Bachelor's project

NTNU
Norwegian University of Science and Technology

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Quality and Traffic Flow in Videoconferencing Infrastructures

Bachelor's project in Computer Science
Supervisor: Ernst Gunnar Gran
Co-supervisor: Henning Elvestad and Bjørn Ludvik Isene

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Kunnskap for en bedre verden

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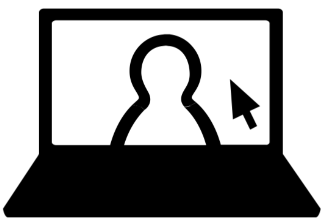


2021/05/20

Preface

The group would like to thank the most involved organizations and individuals in this thesis:

NTNU - Ernst Gunnar Gran
For being an excellent supervisor during our thesis, who guided the group with his experience and knowledge.

Telenor - Henning Elvestad and Bjørn Ludvik Isene
Employers from Telenor. They have been extremely helpful by providing guidance and resources in difficult times. Henning and Bjørn used their expertise and connections to give the group everything needed from equipment to experts in their fields.

Cisco - Espen Berger
Principal Engineer at Cisco Webex. Espen delivered exceptionally great information on Webex's inner workings. Without Espen this thesis would not have come as far as it did.

Abstract

The aim of this thesis was to provide insight into the two platforms Microsoft Teams and Cisco Webex from a network perspective. To achieve this, different virtual meetings were run and analysed. The virtual meetings differ in the number of clients, network topology and more. By collecting data from Cisco Meraki cloud and other instruments, data is compared to a basic virtual meeting, which is called baseline. All of the virtual meetings/test scenarios are also compared to documents which states what is good or poor meeting quality. From the comparison it was found that meeting quality is reliant on the several factors including the first client joining a meeting, the platforms recovery mechanisms and general network parameters.

Sammendrag

Målet med denne bacheloroppgaven har vært å gi innsikt i de to platformene Microsoft Teams og Cisco Webex. For å oppnå målet har forskjellige typer virituelle møter blitt kjørt og analysert. De virtuelle møtene er forskjellige i antall deltakere, bredbånd brukt, nettverksoppsett og mer. Ved å samle data fra Cisco Meraki cloud og lignende instrumenter har data blitt sammenlignet med et møte kalt baseline. Alle de virtuelle møtene/testscenarioer blir i tillegg sammenlignet med dokumenter fra Webex og Teams som sier noe om møtekvalitet. Gjennom disse sammenligningene ble det funnet at møtekvaliteten er avhengig av type bredbånd brukt, første person som starter et møte, en platforms gjennopprettings-mekanismer og generelle nettverksparametere.

Contents

Figures

Tables

Acronyms

ADSL Asymmetric digital subscriber line. 22, 32, 46, 47, 49

AP Access point. viii, 17, 18, 33, 34, 36, 60, 64–66, 73, 74, 76, 77, 89, 92, 94, 95, 244

API Application Programming Interface. 6, 25, 58

AS Autonomous Systems. 15, 16

ASN Autonomous System Numbers. 15, 16

AVC Advanced Video Coding. 7

BGP Border Gateway Protocol. 16

CBR Constant Bit Rate. 12

CQD Call Quality Dashboard. 6

dB decibel. 64, 73, 92

dBm decibel-milliwats. 64, 65, 73, 74, 92

DHCP Dynamic Host Configuration Protocol. 24

DNS Domain Name System. 24, 36, 42

FEC Forward Error Correction. 74

FPS Frames Per Second. 31, 37, 40, 42, 54, 61, 62, 70, 72, 74, 81, 84, 89, 90, 93

Gbps Gigabit Per Second. 10

HEVC High Efficiency Video Coding. 7

HTTPS Hypertext Transfer Protocol Secure. 8, 42

ICMP Internet Control Message Protocol. 14

IETF Internet Engineering Task Force. 12, 13

IP Internet Protocol. 6, 9, 11, 16, 30

ISP Internet service provider. 4, 15, 16, 21–24, 28, 45

JSON JavaScript Object Notation. 6, 31, 40, 98

Kbps kilobit per second. 38–40, 61, 70, 78–80, 84, 88

LAN Local Area Network. 11, 15

Mbps megabit per second. 10, 19, 24, 32, 38, 43, 45, 47, 49, 50, 52, 60, 68, 74, 78, 87, 88

MCU Multi-point Conferencing Unit. 12

MS Microsoft. xiii, 2–4, 6, 8–10, 12, 14–17, 22, 24, 28, 31, 37, 38, 45, 46, 56, 58, 60, 95

N/A not applicable. 31

NTNU Norges teknisk-naturvitenskapelige universitet. 4, 23

OPS Optical Packet Switching. 13

OSI Open Systems Interconnection. 9

PLR Packet Loss Rate. 13

POP Point-of-Presence. 16, 46

QoS Quality of Service. 13, 25

RSRP Reference Signal Received Power. 19, 92

RSRQ Reference Signal Received Quality. 19, 92

RTP Real-time Transport Protocol. 9

RTT Round-Trip Time. 31, 40–42, 44, 45, 48–50, 52–54, 61–63, 70–72, 74, 81–83, 89–91, 93, 234, 235

SD-WAN Software-defined networking in a wide area network. 23

SNR Signal-to-Noise Ratio. 18, 19, 34, 64–66, 73, 74, 76, 92

TCP Transmission Control Protocol. 6, 8, 9, 14, 26, 32, 38, 39, 42–44, 50–52, 54–58, 87, 92, 96

UDP User Datagram Protocol. x, xii, 8, 9, 11, 14, 22, 26, 33, 38, 39, 42–46, 50–52, 54–59, 79, 87, 89, 96, 100, 235

VLAN Virtuelt LAN. 24

VoIP Voice over Internet Protocol. 12

VPN Virtual Private Network. 100

WLAN Wireless Local Area Network. 17

WVC Web videoconferencing. 1, 5, 6, 9

Glossary

4G Fourth generation of broadband cellular network technology. 10, 19, 20, 23, 36, 87, 92, 94

Avg jitter Average network jitter for streams in milliseconds [1]. 31, 46, 57, 58, 66, 93

Avg Packet-Loss Average packet loss rate for a stream in %. [1]. 31, 46, 57, 58, 66

Avg RTT Average of average network propagation round-trip time based on RFC3550 [1] [2]. 31, 45, 57, 58, 66, 67, 93

barneparken Name of a Microsoft teams tenant. x–xiii, 2, 4, 43–45, 47–49, 55–57, 59, 66, 67, 75, 85, 93, 94

bit rate A measurement for the speed data is processed usually measured in bits per second [3]. xii, 7, 10, 31, 39–42, 50, 52, 61, 70, 74, 79, 80, 84, 89, 233

CC BY A license that allows re-users to distribute, remix, adapt, and build upon the material in any medium or format, so long as attribution is given to the creator. The license allows for commercial use. [4]. ix, 3, 8, 11, 13, 16, 18, 22, 28–30, 32–36

codec Device or software that encodes or decodes a digital data stream or signal. 7, 8, 11

CSV A CSV is a comma-separated values file, which allows data to be saved in a tabular format. CSVs look like a garden-variety spreadsheet but with a . csv extension. CSV files can be used with most any spreadsheet program, such as Microsoft Excel or Google Spreadsheets [5]. 31

group-policies Group policies define a list of rules, restrictions, and other settings that can be applied to devices in order to change how they are treated by the network. 6, 24

hoyskolestudent Name of a Webex tenant. 2, 4

jitter Intermittent delays during data transfers. 12–14, 26, 30, 31, 37, 40–42, 44, 48–50, 52–54, 61–63, 66, 71, 72, 81–84, 89–91, 97, 234, 235

latency A measuring of the time it takes for data to get to its destination in the network. 12, 26, 30, 37

load balancing Network load balancing is an efficient distribution of traffic over multiple paths [6]. 24

packet loss One or more packets of data travelling across a computer network fail to reach their destination [7]. ix, 6, 9, 12–14, 26, 30, 31, 37, 40, 44, 48–50, 57, 58, 61, 70, 74, 89, 90, 97

pcap Is an API for capturing network traffic. 6, 10, 67

playbook A document that describes the most interesting elements from the bachelor thesis in order to recreate scenarios. 2

port mirroring Port mirroring is used on a network switch to send a copy of network packets seen on one switch port to another port [8]. 24, 25

scope creep Refers to changes, continuous or uncontrolled growth in a project's scope, at any point after the project begins. 3

skytjenester Name of a Microsoft teams tenant. x, xiii, 2, 4, 43–45, 47–49, 56–59, 96

tenant An administrative domain that a customer can control from the cloud. In a sense this is your "virtual data center in the cloud". xviii, xix, 2, 4, 26, 28, 38, 45, 49, 58, 59

throughput Is "the amount of something (such as material, data, etc.) that passes through something (such as a machine or system)" according to Merriam-Webster Dictionary [9]. 10

Wi-Fi A family of wireless network protocols. 6, 10, 17–19, 22, 34, 61, 65, 67–69, 72, 74, 92, 96

Chapter 1

Introduction

The use of videoconferencing applications has been through an exponential growth of users due to the COVID-19 pandemic [10]. As a consequence of the pandemic millions of people have been forced to stay home. These people are now using virtual meeting solutions to replace a physical meetup at the office and has caused platforms like Microsoft Teams to have a growth of 75 million daily active users [11].

At schools and universities around the world student's educational activities changed to only offering courses online that utilized web videoconferencing (WVC) [12]. Businesses has experienced the change from physical meetings to use WVC, and the workspace has moved to the employees own living room. Given the uncertainties of WVC quality and the challenge of shifting to an online environment, this study aimed to evaluate videoconferencing quality in regards to network performance with the transformation from physical meetings to sessions conducted through WVC.

1.1 Background

The bachelor thesis is a result of the project given by the Norwegian telecommunications company Telenor. The client is one of the world's largest mobile telecommunications companies with operations worldwide [13]. Telenor offers small to mid-size businesses a Cisco Meraki[1] network solution with everything from routers, firewalls, and access points to security cameras [14]. Alongside the different equipment, Telenor sells network plans that are optimized for videoconferencing calls, such as Microsoft (MS) Teams[2] and Cisco Webex[3]. Telenor also provides consultancy regarding network troubleshooting.

Telenor is in collaboration with Microsoft and Cisco to improve quality and performance for videoconferencing software. The collaboration utilizes peering between Telenor and the two corporations. Peering is an agreement between organizations that creates a path for digital communication between two networks for improving users experience [15]. As Telenor provides these different networking solutions towards Microsoft and Cisco, Telenor would like to run a set of different test scenarios with Meraki network equipment. The network equipment is the same that are used by small to mid-size business clients of Telenor. The test scenarios revolves around video and voice quality in videoconferencing.

Microsoft and Cisco are selling tenants for corporations which can be bought for additional administrative control and contribute supplementary data on network quality. "A *tenant* can be viewed as a group of users sharing the same view on the application they use. This view includes the data they access, the configuration, the user management, particular functionality and related non-functional properties", described in Krebs *et al.* [16].

In the case of this bachelor thesis a total number of three tenants were provided by Telenor for testing purposes; *barneparken*, *skytjenester* and *hoyskolestudent*. The two tenants *barneparken* and *skytjenester* are tenants in MS Teams while *hoyskolestudent* belongs to Webex. Telenor provided two tenants to find out if the geographical location of the tenants matter. The two MS tenants geographical location are as follows: *barneparken* is registered as a European tenant, while *skytjenester* is registered as a Norwegian tenant.

Telenor wants a playbook that includes one testing scenario from the bachelor thesis to replicate the same results, that will be presented at the yearly event, Telenor Expo. The playbook should also be detailed enough for Telenor employees to replicate the same scenario at a later time.

[1] see `https://meraki.cisco.com/` for Cisco Meraki's homepage.
[2] see `https://www.microsoft.com/en/microsoft-teams/` for Microsoft Teams homepage.
[3] see `https://www.webex.com/` for Cisco Webex's homepage.

1.2 Problem statement

The purpose of the bachelor thesis is to evaluate the quality and traffic flow in networks with regards to Cisco Webex and MS Teams. The bachelor thesis is twofold. First, to conduct a set of test scenarios. Each test scenario is unique and accordingly affects the videoconferencing platform differently. These experimental tests should be conducted by using Meraki network equipment. An illustration of a basic test scenario can be seen below in figure 1.1. The second step is to present each test scenario and describe the effects on the videoconferencing platform in regards to quality parameters. All of these test scenarios differ in a certain way by for example using a different type of broadband, having extra participants in a meeting or having introduced a poor network connection.

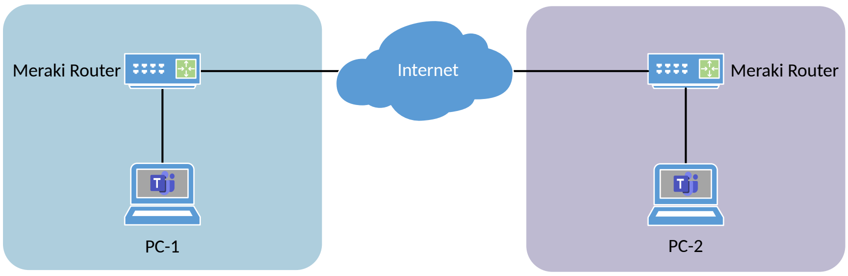


Figure 1.1: Basic test scenario using Microsoft Teams, License: Andreas Rømo, CC BY.

1.3 Scope

As the testing scenarios can be wide in scope it is important to decide what the main focus will be to prevent scope creep. The given task is about network quality, but there are other areas to consider as well.

An area to consider with applications such as MS Teams and Webex (tools are explained in section 2.1) is network security. As the threat landscape in the digital world is expanding [17] it only makes sense to question an applications protective mechanisms. Nevertheless, network security is excluded as a subject in this thesis because of the limited time of the bachelor project.

Another possible area is software. Software can be reviewed for performance issues that occur in a meeting. Doing so would require time which the bachelor group does not have. However, a software related topic that must be touched upon is resolution and frames rate as it is a deciding factor in bandwidth consumption. Software is therefore barely touched upon in the bachelor project.

In this bachelor project it was decided that the applications residing on clients and servers will not be investigated, only the communication between them and its effects on quality. However, there is still too much to take account for in a single thesis. In collaboration with the project owner Telenor, the test scenarios are thus limited to a given amount.

The limitations put upon the scenarios are as follows:

- Tests will only be conducted using three Internet Service Providers (ISPs); Uninett, Telia and Telenor.
- Test locations will be restricted to a single area, such as NTNU campus and student homes.
- Tests will only use MS Teams and Cisco Webex as meeting platforms.
- Tests will only run on the tenants *barneparken* and *skytjenester* for MS Teams and *hoyskolestudent* for Webex.
- Webex tenants and Teams tenants will not be compared.
- Analyze only a single test-run/iteration from each test scenario presented.

1.4 Thesis Structure

This thesis is based on the scientific writing structure called IMRaD. IMRaD is an acronym for introduction, method, results, and discussion.

Chapter 1 covers the introduction part. The next chapter, technical background, will cover the technical knowledge required for further reading. Method in the bachelor thesis consists of the procedure regarding the baseline, testing scenarios and the topology for all of the test scenarios. Chapter 4 will cover results and discussion for the testing scenarios. In chapter 5 there will be an overall discussion surrounding the testing scenarios and choices made in the thesis. Chapter 6 is the conclusion for the bachelor thesis, which gives suggested further work, improvements and a concluding summary.

1.5 Related Work

For network testing a masters thesis titled NPT Online Broadband Test Tool [18] is used as reference on test scenarios. It contains a set of test scenarios in chapter 9 regarding a network testing tool called NPT Broadband Test Tool. Key notes from the master thesis is their listing of clients used when performing tests and the explanation of the environment setup before each performed test. The environment setup has been copied and expanded in this thesis by using a baseline environment instead. The baseline environment is explained in further detail in subsection 3.2.1 and in section 3.1.

Chapter 2

Technical background

The technical background chapter introduces existing tools and technical background needed for reading the bachelor thesis. The chapter explains aspects of how videoconferencing is able to function, how it is built up from a technical point of view and gives network insight.

2.1 Existing Tools and Systems

In the following section the technologies used in the bachelor thesis are explained and also what their purpose is.

2.1.1 Cisco Webex

Cisco Webex Meetings is a videoconferencing service that provides video meetings with schedule and joined experiences. It's possible to join meetings from the browser, applications, mobile and video room devices [19].

Cisco Webex (formerly Cisco Spark and Cisco Webex Teams) is a business communication platform that provides messaging, file sharing, video meetings, white boarding, calling, and other tools [20]. Both Webex (Teams) and Meetings are strongly coherent and are often referred to as simply Webex. Webex is one of the Web videoconferencing (WVC) applications used for testing.

2.1.2 Cisco Webex Control Hub

Cisco Webex Control Hub[1] is a management and analytics tool. It is used to monitor and manage different Cisco devices and workplaces created from the Control hub to simulate physical locations. The control hub will be used for collecting data for the Cisco Webex test scenarios.

[1] see `https://www.cisco.com/c/en/us/products/conferencing/webex-control-hub/index.html#~for-partners` for info on Webex Control Hub.

2.1.3 Cisco Meraki Dashboard

Cisco Meraki Dashboard is a network management interface [21] for Cisco Meraki equipment. It is used as a cloud controller for Wi-Fi, routing, and security. The dashboard will be used to apply different group-policies accordingly to the test scenarios, and to administrate the equipment. A group-policy is a way for an administrator to set rules like access lists and tagging [22].

2.1.4 Microsoft Teams

Microsoft Teams is a business communication platform developed by Microsoft that provides a workspace chat, videoconferencing, file storage, and application integration [23][24]. The MS Teams platform will be used as a WVC application during the test scenarios.

2.1.5 Power BI Desktop

To extract data from MS Teams the tool Power BI desktop[2] is used. Power BI is a visualization tool, and MS Teams had a connector developed to directly integrate data into Power BI [25]. A connector can be seen as a module for power BI. The connector is able to open a link to the Call Quality Dashboard (CQD) API to get information. The Call Quality Dashboard (CQD) is the dashboard for tenant admins to use for inspecting call quality for meetings held on that tenant. Power BI will be used to analyse and visualise the data from the CQDs API.

2.1.6 Wireshark

One of the functions provided by Cisco Meraki equipment is its pcap functionality, that captures network data. The pcap files are visualised using Wireshark[3] to inspect the raw data collected under a test scenario. Where Wireshark's tools are used to analyse the raw data.

2.1.7 Ntopng

Ntopng[4] is a flow collector and a software monitoring tool that shows traffic in real-time. The software can detect clients in the network and monitor their network traffic. Some of the data collected are protocols sent and received on the host, real-time throughput, Transmission Control Protocol (TCP) packet loss, the most used IP protocol, usage sorted by protocol type, and much more [26]. It is also possible to dump network traffic data into a JSON file. Ntopng will be used for inspecting a test in real time and analysis of data collected during the thesis.

[2] See `https://powerbi.microsoft.com/en-us/desktop/` power BI' homepage.
[3] See `https://www.wireshark.org/` for Wireshark's homepage.
[4] See `https://www.ntop.org/products/traffic-analysis/ntop/` for ntopng's homepage.

2.1.8 Codec

Codec is a combination of the words encoder and decoder, but it also compresses or reduces the size of the video/audio based on the codec type [27]. To encode and decode audio and video information recorded codec uses hardware or software-based processes [28]. It is important to utilize a codec when sending or receiving any kind of audio or video over the Internet.

2.1.8.1 Video Codec

A video codec is either a software or a hardware with the function of compressing and decompressing digital video [29]. A video file consists of parameters like bit rate, resolution and the activity in the frame [30] and all of these contribute to increasing the size of the data stream. Some form of compression is needed to send it over the Internet without using too much time and network capacity. This is where a video codec comes in handy to compress the video on one side of the network and then decompresses it on the other side. The compression reduces the size of the data stream sent and helps when trying to send seamless video without stuttering. This helps with needing less bandwidth to send good quality video.

The latest, May 2021, video compression used by videoconferencing systems is H.265[5], also known as High Efficiency Video Coding (HEVC). As a guideline, HEVC provides more efficient encoding, cutting the required bandwidth needed to send a video signal by 50% compared to its predecessor H.264, or advanced video coding (AVC) [31]. None of the videoconferencing platforms used in this bachelor project support HEVC as of this writing.

Platform	Audio Codec	Video Codec(s)
Cisco Webex	Opus	OpenH264
Microsoft Teams	Silk	H.264

Table 2.1: The codecs that each platform uses.

As stated in table 2.1 Teams[6] and Webex[7] both use a type of the H.264 codec. The H.264 codec[8] supports using Intra-frames (I-frames) and Predicted frames (P-frames) [32], an illustration of I- and P frames are shown in figure 2.1. An I-frame is a still image/frame, and to record changes from the I-frame, P-frames are used. A P-frame is therefore only a change from the I-frame. When a new

[5]see `http://hevc.info/` for H.265's homepage.

[6]see `https://docs.microsoft.com/en-us/microsoftteams/platform/bots/calls-and-meetings/real-time-media-concepts` for more info on the codecs used in Teams.

[7]see `https://help.webex.com/en-us/nckc1aeb/Webex-Video-Specifications-for-Calls-and-Meetings` for more info on on the codecs used in Webex.

[8]see `https://shopdelta.eu/h-264-image-coding-standard_l2_aid734.html` for more info on the AVC/h.264 codec.

image is formed by the P-frame, the newly created frame is used for reference for the next P-frame. This can be called incremental changes. When the incremental changes require more data than it would take to recreate the whole image the codec outputs a new image.

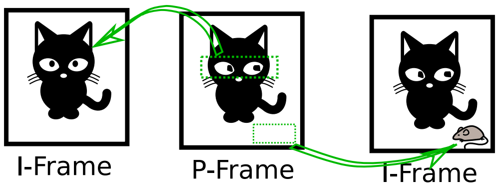


Figure 2.1: The figure illustrates how the different frame types encodes movement in a picture, license: Andreas Kilde Lien, CC BY. Inspired by Cloudflare [33].

More specifically a P-frame, also known as delta-frames, stores only the changes from the previous frame, so it bases its predictions on moving parts at previous frame and then only encodes those parts. This saves space and network bandwidth by not having to encode all the frames again [32].

The loss of a P-frame will trigger a new request for an I-frame. As the request for an I-frame makes a larger change than the continuous P-frames, the I-frame will demand more capacity or in this case bandwidth [34]. This increased demand can be used to identify tops in a graph showing the bandwidth usage throughout a meeting. For Webex it is possible to figure this out in another way by inspecting packets. However, exactly how this is figured out is not allowed to be shared as per request from one of Cisco Webex's employees.

2.1.8.2 Audio codec

Audio codec is much the same as a video codec only it functions on audio rather than video. With the same goal in mind to reduce the size of the audio stream while trying to maintain as good quality as possible [35].

Webex uses Opus[9] as an audio codec [36] and MS Teams uses Silk which is a basis of voice mode in Opus [37]. Opus is efficient code speech (data compression of digital audio) and general audio in a single format. It features low-latency for real-time interactive communication and low-complexity for low-end embedded processors [38].

2.1.9 Networking and protocols

Computer networking is a group of nodes that communicate through protocols using digital interconnections. Hypertext Transfer Protocol Secure (HTTPS), Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) are a few of

[9] See `https://opus-codec.org/` for Opus' homepage.

the many protocols that are used in networks. Protocols that rely on the Internet Protocol (IP) can be put into the Open Systems Interconnection (OSI) model or TCP/IP model [39][40]. The OSI model represents a stack of protocols and visualizes a way protocols can rely on each other. In figure 2.2 UDP is at the transport layer, which means UDP cannot function without a protocol from the network layer. In this case UDP relies on IP.

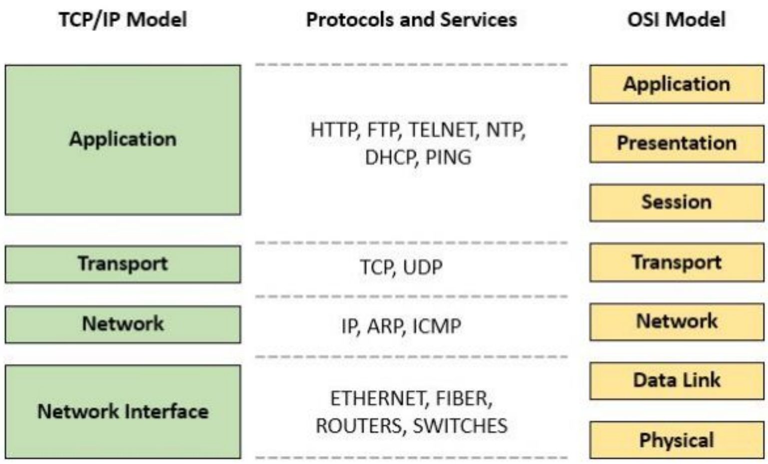


Figure 2.2: The figure shows the OSI model with different protocols on the stack.

