1 and a random segment number
2. The server allocates space for variables and buffer and send a segment with SYN = 1, ACK = segment number +1 and a segment number the server itself chooses randomly
3. The client allocates space for variables and buffer and sends the first data with ACK = server segment number +1 and SYN = 0

When a TCP connection is shutdown, the following is done

1. Client send a segment with FIN flag = 1
2. Server acknowledges the FIN flag
3. Server send a segment with FIN flag = 1
4. Client acknowledges the FIN flag

When the last step is done 30 seconds is waited before closing fully down.
When TCP acknowledges a FIN segment the variables and buffer space deallocated.
The TCP from client has 6 stages

- SYN_SENT - Send SYN
- ESTABLISHED - Receive SYN & ACK, send ACK

- FIN_WAIT_1 - Send Fin
- FIN_WAIT_2 - Receiver ACK
- TIME_WAIT - Receive FIN, send ACK
- CLOSED - Wait 30 seconds

The TCP for server has 6 stages

- Listen - new open socket
- SYN_RCVD - Receive SYN, send SYN & ACK
- ESTABLISHED - Receive ACK
- CLOSE_WAIT - Receive FIN, send ACK
- LAST_ACK - Send FIN
- CLOSED - Receive ACK

In the case of the server receiving a TCP request on a non-TCP port a segment with RST flag is sent back.
For a server to prevent SYN flooding, a hash of the client is made from IP, source and port number called cookie and is sent as segment numbering and forgotten.
When receiving the SYNACK the cookie is recalculated and only if the ack is cookie +1 will the server allocate space and recognize it as a legitimate request.

5.3.5 Congestion Control

Classic TCP uses end-to-end congestion control rather than network assisted (Where router indicates to host if congestion is a problem).
To exercise congestion, control the sending rate is limited to

$$\text{LastByteSent} - \text{LastByteAcked} \leq min(\text{cwnd}, \text{rwnd})$$

Where cwnd is congestion window.
cwnd is determined by

- If a timeout occurs or a retransmission the cwnd is lowered
- A successful delivery will increase the cwnd

The algorithm therefore works by first starting in slow start with cwnd = 1 and is known as additive-increase, multiplicative-decrease (AIMD)

- Slow start - First the rate is 1 MSS, then for every acknowledged segment 1 MSS is added to cwnd essentially doubling it until a problem occurs. When that happens a variable ssthresh is initiated to cwnd/2 and cwnd is set to 1. Then again slow start is begun on it until it reaches sshtresh from which congestion avoidance is begun. In case of 3 duplicates fast recovery is ran with ssthresh = cwnd/2 and cwnd = sshtresh + 3MSS

- Congestion avoidance - Now cwnd is incremented only by MSS/cwnd until a problem occurs. If a duplicate occurs three times sshtresh is updated to cwnd/2, cwnd is set to ssthresh + 3 MSS and fast recovery is ran. If a timeout occurs sshtresh is set to cwnd/2 and cwnd is set to 1.

- Fast recovery - cwnd is incremented by a single MSS. In a timeout sshtresh = cwnd/2, cwnd=1 and slow start is begun.

The Cubic version is an alternative where the congestion avoidance saves the last max limit it reached and then cubically reaches it such the farther the distance to the maximum the more cwnd is increased.

The network assisted version works by the router sets the ECN (Explicit congestion notification echo) flag on a segment to the receiver, which also sets it in the acknowledgment. Which result in halving the congestion window
Delay based congestion control works by using the RTT, if the sender then measures a significantly less than the uncongested throughput rate, then it slows down the sending rate. Essentially it works by avoiding queues.

It can be seen TCP will fairly share bandwidth and congestion control will be equal in multiple TCP connections over time.
This is unlike UDP which will not behave fair and force TCP to get less bandwidth without itself taking less.

5.4 QUIC

QUIC is an extension to the UDP protocol, which provides features from TCP.
A handshake is initiated, but it only needs a single send and receive to setup therefore saving a whole RTT over TCP.
Even QUIC-0 can be used in case of recent connection where no handshake

is needed as saved params are used.
QUIC provides encryption of packages and uses http3.
QUIC uses reliable, and congestion-controlled data transfer features from TCP

5.5 TLS

Transport Layer Security (TLS) is an extension to TCP which gives it security in the form of: encryption, integrity and authentication.
Websites using TLS is recognized by the use of https.
TLS acts as a sublayer in the TLS socket and applies it security.
When running TLS, the data is split up into records which is encrypted and hashed using HMAC.
The TLS record consist of: Type, Version, Length, Data, and HMAC, where data and HMAC is encrypted with E_B
The type of field is used for a copy of the TCP type, such as FIN flags but by using it in the TLS, it is included in the hashing such no tampering of the TCP flags will destroy the connection.
Likewise, is the sequence number and length of message also part of the hash
.
A TLS connection is as follows:

1. A list of supported cryptographic algos and a nonce is sent from client
2. Server chooses a cryptographic alg. for symmetric key, public key and hmac. Then responds with choices, certificate and server nonce
3. Client confirms certificate and gets server public key, generates Pre-Master Secret (PMS), and sends encrypted PMS with server public key
4. Using the chosen algos the server and client generates Master Secret (MS) from PMS and nonce. The MS is sliced up into: 2 encryption keys, 2 HMAC keys, initialization vectors in case of symmetric cipher
5. Client send HMAC of all handshake messages
6. Server send HMAC of all handshake messages

The HMAC handshake messages are used to ensure integrity of the non-encrypted handshakes before. Likewise, is the nonce used to prevent replay attacks.
The keys generated from the MS is denoted

- E_B session encrypted key for data from client to server
- M_B session HMAC key for data from client to server
- E_A session encryption key for data from server to client
- M_A session HMAC key for data from server to client

6 OWASP - Top ten security risk

Open Web Application Security Project conducted a list of biggest security risk based on data on attacks. The following is the top in order.

6.1 Broken Access Control

The biggest problem is allowing attacker access to data or control which was not intended.
This may be through websites, API or any other access to data or control.
Problem often occurs due to amount of access points and lack of testing.
Attackers often use modification of accessible data, such as cookies or url which may not be sanitized.

6.2 Cryptographic Failures

Many systems and companies may use old cryptography or none at all on sensitive data.
This can also be in form of transportation of data both internal and external.
Lack of this will result in case of leaks actually exposed data rather than encrypted data.

6.3 Injection

Injections occur when user data is sent to an interpreter without sanitizing the input.
They can be queries for databases or bad designs for inputs which can result in bad input.
This can be in buffer overflows which may allow injection of code and gaining access to higher privilege.

