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INTEGRATED SERVICES ACCESS NODE WITH LINUX

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ABSTRACT

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The objective of this thesis was to study an integrated all-IP device for the most common network infrastructure services. The thesis studied various services that could be added into the device. Finally it demonstrated such physical device, with data, VoIP and security focus. The work was commissioned by Tele-entre Oy

The work was done using regular, server grade components and freely available open source applications. First different services that an integrated access node would need were studied. Secondly we decided which particular operating system and applications we would use. At the end all components were installed, configured and a test plan was made for checking the functionality.

The result of the study is a functional, multiservice router and firewall with basic VoIP services. The platform can flexibly accommodate any additional services; as well it can also be easily fitted in any PC hardware. If this device would be productized in the future, it would need a more detailed documentation and throughput measurements. Also the hardware would have to be more long term supported. A self-built device is mostly suitable for small and medium size installations, large scale operations are always better off with commercial devices.

Keywords: TCP/IP, Linux, Router, Firewall

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VOCABULARY

AAA	Authentication, Authorization and Accounting
AS	Autonomous System. An Internet domain or entity, which advertises its address space to other domains. Identified by an Autonomous System Number (ASN)
Broadcast domain	The area in which a broadcast message is heard. Separated by routers.
CIDR	Classless Inter-Domain Routing. A system for subnetting the classful IP networks into different size networks.
Collision domain	An area where a packet collision is affecting traffic.
Cost	Cost is a synonym for metric. It describes a routes desirability. It can be used to distinguish the best route from several candidates.
CRL	Certificate Revokation List. A list where the certification authority can put information on revoked certificates. This can be utilized to check certificate validity.
DHCP	Dynamic Host Configuration Protocol. An automated host IP configuration service.
DNS	Domain Name Service. A service for resolving a name to an IP address and vice versa.
DoS/DDoS	(Distributed) Denial of Service. Attacks that aim to paralyze services from or towards the Internet.
FXO	Analogue phone interface (Foreign eXchange Office). An interface which can send off-hook/on-hook signals. This interface connects towards the operator.

FXS	Analogue phone interface (Foreign eXchange Subscriber). An interface that provides a dial tone and supplies battery power and ringing voltage. This interface connects towards the end user.
HFSC	Hierarchical Fair-Service Curve. An algorithm for a queueing discipline where bandwidth and delay constraints can be set.
IDS	Intrusion Detection System. A system for tracking intrusion events. It can be used to alert in case of preset events happen. This service is parallel to the communication system.
IFB	Intermediate Functional Block. An iptables module that allows incoming traffic to be shaped or policed.
IPS	Intrusion Prevention System. A system for tracking and dropping intrusion events. This service is inline with the communication system.
ISP	Internet Service Provider.
Metric	Look at Cost.
MPLS	Multi-Protocol Label Switching. A way to assign paths for a packet based on an outer label on the packet.
NAT	Network Address Translation. It changes the source or destination IP address of a packet before forwarding it. A variety of NAT is NAPT, where the same address is used under different layer 4 ports.
NGFW	Next generation firewall
NTP	Network Time Protocol. This is a method of getting autoconfiguration of accurate time from time servers.

PAN	Personal Area Network.
PKI	Public Key Infrastructure. An infrastructure for authen-tication and secure transfer of information.
PRI	Primary rate interface. A T1/E1 line for voice, which has 23/30 voice channels available.
PSK	Pre-shared key. An authentication key that is shared between peers.
QoS	Quality of service. This is the tool for ensuring that each IP service gets their individual communication re-quirements fulfilled, like throughput or delay.
RTP	Real-Time Protocol. A best effort protocol for delivering real-time data.
SFQ	Stochastic Fairness Queueing. A queueing discipline for dividing the available bandwidth fairly to different users.
SIP	Session Initiation Protocol. A protocol for handling call setup.
SOFO	Small Office/Home Office.
TCP/IP	Transmission Control Protocol/Intenet Protocol. This is the suite that IP communication is based on, and it in-cludes a variety of other protocols.
TLS	Traffic Layer Security. A security protocol for encrypting data on network layer 4. It used to be called SSL.
VLAN	Virtual Local Area Network. A way to convey traffic for several different networks in one interface. Here each network is tagged with a VLAN ID, which identifies which network the traffic is for.

1 INTRODUCTION

1.1 Environment

Data networks and the Internet are growing bigger and more complicated all the time. All services tend to run over IP nowadays, and all premises - both private and commercial - are continuously connected to the Internet. While this is enabling more sophisticated businesses and services, this has also become a threat for businesses. Many companies are susceptible to data communication failures, both intentional and unintentional. This is because companies use cloud services and IP-based ordering systems. For example, if a bakery has a one day outage in Internet services, it could in theory lose a whole day of sales. Security breaches are much more likely today than they were before, as the exploit tools have developed. Anyone can download an exploit kit and start exploiting vulnerable devices, and it does not need much technical savviness.

Internet and security services will incur cost in the terms of continuous service and/or hardware purchase. Usually, different services will add on the cost, meaning that each provider brings in their own device or online services. To avoid some of these costs, private persons, small and even midsized organizations can build their own equipment for the services and Internet edge. This will have associated labor cost, but if there is technical knowhow and spare time, this can save some external costs. There are additional benefits, like custom configurations and rules that can be set up. Infrastructure know-how is kept within the company. On top of that, any kind of service can be added into the network infrastructure.

1.2 Target

The purpose of this thesis was to build an all-IP integrated device, which would fulfill various tasks using inexpensive software solutions. The intention was that this device could safely connect to the Internet, and it would give adequate security and filtering against hostile attacks and denial of service.

The main areas of this device would be:

- IP connectivity and routing
- Network services
- End user services and applications
- Security; threat detection and prevention
- Actual access node hardware and software selection
- Additional service elements

The feature set that was selected for the device is following:

- IP router with static routing, option for dynamic protocols
- DHCP server
- DNS server
- NTP server
- VoIP server
- VPN server
- Firewall and IPS device

As an outcome, we will have an access node device which can take care of these various tasks in one inexpensive open source device.

2 IP NETWORK BASICS

2.1 Network concepts

Network is a wildly used term with different meaning in different contexts. In communications, the term is divided into sub terms, which describe the size and function of the network more accurately. The size comparison is illustrated in figure 1.

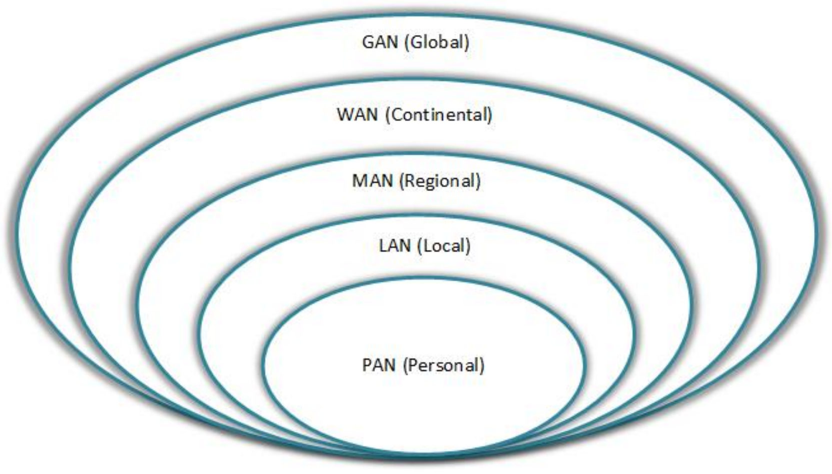


FIGURE 1. Network size comparison

PAN is a personal area network, its radius is up to ≈10m. Current technologies used in this area are e.g. Bluetooth, RFID, IEEE 802.15.4 based protocols, DASH7.

LAN is a local area network, ranging from 10m…1km. Usually, this network is privately owned, belonging to an individual property or organization. The wire-less version of this is called WLAN, using some IEEE802.11 based protocol. Several LANs in the same area comprise a campus network, covering multiple buildings.

MAN is a metropolitan area network, i.e. it is regional, and its coverage is around tens of kilometers. It could be a city wide network, serving a regional operator or municipality. The wireless version of this is WMAN and a common technology is among others WiMax.

WAN is a wide area network. It serves a state or a country and its size is hun-

dreds of kilometers. WWAN is the wireless version of it. PLMN (Public Landline Mobile Network) is an example of a WWAN.
GAN is a global area network, covering the whole globe. The satellite network is one such network. GAN is also a loose synonym for the Internet, which is a global network too. However, the Internet consists of a combination of millions of these smaller network blocks that are just interconnected.

2.2 TCP/IP briefly

TCP/IP is a protocol suite, a kind of an umbrella, where underlying protocols do their tasks. Different TCP/IP protocols are defined as RFCs, requests for comments. These RFCs are hosted and managed by IETF, the Internet Engineering Task Force.

The TCP/IP suite is derived from the OSI-model, a 7-layer abstraction of data layers. The correspondence of each TCP/IP layer to an OSI layer is represented in figure 2.

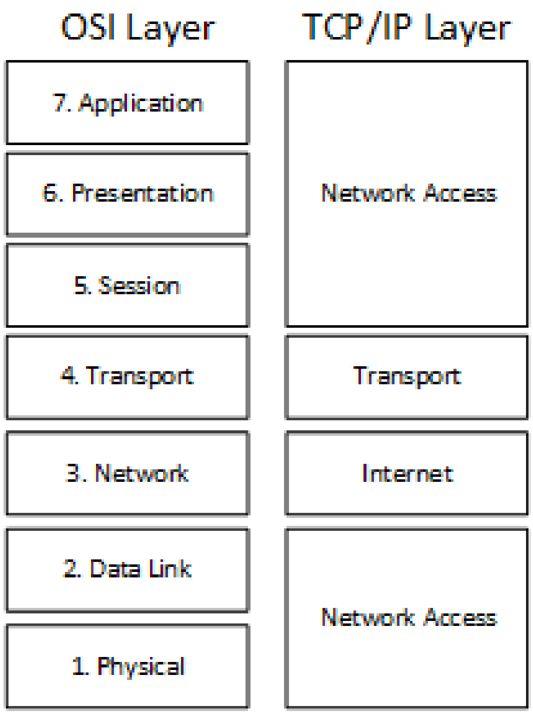


FIGURE 2. OSI-model compared to TCP/IP model

Each of the layers can consist of several protocol options. The basic idea in TCP/IP is encapsulation; Each layer has its own data header and the layer above is encapsulated as the payload. When data moves downwards on the stack, headers are added. When data moves upwards, headers are stripped. When addressing a particular layer, usually the OSI-model layers are referred to, not the TCP/IP layers.

L1 – Physical. This is the medium, e.g. copper, fiber or air.
L2 – Data Link. This is the data representation on the physical media. The layer includes modulation, framing, timing and physical addressing. As physical and data link layers are often very tightly defined together, they have been combined in the TCP/IP model. The most common IP L1/L2 technology is Ethernet (IEEE 802.3 and its wireless companion IEEE 802.11).
L3 – Network. This is the logical address layer. In case of TCP/IP it has the IP address. It provides an unreliable, best effort service.
L4 – Transport. It provides a session based, reliable or unreliable transportation service. The reliable protocol is called the Transmission Control Protocol (TCP, RFC 793) and the unreliable one is the User Datagram Protocol (UDP, RFC 768)
L5 through L7 – Application. These comprise the TCP/IP application layer, which is mostly not relevant to the network operation. An exception is modern network equipment, which does check the application layer headers and data for making decisions on the treatment.

2.3 IP addressing

There are entire books written on IP addressing, and to understand it completely one would need more information that this thesis can cover. A short introduction is put here.

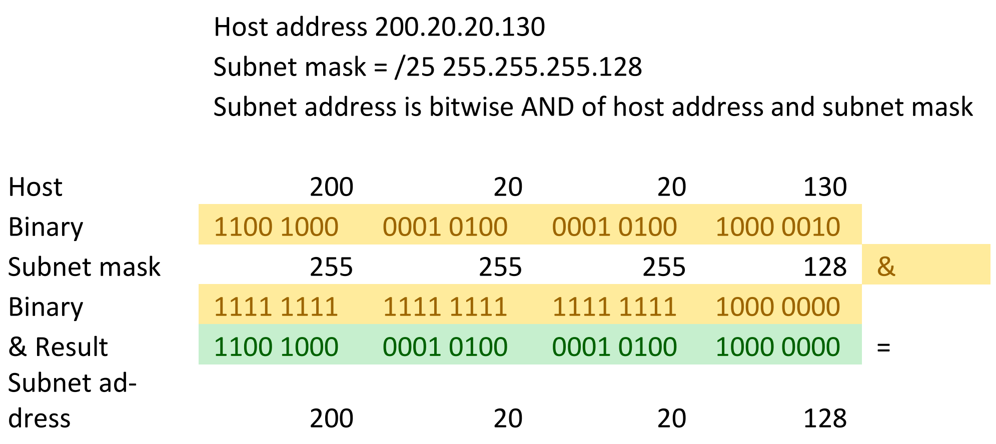
The IP address is the logical address of a host or a network. This is how the device is known to the domain it resides in, whether it is local or global. There are two generations of IP addressing in use currently: IPv4 (RFC 791) and IPv6 (RFC 2460). IPv4 is a 32-bit number, whereas the length of an IPv6 address is 128 bits. IPv4 is still the major addressing scheme, but it is running out of free numbers. That is the main reason why the new version IPv6 was developed. This should provide enough numbers for a long time, maybe indefinitely. Currently both addressing systems are existing side-by-side and various techniques exist to allow traffic to flow between these networks.

2.3.1 IPv4

IPv4 is the original specification, initially built for DARPA to provide a network capability for the ARPA-net. An IPv4 address is 32 bits long and provides over 4 billion addresses. In practice the address space is much smaller, as it was quite generously given out in the beginning, and big blocks remain unusable as they were assigned for special purposes.

The IPv4 address is presented in octets, four groups of 8 bits each. Each group is written out in decimal numbers in a range from 0 to 255. An example of such an address is: ***200.20.20.130.***

Leading zeros in each octet can be left out. The address is divided into 2 parts, the network portion and the host portion. The network portion is the leftmost portion and host portion is the leftover. When referring to the network portion, a network mask or prefix is used. The prefix is also known as a CIDR notation. In the mask the network portion is given as a bit mask, i.e. the network portion is all ‘1’ and host portion zeros. When a logical AND operation between the host address and mask is performed, the result will be the network address. The prefix notation corresponds to the mask notation by telling how many bits in the mask are ‘1’ from the beginning. (see figure 3)

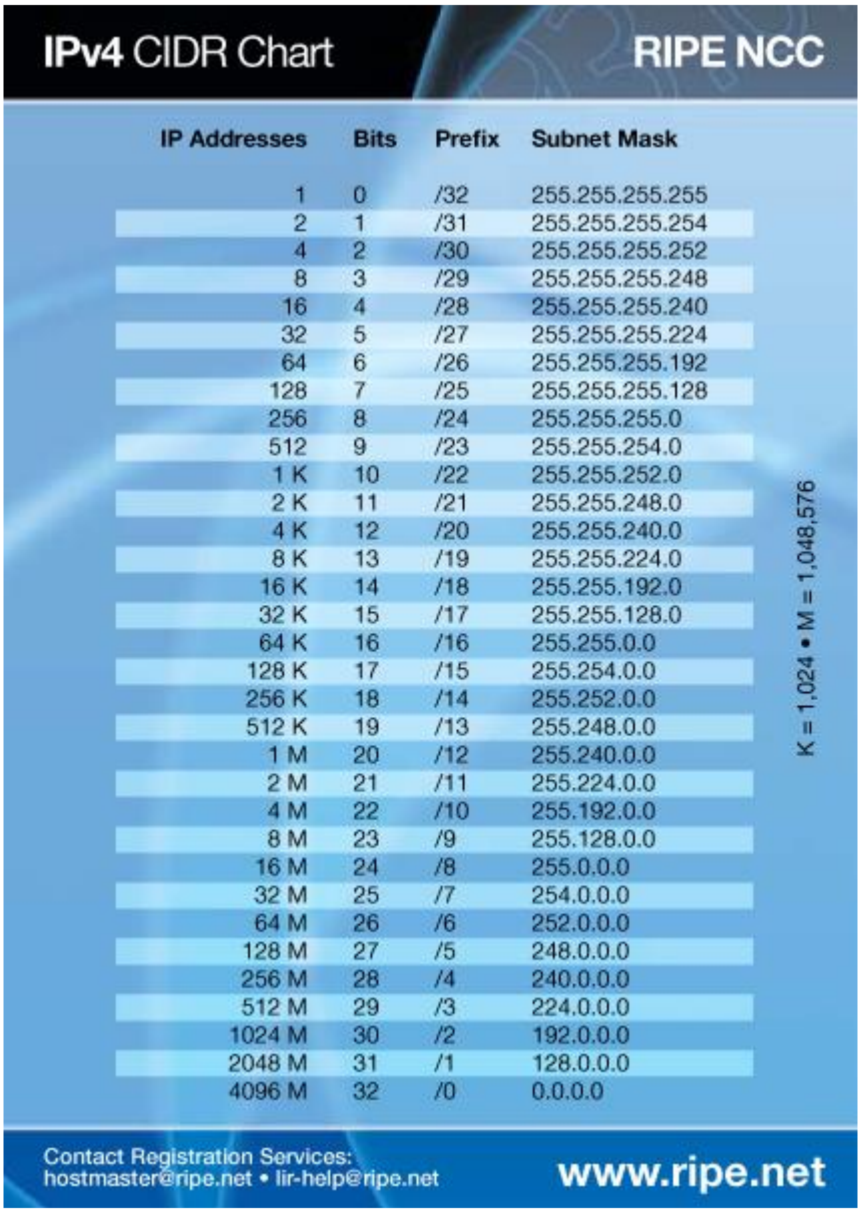


Host address 200.20.20.130
Subnet mask = /25 255.255.255.128
Subnet address is bitwise AND of host address and subnet mask

Host	200	20	20	130	
Binary	1100 1000	0001 0100	0001 0100	1000 0010	
Subnet mask	255	255	255	128	&
Binary	1111 1111	1111 1111	1111 1111	1000 0000	
& Result	1100 1000	0001 0100	0001 0100	1000 0000	=
Subnet ad-dress	200	20	20	128	

FIGURE 3. IPv4 subnet determination example

The correspondence between bitmask and prefix is shown in the following figure 4.



IPv4 CIDR Chart

RIPE NCC

IP Addresses	Bits	Prefix	Subnet Mask
1	0	/32	255.255.255.255
2	1	/31	255.255.255.254
4	2	/30	255.255.255.252
8	3	/29	255.255.255.248
16	4	/28	255.255.255.240
32	5	/27	255.255.255.224
64	6	/26	255.255.255.192
128	7	/25	255.255.255.128
256	8	/24	255.255.255.0
512	9	/23	255.255.254.0
1 K	10	/22	255.255.252.0
2 K	11	/21	255.255.248.0
4 K	12	/20	255.255.240.0
8 K	13	/19	255.255.224.0
16 K	14	/18	255.255.192.0
32 K	15	/17	255.255.128.0
64 K	16	/16	255.255.0.0
128 K	17	/15	255.254.0.0
256 K	18	/14	255.252.0.0
512 K	19	/13	255.248.0.0
1 M	20	/12	255.240.0.0
2 M	21	/11	255.224.0.0
4 M	22	/10	255.192.0.0
8 M	23	/9	255.128.0.0
16 M	24	/8	255.0.0.0
32 M	25	/7	254.0.0.0
64 M	26	/6	252.0.0.0
128 M	27	/5	248.0.0.0
256 M	28	/4	240.0.0.0
512 M	29	/3	224.0.0.0
1024 M	30	/2	192.0.0.0
2048 M	31	/1	128.0.0.0
4096 M	32	/0	0.0.0.0

K = 1,024 • M = 1,048,576

Contact Registration Services:
hostmaster@ripe.net • lir-help@ripe.net

www.ripe.net

FIGURE 4. IPv4 classless network chart (19)

2.3.2 IPv4 address types

Originally, the IP address space was divided into different classes: A, B and C-class addresses were regular unicast addresses handed over to localities needing them. A, B and C class addresses have different network lengths, but in today's classless network concept it has little or no meaning. The D-class is multicast and E-class is for experimental/future use.

These addresses are divided into different types of addresses based on their locality or purpose.

Loopback address (127.0.0.0/8) - This is an address, that points to the device itself. It is not used for communication between devices. It can be used to send traffic internally from an application to another, or to test the TCP/IP stack.

Link Local (169.254.0.0/16) - This is a non-routable link specific address. It can only be used within the same layer2 network (broadcast domain).

Private address space is unique within a certain scope, such as a site. It can be used within one entity; this kind of address is not routable towards the Internet. Operators will block them from entering. Private networks are 10.0.0.0/8, 172.16.0.0/12 and 192.168.0.0/16, and they are defined in RFC1918.