WVC applications use mainly UDP as a transport protocol [41][42]. UDP is a layer 4 protocol which is a best-effort service, commonly referred to as a *connectionless protocol* [43]. This means a node don't know if the packets actually got to their destination. For WVCs this is an advantage as a few lost packets won't affect the quality of the service. TCP is a protocol which works against losing packets. TCP re-sends lost packets, but this will make services which relies on a live feed seem like it is starting and stopping. Also, UDP has less overhead than TCP. Because of these reasons live audio and/or video streams in both MS Teams and Webex primarily use UDP. Webex and Teams does however provide counter measures to packet loss and similar on layers above layer 4.

2.1.9.1 RTP

Real-Time transport protocol (RTP) is a protocol that both Cisco Webex and MS Teams utilizes. This protocol is defined in RFC 3550[10]. RTP tries to move data from one point to another as efficiently as possible and typically uses UDP as the layer 4 protocol. From the RTP headers[11] it is possible to deduct I and P frames.

[10]See `https://tools.ietf.org/html/rfc3550` for RFC3550.
[11]See `https://datatracker.ietf.org/doc/html/rfc3550#section-5.1` for RTP header.

However, RTP can be used differently for each application and therefore deducting I and P frames can be time consuming.

2.1.9.2 Connections

To make a connection between nodes, mediums like Ethernet cables, fiber optic cables and copper cables are used. These mediums have an influence on the possible throughput bit rate. It is also possible to use the non-physical medium radio waves in the 900 MHz to 60 GHz [44] range to communicate. In the test scenarios fiber and 4G are used for internet access. For connecting to the Meraki routers, Wi-Fi and Ethernet cables are used.

2.1.9.3 Bandwidth and throughput

When the client makes use of a network there are several variables that indicate if the network connection is satisfactory or poor. One of those are bandwidth. Bandwidth is often measured in Megabit per second (Mbps) or Gigabit per second (Gbps). If bandwidth is low videoconferencing are affected. Bandwidth is often mistaken for throughput. There is a distinction as bandwidth is theoretical, while throughput is the actual measured speed. The applications has algorithms built into them to figure out the available bandwidth and regulates variables such as frames per second and resolution to relate itself to the available bandwidth. In a test scenario/meeting which will be presented later a limit on the bandwidth will be induced in the middle of the meeting.

For MS Teams the required bandwidth is stated in the documentation shown in table 2.2. The table shows the bandwidth required in the first column and what type of call it relates to in the second column. When inspecting a pcap file from a meeting the bandwidth can be seen, and by putting the bandwidth in relation to this table an assumption of the meeting quality can be made.

Bandwidth (up/down)	**Scenarios**
30 kbps	Peer-to-peer audio calling
130 kbps	Peer-to-peer audio calling and screen sharing
500 kbps	Peer-to-peer quality video calling 360p at 30fps
1.2 Mbps	Peer-to-peer HD quality video calling with resolution of HD 720p at 30fps
1.5 Mbps	Peer-to-peer HD quality video calling with resolution of HD 1080p at 30fps
500kbps/1Mbps	Group Video calling
1Mbps/2Mbps	HD Group video calling (540p videos on 1080p screen)

Table 2.2: Bandwidth requirements for MS Teams [45].

For Cisco Webex the requirements are stated in Cisco's help page and documentation, some of the information is listed in the table below 2.3. [46] [47] [48]

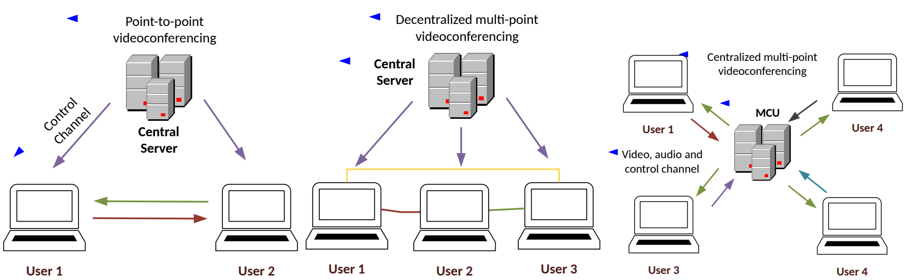
Layer	Bandwidth Range
90 active thumbnail (each)	~60-100 kb/s
180p main video	125-200 kb/s
360p main video	470-640 kb/s
720p main video	768k-2.5 mb/s
Content sharing (sharpness, 1080p/15)	120k - 2.0 mb/s
Content sharing (motion, 720p/30)	900k - 2.5 mb/s

Table 2.3: Bandwidth requirements for Webex [48] with updates from a Cisco Webex employee.

2.1.10 Videoconference

Videoconferencing is a virtual meeting with two or more participants who are often participating from different locations. Participants use devices that has audio, video and internet access to participate in a video call. The main purpose of videoconferencing is to help people connect, regardless of where they are located [49][50].

Videoconferencing in video-enabled meetings happen in two distinct ways: either point-to-point or with multi-point [51][52]. The H.323 standard have roots back to 1996 and was published by the International Telecommunication Union [53]. It was meant for enabling videoconferencing capabilities over a local area network (LAN). With the H.323 standard people could call someone's IP address. Both parts could then see each other's video and talk like shown in figure 2.3a. Over the Internet the audio and video are digitized [54], assembled into UDP packets and sent across the network to the other end, where they are converted back into analog signals. Both video and audio are often compressed with a codec to save bandwidth, but they do have to be on a good link, because compression introduces delay [55], as well as some distortion, so it makes it harder to hold a conversation.



(a) Point-to-point videoconferencing. **(b)** Decentralized multi-point videoconferencing. **(c)** Centralized multi-point videoconferencing.

Figure 2.3: Videoconferencing models. License: Andreas Kilde Lien, CC BY.

The other type of call is multi-point call. This is when two or more locations are linked together in a centralized multi-point video call, where all participants can see and hear each other. In this type of call voice, video and content streams of digital information are sent to a Multi-point Conferencing Unit (MCU) to tie the location together. The program re-sends a collective data stream back to meeting participants in the form of real-time audio and video by combining the individual participant's video and voice traffic [56] as shown in figure 2.3b. Some systems can perform multi-point conferencing without the assistance of an MCU. Because there is no central repository of MCU availability, this approach can provide higher quality. This method is known as decentralized multi-point videoconferencing [57]. Like shown in figure 2.3c.

Videoconference - audio tolerance
Audio performance is one of the important features of videoconferencing systems. People can more easily tolerate video that is jittery and motion that is less smooth. However, it is necessary to maintain an unbroken audio stream for effective communication [58].

2.1.11 VoIP

VoIP is used in a videoconference to deliver the voice aspect of the online conversation. Rather than needing a special device to speak in an audio conference, participants can use their devices' sound and microphones to receive audio and send audio. When using VoIP in a videoconference it is recommended using a headset for getting the best audio possible as PC's microphones are very sensitive to sound and pick up a lot of background noise [59]. In all of the meetings to be run there will not be used headset.

2.1.12 Network Deviation

The quality of video and audio achievable can be indicated by three network characteristics measurable through a variety of values such as jitter, packet loss and latency.

2.1.12.1 Jitter

In videoconferencing calls real-time speech and/or video are generating Constant Bit Rate (CBR) traffic streams [60]. An important performance measure for such traffic is the jitter which can be defined as the measure of the packets' transfer delay variation. It can depend on the packets' routes and is caused by multiplexing several flows in the node queues [61]. There are many definitions of jitter that try to capture the delay variation of packets. In this Bachelor thesis there will be an adaptation of the IETF definition of jitter [62]. It is based on the transit delay between the entry and the exit nodes. It is not known exactly how Webex and MS

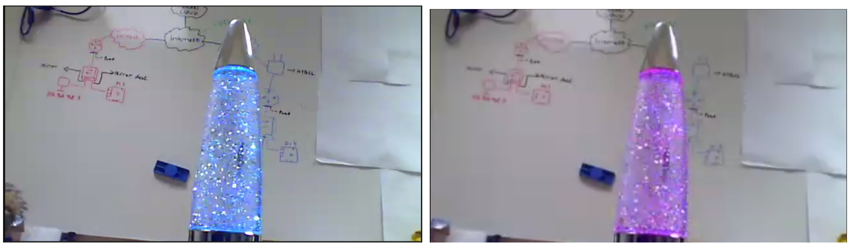
Teams calculate jitter, but something closely related to the IETF is expected to be used by the platforms.

2.1.12.2 Packet Loss

Packet loss at the network layer due to contention [63] is a crucial problem in Optical Packet Switching (OPS) networks, that is used on both locations for this bachelor thesis. Contention occurs in asynchronous OPS when a packet is intended for an output wavelength that is currently being transmitted by another packet. Contention also occurs in slotted OPS when two or more packets in the same time slot are destined for the same output wavelength. Contending packets would be lost in both instances and lead to an increased rate of packet loss (PLR). The rise in the PLR due to contention is largely determined by the frequency of disputes and the total number of packets lost each time a dispute arises. PLR is given by

$$Packet\ Loss\ Rate = \frac{Packets\ sent - Packets\ received}{Packets\ sent} * 100 \tag{2.1}$$

In real-time videoconferencing calls, if packets arrive at the destination out-of-order, the codec has to discard the packet because it is not received in the right sequence and cannot be combined back together. This is packet loss that removes an I-frame or a large part within an I-frame near a scene. This creates ghosting, which is a significant visual impairment as shown in figure 2.4 [64].



(a) Image of none Packet Loss **(b)** Image of 7% packet loss Rate

Figure 2.4: Visualisation of packet loss. The lamps color variation is not caused by packet loss. Observe the increase of blur/ghosting in the image at 7% package loss. License: Andreas Kilde Lien, CC BY

2.1.12.3 Latency

Real-time applications rely on timely delivery, such as videoconferencing. Using a deterministic communication path is the usual way of ensuring timely delivery. There are several industrial networks that are design-deterministic, which means that a maximum and/or minimum quantity can be determined for all aspects, including both delay and bandwidth. In terms of Quality of Service (QoS) and the reliability metric, within the designed limits, deterministic communication networks generally have 100 percent reliability. On the other hand, the Internet is not inherently deterministic and it is therefore difficult to ensure that critical information is passed on to the receiver in time. Reliability defined by Ekram Hossain *et al.* [65] as the probability that the package is successfully received within the latency limit.

UDP latency is hard to calculate. Because UDP is designed to be quick but is unreliable, it does not have any built-in features to detect and recover from latency. Instead, it relies on the application layer protocols (and ICMP) that it's paired with to handle data delivery reliability [66].

2.1.12.4 Cisco Webex - requirements and recommendations

Cisco prefers and strongly recommends UDP as the transport protocol for all Webex voice and video media streams. If this is blocked then Cisco Webex still works using TCP, but audio and video will be impaired [67]. Cisco recommends a packet loss rate that is equal or lower than 1.00% for voice and 0.50% for video. Measured UDP latency should be equal or lower than 100ms for voice and 150 to 300ms for video, and measured UDP jitter should be equal or lower than 30ms for voice and 10ms to 50ms for video in one-way [41].

2.1.12.5 Microsoft Teams - requirements and recommendations

Microsoft Teams should be available to use UDP connectivity, according to Microsoft. If this is blocked then MS Teams will switch to using TCP, but audio and video will be impaired [42]. Microsoft recommends a packet loss rate that is lower than 1.00%. Measured UDP latency should be lower than 100ms, and measured UDP jitter should be lower than 30ms in one-way (from a single source to a destination) [42].

2.1.13 Traffic flow

Traffic flow depends on a several different factors: *Application*, *Amount of users*, *Call type*, *ISP* and *LAN Configuration*.

Server location
The server location the client needs to connect to will depend on which application is used and where MS Teams or Webex has deployed servers.

The amount of users
The amount of clients connected to a call, as mentioned in section 2.1.10, and "Call type" will also play a part on how traffic is routed network wise. For example, a private call in MS Teams and Webex will use a point-to-point call, while a "scheduled meeting" for MS Teams or a meeting in a "room space" for Webex uses a centralized call method. For each test scenario there will be use of a centralized call method.

Call type
As Webex Control Hub will be used for Webex and Power BI Desktop for MS Teams, it is required that the meetings will not be Point-to-Point, even though only two clients is in the meeting. If a Point-to-Point videoconference is not hosted on a server, information about the meeting might not be acquired by the Control Hub or Power BI Desktop, see figure 2.3a. Therefore the meetings will be centralized multi-point videoconferencing meetings. This is done by utilizing MS Teams "scheduled meeting" and Webex's "Room spaces".

ISP
As videoconference meetings require Internet connectivity, the clients needs to be on a LAN connected to an ISP [48][45]. Which ISP used and the configuration of the LAN the client is on, will influence which route the traffic takes through the network. In this thesis Uninett, Telenor and Telia are the ISP's used.

As mentioned in section 1.1, Telenor utilizes peering to improve quality of videoconference meetings held on MS Teams and Webex.

2.1.13.1 Peering and Autonomous systems

Direct peering is an agreement between two networks that is registered as Autonomous Systems (AS). An AS[12] is a network that owns an Autonomous System Number (ASN) and is run by a single entity. Each AS must have its own unique ASN. Without one it would not be possible to route traffic, as it would be impossible to distinguish networks on the Internet.

[12]See Definition on `https://www.ripe.net/publications/docs/ripe-679`.

To be able to connect different networks together with the use of AS numbers, a protocol called Border Gateway Protocol (BGP), is used for exchanging routing information between Autonomous Systems (AS). BGP differ from the standard interior routing protocols as it is an exterior routing protocol and uses Autonomous System Numbers (ASN) for looping prevention [68]

In figure 2.5 there are three AS. LeftNet, MidNet and RightNet. LeftNet shares its internal known routes to MidNet. RightNet also shares its routes to MidNet. MidNet shares its internal known routes to RightNet and LeftNet. MidNet does not share its external discovered routes to the other networks.

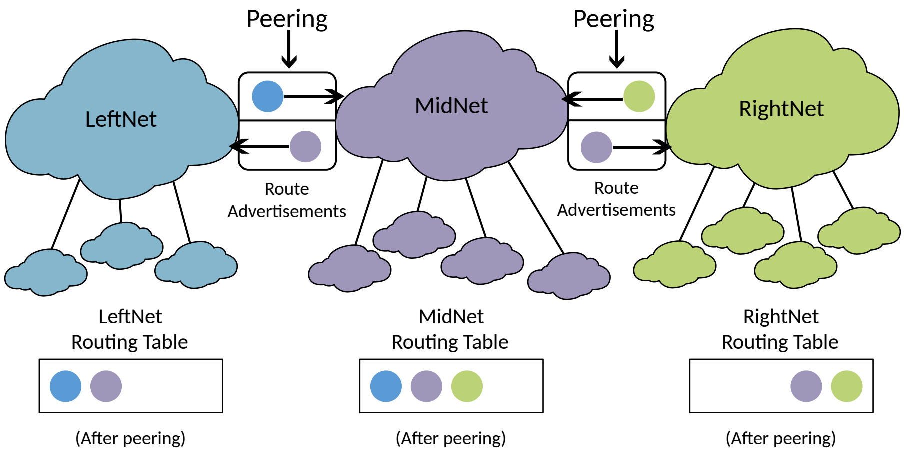


Figure 2.5: The figure illustrate two peering agreements between three networks. License: Andreas Rømo, CC BY. Inspired by drpeering [69].

In PeeringDB[13] a list of AS and their Point-of-Presence (POP) is listed. It is not a given that an AS is listed in this database or that they share peers, but many AS, such as Telenor and Webex[14], has a policy to only peer with other AS that are listed in PeeringDB. In PeeringDB Webex, MS Teams, Telenor and Uninett are listed. By inspecting the fields "Public Peering Exchange Points" and "Private Peering Facilities" and by having dialogue with ISPs, Cisco and MS Teams, the peering agreements are as follows. Telenor utilizes public peering with Webex on Amsterdam-Internet Exchange Point (AMS-IX). For MS Teams, Telenor is connected to a POP in Oslo [70].

With Uninett a transit agreement with NORDUnet brings traffic to Webex. It is not known if NORDUnet utilizes private or public peering with Webex. A traceroute to Webex does not show any IP prefixes associated with AMS-IX. For MS Teams Uninett has private peering with MS Teams from Digiplex Oslo Ulvenveien (DNAS).

[13] See `https://www.peeringdb.com/` for PeeringDB's website.
[14] See `https://www.webex.com/peering-policy.html` for Cisco documentation on peering.

2.1.14 Wireless network

Wireless networks do not require a physical cable from a client to be connected. There exists a range of different wireless networks, but the bachelor thesis will only cover the two wireless network connections that will be used in the testing scenarios.

2.1.14.1 Wireless LAN

A Wireless Local Area Network (WLAN), also known as Wireless Ethernet, is used when there are devices that supports network access and does not have an Ethernet port or just for convenience. To communicate among devices within a WLAN, the devices communicate with radio waves to transmit data. The radio waves for Wi-Fi are transmitted in the 2.4 GHz or 5 GHz frequency range. When testing media quality for Webex or MS Teams on Wi-Fi 5 GHz will be used.

WLAN equipment used in this thesis is based on the IEEE 802.11 standard[15], also known as Wi-Fi. There are multiple IEE 802.11 standards, but the MR-36 (AP) supports up to 802.11AX (Wi-Fi 6)[16], and the laptops used in the tests only supports up to 802.11ac (Wi-Fi 5).

To transmit or receive data over Wi-Fi, antenna(s) are required. How strong and far the signal can reach depends on multiple factors, such as: The transmitter, distance, obstacles, and receiver. To calculate the signal strength between the transmitter (Tx) and receiver (Rx), decibel milliwatts (dBm) is typically used. dBm measures the ratio of electrical power in reference to one milliwatt (mW). Illustration 2.6 should clarify this a bit more.

In the example figure 2.6, the signal has a power of -40 dBm when the receiver receives the signal. However, even with a strong signal power as shown in the figure the signal can still be heavily disturbed by noise.

[15] See `https://www.ieee802.org/11/` for information about IEEE 802.11 standard.
[16] See `https://meraki.cisco.com/product/wi-fi/indoor-access-points/mr36/` for specifications regarding MR36

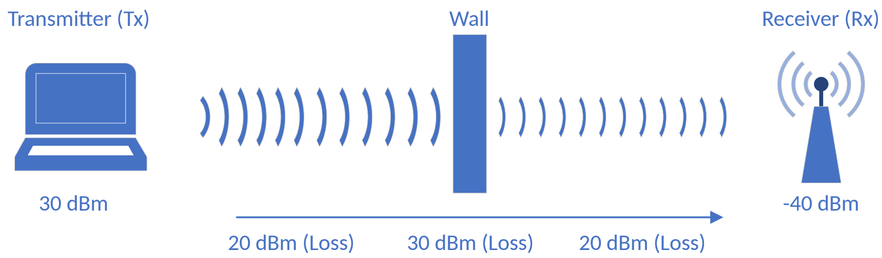


Figure 2.6: The figure illustrates signal power/loss using dBm. License: Andreas Rømo, CC BY.

A more reliable indicator than signal strength is Signal to Noise Ratio (SNR) as illustrated in figure 2.7. SNR compares the desired signal the devices communicates over and signals that is background noises for the two devices exchanging data. For the Meraki AP, SNR is measured in decibels (dB), and in network context, starts from 0 dB. Higher dB value equals greater signal quality indication [71].

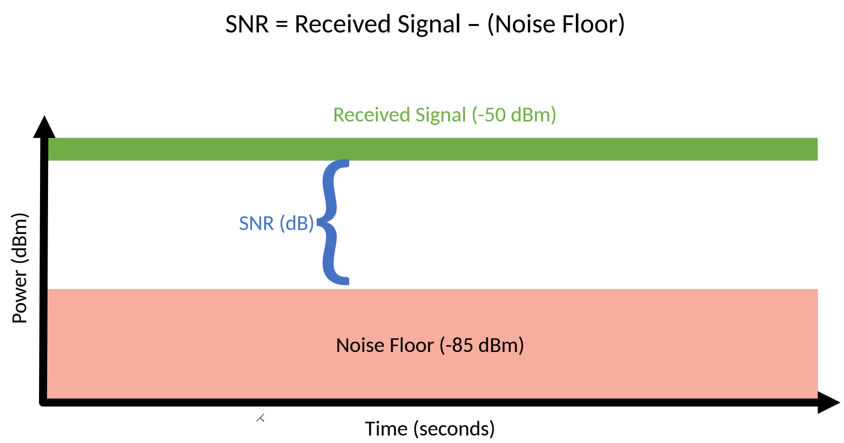


Figure 2.7: The figure illustrates how SNR is the measured. License: Andreas Rømo, CC BY. Inspired by Meraki [72].

Data reliability

In test scenarios 2.1, 2.2 and 4.1 Wi-Fi is used by one of the meeting participants. The data collected from these scenarios are SNR from the AP and dBm from the participant. This data is not sufficient enough to tell if the signal is good or poor for a video call. The participant using Wi-Fi has unknown characteristics such as antenna gain and noise received. As an example if the dBm recorded was -92 dBm for a participant it is hard to determine if the signal strength is unusable or not. For the client that is going to be used for test scenario 2.1, 2.2 and 4.1 it is tested that -92 dBm works for a video call, but unreliably, so if it was possible to calculate the SNR for the participants it would be a more reliable indicator. This could be attempted by using equipment to record the noise.

A passive wireless site survey will be done for the wireless participant in test scenario 2.1, 2.2 and 4.1, however the equipment, an Apple laptop, used for the survey is sub-optimal and doesn't overlap with the receivers for the clients in the meeting. In the event there was funding and time a Wireless Survey with the use of calibrated equipment could help to further determine the quality of the signal. Such equipment is in the price range of 300.000 NOK[17].

2.1.14.2 Mobile networks - 4G

To be able to connect to the Internet using a phone there are different technologies such as 3G, 4G and 5G. The former technologies are sorted chronologically from the oldest to the newest generation of cellular communications.

4G communications
Today 4G is the typical communication mobile devices use for Internet access when Wi-Fi is not available. In 2018 Ookla released speed tests for Norway's 4G network with a download speed of 72.05 Mbps [73]. A download speed at 72 Mbps is on paper able to stream High Definition (HD) video 72 Mbps / 5 Mbps $\approx$ 14 times [74]. That is more than enough for a regular consumer.

4G uses several ranges to communicate. An example is the 694–790 megahertz (MHz) range for 4G and 5G. To use a specified range a company must buy the rights for that specific range. For instance in the 700 MHz range Telenor has bought the rights to use the 703–713 MHz range for upload and 758–768 MHz range for download [75].

4G Signal Quality
The Meraki 4G gateway uses RSRP and RSRQ to measure the signal strength and signal quality. RSRQ is similar to SNR in a way that it is a ratio from noise, but RSRQ is more specific with the resource blocks and antennas. RSRP is also specific with the resource blocks and antennas.

[17] See `https://www.rohde-schwarz.com/ph/product/cmw270-productstartpage_63493-9552.html` for information about the equipment.

4G RSRQ / RSRP	< -112	> -112	> -108	> -104	> -100	> -96	> -92	> -88	> -84	> -80
> -4	0	3	4	7	8	8	9	9	9	9
> -5	0	3	4	6	7	8	8	9	9	9
> -6	0	1	3	4	6	7	8	8	9	9
> -8	0	1	3	4	5	6	7	8	8	9
> -10	0	1	2	3	4	5	6	7	8	8
> -12	0	1	2	2	3	4	5	6	7	8
> -14	0	1	1	2	2	3	4	4	6	7
> -16	0	0	1	1	2	2	3	3	4	4
> -18	0	0	0	1	1	1	1	1	3	3
< - 18	0	0	0	0	0	0	0	0	0	0

Figure 2.8: The figure shows a table for determining signal quality in a 4G network. License: Unknown.

Testing

In test scenario 4.1 4G testing is done with a Telenor 4G wireless mobile broadband/Mobile gateway. This 4G wireless mobile broadband/Mobile gateway probably uses the closest tower located at Niels Ødegaards gate 10-8 in Gjøvik. It would be possible to gather information and calculate which exact tower is used [76], but this takes time and is as such not done. At finnsenderen.no[18] it is shown that this tower supports 800 MHz and 1,800 MHz 4G communication.

[18]See `https://finnsenderen.no/#/main` for towers in Norway.

Chapter 3

Method

With the groundwork of technical background the basis for understanding the method, results and discussion is now set.

In this chapter the method used to perform the tests is explained in detail. The chapter is divided into first a testing overview, which briefly explains the different scenarios. Next the baseline is described in detail. Lastly the test scenarios setup and environment is explained. Summarized this will give insight into how the test scenarios are built up.

3.1 Testing Overview

Seven different test scenarios will be conducted and analyzed. These test scenarios are video call meetings using either Webex or Teams. Each and every test scenario is different, but realistic. All of the meetings/testing scenarios are built upon the first test scenario, the baseline, see section 3.3.1. The reason for this is to avoid explaining each scenario from scratch. Instead, all test scenarios explains its differences from the baseline. In short the baseline is a typical videoconference between two clients, with no artificially introduced characteristics, like low bandwidth or/and high latency.

The basic setup of baseline is shown in figure 3.1, where there will be two PC's that from here on out will be referred to as PC-1 and PC-2. PC-1 uses Telenor as an ISP, while PC-2 uses Uninett as an ISP.