6.4 Insecure Design

Insecure design is not the implementation but rather the security choices.
This can be password recoveries which is too weak or intern protocols which may be too weak when handling sensitive data

6.5 Security misconfiguration

This may be outdated libraries or default setting which is unchanged and therefore leaves a vulnerability.
This may also be in case of settings for development end up in production leaving an open backdoor or likewise.

6.6 Vulnerable and outdated components

Likewise, security misconfiguration but this focus more on the outdated packages or libraries which may be in use.
The libraries may contain vulnerabilities which can gain access to the code or data.
This is prevented by subscribing to bulletins of included libraries or packages and keeping them up to date.

6.7 Identification and authentication failures

Bad implementation of authentication which gives attackers opportunities to gain access via session tokens, keys or passwords.
This can be in the form of, allowing brute force attacks, allowing weak passwords or having a weak password recovery
A good solution is often implementing two step authentication.

6.8 Software and data integrity failures

This happens when an attacker identifies as software or plugin and forces an update upon working code.
This may also be if an attacker simply notices a data stream which does not check for integrity and themselves may send attacking data.

6.9 Security logging and monitoring failures

Most attacks are not detected before 200 days after.
This is due to lack of security logging and monitoring attacking attempts.

A better monitor or log may capture attempts and warn of risky areas before an attack happen

6.10 Server-side request forgery

When a web application is fetching remote resource without validation.
This may lead to an attacker being able to load attack resources onto the application.
This is prevented having a firewall which block all traffic beside the known remote places.

7 Penetration testing

The two most common forms of penetration tests are: Application penetrations testing and infrastructure penetration testing.
Other types are

- Mobile application
- Device penetration testing (laptops, consumer devices)
- Wireless penetration testing
- Telephony or VoIP

There are three different types of intruders

- Hackers - Experimentally minded programmers targets security loopholes in its system
- Crackers - Exploits weak points of its systems to gain illegal advantages
- Insiders - Type of crackers with inside knowledge
- Script kiddies - Uses attack tools

There are mainly 3 different types of attacks

- Network based attacks - attacks using network protocol functionalities this includes port scanning, IP spoofing, sniffing, session hijacking, DoS attacks, buffer overflow and format string attacks
- Social engineering - Manipulating people to reveal security related information

- Circumventing of physical security measures - By using physical attacks and gaining access to physical data or authority

A company should have security policies and security concepts, without that a pen test would not even be helpful.
The procedure of pen testing is in the following steps

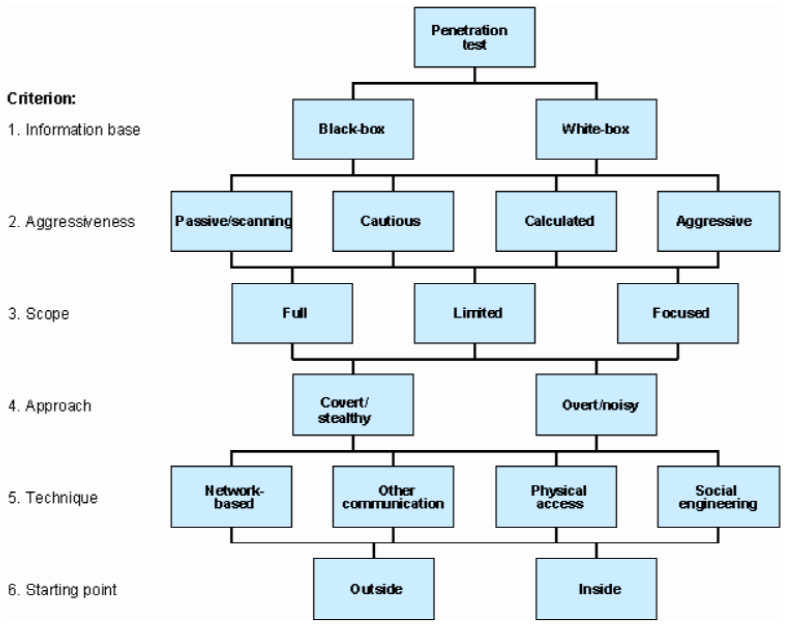
1. Research information about the target system - information like official IP address
2. Scan target systems for service on offer - Using a port scan internet services can be found
3. Identify systems and applications - By using fingerprints in the prt scan
4. Research vulnerabilities - Finding exploits in systems with open ports from fingerprints
5. Exploit vulnerabilities - Using the known vulnerabilities and gaining access for further attacks

7.1 Classifying pen tests

Starting point - Attack of penetration most common: firewalls, web servers, RAS access points and wireless networks.
External servers are known as normal workstations.
Web servers are often vulnerable due to their manifold of functions
Pen testing is useful since security audits and IT audits most often focus on IT infrastructure in terms of compliance, efficiency and effectiveness.
The goals of pen testing should be clear and can be divided into four categories

- Improving security of technical systems - This can be finding improvements in firewalls, routers, web servers, etc.
- Identifying vulnerabilities - This is the actual objective of the test
- Having IT security confirmed by an external third party - The IT system and infrastructures current security
- Improving security of organizational and personnel infrastructure - This is often in form of social engineering

The limits of a pen test are the fast discovery of new vulnerabilities making pen test only validation of security currently but not in the future.
Likewise does the pen test reduce the probability of a successful attack.
The type of attack is also classified to test a given scenario



- Information base - The amount of known information, where black box represents the unknown insider information unlike white box
- Aggressiveness
  - Passively - Vulnerabilities are detected but not exploited
  - Cautious - Vulnerabilities are only tested when it will not result in system suffering
  - Calculated - Exploits vulnerabilities which may result in system disruptions
  - Aggressive - Exploit every vulnerability even if deactivating systems or overloading
- Scope - The amount of testing which directly relates to time spent, focused sub network after modification, limited number of systems, full everything available