Global – Globally routable unicast. This address type can be routed over the Internet. This group includes the A,B and C classes, i.e. addresses where first octet is 1 to 126, and 128 to 223, with the exception of the private networks, link local addresses and some other exceptions. This address type enables Internet communication. It is referred to as a one-to-one address.

Anycast – This is closely related to unicast, and it uses the same address space. This is referred to as one-to-nearest. It is useful if an identical service exists in multiple geographical locations and clients are wanted to use the nearest advertised address. In practice this address is a regular unicast address that is advertised from several different Autonomous Systems.

Multicast – This is a group address type. It could be referred to as one-to-many. A device can join a group which has been given a certain multicast address. All participants of this group then receive the messages destined to the group. The first octet of a multicast address is 224…239. Some of the addresses are reserved, but most of them can be used freely.

Broadcast – This is the last address of a network. It is used to deliver messages that have no set recipient.

2.3.3 IPv6 address composition

An IPv6 address is 128 bits long, and it provides a vast amount of different addresses, more than 1 address per grain of sand on earth. This is enormous compared to IPv4's 32-bit long addresses.

The format of the number is hexadecimal and it is arranged in 16-bit blocks separated by a colon. An example of an IPv6 address looks as follow:

2a00:5240:1111:2222:3333:4444:5555:6666

It can be abbreviated by removing leading zeros in a block. Also16-bit blocks of zeros can be left out once in an address and they can be replaced with "::" as shown below:

2a00:5240:0000:0000:0000:0002:0000:0001 -> 2a00:5240::2:0:1

It should be noticed that zeros can be removed only once, otherwise the address presentation is not unique anymore.

Like IPv4, an IPv6 address consists of a network portion and a host portion. The network portion can further be divided into subnets and aggregated into summarized networks. The routing is done based on the network portion, and the host in a target network is identified based on the host portion. The network portion of an address is given in a CIDR notation, which is a slash followed by a number, telling how many bits the network portion consists of, starting from the beginning. Figure 5 shows how many subnets are available using certain network masks.

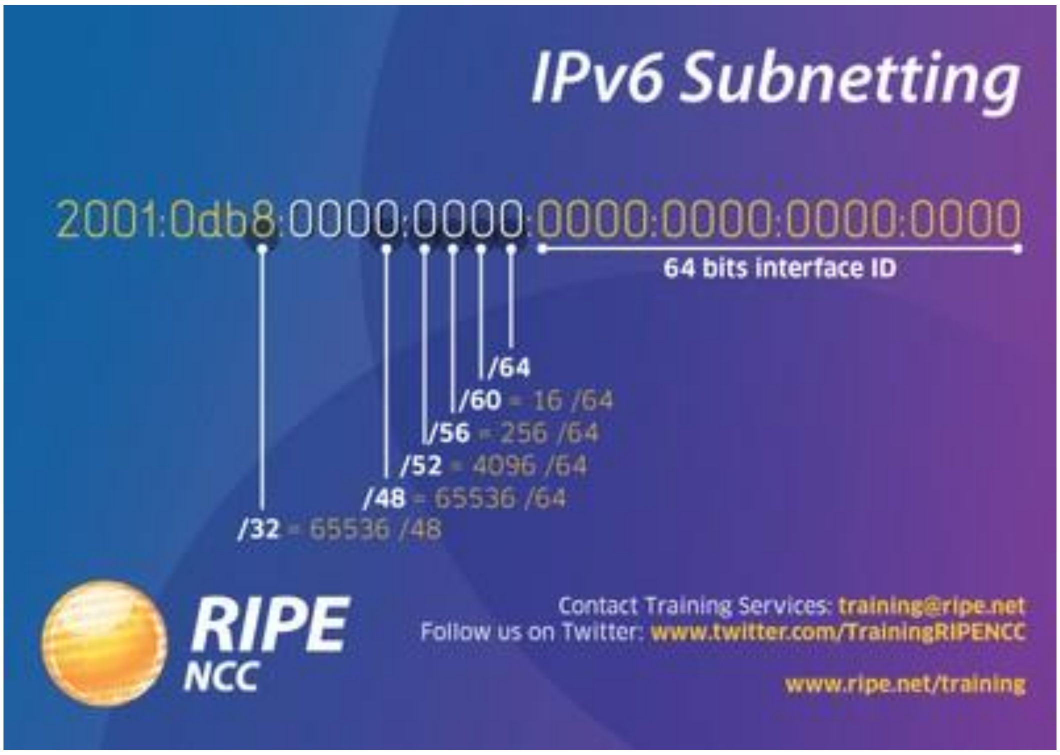

IPv6 Subnetting
2001:0db8:0000:0000:0000:0000:0000:0000
64 bits interface ID
/64
/60 = 16 /64
/56 = 256 /64
/52 = 4096 /64
/48 = 65536 /64
/32 = 65536 /48
RIPE NCC
Contact Training Services: training@ripe.net
Follow us on Twitter: www.twitter.com/TrainingRIPENCC
www.ripe.net/training

FIGURE 5. Subnets of an IPv6 address (20)

The beginning of the address tells us the type of address, which in this case is global unicast (starting with 2). Unicast addresses are globally routable addresses used for one-to-one connections. There are also other types of addresses, which will be described later. The length of the network mask tells how many bits in the address belong to the network portion. In practice, usually an operator gets an assignment of a /32, of course based on the need and request. Then the operator usually assigns its' customer a /48. A common practice is that the final network portion is a /64, so that the host portion consists of 64 last bits of the address. It could be considered a waste of addresses, but then again there are a lot of them available. A 64-bit mask provides for stateless auto configuration (SLAAC), as explained in chapter 2.3.5.

2.3.4 IPv6 address types

IPv6 address types are roughly the same as in IPv4, with some differences. Private addresses (RFC 1918) are called Unique Local (RFC 4193). The broadcast category is also obsoleted and does not exist.

Loopback address is presented as ::1/128. This corresponds to the IPv4 127.0.0.0/8 range.

Link local range is FE80::/10. This corresponds to the IPv4 169.254.0.0/16 range. As the network portion is 10 bits, it means that an address can start with FE8…FEB.

Unique local range – which corresponds to IPv4 private addresses – is FC00::/7. Here there is a 7 bit mask for the network, which gives a beginning of FC all the way to FD.

Global unicast addresses are in the range of 2000::/3. This gives a start number of 2 up to 3. There is no equivalent block in IPv4, but this corresponds to every A, B and C-class address not belonging to some special type group.

Multicast addresses have a range of FF00::/8, which is everything starting with FF. This corresponds to an IPv4 D-class address.

There are also other types of addresses, reserved for tunneling and other special purposes.

2.3.5 IPv6 address assignment

In every case, an IPv6 enabled interface creates a link-local address. This address is used in all initial communication. This is a self-generated address that does not require information from outside, although the host validates the address by checking that it is not used by anyone else.

The assignment of unicast addresses in IPv6 can be done in several different ways. Basically all dynamic assignment methods use the Neighbor Discovery Protocol. Here, the node that connects to the network sends an RS (Router Solicitation) message into the network. The router responds with an RA (Router Advertisement) message. This message contains information about the router, including the autoconfig options. The options are stateless, stateful or both (or none). In a stateless autoconfiguration, also known as SLAAC, the router sends the network prefix, i.e. the network portion of the IPv6 address. The host then generates an address based on this prefix. The other option is stateful, in which case there is a separate server, DHCPv6 server, taking care of the address assignment. If this is the case, the host sends a DHCP solicit messages to find the server, and acquires the address that way. The last option is static, where the address is manually put into the interface. There are many improvements in IPv6 compared to IPv4. A well-structured guide to IPv6 is available from Cisco(15).

2.4 Internet

As it was concluded in the beginning, the Internet is a worldwide network consisting of interconnected smaller networks. A vital part of the networks is the global IP address. The governing agency of Internet addressing is IANA, which is allocating addresses for regional Internet registries, RIRs, such as ARIN and RIPE. The regional registries allocate address space to local Internet registries, LIRs. These local registries are either various size Internet operators or companies, and their task is to assign these addresses to end users or customers. In this way we ensure that there are globally no unintended overlaps in addresses.

LIRs can apply for an ASN (autonomous system number) which allows it to participate in the Internet routing. The Internet consists of either peering or transit relationships between different ASNs. Peering means that an ASN exchanges its own network and customer information with a peer. The ASN is allowed to send data towards the neighbor ASN’s networks and its’ transit customers only. Peering is free - once it has been established. Transit means that an ASN is allowed to send data through the provider ASN to any other ASN, and the transit customer pays for the transit according to a contract. In figure 6, the company D has transit relationships with an operator B and the company E with an operator A. Traffic between them would go E-A-B-D and vice versa. However, if the-operator A’s connection to the operator C would go down, it and its customers would not reach C through the operator B, as B is not advertising C’s routes to A.

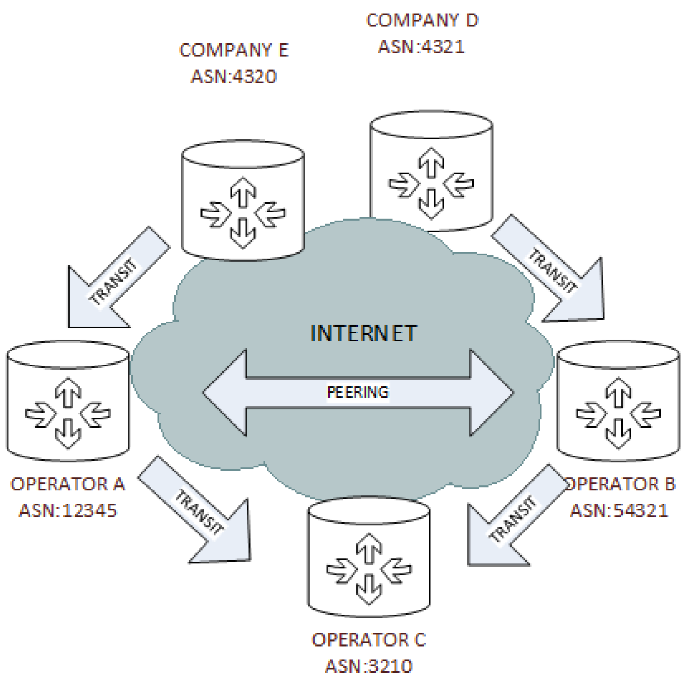


FIGURE 6. Internet relationship principal

The protocol that the Internet route exchange uses is called BGP, the border gateway protocol.

2.5 Network devices

To deliver an IP packet from a device to another through the data network, there has to exist some intermediate devices in between. The devices are conceptually put on some layer, but most modern devices can handle operations on several layers. The network devices are categorized roughly as following:

HUB. This is essentially a multiport repeater. It has several ports, and when it receives a packet on one port, it sends it out on all other ports. It works only at half duplex, meaning that it can either send or receive at a given time, not both. This is a layer 1 device, which is not seen any more in modern networks.

Bridge/switch. This is a device that combines at least 2 different network segments into one physical network, i.e. it grows the broadcast domain but retains the collision domains. Functionally, it listens to ongoing traffic, learning layer 2 addresses. If it has learned an address, it will forward traffic destined to that address only through the port where it was learned on. These are mostly full duplex devices, though a port can be configured as half duplex. Conceptually these devices are L2 devices. The difference between a bridge and a switch is a bit obscure, but essentially a switch is a multiport bridge.

Router. A router combines different networks. It has segments in multiple networks and route traffic between them. This is a layer 3 device, and it performs routing based on the logical address information in a routing table or a table derived from it, called a forwarding information base (FIB). It can be a separate device or part of a so called multilayer switch.

Firewall. This is a device that permits or denies traffic in a network and does network address translation. It operates on the layer 4, though modern firewalls can inspect traffic all the way to the layer 7. This can be a separate device or part of a router.

VPN gateway. This is a device that functions as an endpoint for outside connections. It can be used for allowing access to a trusted network via an untrusted network. It can be a separate device or server, or it can be part of a router or firewall.

Connections between devices are usually made by using different types of Ethernet cabling. There exist other protocols besides Ethernet, however Ethernet is becoming more and more dominant. Ethernet media is nowadays using twisted pair copper cable, fiber-optic cable or it is wireless. There are several categories of Ethernet, all of which are defined in IEEE 802.3 and IEEE802.11. Ethernet can currently reach speeds up to 100Gbps, and 400Gbps is under development (it uses multiplexing).

3 NETWORK SERVICES AND APPLICABLE PROTOCOLS

3.1 Routing – the core of network traffic

For an IP packet to reach the receiver, it needs to know the route to it. For this purpose, there can be either static routes or dynamically learned ones. The latter becomes more important as the network grows. Usually, smaller networks will imply the static routing. Major elements of a route are a destination network, a next hop address or interface, a cost and an administrative distance. The destination network is the network where the packet is headed for. The next hop address is the address of the adjacent router. The cost is the value that is given to a route, i.e. the most preferred way of getting to the network in question. The administrative distance is the route trustworthiness. This is describing the reliability of the manner the route was injected into the system. Directly connected networks have the best trustworthiness, after that comes static routing. The different dynamic routing protocols come after them. It should be remembered that the AD values may be vendor specific.

The routing information is used to build up a routing table. This table exists in the router memory and it is used for looking up routes for packets. Some purpose-built routers use the routing table to build faster lookup methods, like a FIB, the forwarding information base , and an adjacency table. The FIB is a hardware optimized routing table for quicker lookups and the adjacency table includes the hardware addresses of the next-hop IP addresses.

When evaluating routes, it is important to understand the priority order in which a route is selected.

1. Longest match. The route having the longest match in the network portion of the address is preferred
2. Lowest administrative distance (AD). The lower the AD, the more trustworthy the route source is.
3. Lowest cost. If a route has the same network mask and AD, the cost or metric is used to determine the preference of the route. In dynamic pro-

tocols the cost is created based on different network characteristics, such as bandwidth, a hop count or a delay.

4. If all the above are the same, the system usually starts using equal-cost load balancing, i.e. some packets are sent using the other route and some using the other. Some routing protocols allow using unequal cost load balancing by some predefined factor.

3.1.1 Static routing

A static route consists of a statement that tells what the destination IP network is, and where it can be reached from. The target for the packet is given either by a so called next hop address, or by an interface name (or both on some systems). An interface name can be used alone only if the interfaces are directly connected point-to-point. In a multi-access network the next hop method, which is the logical address of the adjacent router, should be used. A static route can also have a metric associated with it. This way a more preferred route to the same network can be assigned.

A major concept in routing is a default route. This is the route that is used if there are no better matching routes. A device can have multiple floating routes. Floating in this context means that there is a route with a worse AD than the one in use. If the currently used route goes down, the other one can be taken into use instead.

3.1.2 Dynamic routing

The routing protocol suite is large, and the details of it can easily take couple of books to explain. The major categories of dynamic routing protocols are distance vector routing protocols, link state routing protocols and path vector routing protocols. The two first ones are used mostly in internal networks. A distance vector as a term means that a single node has no complete picture of the network. It rather relies on the information given from its neighbours. Link state protocols on the other hand have a link state database, where it can independently build a topology of the network.

There are 2 flavours of distance vector protocols in use today: RIP (Routing Information Protocol) and EIGRP (Enhanced Interior Gateway Routing Protocol). RIP is a protocol which relies on the hop count as a metric. It is the least trustworthy IGP, as the convergence is very slow, and the hop count as a metric is not the best one. However, it is fast to put up and it works in smaller networks. EIGRP is a Cisco protocol, which used to be proprietary. The license was released recently, allowing anyone to use it. It is a rather clever protocol, which has several factors affecting the cost: bandwidth, delay, load, reliability and MTU. However, by default only bandwidth and delay are used. It has as well a smart algorithm for having backup routes available for an immediate use.

Link state protocols are OSPF (Open Shortest Path First) and IS-IS (Intermediate System to Intermediate System). They have lots of similarities, however , OSPF was built for IP and IS-IS for any protocol. Both uses a concept of areas, but their definitions are different. OSPF calculates route cost based on bandwidth, IS-IS has a fixed cost in most implementations. The benefit of link state protocols is the convergence speed; they converge very fast. As they are built from different areas, the networks can be really big without causing too much overhead. Also the routing tables are kept reasonable. On the downside, the route calculation algorithm is more processor intensive than those of distance vector protocols. Also configuration along with fine tuning is more difficult.

Path vector protocols actually include only one major protocol, BGP, which is the routing protocol of the Internet. It is also used in MPLS networks for distinguishing virtual routing tables. This is the slowest converging protocol, and it has the most complex parameters for selecting a route, too.

The categories and flavours of dynamic protocols are shown in figure 7.

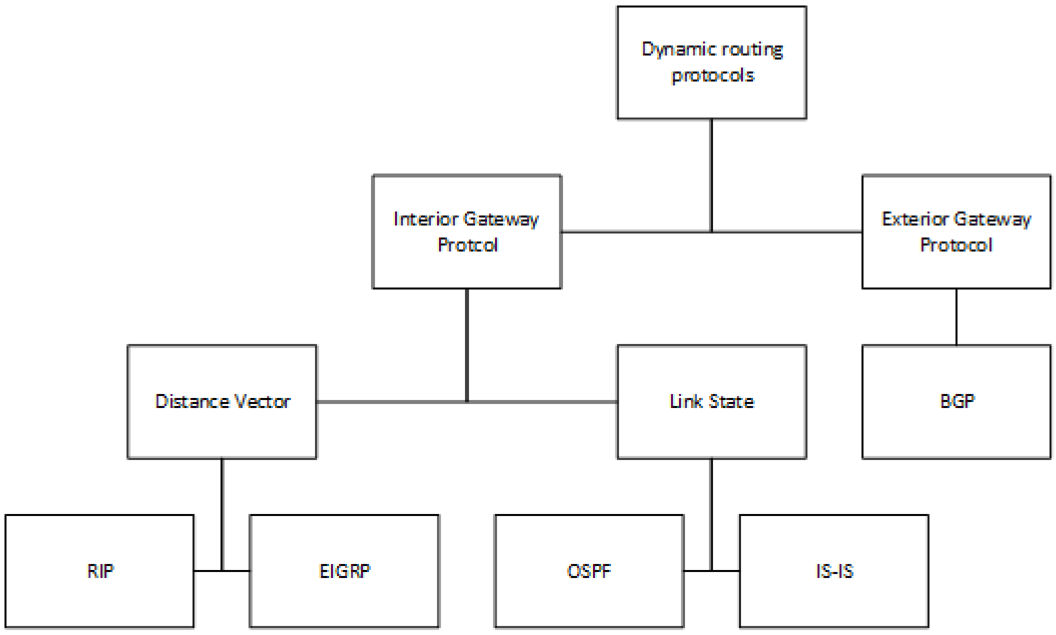


FIGURE 7. Dynamic routing protocols.

3.2 DHCP

The dynamic host configuration protocol is a network service, which can automatically assign necessary IP addresses for computers and other end devices. Therefore, any user connecting to the network gets connectivity automatically, and they do not need to assign static IP addresses for themselves. Mandatory configuration items dictated by the protocol are an IP address and a netmask. However, to be better usable, other configuration values can also be assigned, as the gateway to reach other networks, a search domain name, TFTP or NTP server addresses and other specific options.

DHCP works in the following way, it is also illustrated in figure 8:

The connecting host sends a broadcast message called a DHCP DISCOVER. This is intercepted by the DHCP server. The DHCP server acknowledges this request by sending a DHCP OFFER, which includes the IP address that it is proposing for the host. After receiving this, the host sends out a broadcast message, called a DHCP REQUEST, for that particular address. When it has been received by the server, it reserves the address for that particular node’s use and sends out a DHCP acknowledgement message.

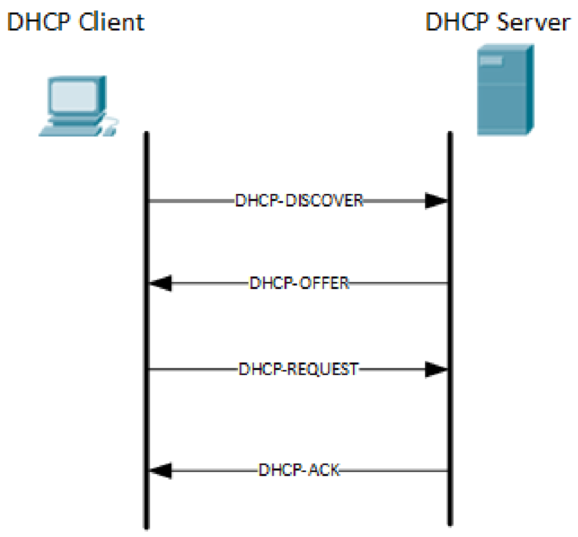


FIGURE 8. DHCP address allocation

If a DHCP server is in a different network from a host's perspective, the router can forward the DHCP request to a predefined DHCP server. This service is called an IP-helper.