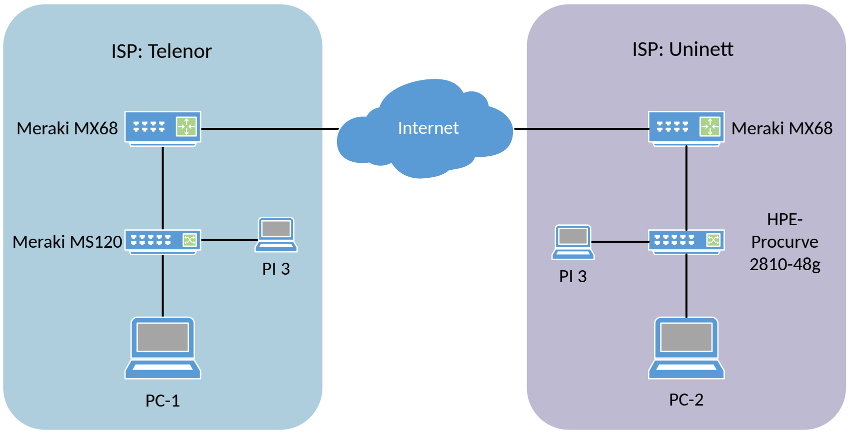


Figure 3.1: The topology for baseline. License: Andreas Rømo, CC BY.

Test scenario 1.2 uses the same setup as baseline, but introduce a predefined ADSL connection in the meeting, test scenario 1.3 also uses the same setup as baseline, but introduces an UDP block on PC-1 halfway into the meeting. See section 3.3.2 and 3.3.3 for further reading.

Test scenario 2.1 changes the setup on PC-1 where instead it will use a wireless connection. Test scenario 2.2 uses the same setup as test scenario 2.1, but will also simulate a bad wireless connectivity on PC-1. See section 3.3.4 and 3.3.5 for further reading.

Test scenario 3.1 uses the same setup as baseline, but will introduce a new client to the meeting. This client will use a student home's Ethernet connection and will be known as PC-3. PC-3 uses Telia as an ISP. See section 3.3.6 for further reading.

Test scenario 4.1 uses a 4G connection combined with Wi-Fi on PC-2, instead of an Ethernet connection. See section 3.3.7 for further reading.

3.2 Baseline

The previous section explained in short the different test scenarios in relation to the baseline. In this section the baseline is described.
The section is split into baseline environment, baseline procedure and extracting data. The subsection for environment gives a generalized view on how the environments in the different applications will be set up, e.g. how the Meraki base environment will be before each test, same for Webex and MS Teams. The subsection baseline procedure will describe how the baseline of the tests is performed, e.g., setting up the equipment, test environment, audio and webcam, the meeting

layout, etc. The subsection for extracting data will describe how data will be documented and stored for examination.

3.2.1 Baseline environment

When the tests are ran they will all be based on the same fundamental environment. Some of the test scenarios will tweak the environment, but that will be specified for each test. The environment is explained by how the equipment is configured and setup, how tools are configured and how clients are configured.

3.2.1.1 Equipment

In this bachelor project the following Meraki network equipment will be used: There will be used two Meraki MX68[1] Cloud-Managed Security and SD-WAN, one Meraki MS120-8[2] switch, one Meraki MR36[3] access point, and one Meraki MG21E[4] 4G router. An additional HPE ProCurve 2810-48g switch will be used for networking on the Uninett location (NTNU Campus, Gjøvik).

As seen in figure 3.1 the Uninett side consists of a Meraki MX68 and HPE Procurve 2810-48g. The MX68 router will be connected to the ISP outlet with an Ethernet cable. All of the Ethernet cables used in the topology is going to be CAT 5e or better. The MX68 router is connected with the HPE Procurve, which again is connected to the client and the Raspberry Pi. When all the equipment is setup in regards to figure 3.1, an equipment health-overview can be checked on the dashboard.

On the Telenor site the MX68 will be connected to a regular home router with an Ethernet cable. At the same time it will be connected to the MS120-8LP switch. The client is then able to use the Internet by connecting to one of the switch ports. Additionally there will be a Raspberry Pi connected to the switch and client, just like the Uninett side.

- PC-1 and the equipment will be located at, Tollerudvegen 1, 2827 Gjøvik (Telenor).
- PC-2 and the equipment will be located at, Teknologivegen 22, 2815 Gjøvik (Uninett).

[1]See `https://meraki.cisco.com/product/security-sd-wan/small-branch/mx68/` for Meraki MX68's sales page.
[2]See `https://meraki.cisco.com/product/switches/access-switches/ms120-8/` for Meraki MS120-8's sales page.
[3]See `https://meraki.cisco.com/product/wi-fi/indoor-access-points/mr36/` for Meraki MR36's sales page.
[4]See `https://meraki.cisco.com/product/cellular/external-antenna/mg21e/` for Meraki MG21E's sales page.

3.2.1.2 Meraki base environment

In Meraki Dashboard there will be created two networks; *Studentnett - hybel* (Telenor) and *Studentnett - skole* (Uninett). The two networks are needed because it can't handle two Meraki MX68 routers in the same network and to differentiate the network traffic. Therefore it is required to have two virtual networks [77]. The MX68's will be named differently for each location. The MX68 for the Telenor location will be named "MX68-Stud-1" and "MX68-Stud-2" for Uninett.

Each of the networks will be located at two different, but static locations during the testing scenarios. A static location provides an advantage by giving each test scenario equal preconditions. Comparing the test scenarios becomes easier as they will have an equal starting point. The Meraki equipment on these locations must be added to their respective networks. The networks parameters such as bandwidth, packet capture, port configuration and group-policies will be changed accordingly to the test scenarios.

On the SD-WAN, Meraki MX68, the following features will be disabled; traffic shaping and load balancing. Doing this prevents traffic prioritization of one platform over the other. None of the test scenarios will use any load balancing between ISPs as there is only a single ISP available on each location. For all the test scenarios there will also be enabled detailed traffic analysis. This gives access to detailed information about the network traffic and hosts in the Meraki Dashboard. The default setting[5] for DHCP is used and the DNS server is set to be Google public DNS[6] 8.8.8.8 and 8.8.4.4 on both of the Meraki MX68 routers. By using a matching public DNS for both clients it is likely to give the same IP for a domain and therefore routing to a destination has a higher chance of going the same route.

The switch, Meraki MS120, will be using VLAN 1 and its default setting. Only exception from default setting is port mirroring. On port one there is a single uplink to the router, Meraki MX68. Port two is a mirror of port three, where PC-1 is connected. By activating port mirroring on port two it will act as a network tap for PC-1. On port two there will be a device running ntopng, which is a network analytics software. The MS120 will be used on the Telenor site.

Both locations will set the bandwidth (from 450 Mbps upper limit) to 100 Mbps in download and upload, this is to keep the bandwidth close to identical on both testing sites. Limiting the bandwidth to 100 Mbps in download and upload should have no impact in the data as both MS Teams and Webex requires far less bandwidth [45][46].

[5]See `https://documentation.meraki.com/MX/DHCP/DHCP_Services` for manual on DHCP service.

[6]See `https://developers.google.com/speed/public-dns/` for more information about the public DNS.

3.2.1.3 Port mirroring and ntopng

When performing the test scenarios, each switch on both ends of the locations will be configured with port mirroring. This information will be collected in real-time on the Raspberry Pi's running ntopng software. At the Uninett location there will be a HPE Procurve 2810-48g[7] switch that will be configured like the Meraki MS120 in regards to port setup.

Ntopng version 2.4.170204 is installed on the operating system Raspbian[8], a Debian-based operating system for Raspberry Pi [78].

The ntopng software can be used for network traffic analysis. There will be use of an older version of the software because of limitation to capture maximal 25000 packets and non-free tiers on newer versions of the software. The traffic collected on both locations will be bandwidth usage on the subnetwork, 192.168.128.0/24.

3.2.1.4 Webex base environment

For all the test scenarios only the Windows client (native) for Webex will be used. There is also a web client available, but it is not optimized since the WebRTC client does not support the API's used to feed detailed media metrics to Webex Control Hub. This is confirmed by a Cisco employee.

In the Windows client all the default settings will be used, except for the "Remove background noise" option which should be disabled to pick up sound.

3.2.1.5 Microsoft Teams base environment

For all the test scenarios only the Windows client (native) for Teams will be used. In the admin panel[9] for Teams QoS should be disabled.

3.2.1.6 Client environment

To reduce traffic interference received by analytics tools all unnecessary applications are shut down on the PC's. The only application that needs to be open is the respective video communication application. All clients run on Windows 10 and must be on the same build, in regards to application and operating systems patches.

[7] See `https://h20195.www2.hpe.com/v2/getpdf.aspx/c04140686.pdf?ver=5` for the manual.
[8] see `https://www.raspberrypi.org/software/` for Raspbian's homepage.
[9] See `https://admin.teams.microsoft.com/` for admin panel on Teams (require login).

3.2.2 Baseline procedure

With the baseline environment detailed the next step is to explain the procedures for conducting the tests.

3.2.2.1 Network procedure

Before a video call can start there will be conducted three speed tests evaluating the capacity of the network on nettfart.no[10], where the average of all the tests will be used for further insight in the scenario. It is important that the three speed tests are consistent in value so that the tests can run within reasonably equal conditions. In a long term view the average values must also be consistent, as to not be influenced by Internet variances when comparing tests.

Another connectivity test will be run towards Cisco Webex and Microsoft teams. For Teams the connectivity.office.com[11] test page will be used to record latency, jitter and packet loss for UDP and give feedback about recommended values. At Webex the mediatest.webex.com[12] test is used to determine if values like jitter, latency and packet loss on both UDP and TCP are within the recommended values. These tests are only run once before a test scenario as its main purpose is showing reachability to the tenants. To ensure that Webex and Microsoft Teams services are operational and are not having any network/service issues, the status of each platform will be checked on status.webex.com[13] and admin.microsoft.com[14] before each test scenario.

After the speed tests are completed a bash script on the Raspberry Pi's will be run after each testing iteration. This is used to restart and delete temporary files for ntopng. This gives each test iteration a clean dataset to view and extract data from.

3.2.2.2 Participants

There will be two participants, PC-1 and PC-2. The participants in the meeting rates the quality of the video, audio and seamlessness from 1 to 6 where 6 is perfect and 1 is terrible. This is based on anecdotal evidence.

In table 3.1 the hardware of the clients is shown.

[10]See `https://nettfart.no` for web page.
[11]See `https://connectivity.office.com` for web page.
[12]See `https://mediatest.webex.com/#/main` for web page.
[13]See `https://status.webex.com/service/status` for web page.
[14]See `https://admin.microsoft.com/AdminPortal/Home#/servicehealth` for web page (require login).

	PC-1	**PC-2**
CPU	Intel Core i7 7700HQ @ 2.80 GHz	Intel Core i5-8330H @ 2.3GHz
RAM	16GB Single Channel @ 2400MHz	8GB Single Channel @ 2400MHz
OS	Windows 10 Home 64-bit	Windows 10 Home 64-bit
Network card	Realtek PCIe GBE	Realtek PCIe GBE @ 2.5G
Wi-Fi network card	Intel Dual Band Wireless AC-8260	Intel Wireless-AC 9560
Ethernet port speed	1Gbps	1Gbps
Webcam	USB2.0 HD UVC Webcam (720p @ 30FPS)	Logitech QuickCam Pro 5000 (480p @ 30FPS)

Table 3.1: Table of the specification of PC-1 and PC-2.

3.2.2.3 Audio and Webcam

Both the participants will enter with the webcam, audio and microphone enabled and run the application in fullscreen. Participants will play audio during the meeting. There will be no sharing of screen or presenting on the call. The audio will be played from a mobile phone and the other participant must be able to hear the audio clearly. The audio for all meeting participants will be of an English dialog between two people[15]. To simulate movement and generate P- and/or I-frames a glitter lamp will be used, placed in the middle of the frame. The reason for acquiring a lamp is to replace human movement. The audio and webcam implementations are to make a consistent repeatable scenario for use in all the test scenarios to rule out any inconsistencies in that regard. Below image 3.2 shows the layout for Webex and the glitter lamps that is used for for both platforms.

[15] See `https://www.youtube.com/watch?v=lhFU5H5KPFE` and `https://www.youtube.com/watch?v=qwaSwHEB-EY` for both videos.

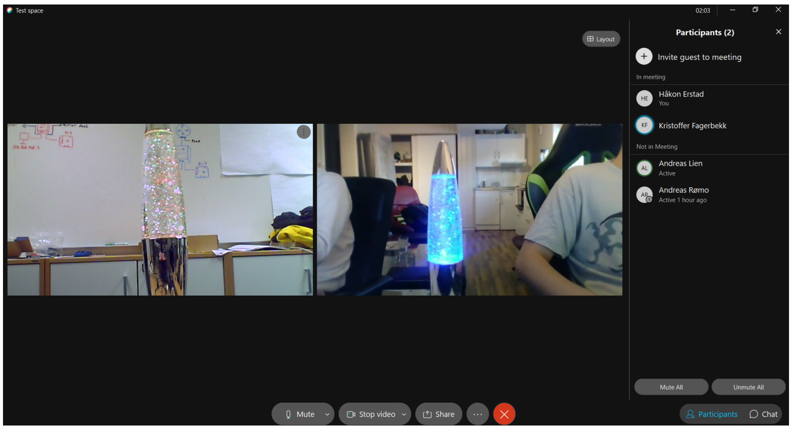


Figure 3.2: The image shows a typical videoconferencing call for a test scenario. License: Cisco Meraki, CC BY.

3.2.2.4 Meeting Layout

PC-1 which has Telenor as ISP will always host the meeting. On MS Teams the host additionally need to schedule a meeting with all the participants in a specific time period. If the meeting is not scheduled, a meeting between two participants could become an point-to-point connection. When that happens traffic might not go through Microsoft Teams servers and the data won't be presented on the Teams admin panel [79].

Baseline meetings will last four minutes to generate enough data for analytics tools, but also to not generate more data than necessary.

The Uninett participant (PC-2) will join the meeting 10 seconds after the host (PC-1) has joined the meeting. When the host joins/starts the meeting, audio will be played for 1 minute by the host. At the 1 minute mark the other party will start playing audio and the host will stop playing audio. This switching of audio playing happens at each 1 minute mark. The host leaves the meeting after four minutes while the other participant will wait 10 seconds then leave. The 10 second gap is just for having a procedure which is easy to follow.

Meetings will at least be repeated three times in a row for each tenants test scenario. This to ensure that all three iterations don't have faulty information or is missing data.

When all is said and done, a baseline has been created for the test scenarios. Changes will be made to fit the different test scenarios, but these are specified later in the description of the specific test scenarios.

3.2.3 Baseline extracting data

When data is produced from a test the next step is to extract it. However, each platform have their own unique way of displaying data, so it will be important to sort out the information accordingly to the scope of the thesis. The different methods on how and what data to extract from each platform is listed below.

3.2.3.1 Meraki Dashboard

To get insight on ports, protocols, hosts and traffic flow, there will be conducted packet capturing of the traffic on clients that are connected to the network on the Meraki cloud. Packet capturing in Meraki Dashboard is shown in figure 3.3 below.

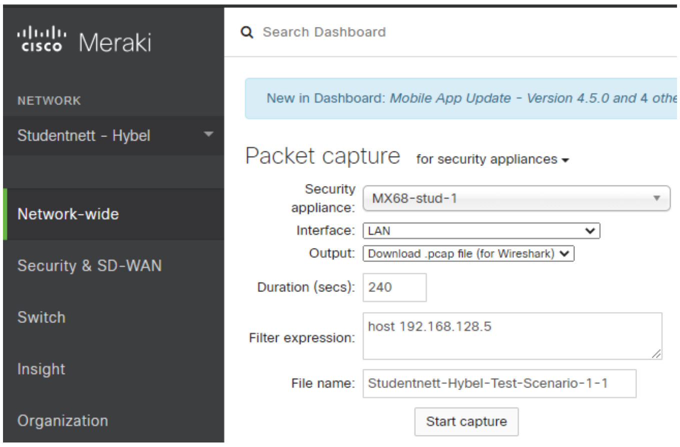


Figure 3.3: Packet capture functionality in Meraki Dashboard. License: Cisco Meraki, CC BY.

3.2.3.2 Webex Control Hub

Webex Control Hub provides information with a timeline, like shown in figure 3.4, that displays information regarding the network connectivity with an overview of different network parameters. In the Control Hub there is an overview of the meeting(s), but also a possibility for a more detailed overview of each participant.

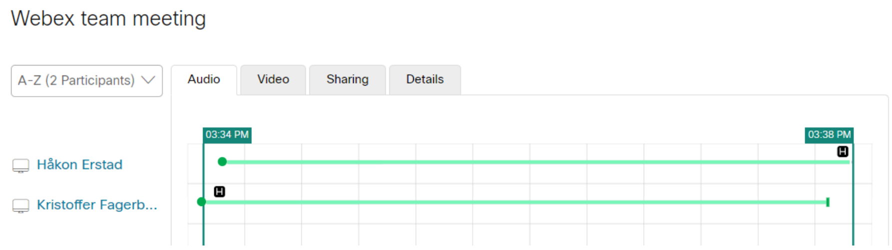


Figure 3.4: A representation of the timeline found in the troubleshooting tab of Webex Control Hub. License: Andreas Rømo, CC BY.

There will be taken multiple screenshots from each meeting. A screenshot will be taken of the overview of "Audio", "Video" and "Details". The three overview tabs for "Audio", "Video" and "Details" are also visible on figure 3.4.
If the timeline for "Audio" or "Video" shows signal-quality below "Good" (not green), it will be necessary to take a screenshot of the particular area by hovering the mouse pointer over that area. Information that is collected while hovering is jitter, packet loss and latency for that specific time frame, an example of this can be seen in figure 3.5.

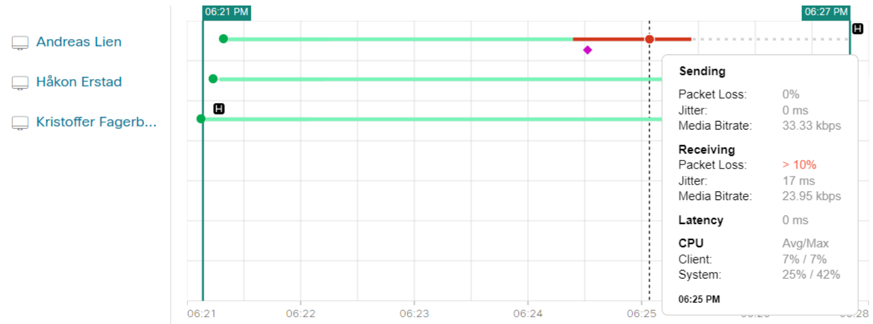


Figure 3.5: Information given by hovering the timeline. License: Andreas Rømo, CC BY.

Also if the timeline shows a purple indicator, a screenshot is taken while hovering over that indicator. This will show the configuration change, such as change in the IP address, protocol type and connection. An example where a protocol change happened can be viewed at figure 3.6.

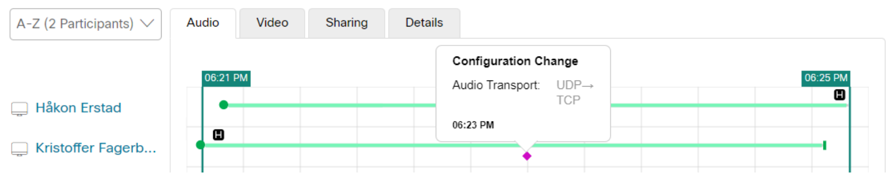


Figure 3.6: Information given by hovering the purple indicator on the timeline. License: Andreas Rømo, CC BY.

Webex Control Hub has the option to download the meeting statistics in JSON format. A JSON file will be downloaded from every meeting that has been held. Information like RTT, jitter, FPS, bit rate and packet loss will be gathered from the JSON file. As there is little to no data being sent for each client, if not specified otherwise only the received data will be extracted and looked at. Except for packet loss where sent traffic will be analysed.

3.2.3.3 Microsoft Teams

When extracting data from Teams there are a set of predefined functions that can be used to gather data [1]. Tools as Power BI and Microsoft Powershell module both use these functions to query data. Powershell is faster than Power BI at requesting data, but Power BI has built in capabilities to represent data in graphs. Therefore Power BI is the tool used for obtaining data like network parameters.

In Power BI each stream of audio and video is presented with information regarding; stream direction (client to server or vice versa), the streams' average packet loss, average max packet loss, average round-trip time, average max round-trip time, avg video frame-rate, conference id, stream type (audio or video) and time and date. All information will be exported to a CSV file.

All the data collected is defined on Microsoft's Dimensions and measurements for call quality page [16]. Table 3.2 below, explains the different stream parameters for upcoming results analysed.

Type	Explanation	Prefferable value
Avg RTT	Average round-trip time calculated in milliseconds	Low
Avg jitter	Average jitter calculated in milliseconds	Low
Avg Packet-Loss	Average packet loss	Low
Avg. Video Frame Rate	Average Frames Per Second (FPS) in a range	High
Media	Type of stream, Audio or Video	N/A
Stream Direction	Direction of stream, Client → Server or Server → Client	N/A
Transport	Layer 4 protocol used	N/A
Classified poor call	Tells if the call was classified poor based on a set of values	N/A

Table 3.2: Explanation for stream parameters looked at for meetings held on MS Teams.

3.2.3.4 Ntopng

Ntopng provides information of all the packets received and sent on the network that it is listening on. The information is visualised as pie charts, graphs and JSON file.

[16]See `https://docs.microsoft.com/en-us/microsoftteams/dimensions-and-measures-available-in-call-quality-das` for all dimensions.

To ensure that the meeting went as expected when applying different group policies, ntopng will be used to monitor real-time traffic.

Traffic that will be collected on both locations is bandwidth usage on the network, protocols used and packet loss (TCP).

3.3 Test Scenarios

Till now this chapter has informed about the environment for the baseline, the meeting procedures for the baseline and data extracting for the baseline.

In this section all performed test scenarios are explained by adding or removing elements from the baseline.

3.3.1 Test scenario 1.1

There will be no changes to the baseline used in test scenario 1.1, see figure 3.7 for network topology.

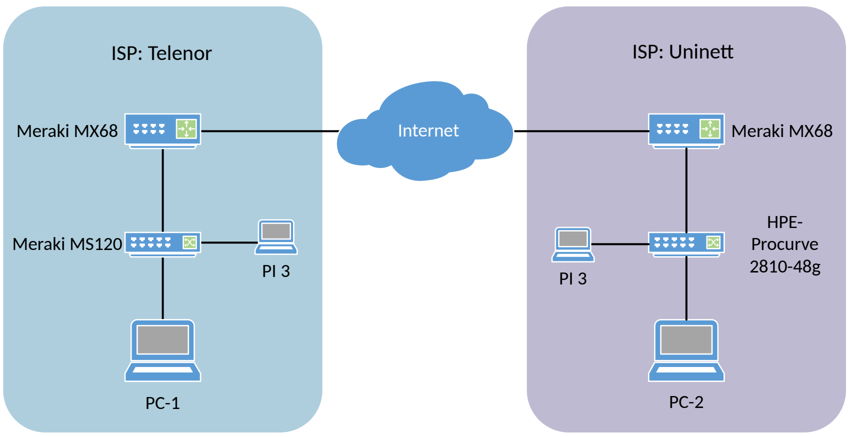


Figure 3.7: Topology for test scenario 1.1. License: Andreas Rømo, CC BY.

3.3.2 Test scenario 1.2

The only change from the baseline is for PC-1, where there will be introduced an ADSL link of 10/2 Mbps for the last 2 minutes. See figure 3.8 network topology.

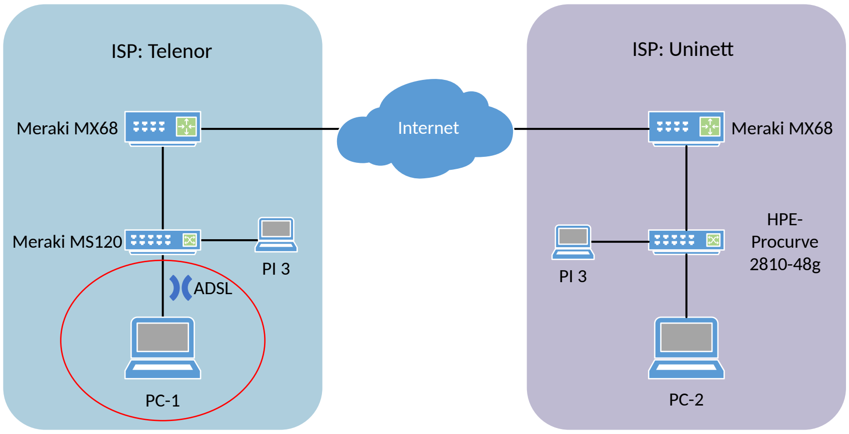


Figure 3.8: Topology for test scenario 1.2. License: Andreas Rømo, CC BY.

3.3.3 Test scenario 1.3

It will be identical to the baseline setup, where the only change is an UDP block for PC-1. The blockage will be introduced 2 minutes into the meeting. See figure 3.9 network topology.