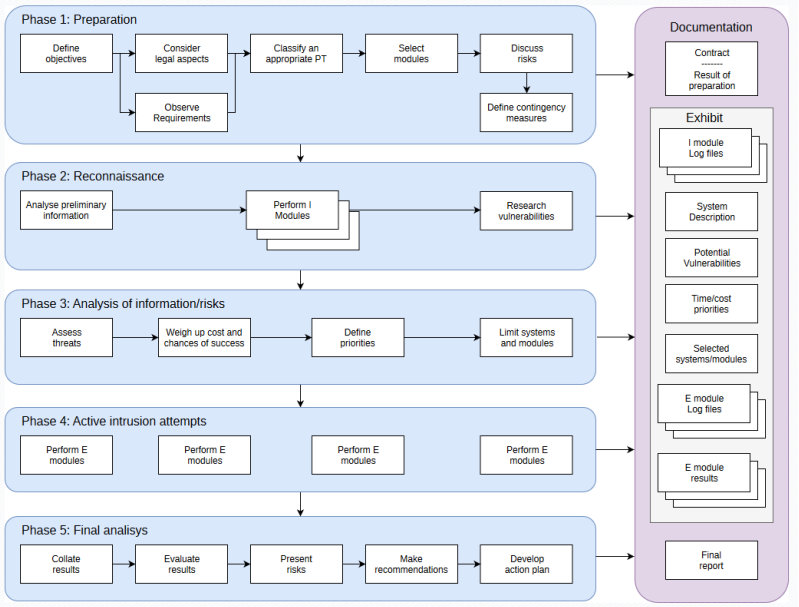
- Approach - How visible the attack is, covert stealthy, overt often with staff such fixes can fast be done
- Technique - Networks based using network protocols also known as IP-based pen test, Other communication networks, mobile, fax, wireless networks etc., physical in case of good enough firewall, social engineering people are frequently the weakest link
- Starting point - Is the test performed from an outside perspective, or inside networks where firewalls do not have to be overcome

Most often a combination of pen tests is advisable.

7.2 Legality and ethical issues

There are different legal issues which rises from pen testing.
For a company pen testing may be needed, to ensure data security to compart with laws.
When a pen test is performed approval by client in all areas is required.
Likewise, often 3rd party also need to approve the test
The test should have a liability to cover claims from 3rd parties.
Ethical issues also arise with social engineering in case of wanted anonymity.

7.3 Phases of pen testing



Preparation - agree on scope and cost based on classification, contracts in place and discuss risks
Reconnaissance - passive test, where information is obtained and overview of system.
Tool for data collection

- whois
- https://website.informer.com/
- namp
- https://spokeo.com/
- osint examples
  - https://www.shodan.io/
  - https://www.spiderfoot.net/
  - https://github.com/laramies/theHarvester

Analyzing information and risks - use data to find risk and determine which to test based on goals and time
Active intrusion attempt - This phase tests the vulnerabilities found, it is here important to consider actual risks and which risk are not affecting the product/organization
Final analysis - Evaluate the vulnerabilities and where they were found and recommendation to eliminate them

7.4 Tools

- Kali Linux - OS with preinstalled tools
- Nmap - port scanning
- NCAT - Read/ write network connection using TCP or UDP
- Metasploit - Framework with most common exploits
- SQLmap - Detecting and exploiting SQL injections flaws
- NIKTO - Scan for harmful files, misconfigurations, outdated software installations on web server
- BURPSUITE - Pen testing web application tool

- John the riper - Tool for cracking password
- NESSUS - vulnerability scanner (not free)
- Wireshark - Packet sniffer and network analyzer tool
- AIRCRACK-NG - wireless network tool
- Tool list - https://sectools.org/

8 The network layer: Data plane

The role of the network layer is the route packets from host to receiver.
Routing is done by all routers by each forwarding the packet from input to correct output link.
This is done using routing algorithms.
The result of routing algorithms is a forwarding table which determines which output link the router should forward to.
There are two approaches to creating the forwarding table either the routers share their tables between each other using a routing protocol.
Or the SDN approach where an external remote controller creates tables for routers and send them it. This can be done by third parties or ISP.
SDN stands for software-defined networking.
The networks service model is defined to be best-effort service therefore many features are not implemented other than trying to deliver.

8.1 What is inside a router

Input ports is used for terminating incoming physical link.
The input performs the lookup function to the forward table via the switching fabric.
In case of a control packet the packet is forwarded to the routing processor.
The switching fabric connects the input and output ports.
The output ports store the sending packets and perform the necessary link-layer and physical layer functions.
The routing processor maintains the router, by updating the forwarding tables or in case of SDN connects to remote and update forward tables.
The forwarding tables matches a prefix of the packets destination address.
If a match does not exist, it exists via the default port.
If multiple matches the longest is chosen.
If the ports are in use the fabric blocks the package and queues it.

All this has to happen in hardware to keep up with transfer speeds.
The input port also has to, physical and link layer processing, check the checksum and time to live, update counters and time to live.

8.1.1 Switching

The act of switching can be done through different methods.
The simplest version switching via memory, copies inputs into processor memory and let the processor due to moving to output.
This is limited to memory bandwidth and therefore the transfer will be limited to half the memory bandwidth.
Switching via a bus transfers packets via a bus, then every packet gets appended a label and the matching output port keeps the packet the rest throws it away.
The label is then removed before sending.
This is then limited to the bus speed since only one package can be worked with at a time.
Switching via an interconnection network, uses a crossbar switch, where a bus from every input goes over output ports bus, such an intersect is present for each.
Then the intersect between input and output is turned on to forward the package.
This results in the package only being blocked in case of the wanted output is taken.
More sophisticated approached uses a stacked interconnect such the output can contain multiple packages

8.1.2 Queue and buffers

Queue can occur both on output and input, input will happen when either too many packages come into one input at once or head of the line (HOL) blocking, where another input waits for output and therefore block the input buffers.
Output will occur if the transmission rate is slower than the forwarding and input.
When queue occur either the latest (drop tail) is dropped or a random picked using an active queue management algorithm
The drop tail has the advantage of faster congestion signaling to sender.
The size of the buffer has to be large enough to prevent package lost too fast in case of a burst, but smaller than queuing delays are too high.

The rule of thumb is a buffer size equal to

$$B = RTT \cdot C/\sqrt{N}$$

Where C is the link capacity and N is the number of independent TCP flows. TCP is part of the equation since in case of a buffer being filled up, since TCP will receive an ACK for every sent from the buffer the buffer will never empty known as buffer bloat.
For emptying the queues different approaches can be taken

- First in first out FIFO - The packages are handled in the same order as arrival
- Priority queuing - The packages are classified upon arrival which then is handled in priority order
- Round Robing and Weighted fair queuing - Like priority queuing, but to ensure every package is sent at some point the queues is served in a circular manner. The class weight then determines the time used pr. queue such higher priority equals more time spent in the queue before moving on.