3.3 DNS

Remembering a bunch of IP addresses is practically impossible for all of the used services. For this purpose, there is a domain name service. It is a feature that interprets IP addresses to names and vice versa. A fully qualified domain name includes several parts separated by a period or a dot, for example an organization could have a registered domain name of myfirm.com. If a publicly available device is added into that network, it could be assigned a name myserver.myfirm.com. Now, when some external user wants to connect to this server, they only need to remember its name, not its IP address.

DNS lookup sequence is presented in figure 9. When a host application wants to reach an outside server, e.g. www.example.com, it sends the name to its local DNS client, called a resolver. The resolver can be a caching one, i.e. if the name has been used before it might remember it from the past. In this case, it first sends a non-recursive query to itself to query the address. If it cannot find the name from the cache, it sends out a recursive (1) query for the address.

Recursive means that the query is sent to another DNS server, which takes care of resolving the address. This server can then start an iterative query, starting from the root server. In this procedure, it will first ask for the .com name server from a root name server (2) that has statically been defined. Then it asks for the example name server from the com name server (4). Finally, it asks for the www name from the example.com server (6). When everything has been resolved, it sends the reply back to the requesting client (8). Iterative queries are normally done by DNS servers, not clients.

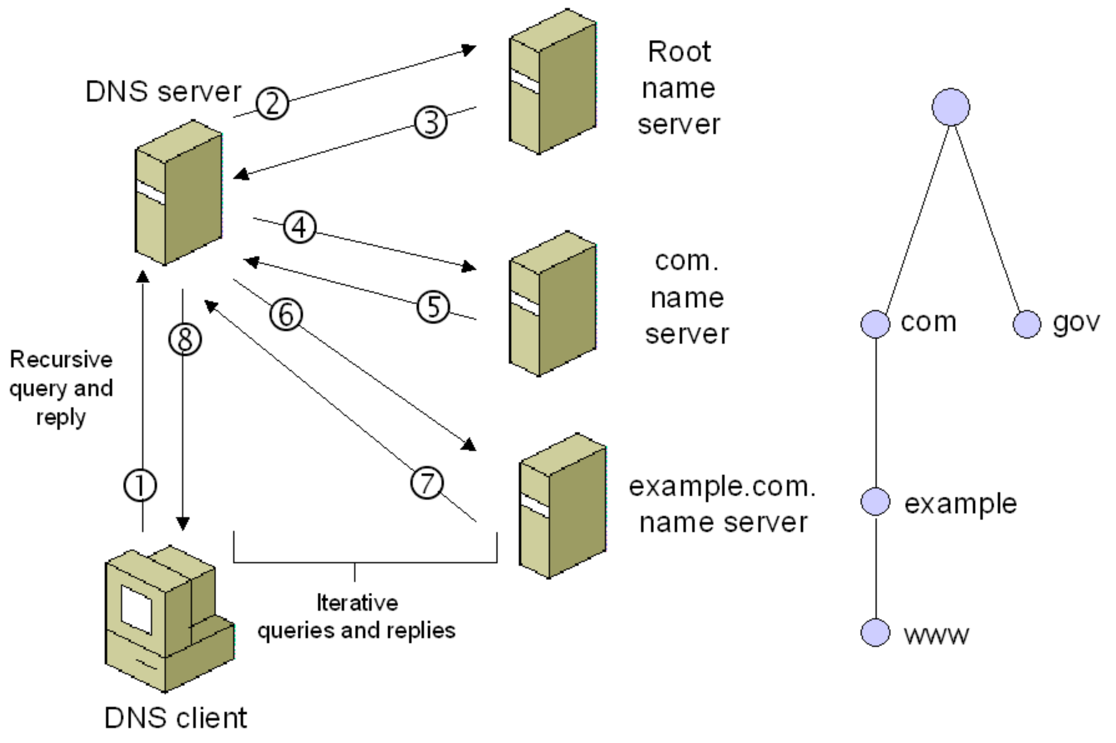


FIGURE 9. DNS query process (21)

3.4 VPN

VPN a.k.a. Virtual Private Network is a concept where two remote networks are made to look being part of the same local network over an unsecure network between them. Usually, these connections are built over the Internet but other applications are found too. Against common belief, a VPN is not secure by de-fault. In practice this is always wanted.

3.4.1 Types of VPN

In the bigger picture there are two types of VPNs by purpose: site-to-site and client /server VPN. A site-to-site VPN is meant to join two remote networks together on a more permanent basis. A practical use case would be a main office and an affiliate office. The traffic from main office would use the VPN tunnel to reach the affiliate one and vice versa. The tunnel can even be made redundant, thus it is possible to have multiple Internet services from different operators, and the tunnel would remain up even if some operator has a service outage.

The other use is the client VPN. This setup is more for a mobile worker or home office. The client program is installed on a desktop, laptop, mobile or SOHO router. It can be invoked when a connection to the office is needed.

3.4.2 CIA

Part of the VPN tunnel security philosophy is CIA. CIA stands for confidentiality, integrity and availability. This triad is used in other security contexts as well, but it is a vital part of a real-life VPN connection. Confidentiality means that only authorized parties should be able to view the data. It is a combination of encryption and authentication. In the VPN context it is predefined who is authorized to see the information, but authentication is needed to receive the data. Integrity means that no-one should be able to alter the data. This is usually ensured with a hash function. A hash ensures that the data has been received in the same format as it was sent. Availability means that the data should always be available. Availability is therefore part of redundancy planning and disaster recovery plans. Different VPN technologies use different technologies to achieve CIA, but pretty much with the same tool sets.

3.4.3 Authentication factors

The authentication of a VPN connection is based on 3 factors: something that is known, something that is possessed and something that the user is (biometrics). It is generally recommended to have at least 2 of these 3 factors used before access will be granted. This ensures that the user asking for access is having a possession of something that has been given to them, either at birth or by the VPN service owner. The most common form of a 2-factor authentication is user/password or certificate and a changing PIN code. The PIN code can be generated by a token, or it can be sent to user's handset via an SMS or a push message.

For site-to-site connections, the most popular method is a pre-shared key authentication. In this case some arbitrary information, called nonce is sent. The initiator is sending the nonce to the responder, and the responder sends its (different) nonce to the initiator. Both respond with the hash result between the received nonce and the PSK. If the results are the same, one can safely assume that the other side has the same PSK.

3.4.4 Encryption

A VPN tunnel can be encrypted with different encryption algorithms. A key exchange is done by asymmetric keys. Asymmetric means that there are two components in a key, which are related to each other, but which are very hard to resolve with one component only. If something is encrypted with the other, it can be decrypted with the other. For a key exchange, usually Diffie-Hellman algorithm is used. Asymmetric keys are explained better in the security chapter when talking about PKI.

The VPN tunnel data traffic is almost always secured by symmetric algorithms. Symmetric means that there is only one key for encrypting and decrypting the data. This key is then changed on the fly at set intervals. There are several algorithms for symmetric encryption, though the most popular one today is AES in different bit lengths.

3.4.5 Hashing in integrity

A hash is a one-way "checksum", which cannot be decrypted back to the original data. It is rather a number, which alters a lot even if just one bit is changed in the original data. In practice, several data values can have the same hash value, but it is very unlikely to happen. In addition to the PSK authentication, a hash is used to ensure the integrity of data. Here, the to-be-sent data is hashed with some shared value, which is not the PSK hash. It can differ depending on the VPN technology. However, the hash of the data will be sent in the packet, and the receiver can then check that their calculated hash value corresponds with the one on the packet.

3.5 QoS – Quality of service

QoS is a method for classifying and transporting IP packets at different importance levels. The target is to transfer real-time data as quickly as possible, while bandwidth hungry data will fill the void, based on importance. Thus a fluent user experience is achieved. Importance levels define what needs to be done with the traffic; is it provided for priority access, a predefined bandwidth or something else. In practice, this means that traffic is identified based on some criteria, e.g. IP-address, port number, application protocol. Based on the criteria, the packets are marked with a QoS value. This value can then be used for different QoS actions; police the rate (inbound or outbound) or shape (outbound only) the traffic below a certain rate. Policing means that there exists a predefined rate that is the absolute maximum transmission/receiving rate and any excess packages will be dropped. Shaping on the other hand is about filling the empty slots in the transmission so that the bandwidth is utilized as well as possible. The principle can be seen in figure 10.

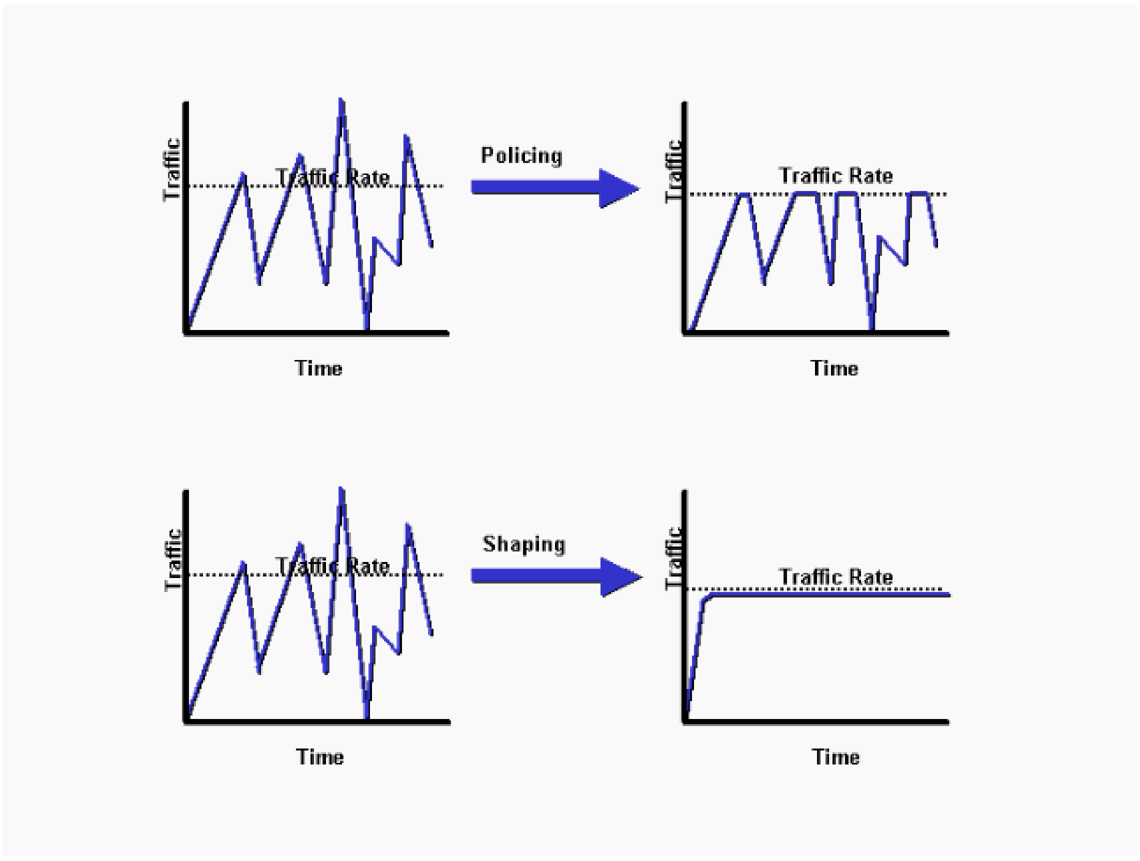


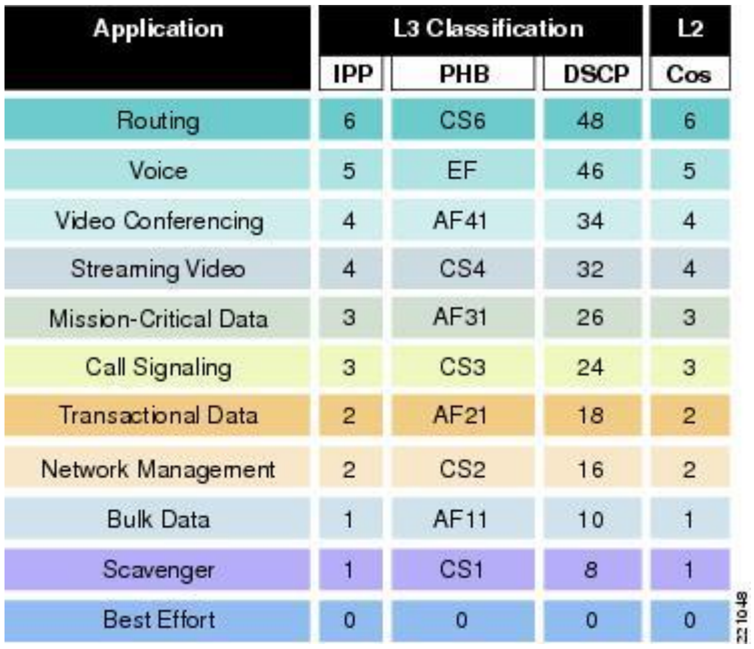
FIGURE 10. Policing vs shaping (2)

Additionally, traffic can be queued based on the markings and start dropping less important packets to avoid congestion.

The main two methods for classifying are the IP precedence (part of RFC 791) and the successor of it, DSCP (Differentiated Services Code Point RFC 2475). DSCP utilizes the 8 bit COS/TOS value for marking purposes, though only 6 bits of the field is currently used. IP precedence uses only three bits of the COS/TOS value, although it is compatible with DSCP. It should be noted that RFC791 lists other options for the three next bits; delay, throughput and reliability. However, these are rarely used. Some examples on what traffic classes could be used for different types of traffic is shown in table 1.

.

TABLE 1. Some DSCP and precedence values (22)



Application	L3 Classification			L2
	IPP	PHB	DSCP	Cos
Routing	6	CS6	48	6
Voice	5	EF	46	5
Video Conferencing	4	AF41	34	4
Streaming Video	4	CS4	32	4
Mission-Critical Data	3	AF31	26	3
Call Signaling	3	CS3	24	3
Transactional Data	2	AF21	18	2
Network Management	2	CS2	16	2
Bulk Data	1	AF11	10	1
Scavenger	1	CS1	8	1
Best Effort	0	0	0	0

3.6 Marking principles

DSCP markings have two parts; the first part is the three bit IPP, IP precedence. The second part is the three bit drop eligibility. Actually only two first bits are used, the third one is always 0. The bigger the number is, the higher the probability to be dropped within the PHB class. The last 2 bits can be used for ECN, Explicit congestion notification. However, this is currently an optional feature. The class zero is best effort (no marking). Assured forwarding has classes AF1x-4x. These values correspond to IPP 1-4. EF stands for expedited forwarding, and it corresponds to IPP 5. Classes 6 and 7 are reserved for internal routing information and such mandatory services for keeping the network running. The second three bit part of the DSCP is drop eligibility.

There are no set classifications for different type of traffic, i.e. the organizations and operators may use the classifications as they see fit. Usually, however, the higher the classification is, the more sensitive the content is for dropping. Class 5 is interactive, delay-sensitive data, i.e. voice and video conferencing. Class 4 is usually video and call control traffic. Class 3 is often some mission critical data, important for business purposes. Network and IT management traffic has often its own class to ensure that IT personnel can access the remote management interfaces.

3.7 Queue discipline

Classification itself cannot achieve anything, it is just a marking. To actually use it for some relevant purpose, different queues are needed. There are several different kinds of queuing techniques based on functionality. The main ones are FIFO queue, a priority queue, a class based queue and a weighted fair queue.

FIFO: First In First Out, i.e. no queue at all. This is the fastest queue with almost no configurability. Only the queue length can be altered.

Priority queue: Everything in the high priority queue is sent before the lesser priority queues.

Weighted Fair Queue: This assigns different flows automatically into different queues. If the set queue limits are hit, the queue starts dropping more aggressive packets before other traffic.

Class based WFQ: Here different rates are assigned for different queues, and then the traffic is guided to different queues based on markings.

In practice a combination of these queues is needed. In Cisco world this would be called LLQ; low latency queuing. In this solution there is a priority queue, which prioritizes a pre-set amount of bandwidth. Then there are different amounts of class-based WFQs, and then the bulk traffic queue.

Similar effects can be achieved with the Linux traffic control (tc), using HFSC and SFQ (17). HFSC can be used to assign bit rates and a predefined maximum delay to classes. SFQ can be used for sharing the queue with multiple users at an equal, fair rate.

3.8 VoIP - Voice over IP

Traditionally voice has been a service that operators bring all the way to our phones. The systems have been proprietary and therefore expensive. Also, subscribers have had to have a double wiring for data and voice. The maintenance of in-house systems have been expensive, and the provided solutions very inflexible. A solution to this is VoIP. It is a technique that allows to send the voice and signalling data over your existing data network. After that, end users can use a desktop device or various applications to playback the voice. The main technology to handle the calls is called SIP, which is introduced later in this chapter. There are other technologies as well, such as SCCP (Cisco), MGCP and H.323, but these are nowadays less popular protocols. Especially open source systems are relying on SIP.

3.8.1 Voice sampling - codec

An analogue voice signal needs to be sampled to get it into a digital format. The purpose is to convert an analogue signal to a digital format, process and send it digitally, and then just before it is played out, it will be converted back to analogue format. This will allow digital equipment to be used for the whole transfer process, and it will also allow a magnitude of operations that can be done to the signal. In sampling, the Nyquist sampling theorem must be obeyed. This theorem states that samples need to be taken on a rate of 2 times the highest frequency of the signal to be sampled.

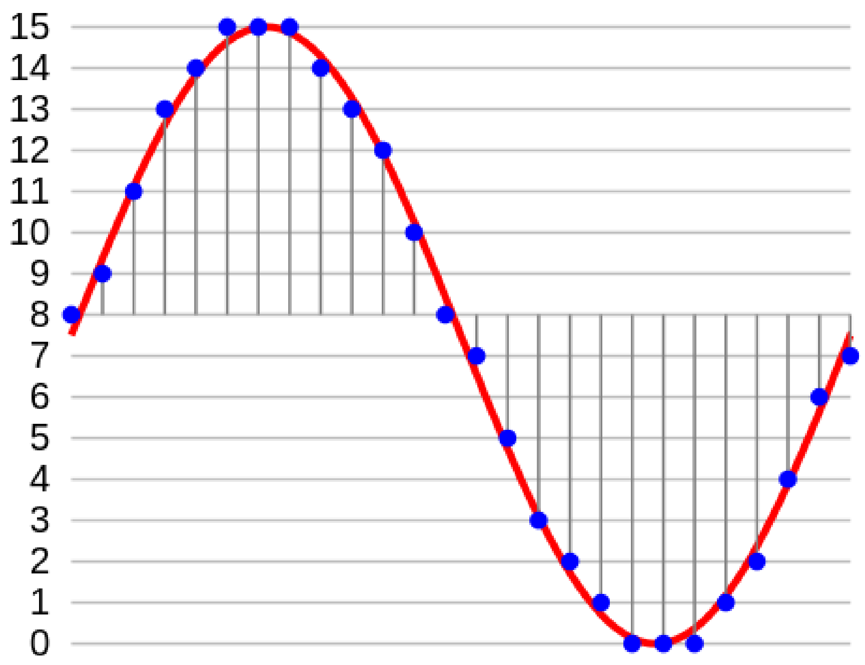
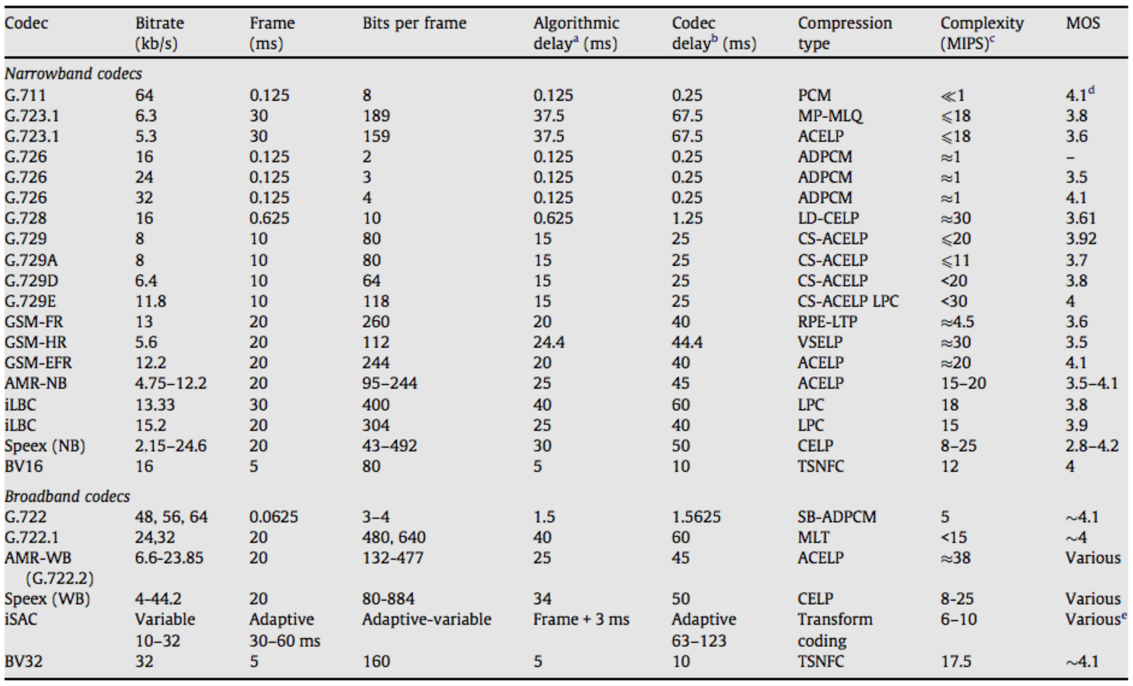


FIGURE 11. Sampling of ananlogue signal. (23)

Figure 11 shows the process of sampling. There exists a certain set of amplitude levels. The analogue voice signal is quantized at the Nyquist rate into the nearest level set by the codec. These levels have some digital representation value, and the sample can now be represented in a digital format for further processing.