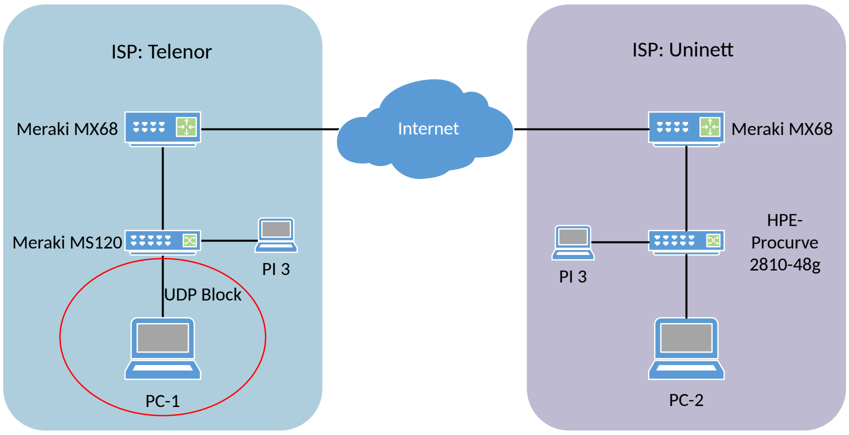


Figure 3.9: Topology for test scenario 1.3. License: Andreas Rømo, CC BY.

3.3.4 Test scenario 2.1

In test scenario 2.1, an Access point (AP) will replace the Ethernet cable for PC-1 about 2 minutes into the meeting, as shown in the network topology figure 3.10.

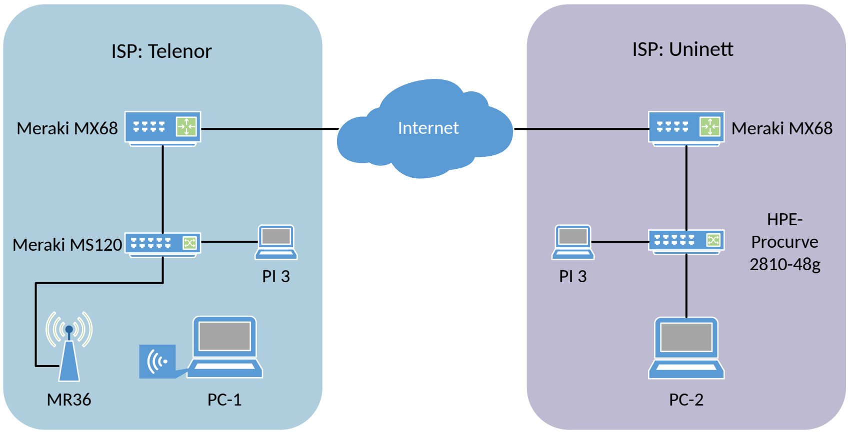


Figure 3.10: Topology for test scenario 2.1. License: Andreas Rømo, CC BY.

Only the frequency 5GHz will be used in test scenario 2.1 as this is the frequency with the highest possible bandwidth and highest channel capacity. A password has also been set for the AP to restrict general access and only give access to the client that needs to use the wireless connection.

There will be conducted a site-survey with the appropriate settings. The site-survey will be conducted following Cisco Meraki's official guide.[80]

Two scripts were made to monitor the connection between the AP and PC-1. These scripts will monitor; Signal strength using Powershell on the PC-1. Signal-to-Noise Ratio (SNR) for the AP (MR36) using the AP Web-GUI.

3.3.5 Test scenario 2.2

Test scenario 2.2 will be identical to test scenario 2.1, an AP will replace the Ethernet cable in the middle of the meeting, as shown in the network topology figure 3.11. The AP will be wrapped in aluminium's foil and be placed as far away as possible to simulate poor Wi-Fi.

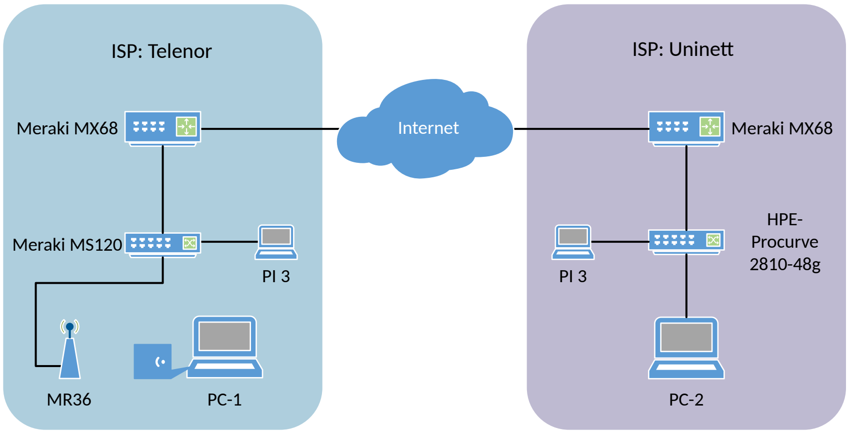


Figure 3.11: Topology for test scenario 2.2. License: Andreas Rømo, CC BY.

3.3.6 Test scenario 3.1

The meetings will have three participants with a total duration of 6 minutes. PC-1 and PC-2 will be identical to baseline, but there will be introduced a third PC, PC-3. As shown in the network topology figure 3.12. PC-3 will be playing the video "First Aid Fail - The Office US"[17]. The specification of PC-3 is shown in table 3.3. PC-2 will swap to its integrated laptop camera, as PC-3 did not have a integrated camera. PC-2's new camera is called "EasyCamera" supports up to 720p @ 30 FPS.

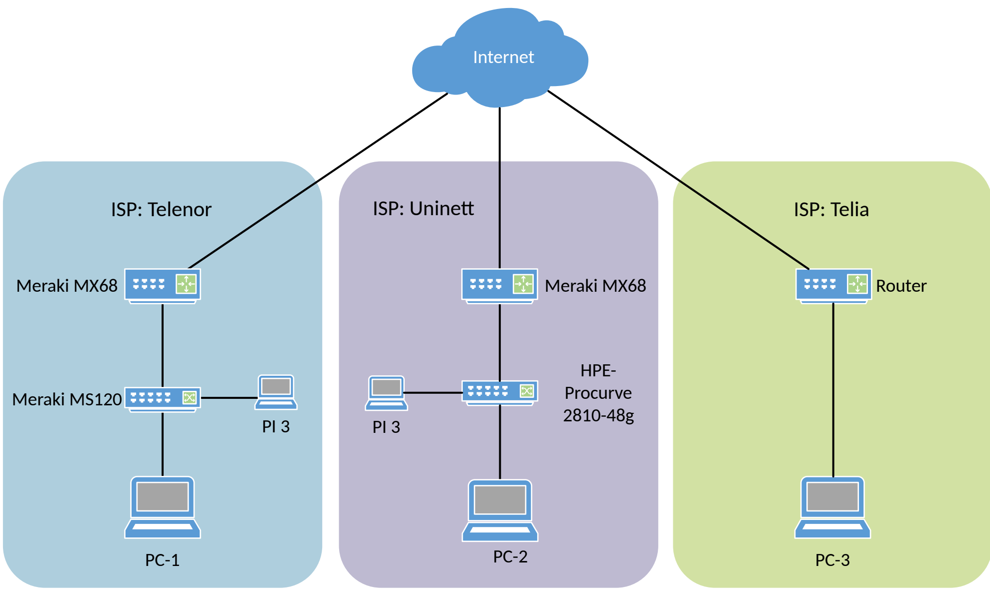


Figure 3.12: Topology for test scenario 3.1. License: Andreas Rømo, CC BY.

[17]See `https://www.youtube.com/watch?v=Vmb1tqYqyII` for video.

	PC-3
CPU	Intel Core i5-3320M @ 2.60 GHz
RAM	8GB Duel Channel @ 1333MHz
OS	Windows 10 Education 64-bit
Network card	Intel 82579LM Gigabyte Network Connection
Wi-Fi network card	Intel Dual Band Wireless AC-7260
Ethernet port speed	1Gbps
Webcam	Logitech QuickCam Pro 5000 (480p @ 30FPS)

Table 3.3: Table of PC-3 specification.

3.3.7 Test scenario 4.1

Test scenario 4.1 will be close to baseline with two participating. PC-2 will use a AP and using a mobile broadband modem (4G), Meraki MG21, as shown in the the network topology figure 3.13. The AP will be configured identical to test scenario 2.1 and the mobile broadband will use Google's DNS.

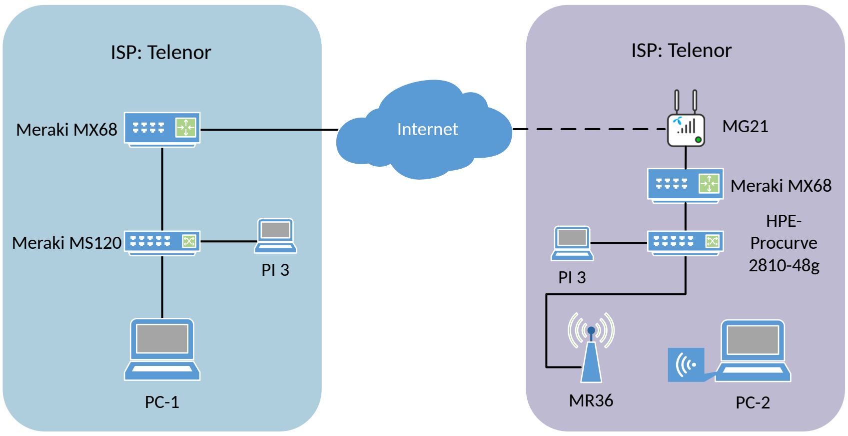


Figure 3.13: Topology for test scenario 4.1. License: Andreas Rømo, CC BY.

Chapter 4

Result and Discussion

The previous chapter Method showed the baseline and its properties, while also describing the test scenarios performed. In this chapter each performed test scenario has been presented with the relevant data collected and analysis of that data. All test scenario sections contains an objective, results and discussion.

The objective describes how to detect the quality of a meeting and what is to be discussed. Analysis presents data and compares the data. The discussion section will elaborate on the quality from the network behavior in relation to the data.

From this chapter the meeting quality for the scenarios is evaluated.

4.1 Test scenario 1.1

Test scenario 1.1 is conducted to simulate a videoconference call between two participants, where both clients are using an Ethernet cable. The scenario reflects an everyday videoconference call where both participants have sufficient quality to cover the requirements from MS Teams and Webex.

4.1.1 Objective

By conducting this test, qualities measured such as frame type (I or P), FPS, jitter, latency, packet loss, protocol usage and bandwidth usage are investigated, and put in relation to the quality documents for Webex (section 2.1.12.4) and MS Teams (section 2.1.12.5). For MS Teams bandwidth requirements see table 2.2 and for Webex see 2.3.

From the results, network behavior is discussed in relation to quality. This gives indication on the meetings quality. This is repeated for every test scenario.

For Webex, PC-1 and PC-2 are compared between each other to observe variations in the aforementioned qualities.

As for MS Teams there are two tenants and therefore the comparison is instead focused on the measured qualities between the tenant's. Comparison between participants for a tenant is excluded as this would heavily increase the work load.

4.1.2 Results and Discussion

Below is the results and discussion for Webex and Teams. The results comes from data extraction procedure and is analyzed. All of the analyzed data is then discussed in regards to the objective. This structure is the same for all coming test scenarios.

4.1.3 Results - Webex 1.1

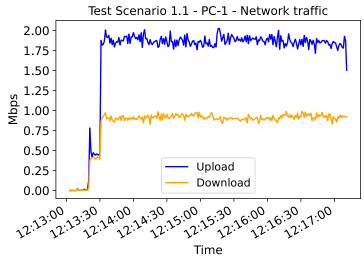
Results from the Webex network test (section 3.2.2) is shown in table 4.1. The values shows a network test to the Webex servers taken before meeting start.

The difference in TCP delay between PC-1 and PC-2 is 45.20 ms. The UDP delay had a difference of 16 ms. Cisco evaluates all delays <=100ms as good and 100 300 ms as fair. The TCP download speed had a difference of 80.15 Mbits/s. The TCP upload speed had a difference of 31.95 Mbits/s. Cisco evaluates all bandwidth >=2 Mbps as good. Both the UDP download and upload lossrate was 0% for PC-1 and PC-2. Cisco evaluates all lossrate <=1% as good [81].

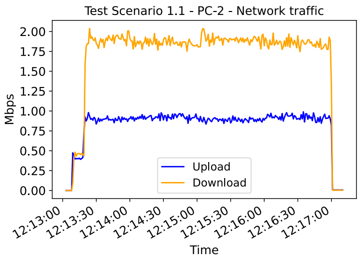
	PC-1	**PC-2**
TCP Delay	93.65 ms	138.85 ms
TCP Download Speed	94.67 Mbits/s	14.52 Mbits/s
TCP Upload Speed	44.58 Mbits/s	12.63 Mbits/s
UDP Delay	85.00 ms	101.00 ms
UDP Download LossRate	0.00%	0.00%
UDP Upload LossRate	0.00%	0.00%
UDP Download Speed	4.61 Mbits/s	3.65 Mbits/s
UDP Upload Speed	5.01 Mbits/s	3.74 Mbits/s

Table 4.1: Table of Cisco Webex Network Test for PC-1 and PC-2.

PC-1's upload is around 1.8-2.0 Mbps with an average of 1.68 Mbps, and a download of around 900-1000 Kbps with an average of 820 Kbps. PC-2 's upload is around 900-980 Kbps with an average of 828 Kbps, and the download around 1.7-2.0 Mbps with an average of 1.68 Mbps, as shown in graphs 4.1a and 4.1b.



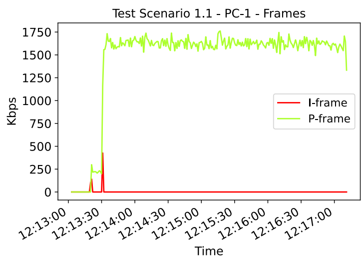
(a) Throughput PC-1 (UDP)



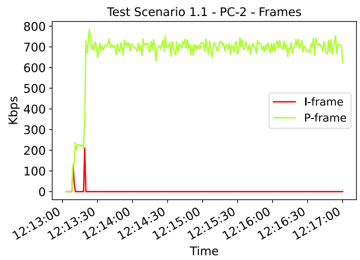
(b) Throughput PC-2 (UDP)

Figure 4.1: Data connected to Webex meeting: 1.1.6 [A.10]. License: Andreas Kilde Lien, CC BY.

The graphs A.6 and 4.2b shows the dispersion of I- and P-frames for PC-1 and PC-2 throughout the meeting. PC-1 is having a bit rate of I-frames around 1500 to 1750 Kbps, while PC-2 has a steady flow of around 650 to 780 Kbps. There is only some I-frames for the first 20 seconds of the meeting for both PCs. The meeting is mainly distributed by P-frames.



(a) The upload dispersion of I- and P-frames for PC-1. The graph is including L3 overhead.



(b) The upload dispersion of I- and P-frames for PC-2. The graph is including L3 overhead.

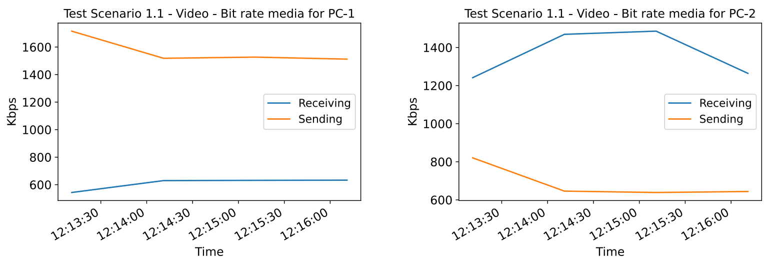
Figure 4.2: Data from Webex meeting: 1.1.6 [A.10]. License: Andreas Kilde Lien, CC BY.

Both locations uses UDP as a primary transport protocol for the videoconference. PC-1 uses UDP 99.53% and TCP 0.47%, this is roughly the same for PC-2, as seen in table 4.2.

Client	Protocol	Sent	Received	Total
PC-1	TCP	179.99 KB	176.26 KB	356.25 KB (0.47%)
	UDP	49.81 MB	24.58 MB	74.39 MB (99.53%)
PC-2	TCP	197.67 KB	204.63 KB	402.31 KB (0.55%)
	UDP	24.52 MB	46.72 MB	71.24 MB (99.45%)

Table 4.2: Table to show protocols used by PC-1 and PC-2 under the meeting. Data connected to Webex meeting: 1.1.6 [A.10].

Webex Control Hub shows that PC-1 has a higher bit rate with between 1512 to 1715 Kbps sent compared to PC-2's 638 to 820 Kbps. The received bit rate on PC-1 ranges from 543 to 632, while PC-2 ranges from 1241 to 1485 Kbps, see graphs 4.3a and 4.3b. The reason for the graphs being a minute shorter then the actual meeting, is because the JSON file that Control Hub provide does not include the last minute.

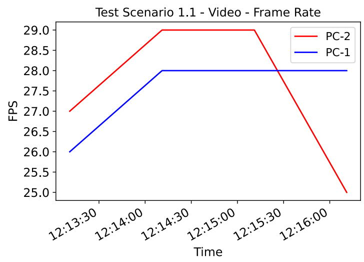


(a) Bit rate for PC-1. **(b)** Bit rate for PC-2.

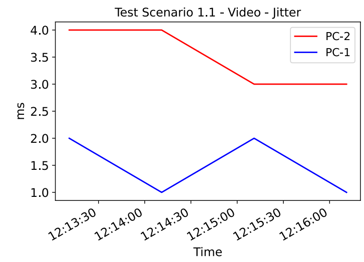
Figure 4.3: Data from Webex meeting: 1.1.6 [A.10]. License: Andreas Kilde Lien, CC BY.

The graphs shown in 4.4 show extracted values for the Webex meeting in test scenario 1.1, these values can be put in relation to the Webex requirements to determine if the meeting had good or poor quality.

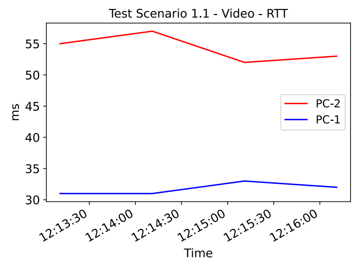
The frame rate throughout the meeting is shown in graph 4.4a and shows PC-1 ranges between 25 to 29 FPS, while PC-2 ranges between 23 to 28 FPS. For the video jitter shown in graph 4.4b, PC-1 ranges between 1 and 2, while PC-2 ranges between 3 and 4. The video RTT shown in graph 4.4c ranges between 31 and 33 on PC-1, while on PC-2 it ranges between 52 and 57 RTT. The packet loss for both video and voice shown in graphs 4.4d and 4.4g on both PC's remain at 0% throughout the meeting. The voice jitter shown in graph 4.4e shows PC-1 going from 6 to 7, while stays at 6 for PC-2. The voice RTT shown in graph 4.4f for PC-1 ranges from 22 to 29, while PC-2 ranges between 42 to 48. All the values are within the requirements stated in section 2.1.12.4.



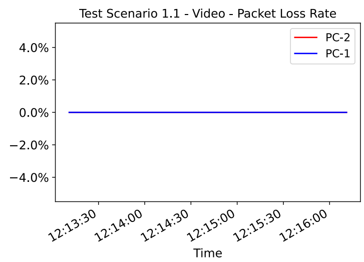
(a) Video - Frame rate.



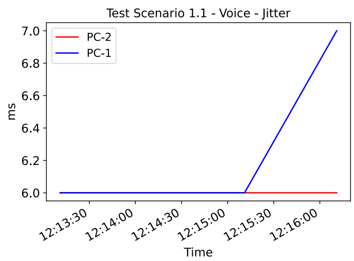
(b) Video - Jitter.



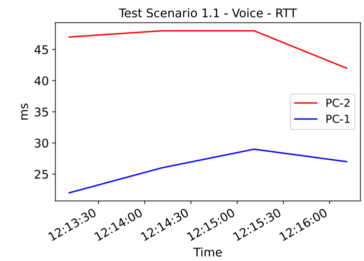
(c) Video - RTT.



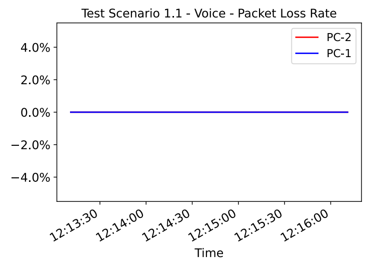
(d) Video - Packet loss rate.



(e) Voice - Jitter.



(f) Voice - RTT.



(g) Voice - Packet loss rate.

Figure 4.4: Data from Webex meeting: 1.1.6 [A.10]. License: Andreas Kilde Lien, CC BY.

4.1.4 Discussion - Webex 1.1

In the results it was presented two different graphs for bit rate, one from Wireshark (4.2) and one from the Webex Control Hub (4.3). From the graphs it is seen that the recorded bandwidth for the PC's are unequal. It is assumed the recorded bandwidth from the Control Hub excludes headers in its calculations, while

Wireshark records all layers and gives a more accurate picture of the bandwidth utilization. It is hard to say if the bandwidth requirements specified by Webex (section 2.3) is in relation to the data in the Control Hub or Wireshark. For now it is presumed that the requirements relates to the results from Wireshark.

There is quite a difference in what is sent of bit rate from PC-1 and what is sent from PC-2 in figure 4.3. A possible explanation is that the difference is a byproduct of the difference in webcam quality as PC-1 uses a 720p30 webcam and PC-2 a 480p30 webcam.

Looking at the protocols used in table 4.2, the meeting used UDP primarily. There is still a small amount of TCP used this is likely caused by the Webex client sending data transfers and loading HTTPS using TCP. Webex uses UDP primarily for DNS lookups aswell, but can use TCP so this may also be a possible cause for the TCP usage [82].

Looking through the other values presented, PC-1 has lower average values on most of the graphs. Most of these values with the exception of FPS are values that are recommended to be as low as possible. This means that the PC-1 performs better in these meetings contrary to PC-2. Part of the explanation for worse network parameters is the difference in where the traffic goes from each client to the server. PC-1 goes directly through Telenor with its peering to Webex, while PC-2 takes a detour first up to Trondheim (Uninett) and then to the Webex server. This is believed to be the cause of the higher RTT.

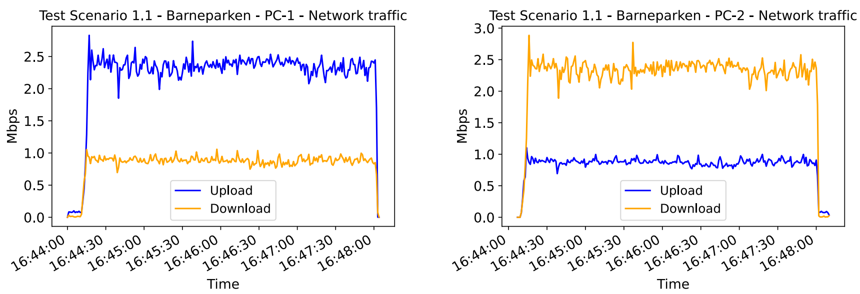
The average latency is still under the threshold for what is seen as poor latency which is around 100 ms or more for voice and 150 to 300 ms or more for video as stated in section 2.1.12.4. The video and voice jitter for both the PC's is also well below the threshold for poor jitter which is also stated in section 2.1.12.4.

The frame types used under the meeting shows two spikes with I-frames at the start of the meeting, one a little bigger than the other. The reason for the difference in size may be because of the rescaling of the videos in the meeting. When joining the meeting only the client's own video will be shown until it receives the data for the other client's video and rescales again to fit both video streams on the screen. After the last spike of I-frames only P-frames are used through the rest of the meeting.

The glitter lamp limits motion to a small area in the middle of the screen and this will not create enough movement to warrant a new I-frame. In a "normal" videoconferencing meeting between people it is normal that the mouth is a object that moves mostly in the middle of the screen and that the rest is mainly static much like the glitter lamp [83]. The test setup therefore resembles more or less a normal videoconferencing meeting between people on a good network connection.

4.1.5 Results - Teams 1.1

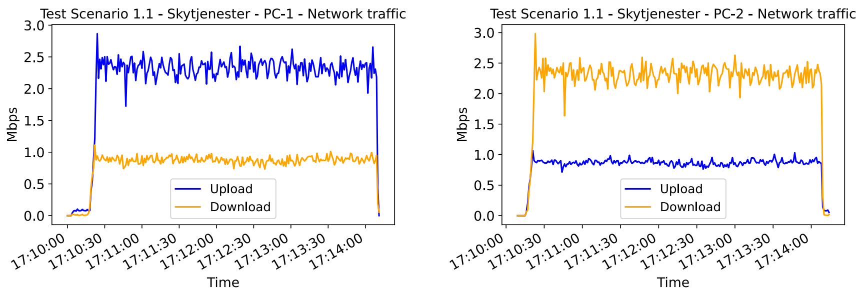
The meeting for *barneparken* was hosted in a data center located in Amsterdam, Netherlands. Where PC-1 had an average upload of 2.38 Mbps and an average download of 0.88 Mbps. PC-2 had an average upload of 0.87 Mbps and download of 2.34 Mbps. This is can be seen in the graph 4.5. The average values are calculated from the first spike to the beginning of the drop at the end.