8.2 The interprotocol (IP)

8.2.1 IPv4

The datagram of an IPv4 is

- Version number - 4 bits to determine version number like IPv6
- Header length - IPv4 datagram contains variable headers therefore 4 bits is used for indicating size
- Type of service - Used to distinguish if the package is a real-time datagram or a non-real-time traffic like FTP, in here two bits are also used for congestion control
- Identifier, flags, fragmentation offset - Used for IP fragmentation
- Time-to-live - Counter for number of routers which the package can be sent through before being dropped
- Protocol - A number representing which protocol is used after the IP the handle the data like TCP or UDP

- Header checksum - Checksum of the integer value of every second header values sum. Recalculated for each router step since headers changes like time to live
- Source and destination IP
- Options - Allow to extend the IP headers further
- Data

The header is 20 bytes without options and with IP the size is 40 bytes
An interface is the boundary between link and host and has each their own IP.
With a 32-bit long IPv4 the total number of IPs is 2^{32} or approx. 4 billion
The IP is formatted into 4 chunks bytes which is written in decimal with dots between the decimals.
Subnets can be defined such a prefix is the identifier and then the whole subnet uses the prefix, and the rest is for intern routing.
This is in the format a.b.c.d/x where x is the number of bits used for the prefix and the rest is the subnet.
This is known as Classless interdomain routing (CIDR). This allows for subnets to have way smaller forwarding tables.
Before CIDR the subnets size was bounded to 8, 16 or 24 bits making the jump from 8 to 16 very large.
255.255.255.255 is used for sending a package to every device in the subnet.
To get a subnet the ISP is contacted which will give part of their subnet to the company.
The ISP subnet is managed by regionals which is managed by Internet corporation for assigned names and numbers (ICANN).
ICANN is a NPO which manages DNS root servers, assigning fmain names and resovlg=ving dfomain name disputes.

8.2.2 Obtaining an address

Dynamic Host Configuration Protocol DHCP is a protocol used to configure devices by admins.
DHCP allows to assign a static IP to devices or temporary IP.
DHCP requires a server or device which relays the DHCP protocol.
The protocol consists of

- DHCP server discovery, which is an udp packet send to port 67 to 255.255.255

- DHCP server offer(s) - Once a discovery is gotten an offer is sent back to the subnetwork, and contains transaction ID, proposed IP, network mask and IP address lease time (The time the ip is valid)
- DHCP request - Client chooses from offers and respond to the wanted by echoing the content
- DHCP ACK - an acknowledgment message is sent confirming the device

The problem with DHCP is the requirement for the protocol for every subnet which is a problem in case of more routers in large areas.

8.2.3 Network address translation (NAT)

NAT is a method for using a routers as a single device.
This allows for a single IP which is shared among the local networks.
This works by when a device sent a package to the router, the request is contains the wanted IP and a socket to the device from route.
The router then rewrites the source and source ip to the wan ip.
This allows up to 60,000 devices on one network with 1 ip address.
This is argued against since is reuses port numbers for addressing purposes.
This also requires for server like application a NAT traversal tool for finding open ports in the local network.
Another problem is it ruins the architecture of the internet, by not assigning devices an IP but rather a number of devices.

8.2.4 IPv6

The problem with IPv4 is the limited number of IPs.
The last IPv4 address space was given out in 2011 to a reginal registry.
There is open IPv4 address in regionals, but the global pool is empty.
IPv6 uses 128 bits to create IPs making sure it will never be problem again.
The format goes as

- Expanded addressing capabilities - This allows to send a package to an IP group
- A streamlined 40-byte header
- Version - numerical value 4 for IPv4 and 6 for IPv6
- Traffic class - 8-bit field for priority certain diagrams

- Flow label - Allowing to label packages as flow and given them optional priority
- Next header - Like to protocol in IPv4
- Hop limit - Like time-to-live in IPv4
- Source and destination addresses
- Data

8.2.5 OpenFlow

OpenFlow is the mechanism which performs the match and action for packages.
The match is done over the whole datagram, including link,- network,- and transport layer.

Ingress port				
src MAC	dst MAC	eth Type	VLAN ID	VLAND Pri
IP src	IP dst			
IP proto	IP TOS	TCP/UDP src port	TCU/UDP dst port	

From this a flow table is generated which states rules, and matching will be used in priority order.
Each row then has a list of action which in done in the given order.
They include forwarding, dropping and modifying fields.
This can be used to create firewalls by blocking IP or methods by dropping the matching columns

8.2.6 Fragmentation

The fragmentation of a packet is the act of splitting up the package such they can fit into the maximum transfer unit (MTU)
The header will have offset set such the packets order can be seen and is divided by 8 bytes to save space.
The offset will come from the previous packages data length (not including header 20 bytes if no options)
The frag flag is not raised on the final package.
The frag flag is raised and the ID will match across the fragmentation.

9 Control plane

9.1 Routing algorithms

Routing algorithms is used to find the least cost routes between sender and receiver.
The cost may reflect, physical length, speed or monetary cost of links.
Two nodes cost is said to be $c(x, y)$ and in case of not being connected the cost is infinite.
The graphs are considered undirected.
Centralize routing algorithms use global knowledge of the network.
To communicate a link-state broadcast algorithm is used. From this the Dijkstras algorithm can be used to find least cost to each node.

The algorithm will run at a centralized unit and the algorithms is referred to as link-state (LS) algorithm
Decentralized routing uses iterative, distributed manner by the routers.
The routers only have knowledge of neighbor routes and the cost to them.
The routers then exchange information about their neighbors to calculate the cost further.
This is called distance-vector (DV) algorithm.
This is done async by to nodes does need to work at same time, and iterative by how each nodes send and updates on changes.
The network cost is then calculated using bellman-fords algorithm.

Algorithms can also be classified by being static which change slowly over time.
And dynamic which changes for every traffic loads or topology change.
Load sensitive algorithms takes the package load into account to congestion.

9.1.1 Problems

Oscillations is where a network of nodes oscillates between two optimal path and therefore never end termination.
To prevent this the routers, send their link advertisement at random intervals to also prevent self-synchronizing.
Count to infinity problem occur if the nodes A,B and C is connected with cost 1.
If B and C is disconnected and does not send an update before A sends an update to B and advertise its length to C at 2.
This will then result in B updating to 2, A updating to 3 since it used B to

get to C. This will then keep happening until they count to infinity.
A solution which works with small node environments is the poisoned reverse, which advertises when routes is broken or increased.

9.1.2 BGP

BGP is a protocol for connection between AS
In the AS there are two types of routers, gateway routers on the edge connecting between AS and internal.
The BGP protocol uses TCP since it needs to transfer large amount of data, here the forwarding tables in other AS.
The BGP protocol comes in two types iBGP and eBGP
iBGP is for internal use, such when a new forwarding table has come into the AS it is shared between internal routers using iBGP
eBGP is external BGP which communicates between different ASs.
A new external forwarding table may include everything to AS1 goes through router A.
The internal then simply uses the optimal path to get a router which is a gateway.
BGP routing is done through different methods

- Hot potato - The AS focuses only on reducing cost internally and ignore outside policies.
- Route-selection - This is the used algorithm used in case of multiple routes and the route is selected in the given order
  - Check local preferences on attributes which is determined by policies
  - The shortest AS path is chosen
  - Hot potato is used
  - BGP identifiers is used to select

BGP is also used to find the nearest DNS server, using IP-anycast.