There are many different codecs. G.711 is the first codec used in the original PCM system. After that there has been several codecs, which concentrate generally on saving bandwidth without affecting the voice quality too much. There are also some adaptive codecs, which alter themselves based on the content or bandwidth available. A codec quality is very difficult to measure. The best system so far is the MOS, Mean Opinion Score. Here a test panel of at least 12-15 people is listening on voice samples produced by different codecs, giving them a score from 1 to 5. A mean value is then calculated based on the score. The higher the score, the better the codec is. Table 2 shows a brief comparison of different codecs.

TABLE 2. Charasteristics of different voice codecs (24)



Codec	Bitrate (kb/s)	Frame (ms)	Bits per frame	Algorithmic delay[a] (ms)	Codec delay[b] (ms)	Compression type	Complexity (MIPS)[c]	MOS
Narrowband codecs								
G.711	64	0.125	8	0.125	0.25	PCM	≪1	4.1[d]
G.723.1	6.3	30	189	37.5	67.5	MP-MLQ	⩽18	3.8
G.723.1	5.3	30	159	37.5	67.5	ACELP	⩽18	3.6
G.726	16	0.125	2	0.125	0.25	ADPCM	≈1	–
G.726	24	0.125	3	0.125	0.25	ADPCM	≈1	3.5
G.726	32	0.125	4	0.125	0.25	ADPCM	≈1	4.1
G.728	16	0.625	10	0.625	1.25	LD-CELP	≈30	3.61
G.729	8	10	80	15	25	CS-ACELP	⩽20	3.92
G.729A	8	10	80	15	25	CS-ACELP	⩽11	3.7
G.729D	6.4	10	64	15	25	CS-ACELP	<20	3.8
G.729E	11.8	10	118	15	25	CS-ACELP LPC	<30	4
GSM-FR	13	20	260	20	40	RPE-LTP	≈4.5	3.6
GSM-HR	5.6	20	112	24.4	44.4	VSELP	≈30	3.5
GSM-EFR	12.2	20	244	20	40	ACELP	≈20	4.1
AMR-NB	4.75–12.2	20	95–244	25	45	ACELP	15–20	3.5–4.1
iLBC	13.33	30	400	40	60	LPC	18	3.8
iLBC	15.2	20	304	25	40	LPC	15	3.9
Speex (NB)	2.15–24.6	20	43–492	30	50	CELP	8–25	2.8–4.2
BV16	16	5	80	5	10	TSNFC	12	4
Broadband codecs								
G.722	48, 56, 64	0.0625	3–4	1.5	1.5625	SB-ADPCM	5	~4.1
G.722.1	24,32	20	480, 640	40	60	MLT	<15	~4
AMR-WB (G.722.2)	6.6-23.85	20	132-477	25	45	ACELP	≈38	Various
Speex (WB)	4-44.2	20	80-884	34	50	CELP	8-25	Various
iSAC	Variable 10–32	Adaptive 30–60 ms	Adaptive-variable	Frame + 3 ms	Adaptive 63–123	Transform coding	6–10	Various[e]
BV32	32	5	160	5	10	TSNFC	17.5	~4.1

3.8.2 SIP – Session Initiation Protocol

SIP is the signaling method in modern open source VoIP networks. SIP takes care of registering clients, and keeping track of their location. The protocol that is transferred in real time can be then something else, SIP does not care. SIP is defined in RFC 3261.

As SIP is very simple in nature, it needs other protocols to function as a media protocol. The most important of those protocols are, e.g. SDP (Session Description Protocol) and RTP (Real Time Protocol). SDP negotiates the necessary parameters for the sessions, and RTP takes care of moving the actual real-time payload.

SIP infrastructure has devices with the following roles and roles can also be combined:

UAC/UAS – User Agent Client/Server. This is usually the origination and termination point of a session. The client function initiates the session and the server function responds to it.

Proxy Server – The most common server. It handles the SIP messages and forwards them to either another proxy or straight to the end point. It can also handle messages between the user agent and the registrar.

Registrar – Registration Server. This device authorizes and registers the devic-es in the environment.

Location Server – The registrar stores the location information of the authenti-cated client on this server.

```
              server1.com  . . . .  server2.com
           .    proxy                 proxy      .
         .                                         .
 User1's  . . . . . . . . . . . . . . . . . . . .  User2's
softphone                                        SIP Phone
   |                |                |                |
   |     INVITE M1  |                |                |
   |--------------->|    INVITE M2   |                |
   |  100 Trying M3 |--------------->|    INVITE M4   |
   |<---------------|  100 Trying M5 |--------------->|
   |                |<-------------- | 180 Ringing M6 |
   |                | 180 Ringing M7 |<---------------|
   | 180 Ringing M8 |<---------------|     200 OK M9  |
   |<---------------|    200 OK M10  |<---------------|
   |    200 OK M11  |<---------------|                |
   |<---------------|                |                |
   |                         ACK M12                  |
   |------------------------------------------------->|
   |                      Media Session               |
   |<================================================>|
   |                         BYE M13                  |
   |<-------------------------------------------------|
   |                       200 OK M14                 |
   |------------------------------------------------->|
   |                                                  |
```

FIGURE 12. Basic SIP session (25)

Figure 12 has a presentation of the messaging involved.
M1. UAC sends an INVITE-message to its configured proxy server, destined for user2
M2. Proxy server 1 does not know the location of the endpoint, thus it sends it to server2 based on the INVITE-messages recipient domain.
M3. Server1 sends a TRYING message back to User1

M4. Server2 sends INVITE to user2, as they know the location of the user.
M5. Server2 sends a TRYING message back to server1
M6-8. User2’s device is alerting, and SIP UAS sends a Ringing notification towards user1
M9-11. User2 answers to the ring and OK message is sent, usually containing the SDP information.
M12. User1 sends an ACK, containing SDP information.
RTP data exchange starts
M13. User2 ends the session with a BYE-message
M14. User1 acknowledges the BYE.

4 NETWORK SECURITY

In this chapter some basic network security devices and protocols are visited. Network security is becoming more and more important, because a growing amount of critical data is available through the Internet. Network security is always a compromise between usability and security; if access is not allowed to anyone, the best security is achieved. However, systems are not very useful in that way. Therefore, a midway between these has to be found, allowing the useful traffic and minimizing the possibilities for harmful traffic. And some measures to track the activity in the network must be available.

4.1 Firewall

A firewall is the basic security device in a network. It intercepts the traffic, and based on some rulesets it either allows packets to pass unmodified, alters or marks the packets, or denies the packets. There are several types of firewalls, and several ways of categorizing them. Several types are defined in NIST 800-41 (26), which divide firewalls to packet filters, stateful firewalls, application firewalls and proxy firewalls. On top of that there are many other special purpose versions listed. However, as many firewall features are available in different kinds of firewalls, it is hard to make a strict categorizing. Some well-known categories are listed below .

Stateless firewall, Packet Filter. This acts more like an access list, where connections need to be statically allowed, both inbound and outbound. This is the quickest form of a firewall, but the least dynamic one, too. A session or flow information is not available for rules, rather the IP addresses, ports or interfaces are used for allow and deny statements.

Stateful firewall. This version of a firewall recognizes traffic flows and makes up the pass decisions based on stateful rules. This means in practice that some IP address or range or interface is allowed to send traffic to the destination, and the return traffic is allowed only for that session. When the session ends, further return traffic is denied. This is more resource consuming, but it makes the rules more efficient and simpler.

Application firewall. This firewall is an enhancement to the stateful firewall. It does a so called deep packet inspection on the packets at the application layer. Therefore, not only the layer 4 information affects the traffic, but also the payload on the application layer.

Application-Proxy gateway. This is a firewall that proxies all connections. This is the safest version of a basic firewall, but as well most resource intensive. In this type of a firewall the connections are always terminated at the firewall, there are no through connections. It is the firewall itself that establishes the remaining connection based on its' rulesets. It includes elements from application firewalls.

Next generation firewall is a term for firewalls that do application level inspection and security inspections. For example a certain application or website can be denied and at the same time scan incoming files for viruses or malware. This version includes elements from the previously listed firewall types, and adds intrusion prevention system functionality on the firewall itself.

4.2 Network Address Translation

One other feature of a network access node is NAT, which stands for Network Address Translation. NAT was originally invented to resolve the problem of using private RFC 1918 addresses for Internet access. The idea is to translate certain inside addresses to publically routable ones and vice versa. NAT was originally defined by RFC 1631, nowadays it has several enhancements and improvements added, thus having several RFCs describing it. There are three major types of NAT (11):

Source NAT, where source address is changed. This type is more common for outwards connections (towards Internet). Here the source address is changed after the packet has been processed by the firewall. When a reply to the sent packet is received, the destination address on that packet will be changed to the original inside address.

Destination NAT, where destination address is changed. This is more common for incoming connections. When the packet arrives at the router, the destination address is altered before the packet is further processed. Whenever a reply to that particular session arrives, the source address is changed to that of the original packet.

NAPT (Network address port translation) is a type of NAT, where multiple hosts inside the LAN network can share one public address. This is made possible by TCP/UDP port translation. For example, if a host initiates a session from port 33333, and the port is free on NAT, the firewall reserves that source port for that host’s use and just changes the source address when sending the packet forward. When a reply is received, the destination is changed to the inside host address, based on a translation table. If some other host tries to use the same source port as this host at the same time, the outgoing packet is given some other free port. When a reply is received on that packet, the firewall will change both destination IP and port based on the translation table.

4.3 Intrusion detection and prevention

Network intrusions are events where someone is intentionally attacking our network in different manners. Attackers usually try to exploit some vulnerability in the devices, or they might use some other methods of attacking to gain access to the network. The reasons for intrusions are various e.g. information gathering, preventing service, building botnets. Nevertheless, whatever the reason, the fact is that attackers can cause a lot of harm, especially in a corporate network. Network traffic analysing is a difficult process, as there are so many harmful exploits, attacks and malware to utilize, and their amount is increasing. Also exploit tools are getting more sophisticated.

The devices performing the analysing are roughly divided into two main categories: Intrusion detection systems (IDS) and Intrusion prevention systems (IPS). The major difference between the systems is that an IDS is parallel to the network, and it is only “sniffing” the network traffic, whereas the IPS is in-line with the data traffic and can drop traffic that it finds harmful. IDS on the other hand is only capable of alerting and showing statistics.

Intrusion detection events can be divided into two main categories: signature based and anomaly based. The signature based inspection tries to find a certain pattern or checksum that always exists in that particular threat. This is fixed and has to be continuously updated as new ones are found. Signatures can be based on the application layer content, traffic patterns, ports utilized or even IP addresses. ´The anomaly-based inspection checks the traffic behaviour. For instance, if some device is suddenly opening a lot of connections to the Internet, an alert can be sent. This type of inspection is tricky to set up but it does not rely on continuous rule updates.

There are organizations that keep track of vulnerabilities, and issue periodical statements on these, one of those being Mitre/CVE. Based on these found vulnerabilities, signature files are generated. The signature files are usually maintained by some security organization or device vendor, and they are updated on regular basis by the security system. The administrators can then choose how strictly these events are treated; what kind of traffic will be dropped and what not. The same applies to the alert rules. On top of vulnerabilities, malware can be filtered using the same kind of signature inspection. The most common malware are viruses, Trojans and worms.

The event management software for a security network is called a SIEM, which stands for Security Information and Event Manager (27). This can consist of multiple separate devices, having different tasks. Some common SIEM tasks are:

- User interface for changing settings
- Dashboard for viewing events
- Alerting rules and targets
- Analysis and correlation. The system analyses the data it gathers, and looks for common attributes between different events.
- Data aggregation and logging. A SIEM system can have several sensors throughout the network, and the software itself aggregates and displays the data from different sensors

4.4 PKI

There is a need to authenticate hosts and users reliably in a network. For this purpose there is a PKI infrastructure (28). PKI, i.e. Public Key Infrastructure is a hierarchical system for “reliably” authenticate a host or person. Based on the information, further authorization can be granted to users or hosts to access the resources

X.509 is the ITU standard for a PKI system. It is a system, where we have a “big boss”, the root certification authority (Root CA). The root CA can authorize intermediate certification authorities. These intermediate authorities can then issue client certificates, which can be used for authenticating clients, and to exchange encryption information. For the system to work properly, the Root CA public certificate must exist and be trusted in all devices in the PKI. The lower we go in the infrastructure, the less security is needed. Also, the validity period of a certificate should shorten the lower we traverse in the hierarchy.

PKI is based on asymmetric key pairs, which consist of a private key and a public key. The keys are mathematically related to each other, but it is very difficult to calculate the pair from the other. Data is encrypted with a public key and decrypted with the private key. A certificate on the other hand is a public key which is signed by the issuing authority’s private key. In addition to the public key and signature, it contains data on the usage, validity and revocation list information etc. In case of authentication, the client certificate’s public key is first encrypted with the authority’s private key. The recipient of the certificate can decrypt it with the CA’s public key. If the decryption result is the public key of the sending client, sender is likely to be who he claims to be. An illustration of certificate retrieval process is in figure 13.

The certificate’s intended use can be restricted to a certain type e.g. the client certificate can be set for authentication only, i.e. it cannot be used for encryption. Likewise, a CA certificate can be set to only be used for digital signature signing.

If a certificate is used to start an encrypted session, the asymmetric keys are used to create a changing symmetrical key. This is because the asymmetric key is very heavy on the processor; symmetrical keys are less burdening for the processor

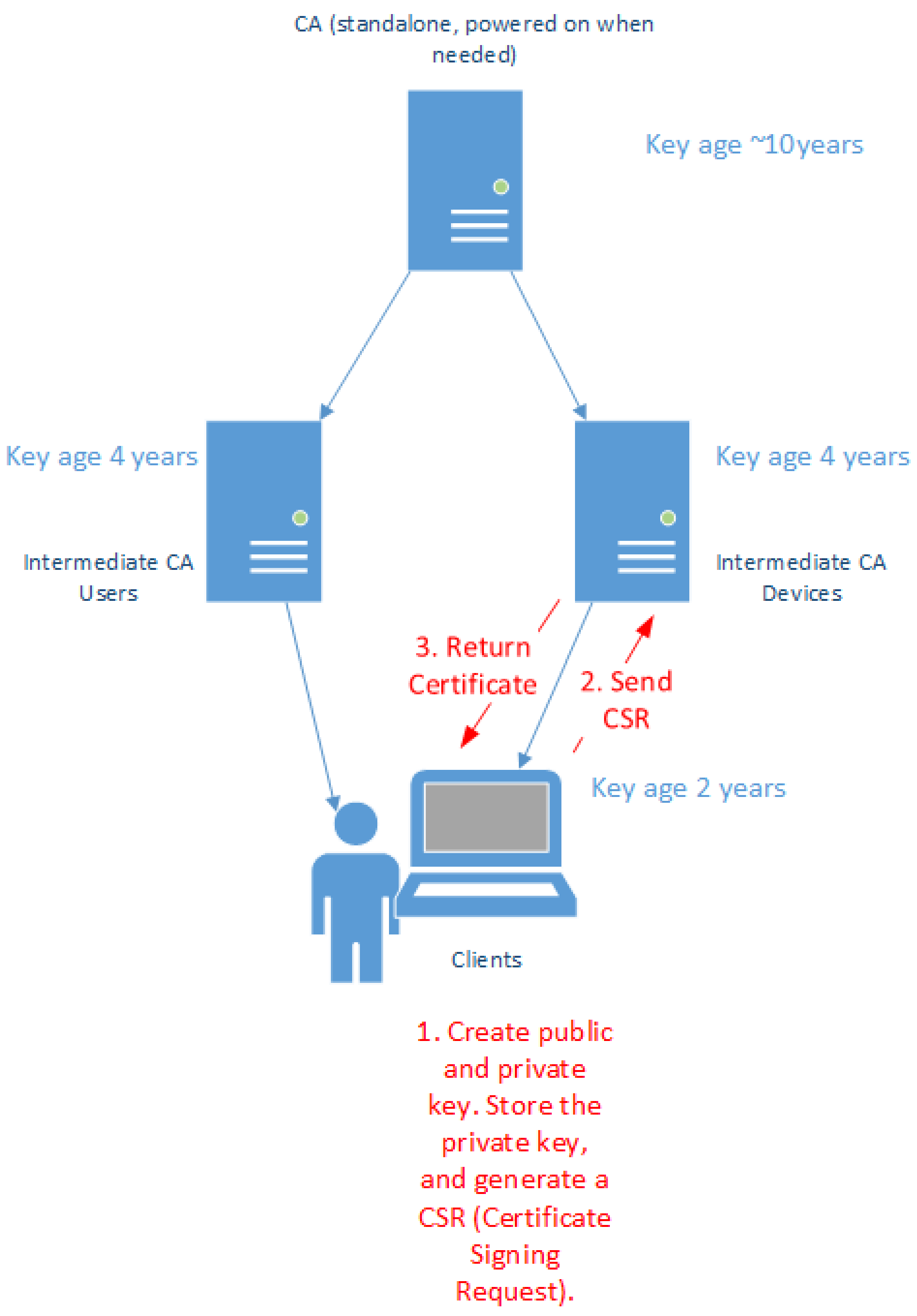

CA (standalone, powered on when needed)
Key age ~10years
Key age 4 years
Intermediate CA Users
Key age 4 years
Intermediate CA Devices
3. Return Certificate
2. Send CSR
Key age 2 years
Clients
1. Create public and private key. Store the private key, and generate a CSR (Certificate Signing Request).

FIGURE 13. Public Key Infrastructure

5 BUILDING BLOCKS FOR THE DEVICE

In this chapter the components to be used for building the device will be selected. There are several options to build a firewall device, and there is no best option. Essential is to pick something that fits the purpose and that builders and maintainers are familiar with.

5.1 General requirements

When a system is selected, there are several general (physical) requirements that need to be evaluated. Some major requirements that the job commissioner had given for a commercial device are below.

Stability. The system needs to stay available as much as possible. Modern systems have a very low MTBF, actually Cisco promises an above 11 years MTBF for some of their ASA firewalls (1). Therefore, the components have to be field proven and of commercial grade.

Support. The support over the intended lifecycle is important for the product. This applies to both hardware and software components. A good network router and firewall will be servicing for at least 3 to 5 years, occasionally even longer. During that time both hardware and software updates are needed. A network card might break down, and it might be safest to have an exactly same model to ensure compatibility. The beauty of a self-built system is, of course, that it is more forgiving if there is a need to use alternate components.

Physical characteristics. The device might have to fit in a certain space, or needs to stay below a preset power consumption. The physical size is usually measured in so called rack units in case of a rack chassis. A rack unit is abbreviated as U or RU. One U-unit is 1.75 inches, which is about 4.45cm. This is the general size of a workgroup switch or a simple router. In this project there is a more sophisticated device, so it could use more space based on the requirements and processing need. But, generally a server or a router is less than 4U high. As the demo device is meant for a small example setup, it can easily fit it in a 1U size.

Usability. A router is attached to a rack; therefore, the access to its interfaces must be in the front. Rear interfaces are not very convenient in a rack environment.

Capabilities. The selected feature set for the device. In this case the device acts as a router, a DHCP-server, a DNS-server, a VPN-server, a VoIP-server and a security device. The device could easily be used for other services too, such as email, web-server and network monitoring, but it would need a better redundancy to be reliable for those operations.

Redundancy. If the device is a critical asset with end user services, it is recommended to have critical parts duplicated, and make regular backups of all data, not only configurations. Redundancy can be achieved with RAID-disks, failover devices, and redundancy protocols. Also power can be redundant. In some installation sites battery backup power might be required (usually 48V)

5.2 Hardware

Based on the set requirements, hardware will be a 1U rack-mount server chassis, which will have a commercial grade Jetway Mini-ITX motherboard. This will ensure a proper installation in a rack and a decent reliability. In this setup there will not be any additional network interfaces, the motherboard and its daughterboard offers enough of them.