(a) Throughput for PC-1 (UDP) **(b)** Throughput for PC-2 (UDP)

Figure 4.5: Data connected to Teams meeting, *barneparken*: 1.1.5 [A.9]. License: Andreas Rømo, CC BY.

The meeting for *skytjenester* was hosted in a data center located in Dublin, Ireland. The upload throughput for PC-1 had an average of 2.31 Mbps and download of 1.1 Mbps, while PC-2 had an average of 2.3 Mbps in download and 0.86 Mbps in upload. The average is calculated from the graph 4.6.



(a) Throughput for PC-1 (UDP) **(b)** Throughput for PC-2 (UDP)

Figure 4.6: Data connected to Teams meeting, *skytjenester*: 1.1.4 [A.9]. License: Andreas Rømo, CC BY.

Below is table 4.3 showing the TCP and UDP distribution for PC-1 and PC-2, during the meeting held on *barneparken*. As shown in the table, PC-1 used UDP as the main transport protocol with 99.48% and PC-2 used 99.3% under this meeting.

Client	Protocol	Sent	Received	Total
PC-1	TCP	277.11 KB	191.28 KB	468.38 KB (0.52%)
	UDP	64.18 MB	22.81 MB	86.99 MB (99.48%)
PC-2	TCP	383.42 KB	250.25 KB	633.67 KB (0.7%)
	UDP	24.15 MB	63.95 MB	88.1 MB (99.3%)

Table 4.3: Data connected to Teams meeting, *barneparken*: 1.1.5 [A.9].

For *skytjenester*, PC-1 utilizes the UDP protocol 98.55%, while PC-2 utilizes the UDP protocol 98.4%. Table 4.4 shows the distribution of the TCP and UDP protocol in the meeting.

Client	Protocol	Sent	Received	Total
PC-1	TCP	345.97 KB	954.31 KB	1.27 MB (1.45%)
	UDP	63.56 MB	22.74 MB	86.31 MB (98.55%)
PC-2	TCP	431.04 KB	1.02 MB	1.42 MB (1.6%)
	UDP	24.05 MB	63.33 MB	87.38 MB (98.4)

Table 4.4: Data connected to Teams meeting, *skytjenester*: 1.1.4 [A.9].

In tables 4.5 and 4.6 each row is a stream that either goes from server to client or client to server. All the parameters are explained in section 3.2.3.3. Average RTT for PC-1, held on *barneparken*, was 26 ms and PC-2 had an average RTT of 36 ms. Average packet loss for both clients were 0. Jitter for PC-1 and PC-2 on audio averaged on 1 ms, while video jitter averaged on 0 ms.

Client	Media	Stream Direction	Avg. RTT	Avg. packet loss	Avg. jitter
PC-1	Audio	Server to client	26ms	0%	1ms
PC-1	Audio	Client to server	26ms	0%	1ms
PC-2	Audio	Server to client	36ms	0%	1ms
PC-2	Audio	Client to server	36ms	0%	0ms
PC-1	Video	Server to client	26ms	0%	0ms
PC-1	Video	Client to server	26ms	0%	0ms
PC-2	Video	Server to client	36ms	0%	0ms
PC-2	Video	Client to server	36ms	0%	0ms

Table 4.5: Data connected to Teams meeting, *barneparken*: 1.1.5 [A.9].

Shown in table 4.6 average RTT for PC-1, held on *skytjenester*, was 40 ms for audio and 40 ms for video, while PC-2 had an average RTT of 50 ms. Average packet loss for both clients were 0. Jitter for PC-1 and PC-2 on audio averaged 1 ms, while video jitter averaged 0 ms.

Client	Media	Stream Direction	Avg. RTT	Avg. packet loss	Avg. jitter
PC-1	Audio	Server to client	40ms	0%	1ms
PC-1	Audio	Client to server	40ms	0%	1ms
PC-2	Audio	Server to client	50ms	0%	1ms
PC-2	Audio	Client to server	50ms	0%	0ms
PC-1	Video	Server to client	42ms	0%	0ms
PC-1	Video	Client to server	42ms	0%	0ms
PC-2	Video	Server to client	50ms	0%	0ms
PC-2	Video	Client to server	50ms	0%	0ms

Table 4.6: Data connected to Teams meeting, *skytjenester*: 1.1.4 [A.9].

There is no I and P frame graphs in Teams. This is not because I and P frames are not used, but rather that decoding the Teams stream was unsuccessful. It is possible to deduct when I frames is sent from the spikes in a bandwidth graph as seen in graph 4.6. But this deduction is prone to be a false positive.

4.1.6 Discussion - Teams 1.1

For test scenario 1.1 there is a insignificant difference between the two tenants for average throughput as PC-1 tenant *barneparken* and *skytjenester* had a difference of 0.07 Mbps in upload and 0.01 Mbps in download. There is also a insignificant difference in throughput between the tenants for PC-2. The throughput exceeds the expected amount of bandwidth required, seen in table 2.2 for the quality the webcams record in. MS Teams states that more bandwidth can be used if available to increase quality of the frames (video) given [45]. This do also prove that there is a small difference between server in Amsterdam and Dublin for participent located in Norway using ether Telenor or Uninett as an ISP.

As shown in table 4.3, both of the meetings used UDP as the transport protocol and there where no significant deviation between the two tenant's. This is to be expected as a MS teams meeting prefers UDP as seen in section 2.1.12.5. There is a bit of TCP used in the meeting, which could be explained by Teams signaling [84]. There is probably other types of noise which causes TCP traffic.

In tables 4.5 and 4.6, the only differences between the tenant is the Avg RTT. A possible reason for the increased Avg RTT between the tenant's according to the data collected, is the server location. The meeting for *barneparken* was hosted in Amsterdam and *skytjenester* was hosted in Dublin. A ping with WiFi and without showed the latency variance between Oslo to Amsterdam and Oslo to Dublin is about 14.76 ms. When comparing the Avg RTT between the tenants, the difference ranges from 14-16 Avg RTT between the tenants. This is a good indication that the difference in Avg RTT for the tenants is due the distance of the data centers. It must also be taken into consideration that it is hard to know, as the ping results from `https://wondernetwork.com/pings/Oslo` most likely do not use POP and utilizes MS teams network infrastructure.

The other values such as Avg Packet-Loss and Avg jitter were also within Microsoft (MS) Teams recommended values [42].

4.2 Test scenario 1.2

In this test scenario the objective is to simulate a 10/2 download/upload ADSL network. The topology for test scenario 1.2 is identical to test scenario 1.1, shown in figure 3.7. The reason for conducting an ADSL test scenario is to reflect on the impact of bandwidth limitation in a small business or home network. In regards to how each platform adjusts to the bandwidth change or major bandwidth drop.

4.2.1 Objective

The main objective for 1.2 is the same as for 1.1. However, Webex and Teams results will now also be compared to values in 1.1.

4.2.2 Results - Webex 1.2

Only PC-1's bandwidth will be presented for test scenario 1.2 since parameters are near identical to test scenario 1.1 (PC-1 and PC-2 can be found in appendix A.8). With an ADSL speed capability of 10/2 in download and upload the bandwidth of PC-1 resembles much of test scenario 1.1s bandwidth graph. As shown in graph A.5.

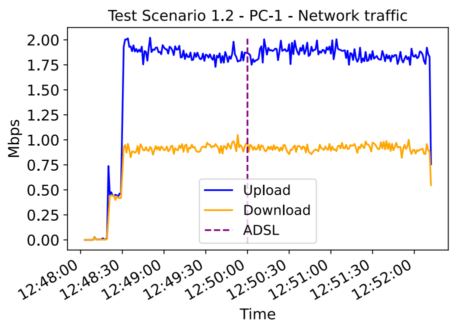


Figure 4.7: Bandwidth usage for PC-1 of packets between PC-1 and Webex service using the UDP protocol. Data from Webex meeting: 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.

4.2.3 Discussion - Webex 1.2

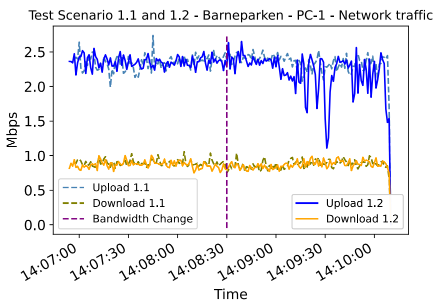
For test scenario 1.2 there are only a few small differences in the data compared to test scenario 1.1, even though this scenario introduces an ADSL connection

2 minutes into the meeting. Changing the bandwidth drastically from 100/100 Mbps in download and upload to 10/2 Mbps in download and upload, there is not a notable change from test scenario 1.1 in the meeting quality or the data extracted. This is believed to be because even with 10/2 Mbps this is enough for upholding the meeting quality, as shown in section 2.3.

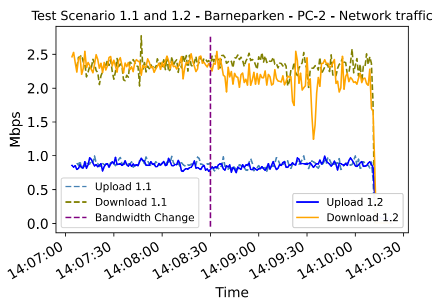
If the test scenario was to be conducted again, there would have been simulated an ADSL network, pressuring the requirement shown in section 2.1.12.4.

4.2.4 Results - Teams 1.2

The meeting for *barneparken* was hosted in Dublin. In graph 4.8a there are multiple drops for upload. For PC-2's download in graph 4.8b traffic oscillations have increased in size towards the end of the meeting. It is a notable change in bandwidth from about 14:08:30 for PC-2.



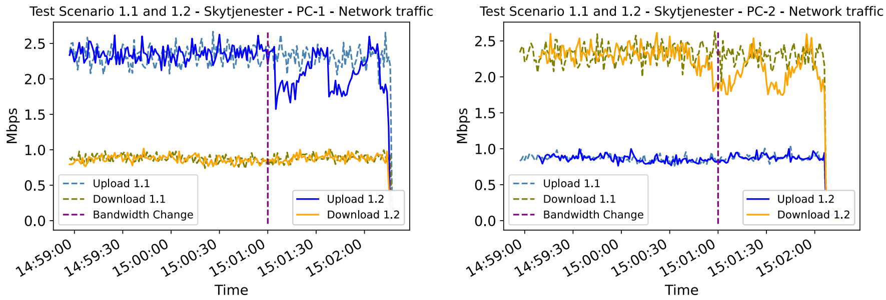
(a) Throughtput for PC-1 (UDP)



(b) Throughtput for PC-2 (UDP)

Figure 4.8: Data connected to Teams meeting, *Barneparken*: 1.2.5 [A.9]. License: Andreas Rømo, CC BY.

For *skytjenester* the meeting was hosted in Amsterdam. In graph 4.9a there are noticeable upload speed drops after 15:01:00 for PC-1. In graph 4.9b the download speed for PC-2 in test scenario 1.2 is reflected by the upload speed for PC-1 in test scenario 1.2.



(a) Throughtput for PC-1 (UDP) **(b)** Throughtput for PC-2 (UDP)

Figure 4.9: Data connected to Teams meeting, *skytjenester*: 1.2.4 [A.9]. License: Andreas Rømo, CC BY.

The tables 4.7 and 4.8 includes the values for both test scenario 1.1 and 1.2. The left side of the arrow represents the value from test scenario 1.1 and the value on the right shows the value collected for scenario 1.2. To compare tenant values for test scenario 1.2 the two tables 4.7 and 4.8 are also compared. From comparing tenants there is 16 to 19 ms difference in latency for audio streams and 17 to 26 ms difference in latency for video streams. There is an exception for one of the streams at *barneparken* where RTT is 98 ms.

Client	Media	Stream direction	Avg. RTT	Avg. packet loss	Avg. jitter
PC-1	Audio	Server to client	26ms → 41ms	0% → 0%	1ms → 1ms
PC-1	Audio	Client to server	26ms → 41ms	0% → 0%	1ms → 2ms
PC-2	Audio	Server to client	36ms → 55ms	0% → 0%	1ms → 1ms
PC-2	Audio	Client to server	36ms → 52ms	0% → 0%	0ms → 0ms
PC-1	Video	Server to client	26ms → 98ms	0% → 0%	0ms → 0ms
PC-1	Video	Client to server	26ms → 63ms	0% → 1%	0ms → 0ms
PC-2	Video	Server to client	36ms → 52ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	36ms → 52ms	0% → 0%	0ms → 0ms

Table 4.7: Data connected to Teams meeting, *barneparken*: 1.2.5 [A.9].

Client	Media	Stream Direction	Avg. RTT	Avg. packet loss	Avg. jitter
PC-1	Audio	Server to client	40ms → 25ms	0% → 0%	1ms → 3ms
PC-1	Audio	Client to server	40ms → 25ms	0% → 0%	1ms → 1ms
PC-2	Audio	Server to client	50ms → 25ms	0% → 0%	1ms → 3ms
PC-2	Audio	Client to server	50ms → 34ms	0% → 0%	0ms → 1ms
PC-1	Video	Server to client	42ms → 37ms	0% → 0%	0ms → 0ms
PC-1	Video	Client to server	42ms → 37ms	0% → 1%	0ms → 0ms
PC-2	Video	Server to client	50ms → 35ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	50ms → 35ms	0% → 0%	0ms → 0ms

Table 4.8: Data connected to Teams meeting, *skytjenester*: 1.2.4 [A.9].

4.2.5 Discussion - Teams 1.2

From the results all parameters were inside the range specified in section 4.2.1. A difference from test scenario 1.1 is where the meeting was hosted. For this test scenario the meeting for *barneparken* was hosted in Dublin and *skytjenester* in Amsterdam, while in test scenario 1.1 it was the opposite. This caused a flip in latency, where now *barneparken* has the latency *skytjenester* had in test scenario 1.1 and vice versa. This strengthens the conclusion that the latency difference is caused by the distance traveled for the packets. There is however an anomaly with the avg. RTT in *barneparken* from PC-1's video stream where the latency was 98 ms. This anomaly could be from the fact that ADSL was introduced in the middle of the meeting. However, there is no direct evidence to support this claim.

In scenario 1.1 there was no jitter or packet loss detected for either tenants, while test scenario 1.2 shows 1 to 3 ms in jitter for audio streams on both tenants and 1% packet loss for a single video stream on both tenants. Notice that both PC-1 and PC-2 in test scenario 1.2 has 1% packet loss when it comes to data uploaded by PC-1. This holds true for both tenants. The packet loss can be caused by limited bandwidth, as the 10/2 line suddenly changes the amount of data Teams can send and receive. When a router receives more packets than it can process, they are thrown away and cause packet loss.

As stated in discussion for test scenario 1.1 in section 4.1.6 Teams can use more bandwidth than stated in the documentation. When the drop in available bandwidth decreased, Teams application probably made changes in its sent bandwidth to stop losing packets. The bandwidth usage as seen in the graphs for both *barneparken* and *skytjenester* (graphs 4.8 and 4.9) had differences from test scenario 1.1. At 14:08:30 *barneparken* and 15:01:00 *skytjenester* is about the time when the group-policy for 10/2 Mbps was activated and shows that the bandwidth utilization by the Teams application reacted by lowering its quality. The quality was not decreased by its frames per second as this can be seen in power BI, therefore the picture resolution most likely changed. This is however not possible to know for certain.

By inspecting the bandwidth graphs for *barneparken*s test scenario 1.2 and *skytjenester*s test scenario 1.2 there are no similarities except that the traffic flow changes from stable to unstable. The graphs for *skytjenester* shows that the limited bandwidth was discovered quickly, and steadily tried to raise the throughput to discover the new bandwidth limitation. While *barneparken* is seemingly more random, with more shifting in the throughput.

4.3 Test scenario 1.3

In this test scenario the objective is to simulate a network where UDP is blocked in the middle of the meeting for services used by the videoconferencing platform. An example for this could be an IT-administrator who has blocked ports used by the videoconferencing platform. Both platform support TCP as a fall-back media transport protocol [42][67]. For the topology see figure 3.9.

4.3.1 Objective

The objective for the test scenario is to see if there are any effects of blocking the UDP protocol, in regards to parameters like jitter, packet loss, bit rate, network protocol usage and RTT.

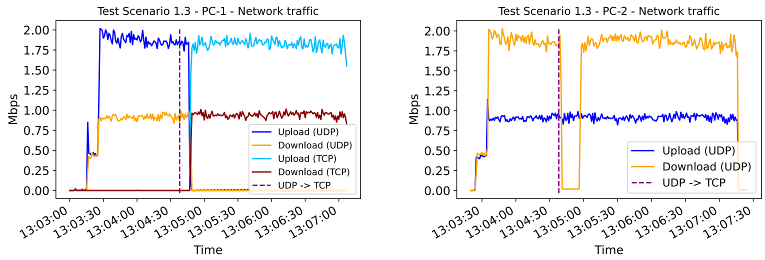
For Webex the objective is the same as in test scenario 1.2 (section 4.2.1).

The objective specified for Teams is the same as in test scenario 1.2 (section 4.2.1).

4.3.2 Results - Webex 1.3

PC-1 has a drop in the packet flow between 13:04:49 and 13:04:50 lasting for 1-2 seconds impacting both upload and download. The drop happens around the same time as when the protocol changes from using the UDP protocol to TCP as the main protocol for connection to the Webex service. The bandwidth usage is around the same as for test scenario 1.1, except for the short drop in network traffic mentioned earlier as shown in 4.10.

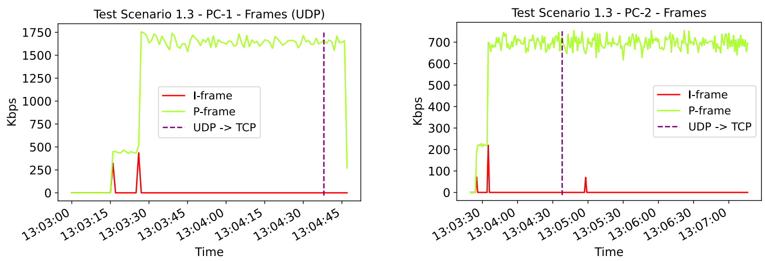
For PC-2 there is a drop in the packet flow between 13:04:41 and 13:04:58 for download lasting for around 17 seconds. PC-2 continues to use the UDP protocol for connection to the Webex service. The bandwidth usage is around the same as for test scenario 1.1, expect for the drop in network traffic mentioned earlier. The UDP bandwidth has 1.14 Mbps as highest value for upload and an average of 0.84 Mbps, for download it has 2 Mbps as the highest value and and an average of 1.57 Mbps. As shown in graph 4.10b.



(a) Throughput PC-1 (UDP & TCP) **(b)** Throughput PC-2 (UDP & TCP)

Figure 4.10: Data connected to Webex meeting: 1.3.4 [A.10]. License: Andreas Kilde Lien, CC BY.

The graphs 4.11a and 4.11b shows the upload dispersion of I- and P-frames for PC-1 and PC-2 throughout the meeting. The graph for PC-1 shows 2 minutes of the meeting, instead of the original 4 minutes. This is because there was no way of extracting the P- and I-frames from a TCP stream. The only visible difference from test scenario 1.1 is that PC-2 gets an I-frames at 13:05:00. The meeting is mainly distributed by P-frames for both PCs.



(a) The dispersion of I- and P-frames using the UDP protocol for PC-1. The graph is including L3 overhead.

(b) The dispersion of I- and P-frames for PC-2. The graph is including L3 overhead.

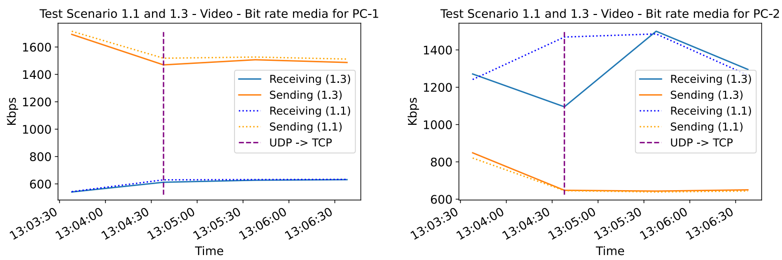
Figure 4.11: Data from Webex meeting: 1.3.4 [A.10]. License: Andreas Kilde Lien, CC BY.

Compared to test scenario 1.1, PC-2 has roughly the same usage of UDP/TCP. PC-1 however has a change in UDP/TCP, TCP was used 62.29% and UDP 37.71% compared to UDP 99.53% and TCP 0.47% in test scenario 1.1.

Client	Protocol	Sent	Received	Total
PC-1	TCP	30.61 MB	15.95 MB	46.56 MB (62.29%)
	UDP	18.67 MB	9.51 MB	28.18 MB (37.71%)
PC-2	TCP	221.37 KB	218.83 KB	440.2 KB (0.63%)
	UDP	24.7 MB	43.37 MB	68.07 MB (99.37%)

Table 4.9: Table to show protocols used by PC-1 and PC-2 under the meeting. Data connected to Webex meeting: 1.3.4 [A.10].

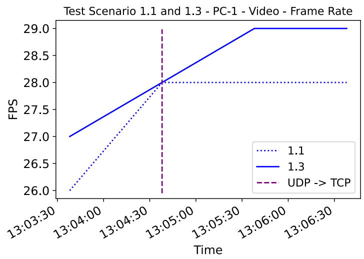
As can be seen in figure 4.12a, the bit rate is close to the same for PC-1, while on PC-2 there is a drop in received Mbps around a minute into the meeting compared to the test scenario 1.1, shown in graph 4.12b.



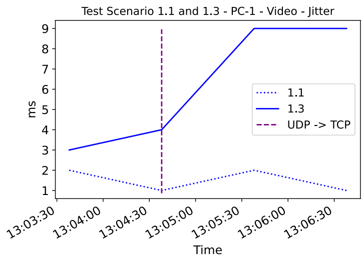
(a) Bit rate for PC-1. **(b)** Bit rate for PC-2.

Figure 4.12: Data from Webex meeting: 1.3.4 [A.10]. License: Andreas Kilde Lien, CC BY.

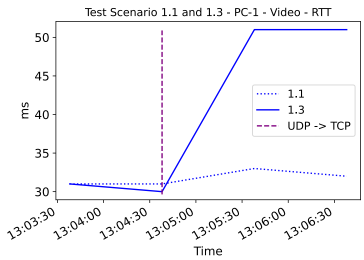
All of the values in the graphs for PC-1 in 4.13 can be seen changing after the switch from UDP to TCP. The biggest changes can be seen in the RTT where for video the values goes up around 20 ms and for voice it goes up around 40 ms. The jitter for voice also makes a change from 6 to 16 ms. Even though there are some changes to the values compared to test scenario 1.1 all of them are still within the requirements for webex meetings specified in section 2.1.12.4.



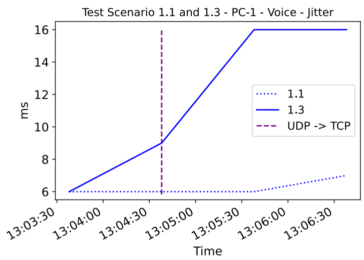
(a) Video - Frame Rate.



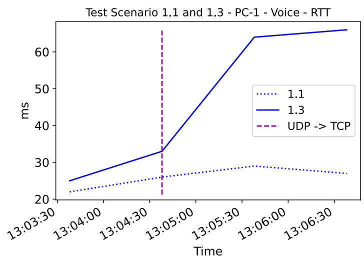
(b) Video - Jitter.



(c) Video - RTT.



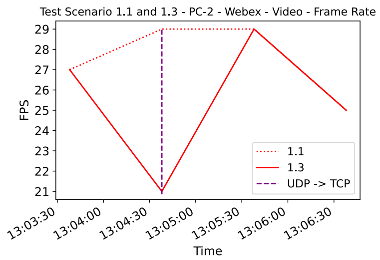
(d) Voice - Jitter.



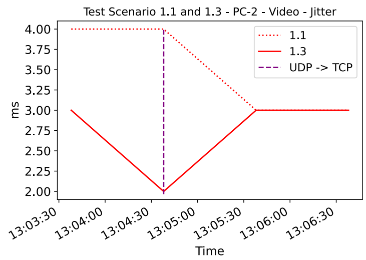
(e) Voice - RTT.

Figure 4.13: Data from Webex meeting: 1.1.6 and 1.3.4 [A.10]. License: Andreas Kilde Lien, CC BY.

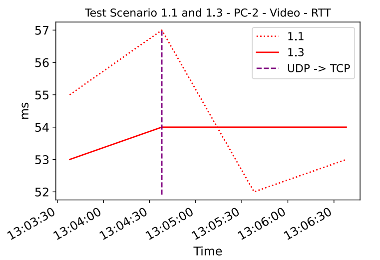
Looking at the graphs for PC-2 seen in 4.14, both the FPS and the jitter takes a dip when the change from UDP to TCP occurs, but return to original value later. Most of the other values stay roughly the same as test scenario 1.1. The voice RTT stays throughout the whole meeting around 5 ms above test scenario 1.1. All the values are within the requirements stated in section 2.1.12.4.