9.1.3 SDN control plane

Software defined network (SDN) network with separated controller.
A SDN network can be identified by the following characteristics

- Flow based forwarding - Packet forwarding can be don based on any number of header fields

- Separation of data plane and control plane - Instead of the routers doing both data and control plane, is the control plane done by a separate entity
- Network control functions - Communicate between a SDN controller and network control applications
- A programmable network - Using API in the SDN controller the network switches can be programmed

A SDN controller consist of

- A communication layer - Communication between SDN controller and controllers using open flow or other protocols
- A network-wide state-management layer - Control decisions, such as flow tables, load balancing or firewalling capability
- The interface to the network-control application layer - Interfaces for read/write network state and flow table

In practice the SDN controller consist of distributed set of servers for fault tolerance, high availability and performance.

9.1.4 OpenFlow

OpenFlow uses the TCP protocol over port 6653
OpenFlow allows for messages from SDN controller to controlled switch

- Configuration - Set a switch configuration params
- Modify-state - Add/delete entries in the flow table
- Read-state - Collect statistics from flow table and ports
- Send-packet - Send a packet using the controller

From the controller switch the SDN controller can get the packages

- Flow-removed - Flow entry on table has been removed
- Port-status - Change in ports
- Packet-in - Packets which does not match table or should be sent to controller

10 IPsec and VPN

IP security protocol (IPsec) is a security protocol applied in network layer on the IP datagrams.
This is what enables virtual private networks (VPNs)
IPsec puts a security layer on top of the payload such as TCP, UDP pr ICMP.
The IPsec can provide, source authentication, data integrity, and replay-attack prevention.
A VPN can be desired in an institution placed in multiple geographical locations.
This will then make is possible to create a secure network connection between locations over the public network.
IPsec can use two protocols: Authentication Header (AH) and Encapsulation Security Payload (ESP)
They both provide source authentication and data integrity, but ESP provide confidentiality and therefore is used more.
Security Associations (SA) is a network layer logic connection which is unidirectional.
If both entities want to send secure datagrams both has to setup a SA connection.
The SA has attributes which both sender and receiver keep track of

- 32-bit identifier for the SA called security parameter index (SPI)
- Origin interface and destination interface (IP addresses)
- Type of encryption
- Encryption key
- Type of integrity
- Authentication key

This information is stored in a Security Association Database (SAD) for each of the systems SAs
The IPsec packet comes in two forms tunnel mode and transport mode, from which the tunnel mode is most used.
The main difference is in tunnel mode the whole ip packet is used as payload whereas transport retains the original IP header.
To create an IPsec datagram in tunnel mode from an ipv4 datagram the following is done

1. Appends ipv4 header in the back in an ESP trailer field
2. Encrypt the result as dictated by SA
3. Adds ESP header information in front of encrypted ipv4 datagram which together is known as enchilada
4. Create an authentication MAC over enchilada as dictated by SA
5. Add Mac in the back of enchilada
6. Create new IP header as a normal ipv4 header in front of the whole package

The ESP header contains an SPI and sequence number.
The ESP trailer contains padding, padding length and NEXT header which identifies the payload type such as UDP.
The padding is used to make the message an integer number of blocks.
To create a connection the Internet Key Exchange (IKE) protocol is used.
It consists of two phases
Phase 1 use Diffie Hellman to create a bidirectional IKE SA.
The IKE SA differ from IPsec SA and provides authentication and encryption.
The exchange keys for encryption, authentication and a master secret to computer IPsec SA is exchanged.
Phase 2 consist of revealing each others identity by signing their messages, in an encrypted connection created in phase 1.
The IPsec encryption and authentication algorithms are negotiated.

11 Firewall

A firewall is a unit of hardware and software which isolates an internal network from the internet.
It does this in three goals

- All traffic from outside to inside and vice versa passes through the firewall
- Only authorized traffic as defined by the local security policy will be allowed to pass
- The firewall itself is immune to penetration

A firewall is split up in three categories

- Packet filtering which filter packets determined on factors such as following which allows for ex. port 80 only be outgoing but not ingoing
  - IP source or destination address
  - Protocol type in IP datagram field: TCP UDP OSPF and so on
  - TCP or UDP source and destination port
  - TCP-flag bits: SYN, ACK...
  - ICMP message type
  - Different rules for datagrams leaving and entering the network
  - Different rule for router interfaces
- Stateful packet filter which tracks TCP connections and use that knowledge for filtering
- Application gateway which is used per application with a set of rules, which uses the data sent to the application for filtering ex. mail server

11.1 Intrusion detection system

Uses deep packet inspection to find potential harmful packets.
There are two types of Intrusion detection system (IDS) which alerts malicious traffic and Intrusion prevent system (IPS) which filter traffic
A system may split traffic in between multiple IDS in case of a lot of traffic, which sends suspicious traffic further to a processing unit.
IDS are most often places as close to the vulnerable points to filter as much traffic before.
IDS use a signature-based system or anomaly-based system which have a database of known attack types which it compares packets with.
Another style is where the IDS learns usual traffic and traffic which stands out is deemed suspicious. By this new attacks can be prevented but it is hard to determine what is usual traffic.
Snort is a public domain open-source IDS with a large community-based database.

12 Link-layer

A device running a link-layer protocol is knows as a node.
Nodes a connected through links.
The purpose is to move a datagram to an adjacent node.

This is done by framing the datagram with a new datagram with new headers.
This is then send using link access which includes medium access control (MAC) which specifies rules for sending from a sender to receiver(s).
A reliable delivery can be part of it with error detection, in case of unreliable link like Wi-Fi.
The error detection is also implemented using hardware.
The link layer has its own hardware like an ethernet chip or in general network interface controller (NIC)
The hardware can also interrupt the CPU and use software.

12.1 Error detection

The biggest goal of error detection is detect an error and be able to correct it and is known as forward error correction (FEC).
This is a big goal for reducing sending time and number of retransmissions.

The simplest error detection is parity bit.
Parity bit works by a single bit representing if the amount of 1 is even or odd.
Problem is errors comes in burst so multiple errors fails the system.
Can also be implemented such the send data is grouped in rows from which the columns also have a parity bit.
By using this the error can be found and corrected.