5.3 Software and applications

5.3.1 Operating system

As the title tells, this device will be a Linux-based device. Linux as such is the core of several so called distributions, distros. There are several factors that affect the choice of distro for a system:

Popularity and support. The system needs to be supported by either a commercial organization or a strong community. Otherwise updates to critical issues might not be available. Also, unpopular distros have a tendency of being more unstable.

Needed modules. Linux features are provided as installable modules. Usually the modules are available as packages for the distro. If not, the source code can be downloaded and compiled on the target system.

Purpose of distro. Distros are usually purpose-built, meaning that different distros are meant for different tasks. There are distros for servers, client hosts, security people, tiny devices etc. A recommendation is to use some server-grade distribution, as those tend to fulfill the above listed requirements as well.

The selection for the access nodes distro is CentOS 7. It is based on a commercial distro called Red Hat, and has a large user base, including world's leading companies. On CentOS, features are available as installable packets, including most features we have included in our requirement list. In case packets are missing, features can be compiled from source code.

5.3.2 Routing protocols

The access node will be in a fairly simple network, and the routing will be handled statically. Thus, for the current needs, the Linux iproute2 –software suite can easily handle the routing. However, to ensure scalability, the router will have a software called Quagga. Quagga is able to control different routing daemons, such as RIP, RIPng, OSPFv2 and v3 and BGPv4. Quagga has a Cisco-like CLI interface, which is used to configure the daemons. Configuring Quagga is left out of the scope of this document. It is good to understand that all kernel routes will by default have a better administrative distance than those Quagga static routes.

5.3.3 DNS server

The most popular DNS server for Linux is Bind. This software is very mature, and its different versions have been used for decades. Especially, if there is no Active Directory server in the network, this server allows internal DNS zones to be established. The system will be installed in a bind-chroot environment, where bind is isolated from the rest of the system.

5.3.4 DHCP server

Linux provides several flavors of DHCP servers. One of the most popular ones is Internet Systems Consortium's (ISC) version. It has versions for both IPv4 and IPv6, though they need different instances running. With this server, mandatory client settings can be automatically assigned along with a magnitude of options. It also allows addresses to be reserved for hosts based on the MAC-address.

The DHCP server can be made to update the local DNS zone, too. For this feature a dynamic DNS setup is needed, and update keys need to be configured on the DHCP server.

5.3.5 VPN software

There are several options for the VPN software. OpenVPN is a very common open source VPN server, which is extremely versatile for different purposes. The added benefit of Linux is that there can be several different VPN servers. OpenVPN can be used for both client/server and point-to-point connections, while Openswan IPsec software can be installed to take care of possible external connections to other vendors' devices.

If an easier configuration is preferred, there is a commercial OpenVPN Access Server available. However, paid user licenses are needed. A less expensive solution is to configure the server yourself, and use the free client software. A very good guide for building a VPN server is in the book OpenVPN 2 Cookbook (16)

5.3.6 VoIP service

When selecting voice services, a good and mature option is a software called Asterisk (*). This software is capable of functioning as a full scale switchboard on top of terminating traditional POTS lines. Own personal messages can be used on it, also interactive voice response (IVR) menus can be used. The software allows meeting rooms and voice mail. Asterisk can be accompanied with a video call service.

If a GUI is wanted for the Asterisk system, there is a framework called FreePBX available. It is built around the Asterisk core. FreePBX makes it easier to assign SIP extensions and to generally control the switchboard and server. It is freeware. If there is a need for a stand-alone VoIP server, FreePBX has a ready-made distribution, which can be deployed virtually or on dedicated hardware.

5.3.7 Firewall

The traditional Linux firewall is called iptables (6). The new Linux versions also have a new zone-based firewall called simply firewalld.. Both versions are available with the CentOS 7. Both firewalls use the same iptables kernel commands, though the command line is different. Rules with the traditional iptables firewall will be created here.

The functionality of iptables is based on three tables, called “filter”,”mangle” and “nat”. There is also a table called raw, which is used on rare occasions for bypassing the connection state tracking. These tables have predefined chains. Figure 14 shows the logic behind different chains.

The nat-table is used for translating addresses based on given rules. Prerouting chain alters the destination address and postrouting chain alters the source address. The mangle table is used for marking packets, directly traffic affecting rules are not allowed here. The actual firewall functionality is on the filter table. On the filter table we have three predefined chains: input, output and forward. The input chain is used when target IP address is on the local machine. The output chain is used when the source address of the packet is on the local machine. And the forward chain is used when neither source nor destination addresses are on the local machine. Is should be noticed that NAT is done before the filter table, and the decision on the filter chain is done after NAT has been processed.

Every iptables rule has conditions and targets (6). Conditions can be matched based on e.g. addresses, ports, protocol, state. When a condition is matched, the target is executed. Commonly used targets or actions in the filter table are ACCEPT, DROP or REJECT. Also custom chains can be written and traffic can be redirected to those chains. This is useful for the traffic categorization and also to make the firewall rules neater. The mangle table target is usually MARK or TOS. This is used to either assign the kernel level marking, or to assign the TOS-byte some value, usually for QoS purposes. The NAT table has SNAT, DNAT or MASQUERADE targets. The SNAT target alters the source address, DNAT the destination address and MASQUERADE uses the interface address for the port address translation.

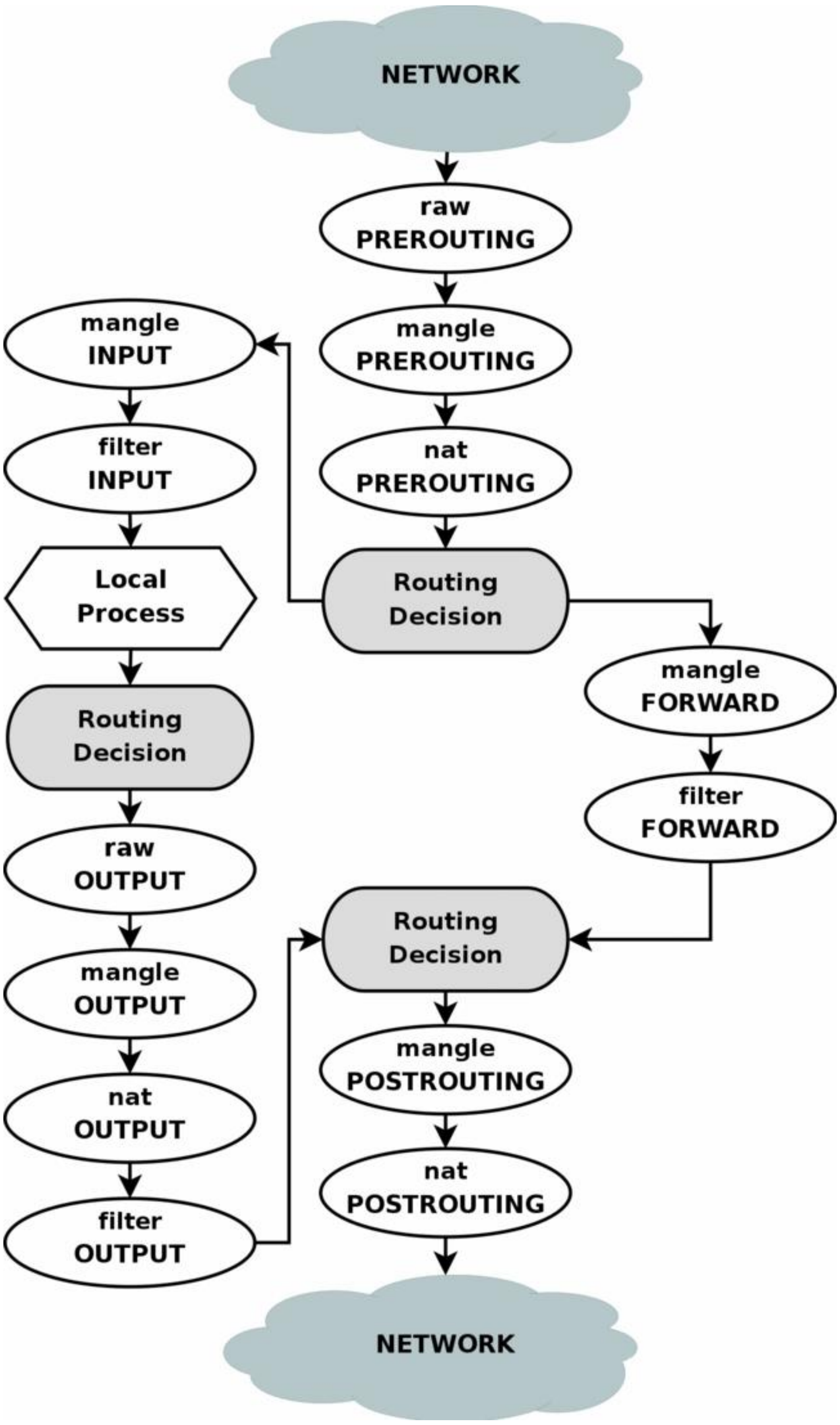


FIGURE 14. Iptables chain processing (6)

5.3.8 IPS software

There are two major Linux IPS services; Snort and Suricata (29). Suricata is the newer one and fully Open Source, whereas Snort is partially open source, but the newest and latest signatures are only for subscribers. The Suricata engine is a bit more modern compared to Snort, whereas Snort has a larger user base. There are no clear pros or cons to prefer one over the other. Just from the open source perspective, Suricata was chosen for this project.

Suricata is a multithreading application, meaning that it can utilize all threads of a processor to process packages. This improves the performance significantly. The system can be monitoring only, in IDS mode. In this mode, it will get all the alerts, but it cannot drop or reject traffic. The admin can though get the alerts, and go through the monitor logs. In an inline - or IPS - mode, it can be configured to affect the traffic flow. This is the more useful mode, as going through the alerts afterwards is a bit waste of time as the bad things have usually already happened. The system uses rules –files, which contain different signatures for different threats. These signatures have an action listed next to them. The action can be PASS, ALERT, REJECT or DROP. Pass lets the packet go through, alert makes an alert to the system while letting the packet pass, reject rejects the packet and generates a message for both the sender and receiver, drop drops the packet without further processing. Managing and tuning these rules can be a big workload, and it takes time to get the proper settings. Rule files are generated by different security communities. Popular ones are those of Snort and Emerging Threats. Custom rules can also be written.

Oinkmaster (12) is a program that can be used for managing the threat files. This is rather a pearl script that can be run periodically to download the latest threat files. It has an associated settings file, which can be used for configuring the rules files. It enables a very granular modification of particular rules. Also it allows changing the actions on entire rule sets. All rules in a rule file defaults to an alert. If the device is situated inline, some of the rules should be modified to drop traffic. Processing rules is processor intensive and time consuming, thus all rules should never be activated at the same time.

6 BUILDING AND CONFIGURING THE SYSTEM

6.1 Topology

The lab objective was to build a system shown in figure 15.

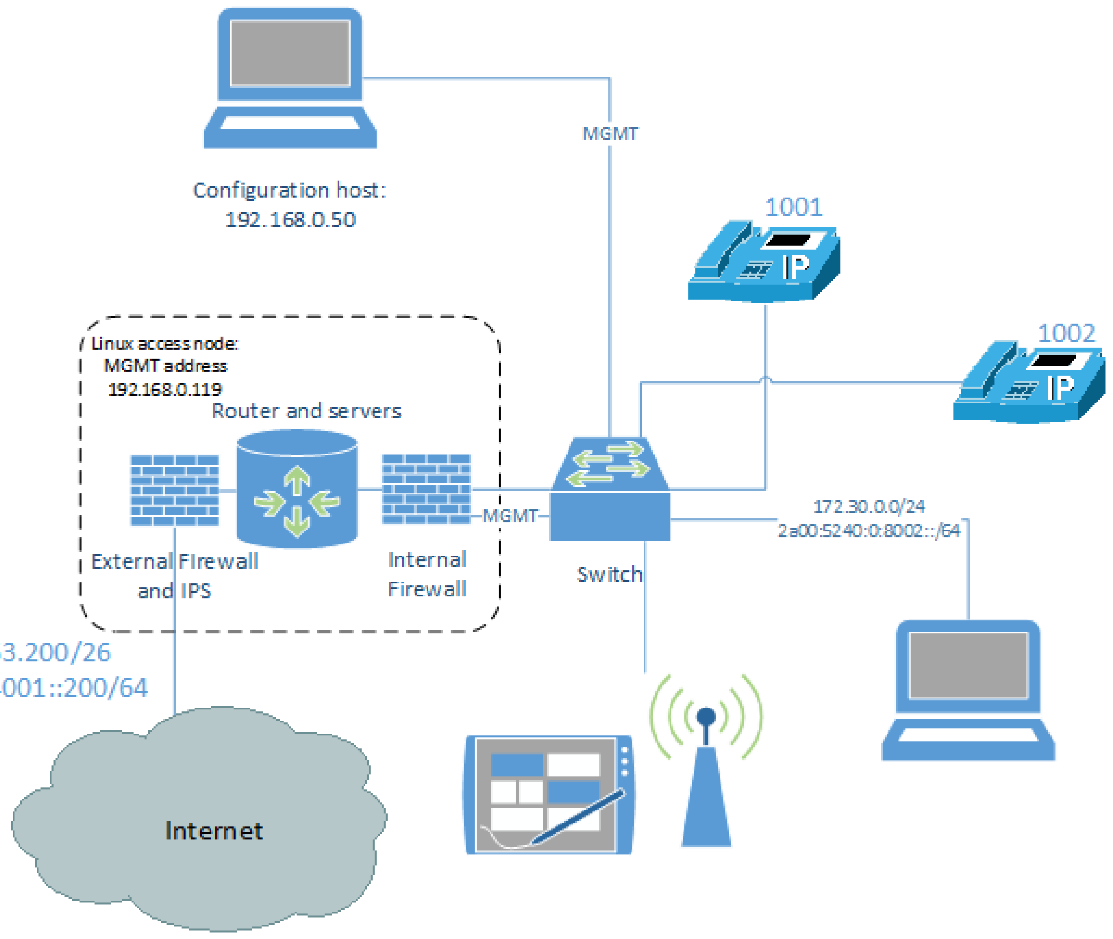


FIGURE 15. Lab network setup

The router device will be connected straight to the Internet with static, public addresses. It is meant to perform IPv4 PAT for the internal devices. 217.116.163.200 will be the NAT address for internal services, and 217.116.163.201 for the guest network. This will have several positive effects: QoS filtering is easier, and if the non-guest users need to access some Internet service, the address can be used as a filter at the other end.

The inside connections could be done from separate interfaces, as there are plenty of them on the router. However, it was decided to use VLAN tagging towards the switch, just to have it demonstrated. In production use an evaluation of the bandwidth need vs.the switchport consumption should be done. In table 3 we have the networks and associated VLANs

TABLE 3. Interface addresses

Network	Name	VLAN
217.116.163.200/26 217.116.160.201/26 2a00:5240:0:4001::200/64	outside	N/A
172.30.0.0/24 2a00:5240:0:8002::ffff/64	inside	71
172.30.10.0/24	guest	72
172.30.20.0/24	voice	73

The wireless access point was bridged, meaning that it is only converting wireless traffic to wired traffic.

6.2 Building hardware

The hardware selected was a rack mount, 1U chassis. It has an industrial mini-ITX board and an SSD drive (see table 4). The motherboard is an old version, which is not on the market anymore. This is generally not recommended for a production setup but for the purpose of demonstration it is more than adequate. For a commercial version, a recent motherboard with enough processing power and memory would be selected.

TABLE 4. Linux router hardware

Hardware	Description	Pros and cons
Case	1U rackmount from plin-kusa	Selected due to 2 charasteristics; size and front layout. Power supply includ-ed.
Motherboard	Jetway NF99-FL	Lots of onboard connectors with an optional network daughter card. Fanless heat sink.
Memory	4GiB SO-DIMM 1333 MT/s	Enough for demonstration. For SIEM environment as much as possible, at least 8GiB.
Hard drive	Intel SSD 40G	Fast and no moving parts

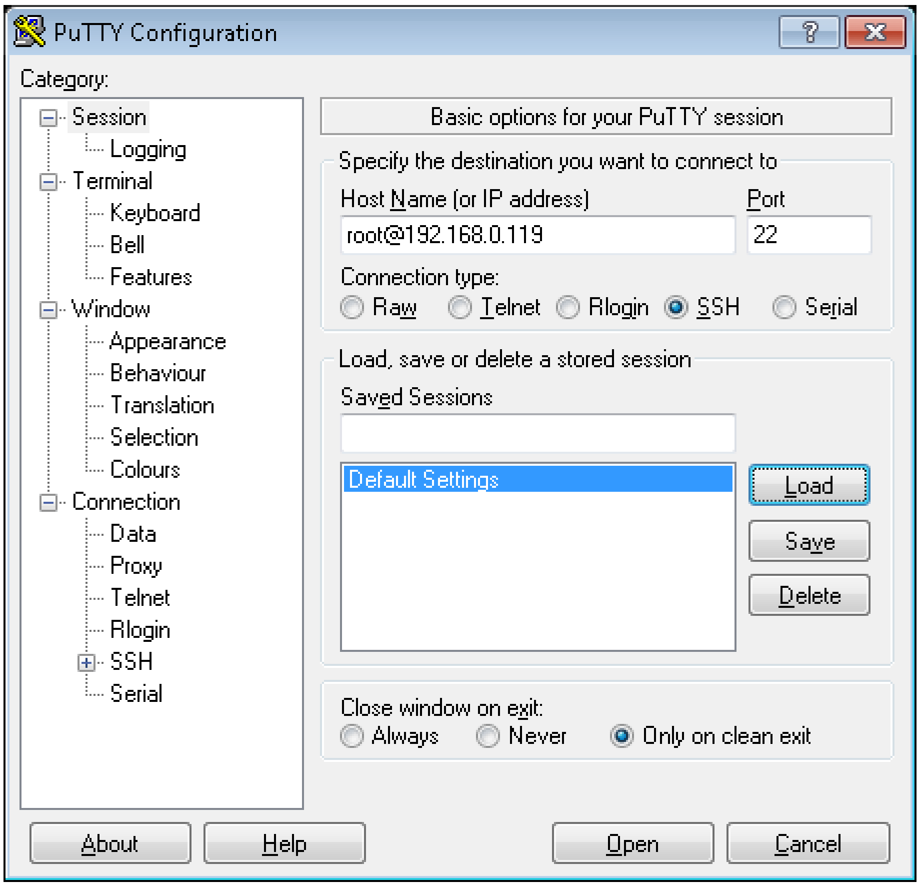
6.3 Installing software

6.3.1 Installing OS

Linux installation is very straight forward. In this case, a minimal version of CentOS 7 was downloaded from CentOS website, burnt on disk and the router hardware was booted from that disk. Just by following the prompts, the operating system was installed. The minimal version was selected to prevent unnecessary packages from being installed. This version has only the essential packages to run the OS and CLI, and any additional packages will be installed afterwards.

Once the OS had been installed, a network cable was installed on the management interface, and an SSH connection initiated to the machine. The root account was used for all configuration. The network interfaces default to dhcp client mode. Therefore, if there is a dhcp-server in the network where the device is connected, an automatic IP address will be given. In this case the assignment was 192.168.0.119.

For initiating an SSH console session, a terminal software was needed. Putty (1310) was used here. There are other popular ones as well, such as Tera Term. The screenshot of a connection initiation is presented in figure 16.



Specify the destination you want to connect to

Host Name (or IP address): root@192.168.0.119

Port: 22

Connection type: Raw Telnet Rlogin [✓ SSH] Serial

Load, save or delete a stored session

Saved Sessions

Default Settings

Close window on exit: Always Never [✓ Only on clean exit]

FIGURE 16. Putty connection settings

The first things first, software needed to be up-to-date, and also a few important packages had to be installed to have some essential network tools available. To address the issue, a packet manager was used. Red Hat packet manager is called RPM, and it can be used by several tools. The most popular one is yum. There are several parameters that can be given to the commands. A useful help for almost all Linux commands can be seen using the command ***man <command>***.

For basic network operations, these packages had to be installed:

Quagga for routing daemon control
nmap for network scanning
tcpdump for tracing network traffic
iftop for monitoring interface load
iperf for measuring network performance

dhcp for DHCP server
nano for text editing (vim is the default)
wget for downloading packages from the web
bind for DNS servicesy
iptables-services for firewall services
openvpn for VPN server
easy-rsa for VPN server certificates
openssl for encryption libraries
openswan for IPSec connections
net-tools for some basic network operations
ntp for network time
radvd for IPv6 address assignment

To install the necessary packages, first a system update is done with ***yum update.*** Then the packages are installed with same command: ***yum install tcpdump dhcp quagga iftop nmap nano wget bind-chroot iperf iptables-services openvpn openssl easy-rsa openswan net-tools ntp radvd***

All software packages are not necessarily found in the default repositories. In this case the iperf package is not found. Therefore an extra yum software repository had to be added into the system, called EPEL (Extra Packages for Enterprise Linux). The installation script can be downloaded online from the fedoraproject.org web site. Extra caution should be used when adding additional repositories, as the installation package provider should be a trusted one. Fedoraproject is a legitimate organization closely related to Red Hat, therefore it is rather trustworthy.
The repository is taken into use by the command ***rpm –i epel-release-latest-7.noarch.rpm***

Then, yum is run again with the missing packages. This time they should install fine. After this, the basic software will be up-to-date.