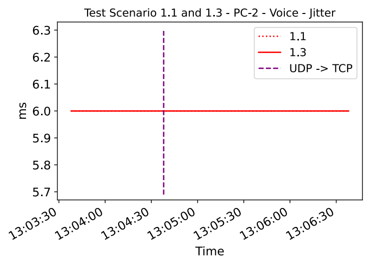
(a) Video - Frame Rate.



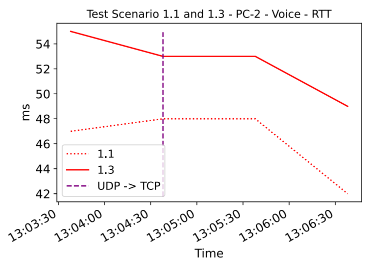
(b) Video - Jitter.



(c) Video - RTT.



(d) Voice - Jitter.



(e) Voice - RTT.

Figure 4.14: Data from Webex meeting: 1.1.6 and 1.3.4 [A.10]. License: Andreas Kilde Lien, CC BY.

4.3.3 Discussion - Webex 1.3

In test scenario 1.3 there was quite a lot of changes compared to test scenario 1.1. The UDP blockage takes effect in around two minutes. Effects from the blockage are well represented on the bandwidth graphs where both locations and participants experience a connection loss.

Interesting finding is that PC-1 uses less time to recover from the blockage compared to PC-2. The reason for this is most likely reconfiguration of packet flow towards PC-2 and the restarting of encryption with PC-2. A question that arises from this is then why it does not take that amount of time to join a meeting or start hosting a meeting. The answer lies in that Webex prepares the allocation of Webex media nodes in the correct region before joining a meeting. This means that if for example a Webex client sees that a meeting is ongoing, the back-end and the client has already begun to make preparations for joining the meeting. When the client joins a meeting the media channels to the meeting instance is allocated and media flows immediately. This type of preparation has been confirmed by a Cisco Webex employee.

From the I- and P-frames graph for PC-2 it is seen that a new set of I-frames are sent after the UDP block is active. The new I-frame had to be sent as the last I-frame was lost by cutting of the UDP stream from PC-1.

When putting the bandwidth and P- and I-frames in relation to each other it is clear that the connection was lost for a period of time. The quality of the video call was therefore affected by the UDP block.

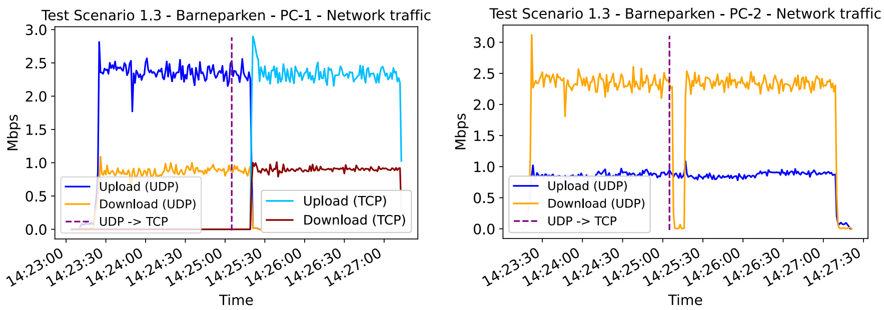
By examining the upload and download usage further it is nearly identical both before and after the blockage. The reason for this could be that Cisco uses an optimisation for packet aggregation (merging multiple packets into a single, to reduce the overhead), and ingress buffering. In general, packet aggregation improves TCP throughput, and the improvement grows with the collected interval. Aggregation can be done at various granularities with packets destined to the same egress server, like the Webex data center, including aggregating all packets (full aggregation), aggregating packets from the same traffic class (per-class aggregation). Webex would most likely want to choose a per-class aggregation for RTP packages. Ingress buffering can improve TCP throughput significantly, and the ingress buffer only needs to be a small size [85]. In all using any of the mentioned approaches is known to reduce the size of packets and could be used to make the TCP packages smaller.

If the test scenario was to be conducted again it would have been interesting to look into a meeting purely using the TCP protocol. Will Webex act differently when it knows it has to use TCP from the beginning of the meeting?

4.3.4 Results - Teams 1.3

The meetings were hosted in Paris, France for *barneparken*, but around the 2 minute mark the server location changes for PC-1 from Paris to Dublin. This is however not the case for PC-2, as it continues its connection with the data center in Paris.

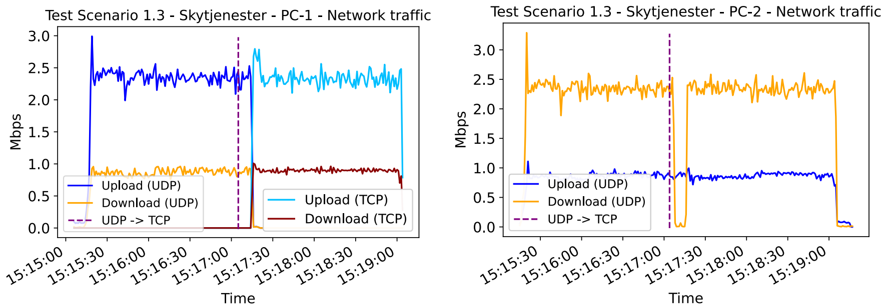
In the graph 4.15, a noticeable drop is shown for both clients in between 14:25:20 and 14:25:30, but for PC-2 there was no drop in upload. There is also a line indicating the moment UDP block was introduced. The drop for PC-2 has a period of 10 seconds, while PC-1 barely has any down time.



(a) Throughtput for PC-1 (UDP & TCP) **(b)** Throughtput for PC-2 (UDP & TCP)

Figure 4.15: Data connected to Teams meeting, *barneparken*: 1.3.4 [A.9]. License: Andreas Rømo, CC BY.

The meeting for *skytjenester* were hosted in Amsterdam, and PC-1 did also for this tenant swapp server. This time to London, England in the middle of the meeting. The graph 4.16 for *skytjenester* also has a noticeable drop, but no upload drop for PC-2 and shows similar results to *barneparken*.



(a) Bandwidth usage for PC-1 of packets between PC-1 and MS Teams service using the UDP and TCP protocol.

(b) Bandwidth usage for PC-2 of packets between PC-2 and MS Teams service using the UDP protocol.

Figure 4.16: Data connected to Teams meeting, *skytjenester*: 1.3.4 [A.9]. License: Andreas Rømo, CC BY.

Distribution among protocols for *barneparken* can be found in table 4.10. PC-1 has a change in User Datagram Protocol (UDP) and TCP compared to test scenario 1.1, the total use of UDP goes from 99.48% to 50.94% and TCP increased from 0.52% to 50.94%. For PC-2 the distribution are insignificant compared to test scenario 1.1 as the change is up 0.07% for UDP.

Client	**Protocol**	**Sent**	Received	Total
PC-1	TCP	33.19 MB	13.55 MB	46.74 MB (50.94%)
	UDP	32.83 MB	12.18 MB	45.01 MB (49.06%)
PC-2	TCP	395.02 KB	244.56 KB	639.58 KB (0.78%)
	UDP	23.88 MB	57.99 MB	81.87 MB (99.22%)

Table 4.10: Table to show protocols used by PC-1 and PC-2 under the meeting. Data connected to Teams meeting, *barneparken*: 1.3.4 [A.9].

For *skytjenester* the change in protocol is quite similiar to *barneparken*. For PC-1 the total usage of UDP has decreased from 98.55% to 50.75% and TCP went up from 1.45% to 49.25%. PC-2 is similar to *barneparken*, no significant change from test scenario 1.1. Like shown in table 4.11.

Client	**Protocol**	**Sent**	Received	Total
PC-1	TCP	31.65 MB	12.72 MB	44.37 MB (49.25%)
	UDP	33.36 MB	12.35 MB	45.71 MB (50.75%)
PC-2	TCP	236.87 KB	105.62 KB	342.49 KB (0.44%)
	UDP	22.93 MB	55.75 MB	78.68 MB (99.56%)

Table 4.11: Table to show protocols used by PC-1 and PC-2 under the meeting. Data connected to Teams meeting, *skytjenester*: 1.3.4 [A.9].

In the table 4.12 for *barneparken* the latency from test scenario 1.2 compared to test scenario 1.1 ranges from 10 to 25 ms. There is video packet loss from PC-1 to the server and from the server to PC-2 at 4%. For *skytjenester* the latency ranges from 10 to 20 ms.

Client	**Media**	**Stream Direction**	**Avg RTT**	**Avg Packet-Loss**	**Avg jitter**
PC-1	Audio	Server to client	26ms → 57ms	0% → 0%	1ms → 2ms
PC-1	Audio	Client to server	26ms → 57ms	0% → 4%	1ms → 1ms
PC-2	Audio	Server to client	36ms → 44ms	0% → 0%	1ms → 2ms
PC-2	Audio	Client to server	36ms → 44ms	0% → 0%	0ms → 0ms
PC-1	Video	Server to client	26ms → 65ms	0% → 0%	0ms → 0ms
PC-1	Video	Client to server	26ms → 65ms	0% → 4%	0ms → 0ms
PC-2	Video	Server to client	36ms → 44ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	36ms → 44ms	0% → 0%	0ms → 0ms

Table 4.12: Data connected to Teams meeting, *barneparken*: 1.3.4 [A.9].

Table 4.13 shows an increase in Avg Packet-Loss for PC-1, from client to server in both audio and video. Avg RTT for PC-1 increases for audio streams by 22ms and video streams with 15ms.

Client	Media	Stream Direction	Avg RTT	Avg Packet-Loss	Avg jitter
PC-1	Audio	Server to client	26ms → 53ms	0% → 0%	1ms → 1ms
PC-1	Audio	Client to server	26ms → 53ms	0% → 4%	1ms → 1ms
PC-2	Audio	Server to client	36ms → 34ms	0% → 0%	1ms → 1ms
PC-2	Audio	Client to server	36ms → 34ms	0% → 0%	0ms → 0ms
PC-1	Video	Server to client	26ms → 41ms	0% → 0%	0ms → 0ms
PC-1	Video	Client to server	26ms → 42ms	0% → 4%	0ms → 0ms
PC-2	Video	Server to client	36ms → 35ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	36ms → 35ms	0% → 0%	0ms → 0ms

Table 4.13: Data connected to Teams meeting, *skytjenester*: 1.3.4 [A.9].

4.3.5 Discussion - Teams 1.3

From all the collected data, the packet loss collected for two streams was above the recommended value of 1%. This is true for both tenants. The bandwidth used for download on PC-2 was 0 in about 10 seconds for both tenants. PC-1 on both tenants has a slight drop at around 2 minutes into the meeting. The applied UDP block on PC-1's router is a plausible explanation for the drop in bandwidth. There was no noticable difference in quality between the tenants.

A bi-effect of the UDP block on PC-1 is when the stream goes over to TCP it also changes server location. It is not certain that it has to change, but in all the tests runs for scenario 1.3 (four iterations A.7) the server location changed for PC-1. There is no direct documentation on why the server location changes for PC-1, but there is documentation on how Microsoft Azure Front Door (AFD) works. By understanding AFD it is possible to narrow down how a call works with two different data centers is used. In figure 4.17 an overview of AFD is displayed. The traffic for a client goes through the AFD and then the Middle-Tier will direct you to the required service, such as MS Teams. The Middle-Tier API depicted is probably what makes it possible to use different MS Teams data centers and still get a high quality meeting. But again, this is highly speculative.

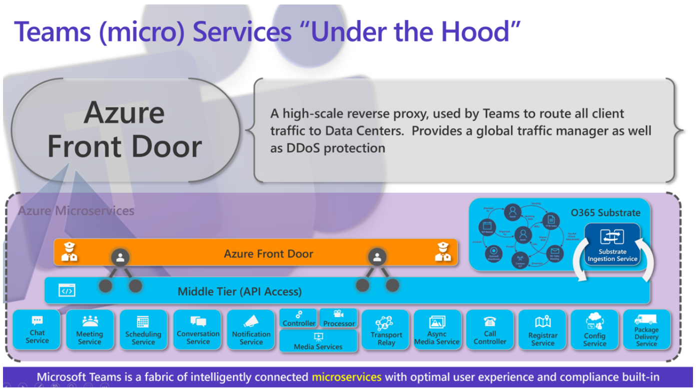


Figure 4.17: Azure front door, Microsoft Teams micro services. License: Microsoft Teams, CC BY.

In the bandwidth usage shown for both tenants it is seen that the upload drop for PC-1 lasts much less than the download drop for PC-2. This could be because of the time it takes for Teams to reconfigure packet sending and restarting encryption using a new underlying protocol. This again is highly speculative as this might not be true for Teams, while it can be for Webex.

All the streams for both *skytjenester* and *barneparken* is stated as UDP streams lasting 4 minutes in power BI, for both PC-1 and PC-2.

4.3.5.1 Tenant quality

By performing all the test iterations of test scenario 1.1 to test scenario 1.3 it was run a total of 30 meetings with a total of 42 server locations on both of the tenants combined. As seen in the bar chart 4.18, it seems that meeting hosted on either *barneparken* or *skytjenester* is both utilizing the same servers spread across Europe, except for one.

The intention for running the same tests on the two tenants was to find out if having a Norwegian tenant versus European tenant affect the network quality of a video call. There are no statistic evidence in our data that choosing either tenant matters in this case, therefore is *skytjenester* excluded for further examination and only *barneparken* will be focused on in the upcoming test scenarios.

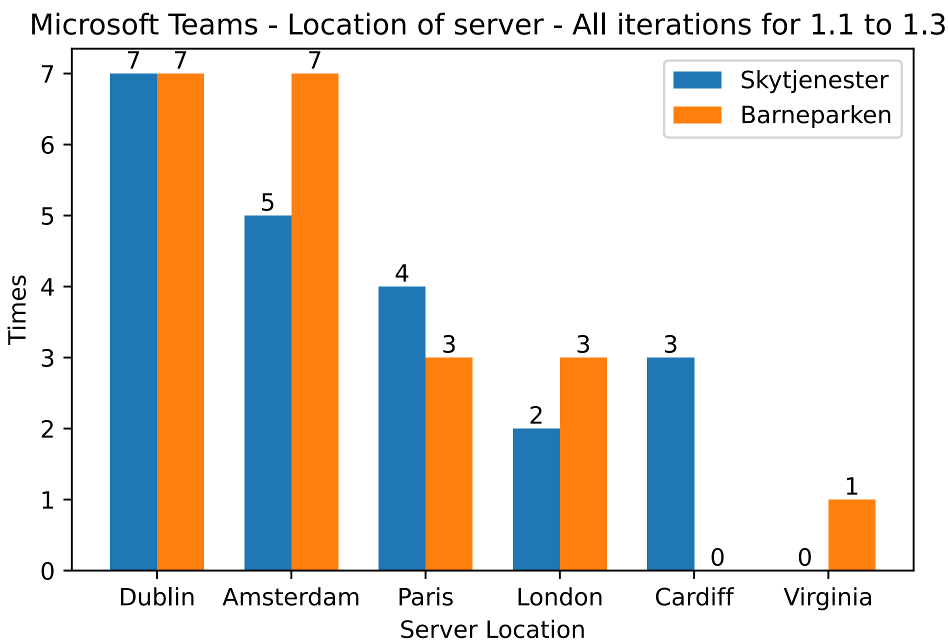


Figure 4.18: Server location for all meetings held during test scenario 1.1 to 1.3, for *barneparken* and *skytjenester*.

4.4 Test scenario 2.1

Test scenario 2.1 is a videoconference meeting where the computer on the Telenor side, PC-1, loses its Ethernet connection and is automatically switched to a wireless connection. Test scenario 2.1 simulates a real life scenario where the client loses its Ethernet connection mid-meeting.

4.4.1 Objective

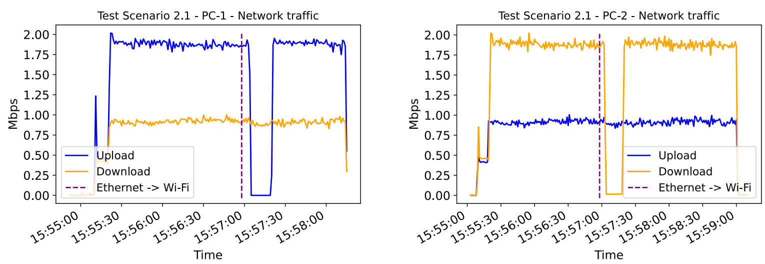
The objective for the test scenario is to see if there are any impacts on quality by switching from Ethernet to a wireless connection. A set of new parameters is introduced in this scenario, such as signal quality and SNR. Since this scenario uses an Meraki AP an additional quality document is used to see if the scenario is above 25 dB for SNR [72]. All the parameters is also put in comparison to test scenario 1.1.

Teams from here on and out will compare PC-1 to PC-2 for tenant barneparken only. The other MS Team tenant, skytjenester, will not be taken into further investigation. See previous subsection 4.3.5.1 for explanation.

4.4.2 Results - Webex 2.1

PC-1 has a drop in throughput from 15:57:05 to 15:57:20 lasting about 30 seconds after the Ethernet cable is disconnected. Compared to test scenario 1.1 the bandwidth values are mostly identical except for this drop. The bandwidth for PC-1 has an average of 1.48 Mbps upload and for download an average of 0.80 Mbps. As shown in graph 4.19a.

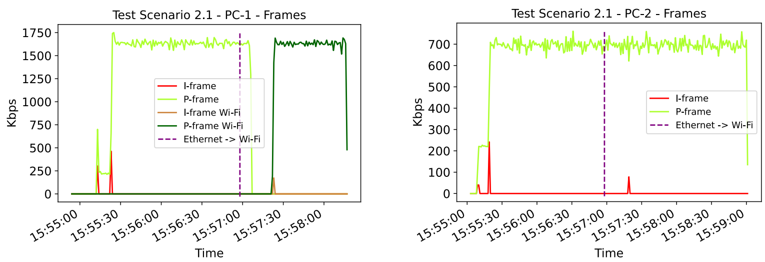
PC-2 has a drop for download between 15:57:03 and 15:57:19. Compared to test scenario 1.1 the bandwidth values are mostly identical except for the drop. The upload bandwidth for PC-2 is on average 0.83 Mbps and on average 1.57 Mbps for download. As shown in graph 4.19b.



(a) Throughput PC-1 (UDP) **(b)** Throughput PC-2 (UDP)

Figure 4.19: Data connected to Webex meeting: 2.1.4 [A.10]. License: Andreas Kilde Lien, CC BY.

The graphs 4.20a and 4.20b shows the upload dispersion of I- and P-frames for PC-1 and PC-2 throughout the meeting. The graph for PC-1 shows both frames over Ethernet and Wi-Fi. The only visible difference from test scenario 1.1 happens when changing from Ethernet to Wi-Fi. Where PC-1 stops sending any frames at 15:57:08, for later at 15:57:22 start by sending P-frames and a I-frame. PC-2 sends a I-frame at 15:57:19. The meeting is mainly distributed by P-frames for both PCs.

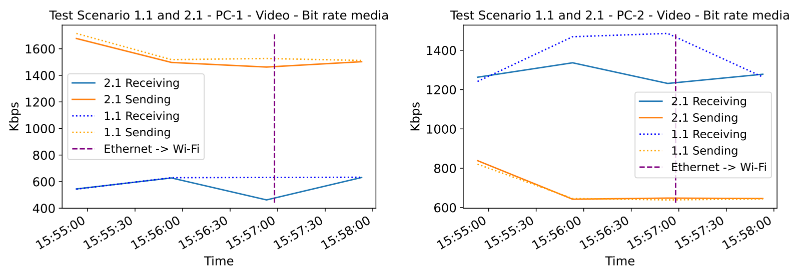


(a) The dispersion of I- and P-frames for PC-1. The graph is including L3 overhead.

(b) The dispersion of I- and P-frames for PC-2. The graph is including L3 overhead.

Figure 4.20: Data from Webex meeting: 2.1.4 [A.10]. License: Andreas Kilde Lien, CC BY.

In graph 4.21 the bit rate for the clients is shown. PC-2 has a noticeable difference in receiving Kbps.

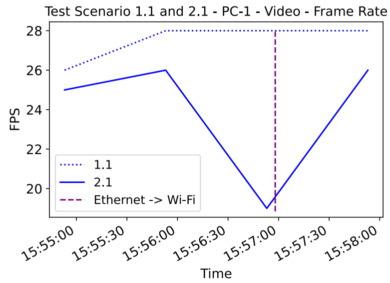


(a) Bit rate for PC-1.

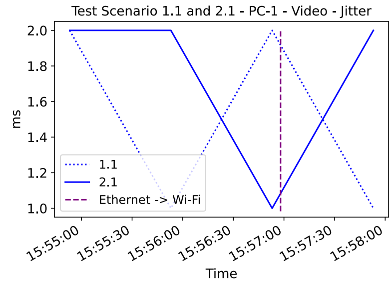
(b) Bit rate for PC-2.

Figure 4.21: Data from Webex meeting: 1.1.6 and 2.1.4 [A.10]. License: Andreas Kilde Lien, CC BY.

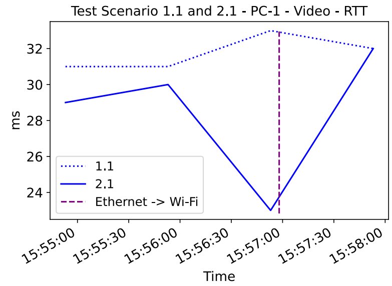
Looking at the graphs for PC-1 in 4.22, the graphs for jitter and packet loss shows little to no change from test scenario 1.1. The graphs for FPS and RTT all three shows a dip around the configuration change. Even with the dips none of the values were outside the webex requirements stated in section 2.1.12.4.



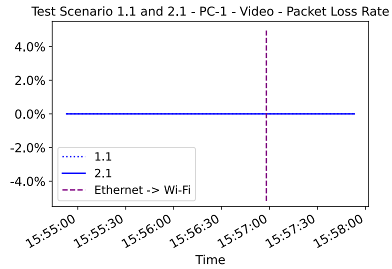
(a) Video - Frame Rate.



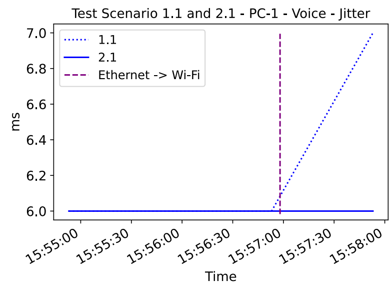
(b) Video - Jitter.



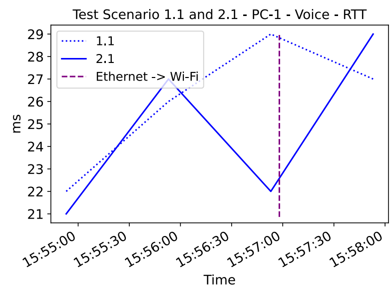
(c) Video - RTT.



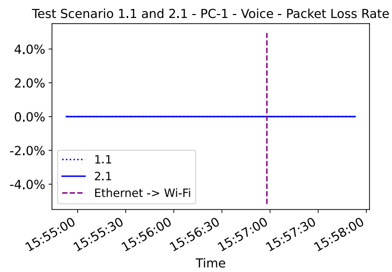
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



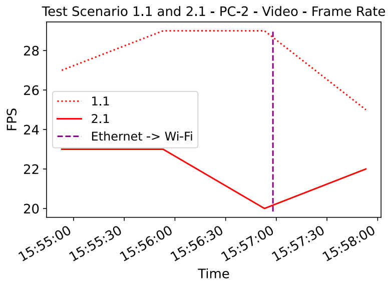
(f) Voice - RTT.



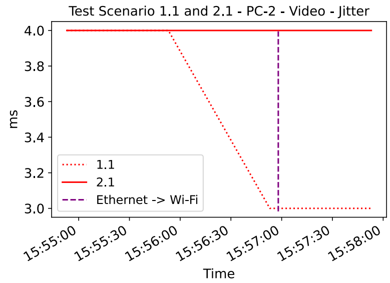
(g) Voice - Packet Loss Rate.

Figure 4.22: Data from Webex meeting: 1.1.6 and 2.1.4 for PC-1 [A.10]. License: Andreas Kilde Lien, CC BY.

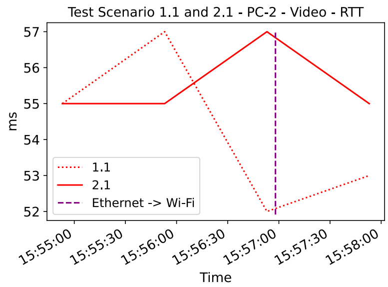
Looking at the graphs for PC-2 in 4.23a, the FPS is consistently lower throughout the meeting compared to test scenario 1.1s values. The rest of the values stay roughly the same as test scenario 1.1 or within 1 or 2, except for the voice RTT which stays around 7 ms above test scenario 1.1. All of the values remained within the webex requirements stated in section 2.1.12.4.