12.1.1 Checksum

For d bits of data, and the checksum k it can be done in multiple ways.
The simplest is the sum of the d bits.
The internet checksum uses this method where bytes of data is treated as 16 bit and the checksum is in the form of 1s compliment.
Cyclic redundancy check (CRS) is a more reliable and is used since in hardware it can be done fast.
It is also known as polynomial codes.
The data d bits are send along r bits. The sender and receiver agree on a g value which the sum of d and r will be dividable by g.
g's most significant bit (leftmost) must be 1.
This means

$$d \cdot 2^r \text{XOR} r = n \cdot g$$

By XOR R on both sides

$$D \cdot 2^r = n \cdot g \text{XOR} r$$

Thus, it can be simplified by

$$R = \text{remainder} \frac{d \cdot 2^r}{g}$$

12.2 Link-layer switches

Switches has the ability to look transparent to hosts and routers.
It serves two purposes filter and forward
Forwarding can be done by IP but the most common is using MAC addresses.
The switches forwards using tables which consist of: MAC address, interface which leads to MAC address and time of entry
There are three cases of forwarding:

- No entry for destination MAC address and it is forwarded to all interfaces
- There is an entry, but the so source is the same as entry, so frame is discarded
- There is an entry and it not from the same source, the frame is forwarded to interface

The table is generated from incoming packages where it saves the interface and the source of frame
This makes the switch plug and play
The switch has advantages over a hub-based star topologies

- Elimination of collisions - by using buffers frames are forwarded one at a time
- Heterogeneous links - interfaces can have different speeds and therefore making it possible to mix equipment
- Management - the switch can detect problems and disconnect malfunctioning devices and gather statistic for debugging

- Security - since frames are forwarded to interfaces package sniffing is more difficult but switch poisoning where the table is flooded with false frames to deny or redirect frames

This is also done fast compared to a router, but require more ARP traffic
The router is also able to have a firewall and therefore is better for large networks since it also can have optimal routing where switches have a three structure.

12.3 Virtual Local Area Networks (VLANs)

The main idea of VLANs is dividing up a switch into virtual LANs to divide and isolate traffic.
This is also to combat having large switches to small groups.
By this frames are only delivered into ither devices in the VLAN group.
The switch then contain a router which can route between VLANs
Some switches also include VLAN trunking for linking switches with a single cable.
The ethernet frame then have a four-byte VLAN tag which identify which VLAN it comes from, in detail it is 2-byte TAG protocol identifier (TPID), 2-byte tag control information field that contains 12-bit VLAN identifier field and 3-bit priority field.

12.4 Multiprotocol Label Switching (MPLS)

MPLS is used for increasing IP routers by using a fixed length label.
This is not replace IP but to work beside in compatible routers call label-switched router.
Here a MPLS header is added between layer-2 (ethernet) and layer-3 (IP).
The header consists of: label, 3 bit for experimental, 1 S bit, and a time-to-live field.
The label can then be used for forwarding without unpacking the IP datagram.
Compatible routers are identified in advertising routes from routers.
The main advantage is unlike IP a single route is not selected rather all routes are considered and rerouting is possible.
This is also smart for VPN since the ip is not considered but rather just a label.

12.5 Data center networking

Data centers have three purposes: provide content, server massively parallel computing infrastructure and cloud computing.
A data center consists of host called blades which consist of CPI, memory and disk storage.
These are stacked in a rack with a top of rack switch (TOR).
TORs are then interconnected to tier 2 switches, connected to tier 1 switches, which is connected to border router.
To the tier 1 switches are load balancers which distributes traffic to the blades such the actual host ip is not needed either.

13 Wireless networks

Wireless hosts are end system devices.
Wireless links connect hosts through a wireless communication link.
Base station is responsible for sending and receiving data.
Hosts to a base station is referred to as operating in infrastructure mode.
Ad hoc networks have no base station and hosts must themselves do routing, address assignment, dns-like name translation and more.
Movement from station to station is known as handoff or handover.
Wireless networks are classified by: whether a packet crosses hops and if a base station is present in the network.
The creates the four possibilities:

- Single-hop infrastructure based - Base station connected to larger wired network, all communication is single hop to base station
- Single-hop, infrastructure less - no base station but a single node coordinate transmission of other nodes (Bluetooth)
- Multi-hop infrastructure based - Base station wired to larger network, may have multiple hops through nodes which relay communication (wireless mesh networks)
- Multi-hop infrastructure less - no base station and communication are between multiple nodes known as mobile ad hoc networks (MANETs)

13.1 Wireless Links and Network characteristics

Wireless networks differ from wired networks by

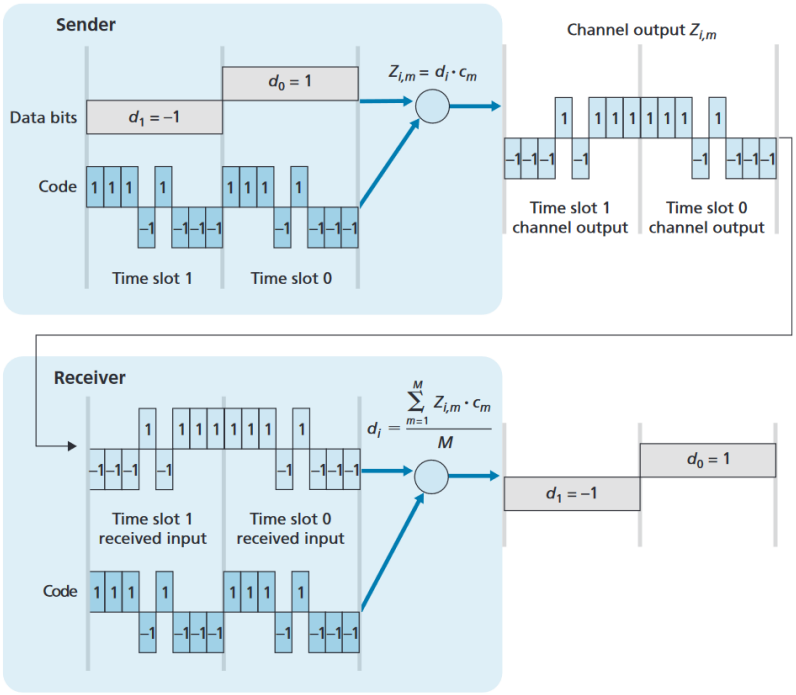
- Decreasing signal strength

- Interference from other sources
- Multipath propagation - occurs when portions of the electromagnetic wave reflect off objects and the ground taking paths of different length between a sender and receiver

Bit errors are more common.
Signal to noise ratio (SNR) is a measurement of the relative strength of the signal to noise, measured in decibel.
Bit error rate (BER) the probability of an error occurring.
Increase the SNR by increasing its transmission power will decrease the probability of BER, but requires more energy and is more likely to interfere
Therefore, a wireless network can select the modulation technique the adapt channel conditions.