6.3.2 Configuring basic network settings

The basic settings on the device were very basic. The very first thing to do, as the device is Internet capable, was to enable the firewall. There are two flavours of firewall in CentOS 7, firewalld and iptables. Both use essentially the same kernel command (iptables), but the configuration is a bit different. The traditional iptables was used for our firewall. To activate this, the following commands were given:

systemctl stop firewalld
systemctl disable firewalld
systemctl enable iptables
systemctll start iptables

Several files had to be configured for a proper routing functionality.
Files in the folder /etc/sysctl.d/ have important system settings. The most important ones related to routing functions are forwarding flags. This setting controls if the device can forward IP traffic or not. By default this is 0, i.e. no routing will happen. This had to be changed to 1. To do this, entries were added to /etc/sysctl.d/99-sysctl.conf:

```
net.ipv4.ip_forward = 1
net.ipv6.conf.all.forwarding = 1
```

In the same file, it was informed that some interfaces are not allowed to participate in IPv6. This is done by entries:

```
net.ipv6.conf.enp2s0.disable_ipv6 = 1
net.ipv6.conf.vlan72.disable_ipv6 = 1
net.ipv6.conf.vlan73.disable_ipv6 = 1
```

Settings were applied with the command
sysctl –system

As default, the network configurations are handled by a daemon called “network manager”. However, the network settings should be configured manually. Therefore, Network manager was disabled using command ***systemctl stop***

NetworkManager and then it was prevented from starting at the next boot by ***systemctl disable NetworkManager.***

Next the hostname of the device was set. This was set in CentOS 7 with host-namectl –command. Thus it was set to “Linux-router” by using the command:
hostnamectl set-hostname “Linux-router”
After that the new hostname was taken into use by typing ***systemctl restart systemd-hostnamed***

Network interfaces are by default configured to get an address by dhcp. All network interfaces that were recognized and present during the installation have a startup script in the folder /etc/sysconfig/network-scripts. Here the interfaces connecting to the Internet and to the switch port were edited.
The Internet connected configuration file would be like below. Here the public IP address and default route are set. The default route is given a metric of 100. Giving a big enough metric enables to afterwards add a default gateway without changing the settings of this interface. The most viable case where this could be needed is if there are multiple Internet connections.

LISTING 1: Interface settings for enp3s0 (Internet interface)

```
--------------------------------------------------------------------------------------------------------------------------

TYPE=Ethernet
BOOTPROTO=static
IPADDR=217.116.163.200
NETMASK=255.255.255.192
GATEWAY=217.116.163.254
IPADDR2=217.116.163.201
NETMASK2=255.255.255.192
DEFROUTE=yes
METRIC=100
IPV6INIT=yes
IPV6ADDR="2a00:5240:0:4001::200/64"
IPV6_DEFAULTGW="2a00:5240:0:4001::1"
NAME=internet
DEVICE=enp3s0
ONBOOT=yes
NOZEROCONF=yes
--------------------------------------------------------------------------------------------------------------------------
```

The management address has to be modified, so that the default route given by DHCP will be removed and above the Internet connection will be used instead.

At the same time a static address is given, so it will not change. It is wise to make a reservation for this address into the DHCP server serving this network.

LISTING 2: Interface settings for enp2s0 (management interface)

```
-----------------------------------------------------------------------------------------------------------------------

TYPE=Ethernet
BOOTPROTO=static
IPADDR=192.168.0.119
NETMASK=255.255.255.0
IPV6INIT=no
NAME=management
DEVICE=enp2s0
ONBOOT=yes
NOZEROCONF=yes
-----------------------------------------------------------------------------------------------------------------------
```

Finally, the internal networks are given some settings, to enable the tagged interface.

LISTING 3: Interface settings for enp5s4 and its' VLANs (LAN interfaces)

```
-----------------------------------------------------------------------------------------------------------------------

DEVICE=vlan71
VLAN=yes
VLAN_NAME_TYPE=VLAN_PLUS_VID_NO_PAD
PHYSDEV=enp5s4
IPADDR=172.30.0.1
NETMASK=255.255.255.0
ONBOOT=yes
IPV6INIT="yes"
IPV6ADDR="fe80::200/64"
IPV6ADDR_SECONDARIES="2a00:5240:0:8002:ffff/64"

DEVICE=vlan72
VLAN=yes
VLAN_NAME_TYPE=VLAN_PLUS_VID_NO_PAD
PHYSDEV=enp5s4
IPADDR=172.30.10.1
NETMASK=255.255.255.0
ONBOOT=yes

DEVICE=vlan73
VLAN=yes
VLAN_NAME_TYPE=VLAN_PLUS_VID_NO_PAD
PHYSDEV=enp5s4
IPADDR=172.30.20.1
NETMASK=255.255.255.0
ONBOOT=yes
```

```
DEVICE=enp5s4
TYPE=Ethernet
BOOTPROTO=none
IPV6INIT=no
NAME=enp5s4
UUID=e54066fe-aaf0-48ae-809a-c716c9be5e15
ONBOOT=yes
-------------------------------------------------------------------------------------------------------------------
```

Now the interfaces have been configured.

A very important thing in a network is having the correct time on all devices, including the router. Therefore, a daemon for tracking the time from time servers is needed. The main file for this service is /etc/ntp.conf. Here, the serving time servers are changed to something closer to Finland:

```
server 0.fi.pool.ntp.org iburst
server 1.fi.pool.ntp.org iburst
server 2.fi.pool.ntp.org iburst
server 3.fi.pool.ntp.org iburst
```

Also, the tight restrictions on who can query for the time are loosened. Here anything in our private network will be allowed to make a time query.

```
restrict 172.30.0.0 mask 255.255.0.0 nomodify notrap
```

Now NTP has been configured. Finally, the Quagga suite will be enabled by commands

systemctl enable zebra.service

systemctl start zebra.service

At this point this can be used for a router-like CLI environment. Later, if needed, dynamic routing protocols can be added to the system. The Quagga CLI can be accessed with the command ***vtysh***

For IPv6, the router advertisements need to be enabled. This is done by modifying the /etc/radvd.conf file

LISTING 4: Router advertisement daemon settings

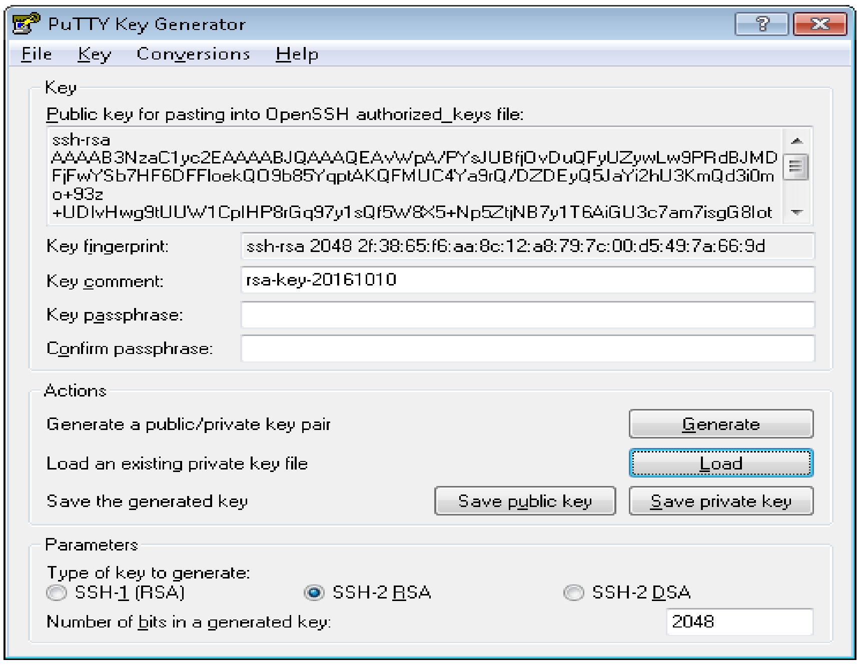
```
-------------------------------------------------------------------------------------------------------------------
interface vlan71

{
        AdvManagedFlag off;
        AdvSendAdvert on;
        MinRtrAdvInterval 3;
        MaxRtrAdvInterval 60;
        prefix  2a00:5240:0:8002::/64
        {
                AdvOnLink on;
                AdvAutonomous on;
                AdvRouterAddr off;
        };
        route ::/0 {
        };
        RDNSS fe80::200 {
        };

};
------------------------------------------------------------------------------------------------------------------------
```

After this there are necessary IP settings.

6.3.3 Securing router access

Routers – as any hardware – should be in locked spaces, accessible only to people maintaining them. All remote access should be limited. For this purpose, SSH-access will only be available from the management network. There are two options of securing root-user access. The first one is to allow only login using a personal account, and then the users have to elevate themselves to the root once inside. The other option is to login as the root using a client whose public key is known to the router. Here, the latter option is used. This is achieved by copying the SSH public key of the authorized client to the router, and then changing settings so that the key is required for logging in.

Depending on the terminal used, there are different options. For Putty, there is an application called Puttygen. It can be used for generating the necessary keys. The screen shot can be seen in figure 17. The generated private key is saved into a file, and the public key shown in the box is copied to file /root/.ssh/authorized_keys on the router box.


PuTTY Key Generator

Key

Public key for pasting into OpenSSH authorized_keys file:

```
ssh-rsa
AAAAB3NzaC1yc2EAAAABJQAAAQEAvWpA/PYsJUBfjOvDuQFyUZywLw9PRdBJMD
FjFwYSb7HF6DFFloekQO9b85YqptAKQFMUC4Ya9rQ/DZDEyQ5JaYi2hU3KmQd3i0m
o+93z
+UDIvHwg9tUUW1CplHP8rGq97y1sQf5W8X5+Np5ZtjNB7y1T6AiGU3c7am7isgG8lot
```

Key fingerprint: ssh-rsa 2048 2f:38:65:f6:aa:8c:12:a8:79:7c:00:d5:49:7a:66:9d

Key comment: rsa-key-20161010

Key passphrase:

Confirm passphrase:

Actions

Generate a public/private key pair

Load an existing private key file

Save the generated key

Parameters

Type of key to generate:
SSH-1 (RSA) [✓ SSH-2 RSA] SSH-2 DSA

Number of bits in a generated key: 2048

FIGURE 17. Puttygen key generator

After this, logging into the router with Putty is attempted. Following result should be seen in case of success:

Using username "root".
Authenticating with public key "rsa-key-20161010"
Last login: Mon Oct 10 10:34:53 2016 from 192.168.0.49

Next, the root login without a saved key is denied. This is done by modifying the following line into /etc/ssh/sshd_config:
PermitRootLogin without-password

This statement does two things: it prevents password authentication on the root's SSH sessions and allows only clients whose authorized key is known to login to the router. This makes it practically impossible to login to the router with brute force attacks. The downside is that if someone steals the computer with the authorized client, they have access to the router without any constraints. They have to be in the management network though, or have access to the VPN.

The setting affects after restarting the router or SSH service. This could be done with commands ***reboot*** or ***systemctl restart sshd.***

6.3.4 Configuring firewall

Configuring iptables is rather straight forward after the logic behind it is understood. There are the different tables(filter, nat and mangle) and within them the fixed chains INPUT, FORWARD, OUTPUT, PREROUTING, POSTROUTING. Some rules will be added on the filter table and two outgoing rules to the nat table. The final target is to have rules that allow inside and the visitor to talk outside, but not between each other. Also the voice VLAN needs to be isolated.

First a port address translation from LAN interfaces to outside will be done. This is achieved using the nat table’s POSTROUTING chain. Everything going out on the interface enp3s0 needs to be translated. First the visitor network is translated to 217.116.163.201, and the rest will be translated to 217.116.163.200. For connections initiated from outside, everything that is not return traffic for a connection initiated from the inside will be dropped. Therefore a drop rule to the appropriate position in the filter table’s INPUT chain will be added. The same applies to the FORWARD chain. Next, all traffic from visitor and inside network to Internet is allowed. All necessary traffic from LAN networks to the INPUT chain is allowed. That traffic would be DHCP, DNS, SIP and RTP for the voice VLAN, DHCP and DNS for the visitor VLAN as well as inside VLAN. The management network will have full access. Finally, traffic between the local interfaces will be denied.

For ipv6 there will be no control traffic, therefore only traffic allowed is for the inside interface to send traffic to the Internet. Also, the return traffic is allowed. IPv6 will be disabled on other interfaces.

Adding the Suricata and VoIP services will cause some additional tweaks to the firewall rules. The final commented rules can be found from the appendix 1.

6.3.5 Configuring DHCP

The DHCP server configuration file is /etc/dhcp/dhcpd.conf. The beginning of the file has some default settings, which can be overridden by the subnet statements. This device will be set authoritative on these networks. The gateway and DNS will be in this router. There is an option to assign a domain name, too. This has the advantage that short names can be used for reaching hosts on the same domain. The domain example.local will be set as the domain name for inside and voip. Visitor addresses will not have the domain name set.

IP phones need TFTP server settings for provisioning info.

DNS will be configured in next chapter, but to update the local DNS files dynamically, some settings related to it need to be put into the files.

The DHCP daemon configuration will now look like this:

LISTING 5: DHCP daemon config

```
#This directive makes the server superior to an non-authoritative server
authoritative;
#This statement sets the syslog facility for logging
log-facility local7;
#Lease times. Can be overruled in subnet statements
default-lease-time 21600;
max-lease-time 86400;
#Checks the availability of the address with a ping on top of it's database
ping-check true;
#Dynamic DNS updates from client are ignored
ignore client-updates;
#Dynamic update key
include "/var/named/chroot/etc/rndc.key";
#If a device has a reservation, it will still be updated into DNS
update-static-leases on;
#Update style. Check man dhcpd.conf for more details on it.
ddns-update-style interim;
#Updates will be done.
ddns-updates on;
#Domain zone to update
ddns-domainname "example.local";
#Reverse zone to update
ddns-rev-domainname "in-addr.arpa";

#here we define the zones and where to send updates to
zone example.local {
        primary 127.0.0.1;
        key rndc-key;
}
```

```
zone 0.30.172.in-addr.arpa {
        primary 127.0.0.1;
        key rndc-key;
}

zone 20.30.172.in-addr.arpa {
        primary 127.0.0.1;
        key rndc-key;
}

#Here are the actual DHCP settings. Network, mask, gateway and other options
#Note that DHCP will only be run on interfaces matching the subnet statments
subnet 172.30.0.0 netmask 255.255.255.0 {
        range 172.30.0.11 172.30.0.253;
        option subnet-mask 255.255.255.0;
        option routers 172.30.0.1;
        option domain-name "example.local";
        option domain-name-servers 172.30.0.1;
        option ntp-servers 172.30.0.1;
}

subnet 172.30.10.0 netmask 255.255.255.0 {
        range 172.30.10.11 172.30.10.253;
        option subnet-mask 255.255.255.0;
        option routers 172.30.10.1;
        option domain-name-servers 172.30.10.1;
        option ntp-servers 172.30.10.1;
}

subnet 172.30.20.0 netmask 255.255.255.0 {
        range 172.30.20.11 172.30.20.253;
        option subnet-mask 255.255.255.0;
        option routers 172.30.20.1;
        option tftp-server-name "172.30.20.1";
        option domain-name "example.local";
        option domain-name-servers 172.30.20.1;
        option ntp-servers 172.30.20.1;
}
----------------------------------------------------------------------------------------------------------------------------
```

After this the daemon needs to be started and also it needs to be run at startup. It must be noted that startup might fail before configuring the DNS portion.

```
systemctl enable dhcpd
systemctl start dhcpd
```

6.3.6 Configuring DNS

Intranet forward and reverse DNS zones will be created for example.local. On top of this, an authoritative record for example.com will exist. Also, the server will take care of iterative DNS requests towards other servers. Visitor devices will not have access to intranet zones. The Bind server will run in a chroot-jail, meaning that even if compromised, it will not have access to other parts of the system. Initial settings files will be generated with a script. The setup will be started with the command ***/usr/libexec/setup-named-chroot.sh /var/named/chroot on***

The configuration files will be in /var/named/chroot/etc and zones in /var/named/chroot/var/named. A baseline for settings was taken from Oracle (7), which were then modified to suite our example network using hints from Geekdudes website (8). The main configuration file, named.conf, can be found from the etc –folder. The most important statements here are the zone definitions, and that the recursion is limited to local networks. Open recursion will likely cause the router to be an instrument for a DDoS attack. Views will be defined, i.e. only the inside and voip networks have access to example.local record, and everyone else have only access to the global one. To enable dynamic updates of intranet DNS records, the rndc.key file generated by the system will be utilized, the same as used for DHCP.

Finally zone files for all zones will be configured. Within these, the zone header and actual records are defined. The dynamic zone files are populated by the system as well, the static ones only manually.

Before the service can be started, some selinux settings have to be altered. Selinux is a security utility designed by NSA, which restricts the process access to files it is by default not needing. Here it is ensured that the named daemon can write into zone files, as dynamic DNS is used. The command needed is ***setsebool -P named_write_master_zones true***

This ensures that the named process can write into the zone files. –P is there to make the changes persistent over reboots.

All DNS files are found in the appendix 3.

6.3.7 Configuring VPN

A VPN server will be used to allow remote workers to connect into the system. For this purpose the OpenVPN –package is downloaded, along with OpenSSL and Easy-RSA.

The first thing to do is to create the PKI infrastructure. The easiest way is to create self-signed certificates. The /usr/share/easy-rsa/2.0 folder content is copied to the /etc/openvpn/PKI folder. Next the variables in the vars file are modified. To ease the certificate issuance, the location and company data can be filled into the template. Also, the certificate destination can be somewhere else, in this device /etc/openvpn/keys was used. When being happy with it, the following commands are given:

source ./vars	This will take the defined variables into use
./clean-all	This resets the system. It should be noted that this WILL erase all existing certificates.
./build-ca	This builds the certificate authority certificates for Openvpn
./build-dh	This builds the Diffie-Hellman key pair for the encryption key exchange
./build-key-server	This builds the VPN server certificates, which will be signed by the certificate authority. The server name is given as an argument.
./build-key	This is used to build client certificates. The username is given as an argument.

There are several ways of doing the authentication. In this example system both a client certificate and username/password authentication will be used. This gives some level of 2-factor authentication, as the client needs to have both the certificate and the username/password. If a user needs to be removed, the certificate can be revoked along with the username. The user can be given different settings based on their certificate name. For example it can be dictated what IP address is given to a certain user, and then allow them to access to a certain device using the firewall. One other commonly used use case is NAT; a certain user needs to access an Internet service from a certain public IP address. In those cases, NAT rules can be made based on the user's local IP address. Also routes, DNS settings etc. can be pushed to the client's device.

After the necessary user certificates have been added, usernames need to be added into the system. This example system uses the regular PAM-authentication, which is the same as is used for logging into the system. Users are added using the following commands:

```
useradd –M –s /sbin/nologin <username>
passwd <username>
```

The downside of this is that there is a local password, which is hard to change. A better option would be to use an LDAP server, such as active directory. This can be utilized by OpenVPN, but this is not handled here. After this, the server settings file will be configured. A good reference to options in the file is in the OpenVPN manual (9).

LISTING 6: OpenVPN server settings

```
port 1194
proto udp
dev tun0

local 217.116.163.200

ca /etc/openvpn/keys/ca.crt
cert /etc/openvpn/keys/vpn.example.com.crt
key /etc/openvpn/keys/vpn.example.com.key
dh /etc/openvpn/keys/dh1024.pem
tls-server
tls-auth /etc/openvpn/keys/ta-example.key 0

crl-verify /etc/openvpn/keys/crl.pem

server 172.30.30.0 255.255.255.0
topology subnet

client-config-dir /etc/openvpn/clients

#Verkkoavaruudet, mihin asiakkaan pitaa paasta kiinni
push "route 172.30.0.0 255.255.255.0"
push "dhcp-option DNS 172.30.30.1"
push "dhcp-option DOMAIN example.com"

cipher BF-CBC
#comp-lzo

user nobody
group nobody

persist-key
persist-tun
keepalive 10 60

status openvpn-status.log
daemon
verb 4

plugin /usr/lib64/openvpn/plugins/openvpn-plugin-auth-pam.so login
------------------------------------------------------------------------------------------------------------------------
```

First it is told which IP to bind to, and in this case it is the IP of the outside interface. By default Openvpn works on the UDP port 1194, and it is left as default. It is explicitly defined which tun interface is to be used. This can then be used for firewall rules and client settings. After defining certificates and CRL, the network for which the server provides addresses us defined. In this case it will be 172.30.30.0/24. The topology is defined as subnet, this allows for a more efficient address usage. Routes and necessary DHCP options that will be pushed to the client are defined. Service is started as user nobody, so hijackers will not get root access. Keepalives are sent over the connections, and the tun interface is made persistent (always up). The server is started in daemon mode, and the PAM plugin is used for for authentication.