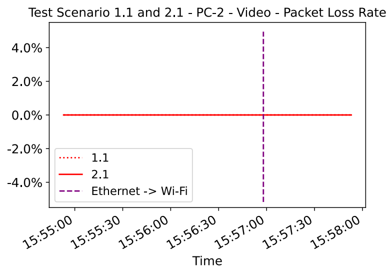
(a) Video - Frame Rate.



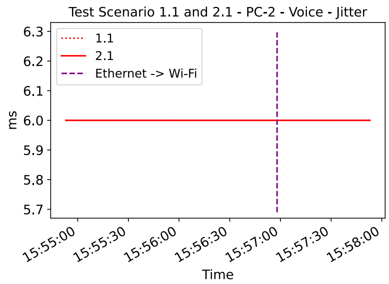
(b) Video - Jitter.



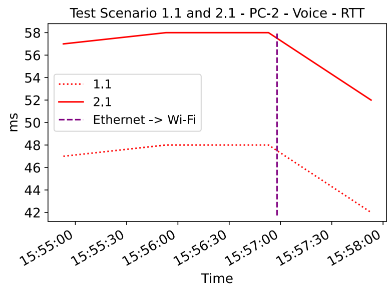
(c) Video - RTT.



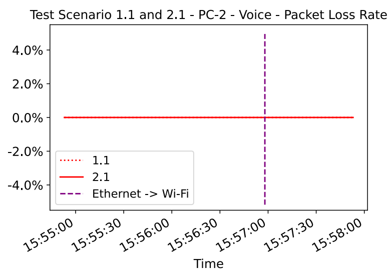
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



(f) Voice - RTT.

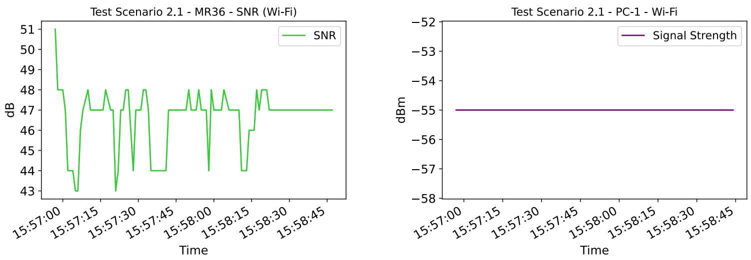


(g) Voice - Packet Loss Rate.

Figure 4.23: Data from Webex meeting: 1.1.6 and 2.1.4 for PC-2 [A.10]. License: Andreas Kilde Lien, CC BY.

The two graphs 4.24a and 4.24b shows the Signal-to-noise ratio (SNR) and signal strength for PC-1 when connected to the Meraki AP. SNR values range between 43 and 51 dB, with an average of 46.59 dB. Cisco's documentation mentions that SNR over 25 dB is recommended voice applications, which the SNR is well above [72]. This is however also a video application, which might require even more.

For signal strength for PC-1 is stable at -55 dBm for the whole meeting. A passive site survey of the location is seen in figure 4.25.



(a) Signal-to-noise ratio for PC-1's connection to AP.

(b) Signal strength for PC-1's connection to AP.

Figure 4.24: Data connected to Webex meeting: 2.1.4 [A.10]. License: Andreas Kilde Lien, CC BY.

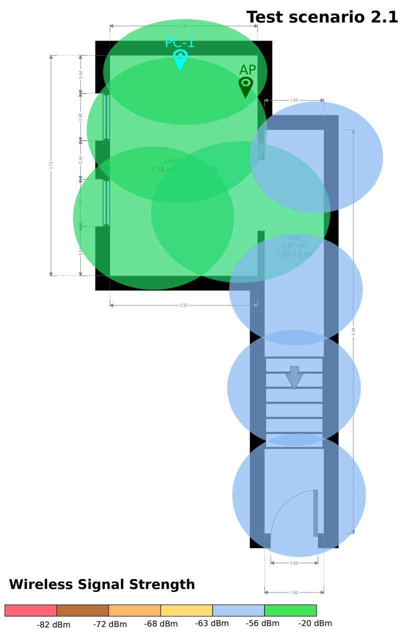


Figure 4.25: Site survey of PC-1 location where lower values are worse and higher values are better. License: Andreas Kilde Lien, CC BY.

4.4.3 Discussion - Webex 2.1

Test scenario 2.1 is the first test scenario to use Wi-Fi. An interesting finding is that while using Wi-Fi on Webex the bandwidth usage was more or less equal to a videoconference using Ethernet. This could be explained with both achieving high enough download and upload speed on Wi-Fi and Ethernet that satisfies the Webex requirements.

Inspecting the bandwidth graph further there is a drop for both PC-1 and PC-2. The reason for PC-2's drop is anticipated to be the same as for test scenario 1.3 (reconfiguration and encryption). There is also a drop shown for PC-1 this time. Since PC-1 looses its connection to the Webex server, the old connection needs to be confirmed dropped and re-established. This is probably what makes the drop last in a longer time frame.

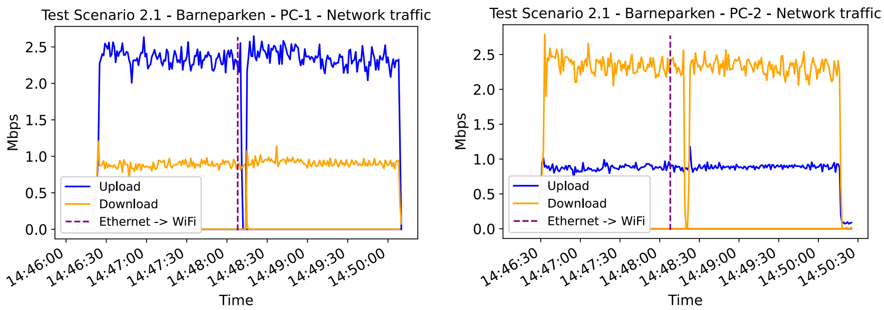
From the I- and P-frames graphs in 4.29 the frames behave mostly the same as test scenario 1.1. The exception comes when PC-1 changes from Ethernet to Wi-Fi, PC-1 then stops sending any frames for around 20 seconds, then both PC's gets a new I-frame when PC-1 is reconnected on the Wi-Fi connection.

For signal quality the AP delivered satisfying SNR values that were above Meraki's documentation for data usage and what is recommended for high speeds [86][72]. The signal strength from PC-1 is harder to determine as this is not put into relation to anything. The dBm signal received can be different for each vendors antenna. It is therefore best to find the signal quality with a calibrated and known reference. But the test scenario is considered to be a simulation of a best performance video-conference so, the values for signal strength could be considered as "good" values.

If test scenario 2.1 was to be conducted again it would have been interesting doing a similar test using the 2.4 GHz band instead of 5 GHz to compare result.

4.4.4 Results - Teams 2.1

The meeting for test scenario 2.1 was hosted in Amsterdam and the throughput graph 4.26 is shown below. The graph compared to test scenario 1.1 is similar expect the drop that happens between 14:48:00 to 14:48:30. This is time the Ethernet cable was removed for PC-1 and auto connects to the AP (MR36).



(a) Throughtput for PC-1 (UDP) **(b)** Throughtput for PC-2 (UDP)

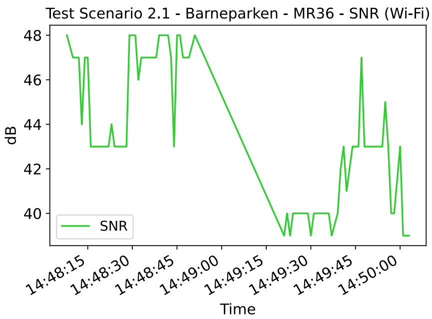
Figure 4.26: Data connected to Teams meeting, Barneparken: 2.1.4 [A.9]. License: Andreas Rømo, CC BY.

Noticeable differences from test scenario 1.1 shown in table 4.14 is the increase in Avg RTT, Avg Packet-Loss and Avg jitter for PC-1. PC-2 only have a 1 ms increase in jitter for audio.

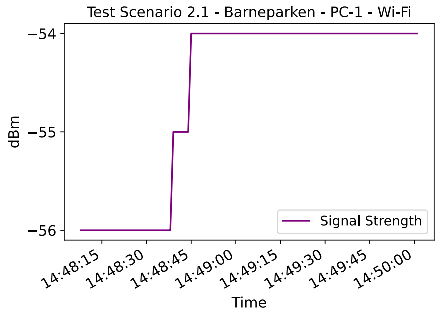
Client	Media	Stream Direction	Avg. RTT	Avg. Packet Loss	Avg. Jitter
PC-1	Audio	Server to client	26ms → 30ms	0% → 2%	1ms → 2ms
PC-1	Audio	Client to server	26ms → 29ms	0% → 2%	1ms → 1ms
PC-2	Audio	Server to client	36ms → 36ms	0% → 0%	1ms → 2ms
PC-2	Audio	Client to server	36ms → 36ms	0% → 0%	0ms → 0ms
PC-1	Video	Server to client	26ms → 30ms	0% → 1%	0ms → 0ms
PC-1	Video	Client to server	26ms → 30ms	0% → 2%	0ms → 0ms
PC-2	Video	Server to client	36ms → 36ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	36ms → 36ms	0% → 0%	0ms → 0ms

Table 4.14: Data connected to Teams meeting, *barneparken*: 2.1.4 [A.9].

A log of the SNR collected at the Meraki AP is shown in figure 4.27a. The SNR graph lasts about 2 minutes and has an average of 43.5 dB. The signal strength in dBm from the Meraki AP to PC-1 is shown in figure 4.27b. The signal strength average is -54.9 dBm.



(a) Signal-to-noise ratio measured by AP.



(b) Signal strength for PC-1's connection to AP.

Figure 4.27: Data connected to Teams meeting, Barneparken: 2.1.4 [A.9]. License: Andreas Rømo, CC BY.

4.4.5 Discussion - Teams 2.1

From all the collected data only three streams from *barneparken* was inside recommended values. The packet loss exceeded the recommended value of 1% with 2%. The recorded SNR values from the meeting was 20 dB above the recommended 25 dB, and therefore qualifies as good enough for a video application.

In the graphs showcasing bandwidth there is a drop to 0 Mbps at about 2 minutes into the meeting. This is the exact time the Ethernet cable was pulled out and the Wifi comes into play. The rest of the meeting is stable. The quality was affected by the drop.

It should also be mentioned that in the throughput graph 4.26 the "Ethernet to WiFi" line, do not correlate. PC-1 drops connection while PC-2 still receives a feed from the server. There could be a buffer to the stream from the server, but in videoconferencing it is highly unlikely that a buffer could be this long (14 seconds). The other test iterations showed similar results, some even more "off". An Network Time Protocol (NTP) error could be the cause when these test iterations were done. The MX68 routers that performs the pcaps however share the same NTP server, and therefore the times should have been synchronized.

The server used for the meeting is the same as test scenario 1.1 (Amsterdam). This makes comparing latency easier. From the streams in table 4.14 PC-1s latency were affected by using Wi-Fi, while PC-2 stayed the same. The most plausible explanation is therefore the extra "hop" for the packets on PC-1, as it increases Avg RTT by 3 to 4 ms. Multiple pings to "vg.no" was performed from PC-1, with Ethernet cable and Wi-Fi, this can be found in appendix (A.15). This showed a similar increase in latency, 2 to 4m s.

The packet loss difference compared to test scenario 1.1 comes from the change from Ethernet to Wi-Fi. This can be argued, as the only streams with packet loss is those involved with PC-1. It could be other disturbances on the Internet, however it is not possible to know. The test procedure does require a connectivity test before each meeting and it came out "good". Although the test came out good it does not mean that MS Teams is flawless. The test only sees if certain parameters falls below a certain maximum. MS Teams could still induce latency, jitter and other relevant parameters. It is shown that Teams performs worse during work hours [87].

The SNR and signal strength is not possible to compare to test scenario 1.1 as there is no SNR or signal strength to compare with.

4.5 Test scenario 2.2

Test scenario 2.2 simulates PC-1 on a poor wireless connection and PC-2 on an Ethernet connection. The scenario shows how a unstable wireless connection will act in a videoconference.

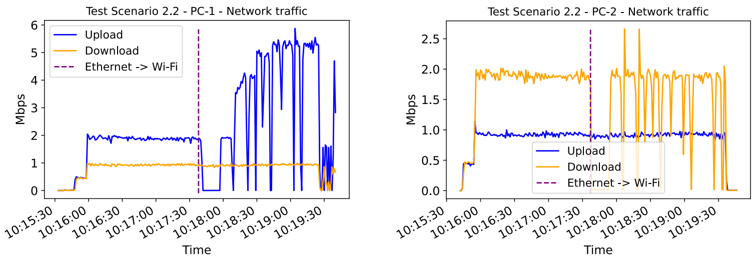
4.5.1 Objective

Test scenario 2.2 will be compared to test scenario 2.1, as apposed to 1.1. See section 3.3.4 for 2.1. Just as in when comparing to 1.1, network parameters are compared to quality documents and compared between scenarios.

4.5.2 Results - Webex 2.2

PC-1 has a drop in throughput about 30 seconds after the Ethernet cable is disconnected and the computer switch to using Wi-Fi. The throughput starts fluctuating throughout the meeting. Compared to test scenario 2.1 the bandwidth values are mostly identical for the first two minutes, but the upload is much more unstable the last two minutes. The upload average is 2.25 Mbps and the download average is 0.81 Mbps. As shown in graph 4.28a.

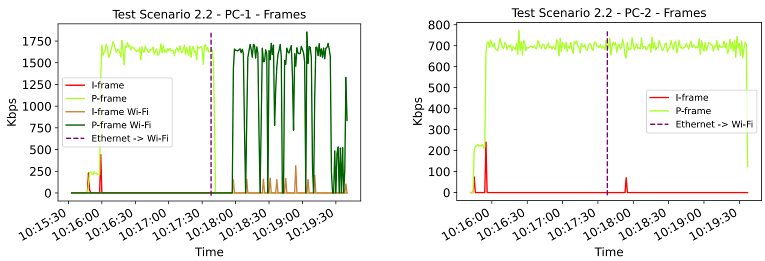
PC-2 has a throughput drop for download between 10:17:30 and 10:18:00. Download stabilises on around 2 Mbps after the drop lasting abut 10 seconds, before the graph starts fluctuating throughout the meeting. Compared to test scenario 2.1 the bandwidth values are mostly identical for the first two minutes, but the download is much more unstable the last two minutes. The bandwidth for PC-2 an average of 0.85 Mbps upload and 1.49 Mbps download. As shown in graph 4.28b.



(a) Throughput PC-1 (UDP) **(b)** Throughput PC-2 (UDP).

Figure 4.28: Data from Webex meeting: 2.2.3 [A.10]. License: Andreas Kilde Lien, CC BY.

The graphs 4.29a and 4.29b shows the upload dispersion of I- and P-frames for PC-1 and PC-2 throughout the meeting. The graph for PC-1 shows both frames over Ethernet and Wi-Fi. The only visible difference from test scenario 1.1 happens when changing from Ethernet to Wi-Fi. Where PC-1 stops sending any frames at 10:17:42 until 10:17:58 where it is connected to Wi-Fi. The graph is shifting a lot on Wi-Fi and sending I-frames everytime the bandwidth increases. PC-2 sends an I-frame at 10:17:55. The meeting is mainly distributed by P-frames for both PCs.

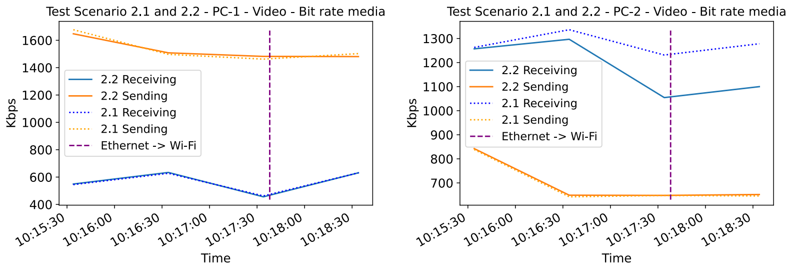


(a) The dispersion of I- and P-frames for PC-1. The graph is including L3 overhead.

(b) The dispersion of I- and P-frames for PC-2. The graph is including L3 overhead.

Figure 4.29: Data from Webex meeting: 2.2.3 [A.10]. License: Andreas Kilde Lien, CC BY.

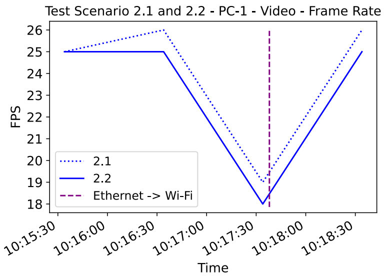
The bit rate seen in graph 4.30a shows PC-1 having almost identical values as in 2.1. For PC-2 in graph 4.30b the sent data is identical to 2.1, while the received data for 2.2 remains close until it takes a dive at around 10:17:00 and goes down from around 1260 Kbps to around 1090 Kbps. The largest difference is around when the switch to Wi-Fi happens where the bit rate has a difference of around 170 Kbps from 2.1.



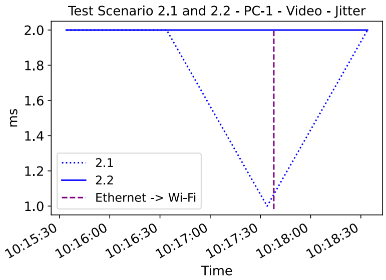
(a) Bit rate for PC-1. **(b)** Bit rate for PC-2.

Figure 4.30: Data from Webex meeting: 2.2.3 [A.10]. License: Andreas Kilde Lien, CC BY.

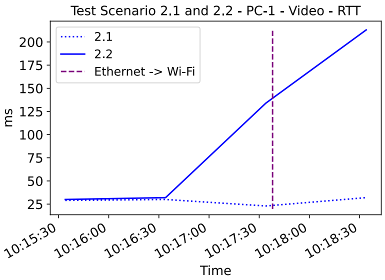
The graphs for PC-1 in 4.31 most of the values starts to rise steadily except for the FPS which takes a drop. Both the RTT for voice and video rises and goes to around 210 ms. The packet loss for video goes up to 14.1%, while for voice it goes up to 1.53%. The RTT for voice and video and the packet loss rates are not within the requirements stated in section 2.1.12.4.



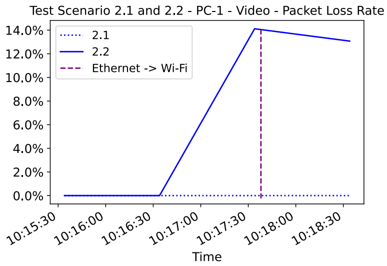
(a) Video - Frame Rate.



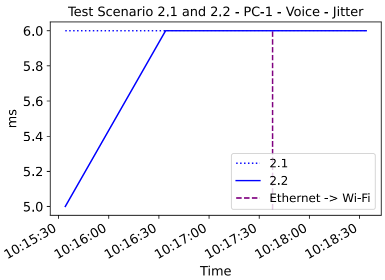
(b) Video - Jitter.



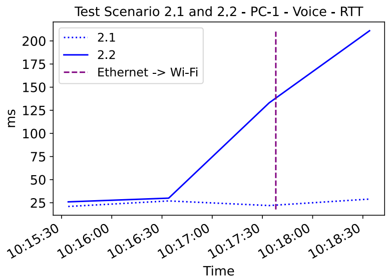
(c) Video - RTT.



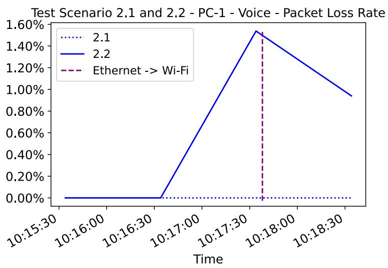
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



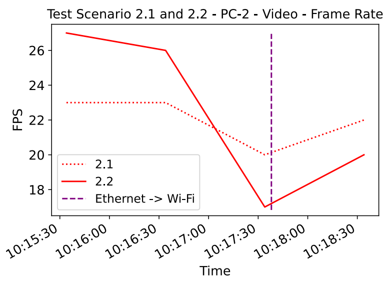
(f) Voice - RTT.



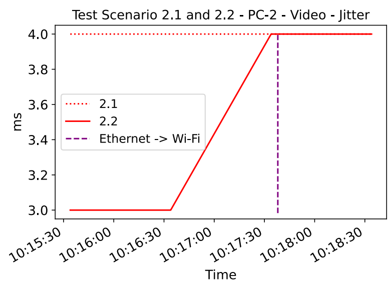
(g) Voice - Packet Loss Rate.

Figure 4.31: Data from Webex meeting: 2.1.4 and 2.2.3 for PC-1 [A.10].

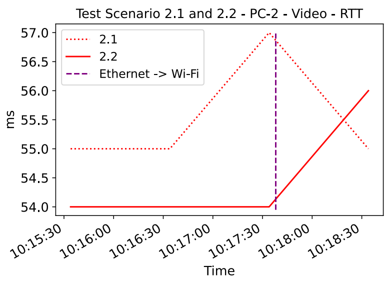
Inspecting the graphs for PC-2 in 4.32, most values stay around roughly the same as in test scenario 2.1. The FPS has a drop from 29 at the highest to 17 at its lowest, this happens around where the change from Ethernet to Wi-Fi takes place. Still the values stay within the requirements for Webex stated in section 2.1.12.4.



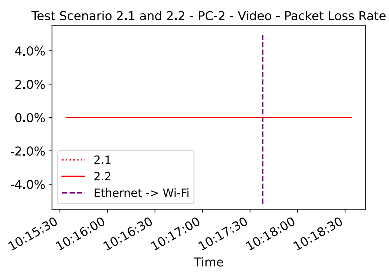
(a) Video - Frame Rate.



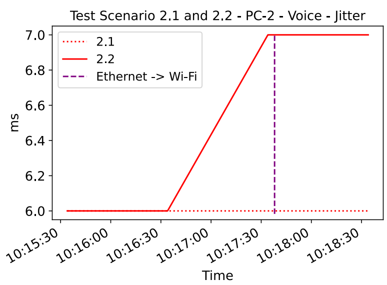
(b) Video - Jitter.



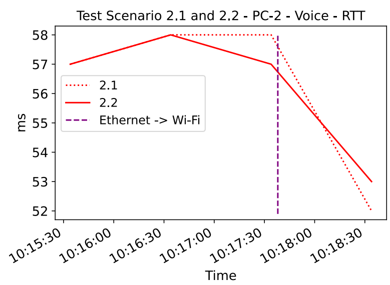
(c) Video - RTT.



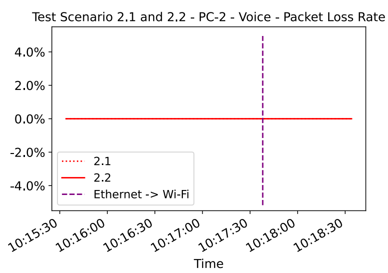
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



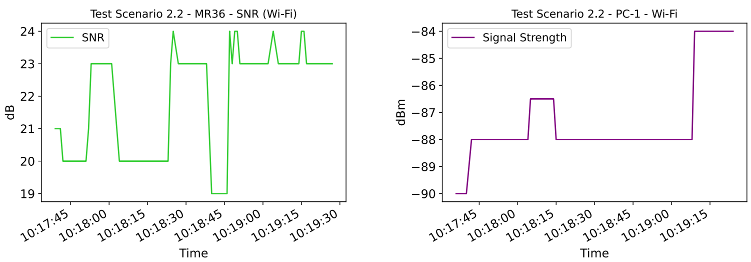
(f) Voice - RTT.



(g) Voice - Packet Loss Rate.

Figure 4.32: Data from Webex meeting: 2.1.4 and 2.2.3 for PC-2 [A.10]. License: Andreas Kilde Lien, CC BY.

The two graphs 4.33a and 4.33b shows the SNR and signal strength for PC-1 when connected to the AP. SNR values range between 19 and 24 dB, with an average of 21.90 dB. Compared to test scenario 2.1 that had an average of 46.59, is it a significantly lower SNR value for test scenario 2.2. For signal strength it range between -90 and -84 dBm, with an average of -87.34 dBm. The value for signal strength compared to test scenario 2.1's average of -55 dBm is significantly lower for test scenario 2.2.



(a) Signal-to-noise ratio for PC-1's connection to AP.

(b) Signal strength for PC-1's connection to AP.

Figure 4.33: Data connected to Webex meeting: 2.2.3 [A.10]. License: Andreas Kilde Lien, CC BY.

A passive site survey of the location was conducted following Cisco's best practices documentation [80]. The signal quality for PC-1 was -85 dBm, as shown in figure 4.34. Compared to test scenario 2.1 the signal strength is vastly more narrow.