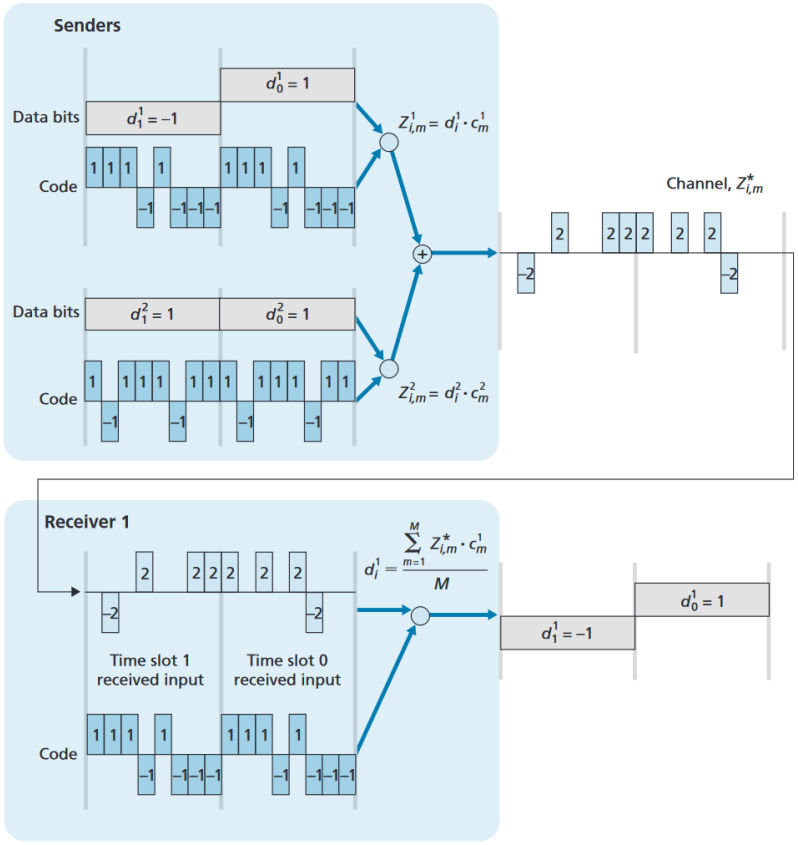
13.1.1 CDMA

Code division multiple access (CDMA) is method for working with the shared medium.
When a bit is sent it is multiplied with a code, where 0 is represented as -1.



To get the value again it is done by $d_i = \frac{1}{M} \sum_{m=1}^{M} Z_{i,m} \cdot c_m$

By this the average can be taking and therefore the longer code decreases the chance of a bit error.
Therefore, even in case of two signal and the signal is additive, then each receiver will still get the correct value by using the correct code.



13.2 Wi-Fi: 802.11 Wireless LANs

Comes in different standard which changes the frame but is backwards compatible.
Wi-Fi comes in two frequency ranges 2.4-2.485 GHz and 5.1-5.8 GHz, where 2.4 is unlicensed and therefore used more, and 5 GHz does not reach as far on the same power level and suffer more from multipath propagation.
Uses CSMA/CA protocol

13.2.1 CSMA/CA protocol

Carriere Sense multiple Access is a method for making hosts not transmit in the way of each other.

Works by the rules: Listen before speaking and if someone else begins talking at the same time stop talking.
CSMA with collision detection (CSMA/CD) is an extension to the CSMA.
CSMA works by in case of a collision and a transmitter does not get its acknowledgment it will wait a random interval and then try to retransmit.
CSMA/CD expands upon this such if it reads a signal while transmitting it will stop the transmission and go to waiting, instead of still sending the whole package.
The time interval is chosen in a binary exponential backoff such it starts low and get exponentially larger for every collision.
The efficiency will then be equal $\frac{1}{1+5d_{prop}/d_{trans}}$

13.2.2 Wi-Fi wireless LAN Architecture

Is a basic serve set (BSS) and contains a one or more wireless stations and central base station known as the access point (AP).
The typical network therefore has a router and AP most often integrated into one unit.
The wireless station has a 6-byte MAC address for connecting, likewise the AP.
Wireless station needs to associate with an AP before it can send or receive network layer data.
An AP has an associated service set identifier (SSID) which identified the AP.
The frequency is the split into smaller ranges for 2.4 GHz there is 11 partially overlapping channels, where channels separated by four or more channels is non overlapping.
The find an AP a beacon frame is sent periodically, with the APs SSID and MAC address, and devices scans the 11 channels for beacon messages known as passive scanning.
For the device to choose the connected APs the host themselves implements an algorithm not specified.
Active scanning the device sends a broadcasting probe frame which is responded to by the AP.
When connected a DHCP discovery message is send into the subnet through the AP.
The AP can authenticate with different methods.

- Devices MAC address
- Username and passwords

This can be handled by an authentication server

13.3 802.11 MAC Protocol

There are three different classes of multiple access protocols

- Channel partitioning including CDMA
- Taking turns
- Random access

13.3.1 Taking turns

Taking turns work by a polling master is chosen, the master makes sure that in turn every node is in turn given access to the medium.
This has drawbacks in case of the master occur some error it will stop the whole protocol until a recover is done.
A decentralized alternative is the token passing protocol, where a fixed order is chosen and once a token is given the host can use the medium and pass on the token after use.
The problem is in case of a host occur an error and does not forward the token it stops the protocol

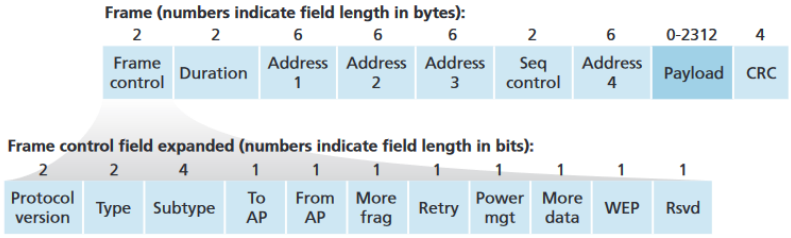
13.3.2 Random access

Tries to transmit frames in case of collision wait a random amount of time.
Example is Slotted ALOHA which retransmit using a probability
The most optimal probability is 37% making 26% chance of collision
The MAC uses this form of protocol known as CSMA without collision avoidance.
This is due to host may not detect other networks transmission because they are opposite of the router and therefore the router receives the transmission before the other host known as hidden terminals.
Therefore, the Wi-Fi uses a link-layer acknowledgment/retransmission (ARQ) scheme.
When the destination receives a frame that passes the CRC waits a short inter-frame spacing (SIFS) and sends an acknowledgment frame.
In case of no ack the host will retransmit.

13.3.3 Hidden terminals

To counteract the problem a reservation of the medium is given by the AP. A host sends a short request to send (RTS) control frame, and if responded by the AP with a Clear to Send (CTS) frame allows the host to send data. Collision will be a little problem due to the frames being short.