For iptables the following rule is needed:
iptables -A FORWARD -i tun0 -o vlan71 -m comment --comment "VPN traffic to home network" -j ACCEPT

The server is enabled with command ***systemctl enable openvpn@servername***
The server is then started with command ***systemctl start openvpn@servername***
The servername is the name of the server configuration file.

When the server side is up, the client application needs to be installed on the mobile worker's PC or laptop. There is an OpenVPN client software available at https://openvpn.net/. This software package is installed on the client, and the public ca-cert file, public and private client cert files and the configuration file are moved into the folder C:\Program Files\OpenVPN\config

LISTING 7: OpenVPN client settings

```
------------------------------------------------------------------------------------------------------------------------

client
dev tun
proto udp
remote 217.116.163.200 1194
resolv-retry infinite
nobind

cipher bf-cbc
ca ca.crt
cert user.crt
key user.key
tls-client
tls-auth ta-example.key 1
tls-timeout 5
ns-cert-type server
verb 2
auth-user-pass
auth-nocache
------------------------------------------------------------------------------------------------------------------------
```

Client uses corresponding settings with the server file. First, the client statement tells this is the client configuration. The connection is made to 217.116.163.200 udp port 1194, and in case of failure, connection is retried until the end of time. Cipher needs to be the same in both the client and the server, default Blowfish encryption is used. The Auth-nocache option tells that credentials should not be stored.

With these settings in place, a VPN-tunnel can be established.

6.3.8 Configuring VoIP

The asterisk system is the one of the most popular open source VoIP systems available. This system can be used for any kind of VoIP purposes. A small system, that can make internal calls using two voip phones, will be set up. It is rather easy to add VoIP capabilities outside, too, using Skype or something similar. PRI or FXO cards can be added to have the POTS line connected to the system. In this example two SIP subscribers are configured, which can call themselves internally. External calls are going through an IAX trunk towards Tele-entre OY’s PBX.

The installation of Asterisk on Centos7 is rather complex, as there is no repository for Asterisk packages for this distro. So, it needs to be compiled from scratch according to the instructions from Asterisk (10). The version 11 will be used, as it is more than enough for this demo, and it is a long term support version.

First the source code is downloaded from Asterisk website.
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-11-current.tar.gz

Then the tar-ball is extracted
tar xzf asterisk-11-current.tar.gz

The extracted source directory is entered with cd –command and start the configuration part. It should be noted that live systems with POTS components, such as FXO/FXS/PRI ports need to have the appropriate drivers installed. The DAHDI package is for POTS channels in general and LIBPRI is additional if digital lines are used. Now, the prerequisites existence is checked:
contrib/scripts/install_prereq test
This command tests if all prerequisites are installed. If not, it gives a command string for yum to install. After those have been sorted out, the initial configuration can be made.
./configure –prefix=/usr/local --libdir=/usr/local/lib64 --sysconfdir=/etc

Next, options will be selected for our Asterisk build. Asterisk has a menuselect configuration utility, where different options are selected for the installation. This is invoked by ***make menuselect***. This gives a lot of options, which can be checked from the Asterisk website if needed. Other languages, database support and mp3 support can be selected among others. For this demo the defaults were just enough.

Next, the compilation process is started. This happens with the command ***make***. Compiling takes some time, thus it is wise to perform it using the console or using the ***screen*** command. This ensures that if the session is terminated for any reason, the build does not stop. After this has been done successfully, the compiled software can be installed with the command ***make install.***

Now the executables exist. After this, template files can be generated by issuing command ***make samples***. In some occasions these are not wanted, as they replace existing files. Also, if there are some previous settings files, which are just being restored, this is an unnecessary step. Initialization scripts are created with the command ***make config*** . This enables starting and termination of the application using regular distro commands. Last but not least is ***make install-logrotate***. This one creates a logrotate configuration. Logrotate is a Linux feature, which archives logs at a certain interval, or at a certain size. This is very useful if there are a lot of calls and/or small disk space, preventing the files from not growing too big.

Now the application has been installed. Next the software must be configured. Good examples are found online (14). All configuration files are in the /etc/asterisk –folder. Configuration is started by defining the SIP settings. There are some general settings, such as what address to bind to, and then the peer configurations. These settings are in the sip.conf –file.

LISTING 8: SIP settings for Asterisk

```
------------------------------------------------------------------------------------------------------------------------

[general]
bindaddr=172.30.20.1
permit = 172.30.20.0/255.255.255.0


[1001]
username = 1001
callerid = "User1"<1001>
secret = abc123
type = friend
host = dynamic
context = from-internal
qualify = yes

[1002]
username = 1002
callerid = "User2"<1002>
secret = abc123
type = friend
host = dynamic
context = from-internal
```

```
qualify = yes
```

Here, all SIP messages are bound to IP address 172.30.20.1. This means that the SIP registrar will not listen to other local addresses. Then there is the end user registration data, which includes e.g. username and callerid.. The context dictates into which context the call is put in when received (see dialplan below). The type can be “peer”, “user” or “friend”. A peer is a device that conveys calls, such as another Asterisk system. A user is an end user device, which registers with asterisk. A friend can be of both types. The host can be either a static host address, or it can be set as “dynamic” like here. If it is dynamic, the host is allowed to register itself with any IP address.

The settings for the IAX protocol are configured next, as a connection to the upstream PBX is needed. The file for these settings is iax.conf.

Listing 9: IAX settings for Asterisk

```
[general]
port = 4569
bindaddr=217.116.163.200

register => thesislab:topsecret123@172.16.1.152

[teleentre]
type=friend
disallow=all
allow=alaw
username=teleentre
secret=topsecret123
context=from-external
host=dynamic
qualify=yes
```

The options are a lot similar to those in the sip.conf file. The biggest difference is the register string, which is used to register with the peer on the other side. Also there is the qualify –statement, which makes the Asterisk system to track the peer. The Asterisk sends keepalives to the other side, and if it does not hear back, it considers the system unreachable. This has mostly relevance if systems are behind NAT.

Last, the rules that make everything tick are introduced, i.e. the dialplan. This file has the calling logic, explaining how different extensions and contexts are handled. File for this is extensions.conf.

LISTING 10: Dialplan settings for Asterisk

```
--------------------------------------------------------------------------------------------------------------------

[from-internal]
exten => _100[12],1,Dial(SIP/${EXTEN},45,t)
same => n,Hangup()
exten => _0.,1,Dial(IAX2/thesislab@teleentre/${EXTEN:1})
same => n,Hangup()

[from-external]
exten => _100[12],1,Dial(SIP/${EXTEN},45,t)
exten => n,Hangup
--------------------------------------------------------------------------------------------------------------------
```

A context is defined here in the brackets. In each context, there are dialplan rules. Here, the rules try to match the dialed extension, and if it matches, the actions are executed. An underscore in the beginning means that there is a pattern match, without the underscore there would be an explicit match. What is missing here is the configuration from the other side of the IAX trunk. However, the system there is GUI-based, so the trunk information cannot be extracted. The logic is that anything internal stays within this system, and all external calls go through the upstream PBX to the PSTN network. Likewise, if someone calls certain numbers from the PSTN, they will be directed to this system and the particular extensions.

When everything is in place, Asterisk can be enabled by command ***systemctl enable asterisk*** and ***systemctl start asterisk.*** As the package is not made for the new systemctl utility, it will redirect the necessary command to the old ***chkconfig*** and ***service*** –utilities.

6.3.9 Configuring Suricata

The purpose is to use Suricata as an IPS system, meaning that it can drop harmful traffic when seen. It will be configured to inspect the traffic coming in from the Internet.

Suricata was installed according to the guidelines given by the Open Information Security Foundation (3) and (4). The guide was followed step by step, starting from the dependencies to a full installation:

```
sudo yum -y install gcc libpcap-devel pcre-devel libyaml-devel file-devel \
  zlib-devel jansson-devel nss-devel libcap-ng-devel libnet-devel tar make \
  libnetfilter_queue-devel lua-devel

wget http://www.openinfosecfoundation.org/download/suricata-3.1.tar.gz

tar -xvzf suricata-3.1.tar.gz

cd suricata-3.1

./configure --prefix=/usr --sysconfdir=/etc --localstatedir=/var --enable-nfqueue --enable-
lua

make

make install-full

ldconfig
```

The configuration was made automatically by the make install-full –command, but the settings were tweaked just a bit:

```
HOME_NET: "[172.30.0.0/16,5240:0:8002::/64]"
```

Next, it is checked that everything works fine. The command suricata –c /etc/suricata/suricata.yaml –i enp3s0 --init-errors-fatal is given. If everything comes up fine, the packet statistics in file /var/log/suricata/stats.log should be growing. The next step is to put suricata inline. This is done by feeding the incoming packets from the Internet to the Suricata engine by the iptables/netfilter NFQUEUE –feature (5). With iptables, it is possible to select what traffic is inspected and what is not. The following settings will be added to iptables:

```
iptables –I FORWARD –i  enp3s0 –m state –state ESTABLISHED,RELATED
–j NFQUEUE
```

This statement will cause the replies for established Internet connections to be inspected. As it is in the forward-chain, only the established pass-through traffic will be inspected.

After that, Suricata can be started with command ***/usr/bin/suricata –c /etc/suricata/suricata.yaml –q 0***. To make the command boot persistent, an init-script is added into the /etc/init.d –folder, as shown in listing 11.

LISTING 11: Startup script for Suricata

```
#!/bin/sh
#
### BEGIN INIT INFO
# Provides:          suricata
# Required-Start:    $time $network $local_fs $remote_fs
# Required-Stop:     $remote_fs
# Default-Start:     2 3 4 5
# Default-Stop:      0 1 6
# Short-Description: Next Generation IDS/IPS
# Description:       Intrusion detection system that will
#                    capture traffic from the network cards and will
#                    match against a set of known attacks.
### END INIT INFO


# Source function library.
. /etc/rc.d/init.d/functions
. /etc/sysconfig/suricata

# We'll add up all the options above and use them
NAME=suricata
DAEMON=/usr/bin/$NAME



check_nfqueue() {
if [ ! -e /proc/net/netfilter/nfnetlink_queue ]; then
    logger -t suricata "NFQUEUE support not found !"
    logger -t "Please ensure the nfnetlink_queue module is loaded or built in kernel"
    exit 5
fi
}

case "$LISTENMODE" in
  nfqueue)
    IDMODE="IPS (nfqueue)"
    LISTEN_OPTIONS=" -q $NFQUEUE"
    check_nfqueue
    ;;
  pcap)
    IDMODE="IDS (pcap)"
    LISTEN_OPTIONS=" -i $IFACE"
    ;;
  af-packet)
    IDMODE="IDS (af-packet)"
    LISTEN_OPTIONS=" --af-packet"
    ;;
  *)
    echo "Unsupported listen mode $LISTENMODE, aborting"
    exit 1
    ;;
esac

SURICATA_OPTIONS=" -c $SURCONF --pidfile $PIDFILE $LISTEN_OPTIONS -D"

# See how we were called.
case "$1" in
  start)
       if [ -f $PIDFILE ]; then
           PID1=`cat $PIDFILE`
           if kill -0 "$PID1" 2>/dev/null; then
               echo "$NAME is already running with PID $PID1"
               exit 0
           fi
       fi
       echo -n "Starting suricata in $IDMODE mode..."
       $DAEMON $SURICATA_OPTIONS > /var/log/suricata/suricata-start.log  2>&1 &
       echo " done."
       ;;
  stop)
       echo -n "Stopping suricata: "
       if [ -f $PIDFILE ]; then
           PID2=`cat $PIDFILE`
       else
           echo " No PID file found; not running?"
           exit 0;
       fi
       killproc `basename $DAEMON`
       if [ -n "$PID2" ]; then
           kill "$PID2"
           ret=$?
           sleep 2
           if kill -0 "$PID2" 2>/dev/null; then
               ret=$?
               echo -n "Waiting . "
               cnt=0
               while kill -0 "$PID2" 2>/dev/null; do
                   ret=$?
                   cnt=`expr "$cnt" + 1`
                   if [ "$cnt" -gt 10 ]; then
                      kill -9 "$PID2"
                      break
                   fi
                   sleep 2
                   echo -n ". "
               done
           fi
       fi
       if [ -e $PIDFILE ]; then
           rm $PIDFILE > /dev/null 2>&1
       fi
       echo " done."
    ;;
  status)
       # Check if running...
       if [ -s $PIDFILE ]; then
```

```
        PID3=`cat $PIDFILE`
        if kill -0 "$PID3" 2>/dev/null; then
          echo "$NAME is running with PID $PID3"
          exit 0
        else
          echo "PID file $PIDFILE exists, but process not running!"
        fi
      else
        echo "$NAME not running!"
      fi
    ;;
  reload-rules)
      echo -n "Reloading rules: "
      if [ -f $PIDFILE ]; then
        PID4=`cat $PIDFILE`
      else
        echo " No PID file found; not running?"
        exit 1;
      fi
      kill -USR2 "$PID4"
      echo "Done."
    ;;
  restart)
      $0 stop
      $0 start
    ;;
  force-reload)
      $0 stop
      $0 start
    ;;
  *)
      echo "Usage: $0 {start|stop|reload-rules|restart|status}"
      exit 1
esac

exit 0
```

After putting the script in place, it is enabled at startup by first making it executable with ***chmod 755 suricata*** (suricata is the file name) and by issuing command ***systemctl enable suricata.***

Last, it is made sure that all suricata rules are updated on a scheduled basis. The rules can be updated either manually, or by a program called oinkmaster. On top of downloading the latest rules, Oinkmaster can modify the rulesets to whatever if preferred. In this case the Emerging Threats rulesets are donloaded. Then, some rulesets are modified to drop traffic. First, the Oinkmaster source is downloaded from the website (12). This website has as well good documentation about the configuration itself. After having the source, the oinkmaster pearl script is copied to /usr/local/bin –folder and the oinkmaster.conf –file is copied to /etc –folder. The oinkmaster.conf is modified will be modified in the following way:

```
#Emerging threats rules:
url = https://rules.emergingthreats.net/open/suricata-1.3/emerging.rules.tar.gz
modifysid * “^alert” | “drop”
```

The first rule defines the download path, the second one modifies all rules to drop traffic. Now, in a real production environment the rules need to be checked more carefully. If there are strict drop rules, it is really necessary to select only the most critical rulesets for the environment. The rulesets in use are selected in suricata.yaml –file. In this example environment, only rule files that have malware, viruses or other clearly harmful signatures in them are selected.

Finally a crontab rule is made to update the rulesets every day. This script is inserted in /etc/cron.d.

LISTING 12: Suricata rule update script

```
00 06 * * * root /usr/local/bin/oinkmaster.pl -o /etc/suricata/inlinerules 2>&1 | logger -t oinkmas-
ter
00 07 * * * root /etc/init.d/suricata reload-rules 2>&1 | logger -t oinkmaster
```

Here updating the rule set is started at 6:00 every morning. Then it is given some time to process, and the reload happens at 7:00am. The logger-command makes syslog entries into the /var/log/messages –file.

6.3.10 Configuring QoS settings

The quality of service is meant for ensuring bandwidth for essential services. It is most efficient when sending information, since it is the only direction where there is full control of the data. But it can be used to some extent as well on received information. The receiving direction is used in this case, meaning that data coming in from the Internet will be categorized. This is the main direction for traffic, because end user networks tend to be more hungry downstream than upstream. Policing the received data relies on the TCP behavior. The TCP throttling kicks in, when TCP packets are dropped. This throttling cuts the speed of that particular TCP session, thus, releasing the bandwidth for the other sessions..

The essential part is measuring the total bandwidth available. This can be done with iperf or a similar bandwidth tool. It is also possible to just start a download of some big file and check the bandwidth with iftop-application. The best moment to test this is when there is no other usage. Essential is that the total bandwidth configured in QoS is a little less than the actual bandwidth. If misconfigured, the QoS will never step in. The downstream Internet trafficwill be limited to 20Mbps in the test system.

The classification hierarchy is shown in the figure 18 below:

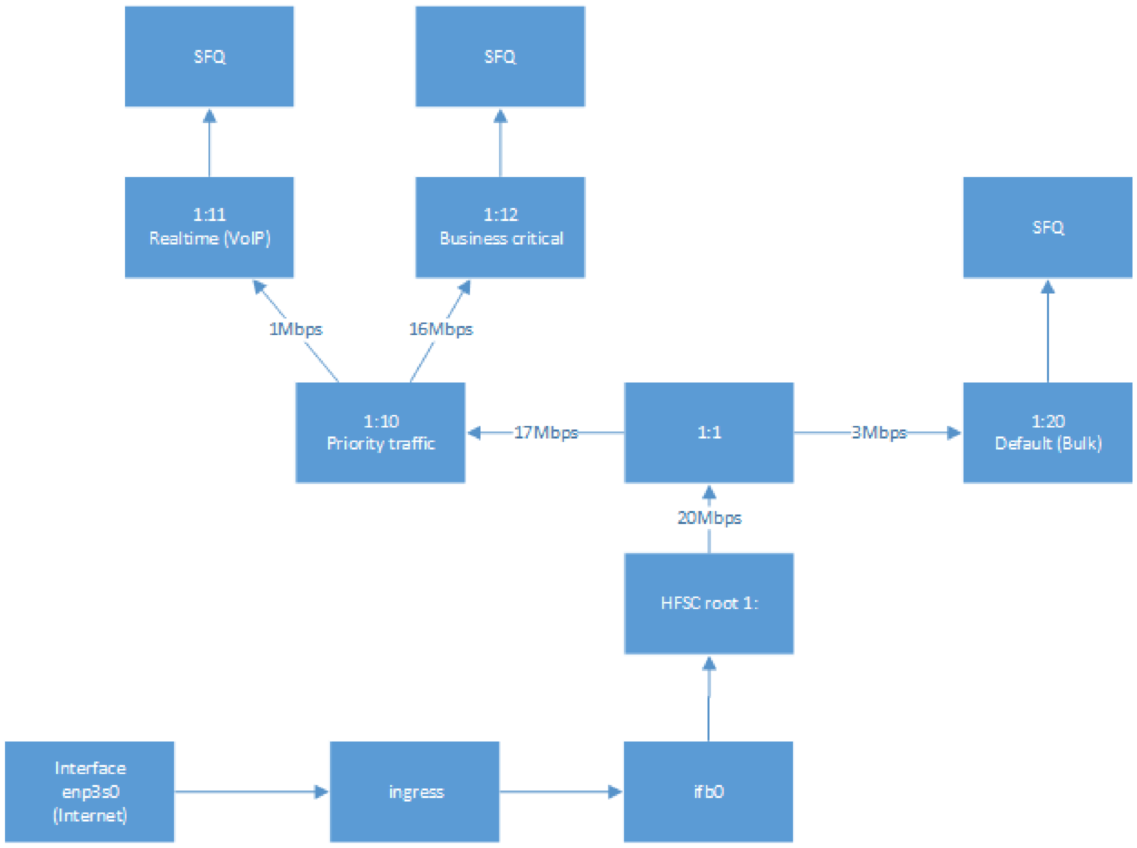


FIGURE 18. QoS incoming classes

All incoming traffic is put into an ingress software queue, using a linux module called IFB (18). This is a trick that is needed due to the fact that there is no managed incoming queue in Linux interfaces, thus, it is pretended that the incoming queue is an outgoing queue. On the root and leaf classes a HSFC scheduling algorithm is put (17). This is a queuing discipline, which allows to adjust both the bandwidth and delay for a class. Then classes are made for different traffic, and traffic is guided to different classes based on filter statements. At the end SFQ scheduling is used for giving all users a fair proportion of the bandwidth. Internal traffic is assigned 17Mbps, and guests can have the rest 3M. One thing to note is that the HSFC allows one single class to use all of the root bandwidth if higher priority classes do not have any traffic. There is a very extensive guide to traffic handling with Traffic Control (tc) available (11). Listing 13 has the applicable settings in a shell script format.