13.3.4 Frame



Payload up to 2312 bytes, but mostly fewer than 1500 bytes.
The frame has four address field

- Mac address of the wireless station that is to receive the frame
- Mac address of the station that transmits the frame
- Mac address of the router which connects the subnet to the router
- Used for forward frames to each other in ad hoc mode

Duration used for reservation of the medium.
The type and subtype fields are used to distinguish the association RTS, CTS, ACK and data frames.
To and From field are used to defined meanings of the different address fields.
WEP indicated if encryptions are being used.

13.3.5 Mobility in the same IP subnet

Mobility between BSSs is straightforward if the BSSs is part of the subnet. This can allow the host to keep the IP and all ongoing TCP connections. Otherwise, a new IP is needed through DHCP and will disrupt all TCP connections.

13.3.6 Advanced features

Rate adaption selects the modulation technique.
If an ode sends two frames in a row without receiving an acknowledgement the transmission rate falls back to the next lower rate.
If 10 frames succeed the increases to a new higher rate.
Power management can be done by managing sleep and wake states.
A node will tell the AP when it goes to sleep by setting a power management bit in the header.
A time in the node is set to wake up just before the AP beacon time.
The AP then buffers frames for the node, and in the beacon tells if nodes have any buffered frames.
If any then fully wakes up otherwise goes back to sleep.

13.3.7 Personal Area networks: Bluetooth

Uses time division multiplexing (TDM) which divides time into time frames and further divides each time frame into N time slots.
Each time slot is then assigned to a node. Time slots size is chosen based on transmission time of a single packet.
TDM eliminates collision and is fair but waste a lot of time in case of nodes not always having to transmit.
It also uses FDM which divides the medium range into channels which a node is assigned to but suffers the same problem as TDM.
Randomized backoff and polling, error detection and correction and reliable data transfer via ACKS and NAKS.
Uses 2.4 GHz.
During a timeslot of the TDM one of 79 channels is used which is changed in a pseudo random way known as frequency hopping spread spectrum (FHSS)

14 Securing wireless LANs and 4g/5g

The wireless nature makes it possible to sniff all packages.
802.11 should handle

- Mutual authentication - verify the identity and access privileges, likewise the device authenticates the network
- Encryption - Using the symmetric key encryption is used in practice since encryption and decryption must be performed at high speed.

To process the mutual authentication and encryption-key derivation

1. Discovery - AP advertise its presence and the forms of authentication and encryption, the device then requests specific forms of auth and encryption

2. Mutual authentication and shared symmetric key derivation - Devices and AP share a common secret (e.g., password), using nonces and cryptographic hashing

3. Shared symmetric session key distribution - A protocol will be needed for the authentication server to inform the PA of the shared symmetric session key

4. Encrypted communication between mobile device and a remote host via the AP - AES used

14.1 Mutual authentication and shared symmetric session key derivation

The first standard Wired equivalent privacy (WEP) contained security flaws.
Wi-Fi protected access (WPA1) came with message integrity check, and avoided attacks which could infer encryption key after some observation
WPA2 mandated the use of AES symmetric key encryption.
WPA is a four-way handshake protocol that performs both mutual authentication and shared symmetric session-key derivation.
The handshake goes as

1. Authentication server (AS) generate a nonce and send it to the mobile device

2. Mobile device M receive the nonce generated the symmetric shared session key using the AS nonce and its own nonce and the known secret and MAC address of M and AS, and ends its own nonce and the AS nonce and secret key in HMAC signed package

WPA3 updates attack on the four-way handshake by using reused nonce and longer key length among other changes.

14.2 Authentication and key agreement in 4g/5g

The 4g authentication protocol AKA consist of

1. Authentication request to HSS - device sends its international mobile subscriber identity (IMSI) which is related to the Mobility Management Entity (MME), MME then sends it to the Home subscriber service (HSS)

2. Authentication response from HSS - HSS uses shared secret key to create an authentication token

3. Authentication response from mobile device - uses the shared secret key to decode the authentication token and sends the answer to the HSS

4. Mobile device authentication - If a match is done HSS inform the base station and mobile device that it is authorized and sends the base station keys used in the authentication token

5. Data plane and control plane key derivation - Base and mobile device agree on encryption protocol using the auth key

For 5g there were changes to

- Two new protocols for authentication and key agreement, meant for IoT environment and does not need shared secret key

- Uses public key cryptography techniques to encrypt a devices permanent identity

15 Anonymity and Privacy

Anonymity can only somewhat be reached using the protocol looked at so far.
Using IPsec, it is possible to anonymize the sender but not the receiver.
It is possible to reach confidentiality with TLS and IPsec (confidentiality has some identifier to pinpoint the connection unlike anonymous which is simply a connection)

15.1 Internet

Unlike the open internet known as the NFS net, there exist different networks which is also connectable.
The deep web consists of non-indexed sites, and therefore will not appear in search engines and the direct url have to be known to access.
The dark web uses different protocol with more encryption and uses multiple hops to get access to sites not visible on the normal net anonymously.
TOR is a browser used for setting up an onion connection.
It is done by selecting 3 onion nodes and generating shared key with each.
Then the message is encoded for each three nodes such the first node decodes the first layer and sends it further, until the last node decodes the actual message and sends it to the wanted web server.
The TOR browser defines

- Message length - to ensure that length could not be used to determine where in the chain the message is
- Message structure
- How to establish keys
- How messages are sent
- How nodes should decrypt and forward messages

Due to the increased hops and encryption the browser is substantial slower.
The nodes use trusted guard/entry nodes such an entity cannot get enough nodes to determine traffic

15.2 Hidden services - Onion websites

Onion websites exists on in the nodes, such it never leaves the TOR network.
This achieves that either the client or server know each other if no authentication.
This will ensure privacy in case of a sniffer at the input and output and a client can be correlated to output, therefore defeating anonymously.
A hidden service is setup by selecting three random nodes as entry nodes i.e., nodes which knows the address of the server.
The service is then added to the global hash table (known as DHT distributed on all nodes) as a descriptor with its onion address as key, and the value pair of ip for entry nodes and its public key for authentication.
When a connection is made after the wanted hops it connects to an entry

point which essentially forwards, the service then goes to the last node known as the rendezvous point (RP) of the chain before the entry point and sends the response through its own number of hops.
The onion address comes in different versions

- Version 1 - Generated from public key
- Version 2 - First half of base32 encoded SHA-1 hash of public key from 1024 RSA
- Version 3 (standard) - Is now 56 characters long, uses SHA-3 instead and curve25519 instead of RSA, making it more private and robust