LISTING 13: QoS configuration script

```
--------------------------------------------------------------------------------------------------------------------------------
#!/bin/sh

#Loading kernel modules
modprobe sch_hfsc  #HFSC-alogrithm
modprobe cls_u32   #u32 classifier
modprobe sch_sfq   #u32 classifier
modprobe ifb       #ifb0 interface (incoming traffic software queue)

#Removing possibly existing QoS settings.
tc qdisc del dev enp3s0 root 2>/dev/null
tc qdisc del dev enp3s0 ingress 2>/dev/null
tc qdisc del dev ifb0 root 2>/dev/null

#Enable ifb0 interface
ip link set ifb0 up

#We make an ingress class into Internet interface
tc qdisc add dev enp3s0 handle ffff: ingress

#All incoming traffic from Internet is put into software interface
tc filter add dev enp3s0 parent ffff: protocol ip prio 2 u32 match u32 0 0 flowid 1:1 action mirred
egress redirect dev ifb0

#We define HFSC queueing discipline, default class 1:20
tc qdisc add dev ifb0 root handle 1: hfsc default 20

#Set limits for classes, both bw and delay can be used
tc class add dev ifb0 parent 1: classid 1:1 hfsc sc rate 20Mbit ul rate 20Mbit
tc class add dev ifb0 parent 1:1 classid 1:10 hfsc sc rate 17Mbit ul rate 20Mbit
tc class add dev ifb0 parent 1:10 classid 1:11 hfsc sc umax 1500b dmax 53ms rate 1Mbit ul rate
20Mbit
tc class add dev ifb0 parent 1:10 classid 1:12 hfsc sc umax 1500b dmax 200ms rate 16Mbit ul
rate 20Mbit
```

```
tc class add dev ifb0 parent 1:1 classid 1:20 hfsc sc rate 3Mbit ul rate 20Mbit

#Add SFQ discipline
tc qdisc add dev ifb0 parent 1:11 handle 11: sfq perturb 10
tc qdisc add dev ifb0 parent 1:12 handle 12: sfq perturb 10
tc qdisc add dev ifb0 parent 1:20 handle 20: sfq perturb 10

#Assign traffic to classes based on IP
tc filter add dev ifb0 parent 1:0 protocol ip prio 1 u32 match ip dst 217.116.163.200 flowid 1:10
tc filter add dev ifb0 parent 1:0 protocol ip prio 4 u32 match ip src 172.16.1.152/32 flowid 1:10

#Within class 1:10, we will move DSCP EF or CS5-marked packets to flowid 1:11
tc filter add dev ifb0 parent 1:10 protocol ip prio 10 u32 match ip tos 0xb8 0xfc flowid 1:11
tc filter add dev ifb0 parent 1:10 protocol ip prio 11 u32 match ip tos 0xa0 0xfc flowid 1:11
#Other internal traffic to 1:12
tc filter add dev ifb0 parent 1:10 protocol ip prio 12 u32 match ip dst 0.0.0.0/0 flowid 1:12

#Guest traffic will go to flow 1:20, as it is default
--------------------------------------------------------------------------------------------------------------------------
```

6.4 Testing the system

Now when our system is complete, a test plan is needed to ensure that the system works as designed. First the basic network operations are tested: internal networks should get an ip address, but they should not be able to communicate with each other. The inside network should get both IPv4 and IPv6 addresses. The inside and voice networks' hosts should automatically be added into the DNS system. And all traffic from the Internet to hosts should be inspected for malware.

Router hardening must be tested, i.e. the root access is only available from certain hosts and only from management network.

VoIP system needs to be tested so that it can create calls between hosts, and to public land lines. Calls from a landline should alert the targeted VoIP phone.

The next step is remote access testing. The user certificate must exist to get a connection. Revoked certificates should not work. Management access should be possible using the VPN client, as well as access to the LAN networks.

Last but not least is the QoS testing. It is necessary to test that the classifying works properly, and that the limitation really kicks in.

The test matrix can be seen in appendix 4.

7 CONCLUSIONS

As witnessed, a Linux router is very versatile in different operations; there are very few tasks that it cannot perform at least at some level. It was made to perform routing, firewalling, security screening and additionally a VPN server and a VoIP server were installed in it. These are just example services seen in these kinds of devices; alternatively an email server, network monitoring service or a webserver could have been installed. Pretty much any service can be installed on a Linux access node, and only thing needed for it is the necessary hardware resources and capable people to maintain it.

Also, it was seen that implementing different services needs a big amount of configuration. The configuration itself is sometimes tricky, and there are several possibilities for errors and incorrect settings. Along with the possibility to install subpar hardware, these facts make the use of Linux devices as routers very controversial. Purpose-built devices are faster in their routing efficiency, and those can easily be backed up and restored. It is easy to find documentation and people understanding their configurations. In case there are problems with these devices, the vendors’ technical assistance is there to help. However, purpose built devices are not even nearly as flexible as Open Source Linux - based devices. As such, a Linux device will not be very suitable for extremely large organizations; those need something that can be outsourced and easily documented and maintained. Additionally, the cost of equipment is usually not a big concern for bigger companies. However for private persons with adequate knowledge and small and even midsized companies this is a viable solution. If IT is maintained internally by the company and there are some spare hours to spend for configuration and support, adding this kind of a device into the infrastructure will reduce costs and give flexibility to add customized services into the network.

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APPENDICES

/etc/sysconfig/iptables -file

```
-----------------------------------------------------------------------------------------------------------------------
*filter
:INPUT ACCEPT [0:0]
:FORWARD ACCEPT [0:0]
:OUTPUT ACCEPT [29611:3936789]
:VOICE - [0:0]
-A INPUT -i lo -j ACCEPT
-A INPUT -m state --state RELATED,ESTABLISHED -m comment --comment "Allow return
traffic" -j ACCEPT
-A INPUT -p icmp -m icmp --icmp-type 8 -m comment --comment "Allow ping" -j ACCEPT
-A INPUT -s 172.16.1.152/32 -i enp3s0 -p udp -m udp --dport 4569 -m comment --comment IAX
-j ACCEPT
-A INPUT -i enp3s0 -p udp -m udp --dport 53 -m comment --comment DNS -j ACCEPT
-A INPUT -i enp3s0 -p udp -m udp --dport 1194 -m comment --comment OpenVPN -j ACCEPT
-A INPUT -i enp3s0 -m comment --comment "Drop Internet traffic if not return traffic or ping" -j
DROP
-A INPUT -i tun0 -j ACCEPT
-A INPUT -i enp2s0 -m comment --comment "Network management unrestricted" -j ACCEPT
-A INPUT -i vlan71 -p udp -m multiport --dports 53,67,123 -m comment --comment "Allow NTP,
DNS and DHCP for inside" -j ACCEPT
-A INPUT -i vlan72 -p udp -m multiport --dports 53,67,123 -m comment --comment "Allow NTP,
DNS and DHCP for visitors" -j ACCEPT
-A INPUT -i vlan73 -j VOICE
-A INPUT -j REJECT --reject-with icmp-host-prohibited
-A FORWARD -i enp3s0 -m state --state RELATED,ESTABLISHED -m comment --comment
"Inspect established connections from Internet" -j NFQUEUE --queue-num 0
-A FORWARD ! -i enp3s0 -m state --state RELATED,ESTABLISHED -m comment --comment
"Allow established and related" -j ACCEPT
-A FORWARD -s 217.116.163.196/32 -m comment --comment "Test access" -j ACCEPT
-A FORWARD -o enp3s0 -m comment --comment "Internet access accepted" -j ACCEPT
-A FORWARD -i enp2s0 -m comment --comment "Management unrestricted" -j ACCEPT
-A FORWARD -i tun0 -o vlan71 -m comment --comment "VPN traffic to home network" -j AC-
CEPT
-A FORWARD -m comment --comment "reject other traffic" -j REJECT --reject-with icmp-host-
prohibited
-A VOICE -p udp -m multiport --dports 53,67,69,123,5060,10000:20000 -j ACCEPT
COMMIT
# Completed on Wed Oct 19 16:04:10 2016
# Generated by iptables-save v1.4.21 on Wed Oct 19 16:04:10 2016
*nat
:PREROUTING ACCEPT [14280:2325286]
:INPUT ACCEPT [6697:1453667]
:OUTPUT ACCEPT [720:60134]
:POSTROUTING ACCEPT [21:4331]
-A POSTROUTING -s 172.30.10.0/24 -o enp3s0 -j SNAT --to-source 217.116.163.201
-A POSTROUTING -o enp3s0 -j SNAT --to-source 217.116.163.200
COMMIT
-----------------------------------------------------------------------------------------------------------------------
```

/etc/suricata/suricata.yaml -file

```
---------------------------------------------------------------------------------------------------------------------------
%YAML 1.1
---

# Suricata configuration file. In addition to the comments describing all
# options in this file, full documentation can be found at:
# https://redmine.openinfosecfoundation.org/projects/suricata/wiki/Suricatayaml

##
## Step 1: inform Suricata about your network
##

vars:
  # more specifc is better for alert accuracy and performance
  address-groups:
    #HOME_NET: "[192.168.0.0/16,10.0.0.0/8,172.16.0.0/12]"
    #HOME_NET: "[192.168.0.0/16]"
    #HOME_NET: "[10.0.0.0/8]"
    HOME_NET: "[172.30.0.0/16,2a00:5240:0:8002::/64]"
    #HOME_NET: "any"

    EXTERNAL_NET: "!$HOME_NET"
    #EXTERNAL_NET: "any"

    HTTP_SERVERS: "$HOME_NET"
    SMTP_SERVERS: "$HOME_NET"
    SQL_SERVERS: "$HOME_NET"
    DNS_SERVERS: "$HOME_NET"
    TELNET_SERVERS: "$HOME_NET"
    AIM_SERVERS: "$EXTERNAL_NET"
    DNP3_SERVER: "$HOME_NET"
    DNP3_CLIENT: "$HOME_NET"
    MODBUS_CLIENT: "$HOME_NET"
    MODBUS_SERVER: "$HOME_NET"
    ENIP_CLIENT: "$HOME_NET"
    ENIP_SERVER: "$HOME_NET"

  port-groups:
    HTTP_PORTS: "80"
    SHELLCODE_PORTS: "!80"
    ORACLE_PORTS: 1521
    SSH_PORTS: 22
    DNP3_PORTS: 20000
    MODBUS_PORTS: 502


##
## Step 2: select the rules to enable or disable
##

default-rule-path: /etc/suricata/inlinerules
rule-files:
 - botcc.rules
# - ciarmy.rules
 - compromised.rules
 - drop.rules
 - dshield.rules
# - emerging-activex.rules
# - emerging-attack_response.rules
# - emerging-chat.rules
# - emerging-current_events.rules
# - emerging-dns.rules
 - emerging-dos.rules
 - emerging-exploit.rules
# - emerging-ftp.rules
# - emerging-games.rules
# - emerging-icmp_info.rules
# - emerging-icmp.rules
# - emerging-imap.rules
# - emerging-inappropriate.rules
 - emerging-malware.rules
# - emerging-misc.rules
# - emerging-mobile_malware.rules
# - emerging-netbios.rules
# - emerging-p2p.rules
# - emerging-policy.rules
# - emerging-pop3.rules
# - emerging-rpc.rules
# - emerging-scada.rules
# - emerging-scan.rules
# - emerging-shellcode.rules
# - emerging-smtp.rules
# - emerging-snmp.rules
# - emerging-sql.rules
# - emerging-telnet.rules
# - emerging-tftp.rules
 - emerging-trojan.rules
# - emerging-user_agents.rules
# - emerging-voip.rules
# - emerging-web_client.rules
# - emerging-web_server.rules
# - emerging-web_specific_apps.rules
 - emerging-worm.rules
 - tor.rules
# - decoder-events.rules # available in suricata sources under rules dir
# - stream-events.rules  # available in suricata sources under rules dir
# - http-events.rules    # available in suricata sources under rules dir
# - smtp-events.rules    # available in suricata sources under rules dir
# - dns-events.rules     # available in suricata sources under rules dir
# - tls-events.rules     # available in suricata sources under rules dir
# - modbus-events.rules  # available in suricata sources under rules dir
# - app-layer-events.rules  # available in suricata sources under rules dir

classification-file: /etc/suricata/classification.config
reference-config-file: /etc/suricata/reference.config
# threshold-file: /etc/suricata/threshold.config
```

<rest of the file omitted>

named.conf –file:

```
-----------------------------------------------------------------------------------------------------------------------
//
// named.conf
//
// Provided by Red Hat bind package to configure the ISC BIND named(8) DNS
// server as a caching only nameserver (as a localhost DNS resolver only).
//
// See /usr/share/doc/bind*/sample/ for example named configuration files.
//

options {
        listen-on port 53 { any; };
        listen-on-v6 port 53 { any; };
        directory       "/var/named";
        dump-file       "/var/named/data/cache_dump.db";
        statistics-file "/var/named/data/named_stats.txt";
        memstatistics-file "/var/named/data/named_mem_stats.txt";
        max-journal-size 50k;
        allow-query     { any; };
        allow-recursion {
        127.0.0.1;
        172.30.0.0/24;
        172.30.10.0/24;
        172.30.20.0/24;
        ::1;
        fe80::/64;
        2a00:5240:0:8002::/64;
        };

        /* Path to ISC DLV key */
        bindkeys-file "/etc/named.iscdlv.key";

        managed-keys-directory "/var/named/dynamic";

        pid-file "/run/named/named.pid";
        session-keyfile "/run/named/session.key";
        check-names master warn;
        check-names slave warn;
        check-names response warn;
};

controls {
        inet 127.0.0.1 allow {127.0.0.1;} keys {"rndc-key"; }; };

#        include "/etc/named.root.key";
        include "/etc/rndc.key";


view trusted {
        match-clients { 172.30.0.0/24;
                        172.30.20.0/24;
                        127.0.0.1; };
        zone "." IN {
                type hint;
                file "named.ca";
        };

        zone  "example.com" {
        type master;
        file "example.com";
        };

        zone  "example.local" {
        type master;
        file "dynamic/example.local";
        allow-update { key rndc-key; };
        };

        zone  "0.30.172.in-addr.arpa" {
        type master;
        file "dynamic/0.30.172.in-addr.arpa";
        allow-update { key rndc-key; };
        };

        zone  "20.30.172.in-addr.arpa" {
        type master;
        file "dynamic/20.30.172.in-addr.arpa";
        allow-update { key rndc-key; };
        };

};


view guest {
        match-clients { any; };

        zone "." IN {
                type hint;
                file "named.ca";
        };

        zone  "example.com" {
        type master;
        file "example.com";
        };
};
```

----

example.com zone file:

```
-------------------------------------------------------------------------------------------------------------------------


$ORIGIN .
$TTL 86400
example.com     IN      SOA     linux.example.com.      root.example.com. (
                        1 ; serial
                        21600      ; refresh after 6 hours
                        3600       ; retry after 1 hour
                        604800     ; expire after 1 week
                        86400 )    ; minimum TTL of 1 day
;
        IN      NS      linux.example.com.
$ORIGIN example.com.
linux   IN      A       217.116.163.200
-------------------------------------------------------------------------------------------------------------------------
```

example.local zone file:

```
-------------------------------------------------------------------------------------------------------------------------


$ORIGIN .
$TTL 43200      ; 12 hours
example.local           IN SOA  localhost. root. (
                                280        ; serial
                                21600      ; refresh (6 hours)
                                3600       ; retry (1 hour)
                                604800     ; expire (1 week)
                                86400      ; minimum (1 day)
                                )
                        NS      localhost.
$ORIGIN example.local.
$TTL 3600       ; 1 hour
```

0.30.172.in-addr.arpa zone file:

```
-------------------------------------------------------------------------------------------------------------------------


$ORIGIN .
$TTL 43200      ; 12 hours
0.30.172.in-addr.arpa   IN SOA  localhost. root. (
                                183        ; serial
                                28800      ; refresh (8 hours)
                                7200       ; retry (2 hours)
                                604800     ; expire (1 week)
                                86400      ; minimum (1 day)
                                )
                        NS      localhost.
$ORIGIN 0.30.172.in-addr.arpa.
$TTL 3600       ; 1 hour
```

Test case	**Descrip-tion**	**Procedure**	**Expected outcome**
1 IP set-tings and connectiv-ity			
1.1 DHCP			
IP address assignme nt	Clients get IP ad-dresses when connect-ed. The settings are com-plete, so that gate-way, DNS server and other set options are given.	Connect a host in each VLAN. Enable DHCP client. Check IP settings and confirm IP settings.	All clients get IP address, mask and DNS. Voip cli-ents get TFTP server.
DNS updates	DNS rec-ords are updated dynami-cally. Both forward and re-verse	Connect a host in each VLAN. Enable DHCP client. Check DNS rec-ords on router with dig -tAXFR example.local, 0.30.172.in-addr.arpa and 20.30.172.in-addr.arpa	Hosts connected to the inside and voip networks are listed in records

	zones.		
1.2 DNS			
Recusive lookups	Recursive lookups work from the inside networks. Lookups are blocked for outside requests.	Ask for www.google.com address from inside client using nslookup. Do the same with an external client, by setting server to 217.116.163.200.	Outside recursive lookup should fail. Inside clients should be able to make requests.
Zone lookups	Check that zone visibility is correct	Start nslookup. Point server to be our linux router. Ask for linux.example.com and client.example.local.	Local public zone example.com can be queried from everywhere. Private zone example.local only from inside and VoIP.
1.3 Connectivity check			
Check network access using IPv4	Check that Internet can be accessed using IPv4	Open browser and open page http://217.116.160.50.	You should see your public IPv4 address (router's public IP).
Check network access	Check that Internet can be	Open address http://[2a00:5240:0:2024::50].	You should see your client's IPv6 address.

using IPv6	accessed using IPv6		
2. Firewall			
Check managem ent access	Check that manage-ment can get to all destina-tions	Plug a computer into management network. Plug end devices into each vlan. Ping devices from management net-work. Then try ssh ac-cess as root to router. First try without client certificate, then with it.	Ping to clients works. Ssh access without certificate should fail. Ssh access with certificate should let you in.
Check connectiv-ity be-tween LAN networks	Check that LAN net-works are restricted	connect end devices into each network. Ping the devices in the other networks from each device.	Ping to other networks should fail.
Check access from outside	Check that outside access is limited to DNS, VoIP and VPN	Try an ssh connection from outside	You should get a connec-tion timeout
Check NAT	Check that LAN net-works are natted properly	Start ping to 8.8.8.8 from both inside and visitor hosts. Check Internet interface with command tcpdump -n -i	Ping source should be 217.116.163.200 for inside and 217.116.163.201 for visitor

		enp3s0 icmp	
3 VoIP			
SIP	Check internal SIP calls	Dial 1002 for phone with extension 1001. Dial 1001 from phone with extension 1002. Dial some wrong extension.	In 2 first dialing scenarios phone should ring. When answered, you should here sounds from both ends. In 3rd scenario you should get an error note
IAX out	Check outgoing call	Dial 0297027100	Tele-entre switchboard should ring. Answer and ensure that sound is heard.
IAX in	Check incoming call	Dial 02970271031 and ..32	Both SIP phones should ring respectively
4. Suricata			
Check traffic inspection	Check that through traffic is inspected	Check suricata log file /var/log/suricata/stats.log	The counters should be increasing in /var/log/suricata/.
Check oinkmaster	Check that oinkmaster updates the rules	Check logfile /var/log/suricata/suricata.log and /var/log/messages	After each update in messages -file the suricata rules should be updated as well
5. OpenVPN			

Check server connectivity	Check that you can access the server	Download the client files to your laptop or computer. Connect to the example-server. Give bogus credentials. Check the connection log.	There should be a log entry: TLS: Initial packet from [AF_INET]217.116.163.200:1194
Check login	Check that you can login to server with valid credentials	Login to server. Use valid credentials, both the ones the certificate is for and some other.	Login with any valid credentials will work. You can see the user's certificate name in log file /etc/openvpn/openvpn-status.log
Check access	Check that you can access the inside networks and the router	Logon to the router. Check that you can reach the inside network. Use putty and the authrorized key to logon to the router using the openvpn address 172.30.30.1. Ping the computer in the inside network from your local computer.	SSH to router should work. Ping should succeed. You can check traffic using tcpdump -n -i tun0 on the router.
6. QoS			
Check rate limits	Check that proper limits are set on	Give command tc -s class ls dev ifb0	Check that the limits on classes are the same as given in the script.

	classes		
Check VoIP classification	Check that voip traffic is going to right class	Ensure all packets to 217.116.163.200 are marked CS5 or EF on remote PBX. Start a ping to 217.116.163.200 from remote pbx. Logon to the router. Give command tc -s class ls dev ifb0.	Packet count in class 1:11 should be increasing
Check bulk traffic	Check that other classes are working	Download something using both inside client and guest client	Both downloads should succeed. When downloading with inside client, the class 1:12 counters should increase. When downloading from guest network, class 1:20 counters should increase
Check voice quality	Check that the QoS queue works and voice is given priority	Tweak the tunings so that total bandwidth is 110k. Allocate 100k to the 1:11 class and 10k for 1:12. Make a call outside with the voip phone and start downloading a file from internet with inside computer.	You should see a lot of dropped packages on 1:12 class, while call stays up and call quality is good.