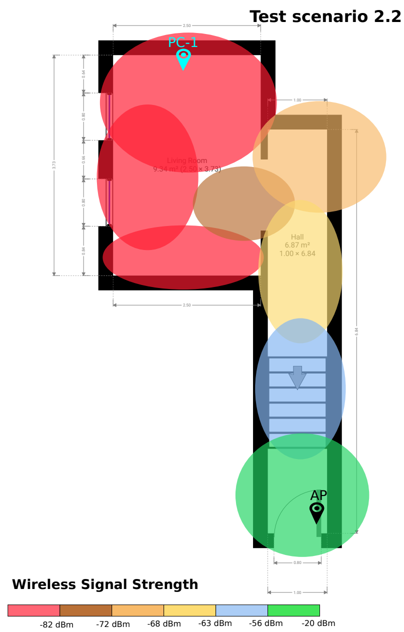


Figure 4.34: Site survey of PC-1 location where lower values are worse and higher values are better. License: Andreas Kilde Lien, CC BY.

4.5.3 Discussion - Webex 2.2

Similarly to test scenario 2.1 when the connection switch is introduced both PC's experience a complete drop to 0 Mbps in bandwidth which lasts for around 20 seconds. While after the switch in this test scenario the bandwidth becomes inconsistent ranging from no bandwidth sent to almost 6 Mbps on PC-1.

When Webex experiences large amount of packet loss it uses both Forward Error Correction (FEC) and Video retransmission (Video RTX) to try and smoothen out the meetings as best as possible [88]. The FEC mechanism tries to recover lost packets by sending extra error-correcting packets for every packet sent, while the video RTX will retransmit lost packets. Both of these mechanisms can send packets on top of the maximum bandwidth of 2 Mbps and this is believed to be the cause of the abnormal high amount of upload throughput.

PC-2 receive inconsistently 2 Mbps with some spikes to 2.5 Mbps and some drops to 0 mbps bandwidth received. This shows the inconsistency of the sent bandwidth from PC-1.

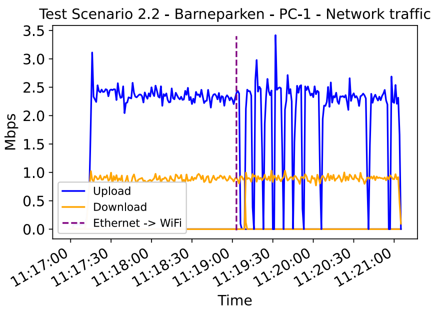
The change of connection is not seen in the bit rate media of PC-1 (figure 4.30) while PC-2 experiences a drop in received bit rate. Most of the values seen in test scenario 2.2 are within the requirements.

An interesting finding is the FPS received on both PC's drops around the time when the configuration switches from Ethernet to Wi-Fi. The drop is likely caused by PC-1 needing to reconnect and the video freezing or dropping out. The values on PC-1 for RTT was around 150-200 ms for voice and video and the packet loss for video was around 14% and for voice it was up to around 1.60% all of these qualify as "bad" values and can result in poor meeting quality when put in comparison to the requirements 2.1.12.4. It is reasonably certain that the values have been caused by the poor Wi-Fi connection introduced as all the values starts to rise after the configuration change from Ethernet to Wi-Fi.

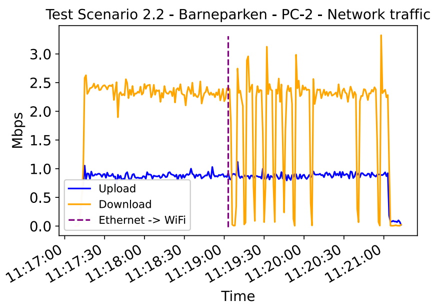
The SNR seen in graph 4.33a is significantly lower than it was in test scenario 2.1. Looking at the figure of the site survey 4.34 this makes sense in how far away the AP is away from the PC-1 combined with using 5 GHz which typically has shorter ranges compared to 2.4 GHz. The signal strength in dBm seen in graph 4.33b is significantly lower compared to test scenario 2.1. Looking at the values collected for the passive site survey 4.34 the low signal strength recorded makes sense as it is also seen in the site survey. A higher values in both SNR and signal strength result in better connection to the Wi-Fi some of the high values noticed in RTT and packet loss may have been caused by the low signal strengths.

4.5.4 Results - Teams 2.2

Just as in test scenario 2.1 the meeting was hosted in Amsterdam. In the graphs 4.35 below, a sudden increase in oscillations is seen at 11:19:00 and onward in upload for PC-1 and download for PC-2. This is the point were PC-1 swaps from Ethernet cable to Wi-Fi.



(a) Throughtput for PC-1 (UDP)



(b) Throughtput for PC-2 (UDP)

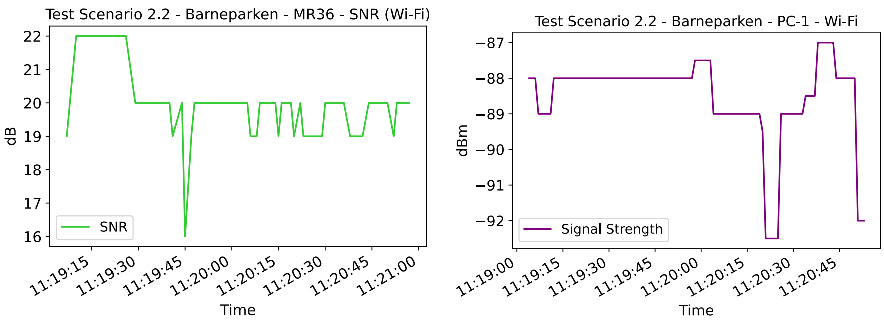
Figure 4.35: Data connected to Teams meeting, *barneparken*: 2.2.3 [A.9]. License: Andreas Rømo, CC BY.

In table 4.15 the values before the → are values that was looked at in the previous test scenario, 2.1 and values after → is for this test scenario, 2.2. There is a further increase of all the values previously seen for PC-1 in scenario 2.1.

Client	Media	Stream Direction	Avg. RTT	Avg. Packet Loss	Avg. Jitter
PC-1	Audio	Server to client	30ms → 43ms	2% → 2%	2ms → 7ms
PC-1	Audio	Client to server	29ms → 45ms	2% → 3%	1ms → 3ms
PC-2	Audio	Server to client	36ms → 36ms	0% → 0%	2ms → 1ms
PC-2	Audio	Client to server	36ms → 36ms	0% → 0%	0ms → 0ms
PC-1	Video	Server to client	30ms → 50ms	1% → 2%	0ms → 0ms
PC-1	Video	Client to server	30ms → 33ms	2% → 9%	0ms → 0ms
PC-2	Video	Server to client	36ms → 36ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	36ms → 36ms	0% → 0%	0ms → 0ms

Table 4.15: Data connected to Teams meeting, *barneparken*: 2.2.3 [A.9].

In figure 4.36b the signal strength for PC-1 is shown, while in figure 4.35 the SNR recorded from the AP is shown. The signal strength averages -88.5 dBm down from 54.9 dBm, in test scenario 2.1. The SNR also drops in average, 20 dB, down from 43 dB. The site survey mentioned in Webex also related to Teams (figure 4.34).



(a) Signal-to-noise ratio measued by the AP(MR36).

(b) Signal strength for PC-1's connection to AP

Figure 4.36: Wi-Fi signal data collected under Teams meeting held on Barneparken: 2.2.3 [A.9]. License: Andreas Rømo, CC BY.

4.5.5 Discussion - Teams 2.2

For the teams quality document every parameter except packet loss was inside the required range for a good quality meeting. The packet loss exceeded the maximum 1% for 4 streams. All these streams were for PC-1, while PC-2 had no packet loss at all. Packet-loss is something that can require a new I-frame to be sent. By inspecting the throughput graphs there are seen spikes which might indicate new I-frames sent. In correlation with the packet-loss this indicates lost video feed for the end users.

A look at the bandwidth for PC-1 in figure 4.35a shows drops in bandwidth. The drops are below the required bandwidth for the clients, which means the clients had problems in the call. The upload throughput from PC-1 is reflected in PC-2's download. This means also PC-2 would notice problems in the call.

The SNR shown in 4.36a is below the required SNR limit. For a video application such as Teams the bandwidth requirements is higher than for a pure voice application, and as such 25 dB for SNR would be too little for Teams. This indicates that the meeting was insufficient. The SNR (AP) and signal strength (PC-1) could explain why the throughput had many drops, as the WiFi connection might have closed and re-established several times.

From the bandwidth graph the upload of PC-1 is the one having frequent drops. The download is pretty similar with scenario 2.1 by having a stable bandwidth usage and a single drop in the middle. The reason for download not having the same drops could be explained by the bandwidth. Both the SNR (AP and signal strength (PC-1) is poor, but there is a difference in the throughput usage. The maximum download bandwidth usage relies on the signal strength and the maximum upload bandwidth relies on the APs SNR. Since this meeting required more upload bandwidth than download only the upload was affected. If the download bandwidth increased it also most likely would cause frequent drops in bandwidth.

4.6 Test scenario 3.1

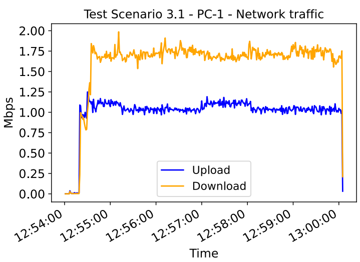
Test scenario 3.1 simulates having an extra third client in the meeting as seen in graph 3.3.6. This extra client is connected to Telia as ISP.

4.6.1 Objective

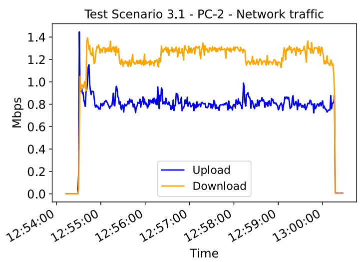
Objective for test scenario 3.1 is to see how PC-1 and PC-2 is compared to test scenario 1.1, and how a extra participant impacts the meeting quality in regards to network parameters. PC-1, PC-2 and PC-3 be put in relation to the requirements stated in sections 2.1.12.5 and 2.1.12.4.

4.6.2 Results - Webex 3.1

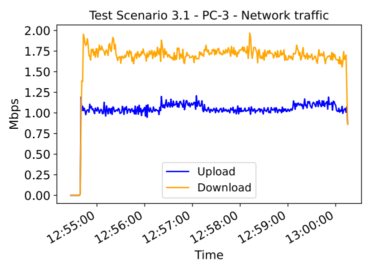
Looking at the graphs for throughput, PC-1 had an average upload speed of 1.05 Mbps and an average download speed of 1.69 Mbps, compared to an average of 1.68 Mbps upload speed and 820 Kbps download speed in test scenario 1.1. PC-2 had an average upload speed of 811 Kbps and an average download speed of 1.24 Mbps compared to an average of 828 Kbps upload speed and an average of 1.68 Mbps download speed in test scenario 1.1. The average upload speed for PC-3 was 1.05 Mbps and the average download speed was 1.71 Mbps. As PC-3 was introduced this test scenario, no previous data is recorded to compare to, but put in relation to both PC-1 and PC-2, PC-3 had the highest throughput of both upload and download.



(a) Throughput PC-1 (UDP)



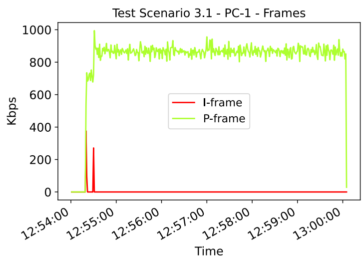
(b) Throughput PC-2 (UDP)



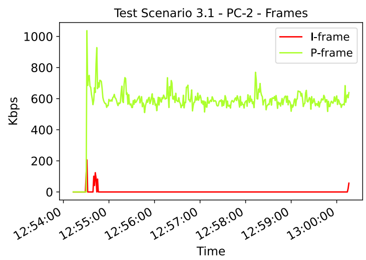
(c) Throughput PC-3 (UDP)

Figure 4.37: Data connected to Webex meeting: 3.1.4 [A.10]. License: Andreas Kilde Lien, CC BY.

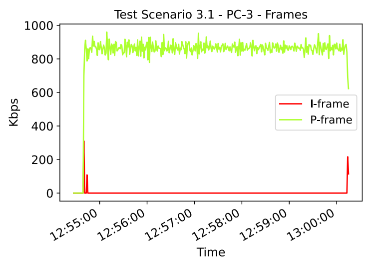
The graphs 4.38a, 4.38b and 4.38c shows the upload dispersion of I- and P-frames for PC-1, PC-2 and PC-3 throughout the meeting. For PC-1 it is mostly identically to test scenario 1.1 except for a much lower bit rate. PC-2 has in the scenario changed to use another webcam so, it is not passable to do any direct compression. But do experience some shifting to begin with all the way from 1000 to 600 Kbps with two I-frames from the start and one at the end. Is mostly stable at around 600 Kbps. For PC-3 that uses the same webcam as PC-2 in test scenario 1.1 it stays close to or above 800 Kbps for P-frames, while sending two I-frames at the beginning and one at the end of the meeting. The meeting is mainly distributed by P-frames for all of the PCs.



(a) The dispersion of I- and P-frames using the UDP protocol for PC-1. The graph is including L3 overhead.



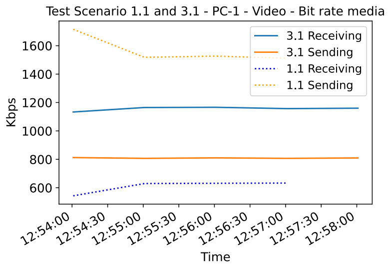
(b) The dispersion of I- and P-frames using the UDP protocol for PC-2. The graph is including L3 overhead.



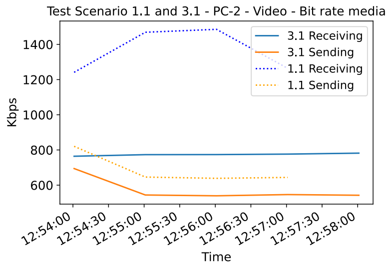
(c) The dispersion of I- and P-frames using the UDP protocol for PC-3. The graph is including L3 overhead.

Figure 4.38: Data from Webex meeting: 3.1.4 [A.10]. License: Andreas Kilde Lien, CC BY.

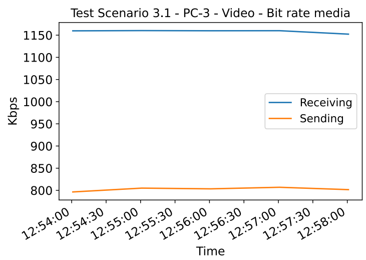
The bit rate received for the clients is shown in graph 4.39. Both the download and upload for PC-1 and PC-2 has changed from test scenario 1.1. The bit rate received on PC-3 shown in graph 4.39c is around 1160 Kbps throughout the meeting while the sent bit rate is around 800 Kbps.



(a) Bit rate for PC-1.



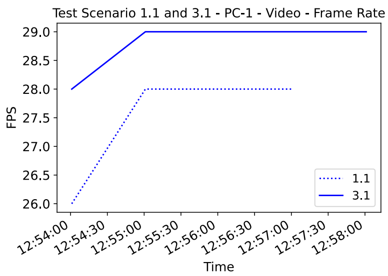
(b) Bit rate for PC-2.



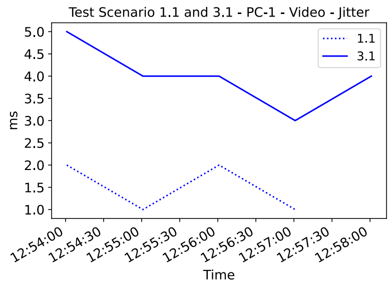
(c) Bit rate for PC-3.

Figure 4.39: Data from Webex meeting: 1.1.6 and 3.1.4 [A.10]. License: Andreas Kilde Lien, CC BY.

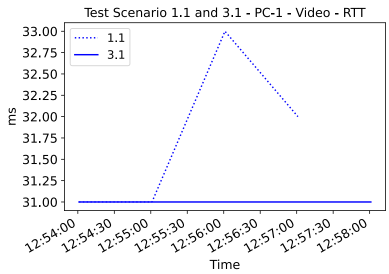
The FPS and jitter for video for PC-1 are both around 2 to 3 ms higher than in test scenario 1.1. The rest of the values stay close to the values of test scenario 1.1 as shown in graphs 4.40. All the values presented are within the requirements stated in section 2.1.12.4.



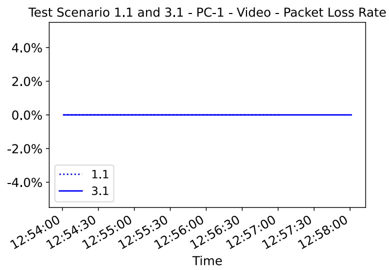
(a) Video - Frame Rate.



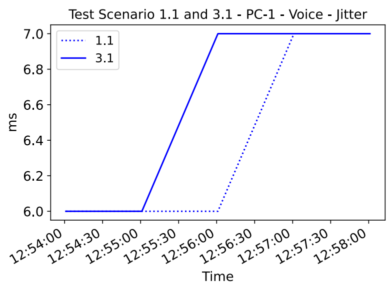
(b) Video - Jitter.



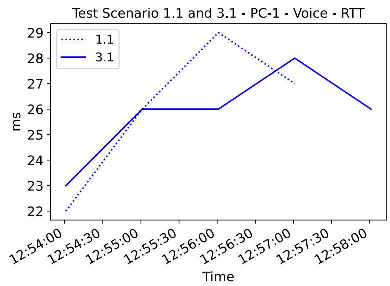
(c) Video - RTT.



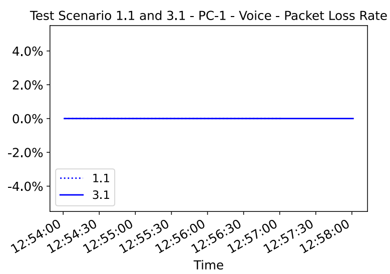
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



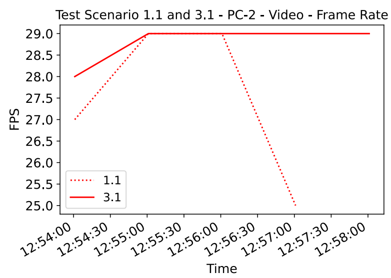
(f) Voice - RTT.



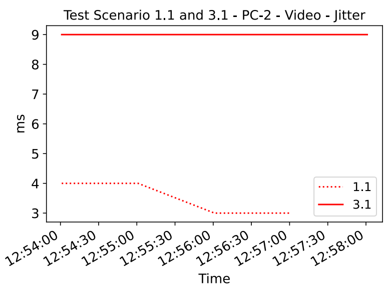
(g) Voice - Packet Loss Rate.

Figure 4.40: Data from Webex meeting: 1.1.6 and 3.1.4 [A.10] for PC-1. License: Andreas Kilde Lien, CC BY.

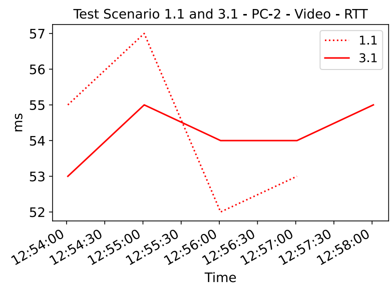
The biggest change from test scenario 1.1 for PC-2 is the higher video jitter which is around 5 too 6 ms higher. The other values stay within an acceptable range of test scenario 1.1 as shown in graphs 4.41. All the values presented are within the requirements stated in section 2.1.12.4.



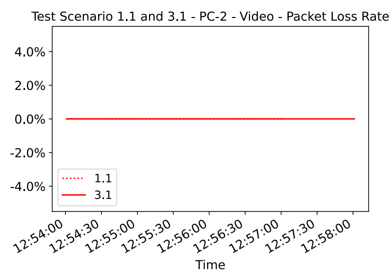
(a) Video - Frame Rate.



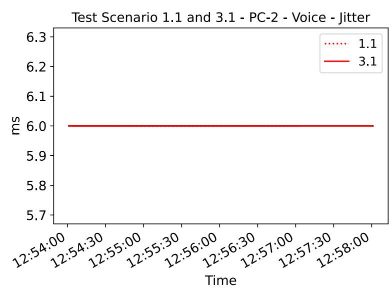
(b) Video - Jitter.



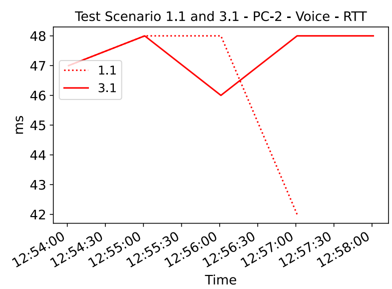
(c) Video - RTT.



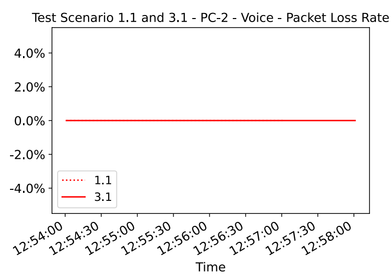
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



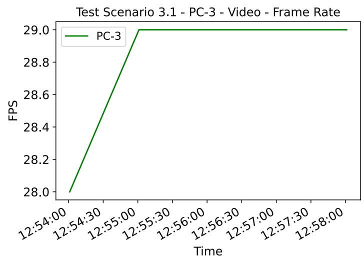
(f) Voice - RTT.



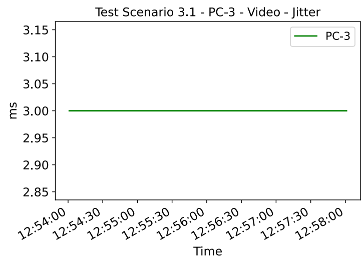
(g) Voice - Packet Loss Rate.

Figure 4.41: Data from Webex meeting: 1.1.6 and 3.1.4 [A.10] for PC-2. License: Andreas Kilde Lien, CC BY.

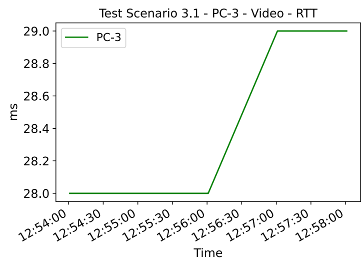
All the values presented in the graphs 4.42 for PC-3 shows the values to be within an acceptable range of at most a change of 3 in the value. Resulting in all the values presented being within the requirements stated in section 2.1.12.4.



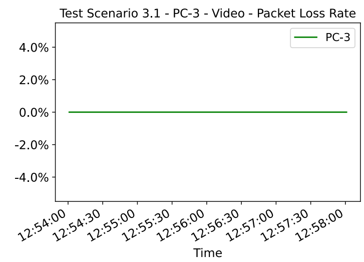
(a) Video - Frame Rate.



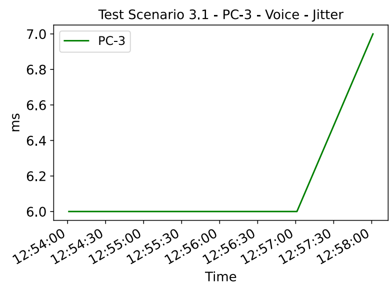
(b) Video - Jitter.



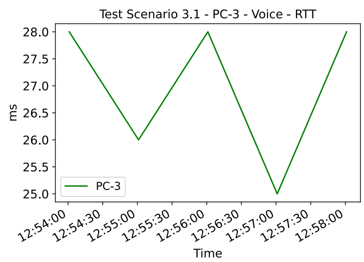
(c) Video - RTT.



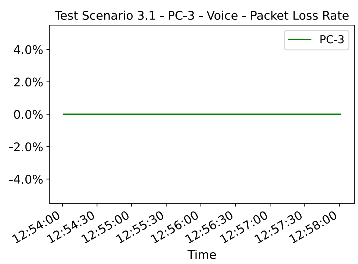
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



(f) Voice - RTT.



(g) Voice - Packet Loss Rate.

Figure 4.42: Data from Webex meeting: 3.1.4 [A.10] for PC-3. License: Andreas Kilde Lien, CC BY.

4.6.3 Discussion - Webex 3.1

Test scenario 3.1 introduced a new participant, which makes a total of three participants. Compared to test scenario 1.1 with two clients this introduces new challenges as there is an extra video and audio stream to send and receive traffic from and to.

An interesting observation is the change in throughout experienced for both PC-1 and PC-2. PC-1 has an decrease of 630 Kbps average upload, while an increase in the average download speed by 870 Kbps. This is likely caused by having another client in the meeting, which will scale down the image sent needing less upload as well as receiving more video streams increasing the download. This is likely the cause for the change in bit rate as well for the PC's. For PC-2 there wasn't a big change in the average upload speed, while the average download speed had a decrease of 440 Kbps.

PC-3 put in relation to the other clients had the highest average upload and download speed, but the values where roughly the same as for PC-1. Both the throughput and the bit rate for PC-2 behaves in a strange way staying at lower values compared to the other PC's. The expected results would be that all the PC's had around the same values for both received and sent as all the PC's used the same resolution.

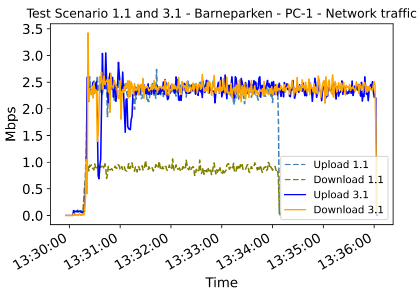
In the frame graphs 4.38a and 4.38b for PC-2 and PC-3 there is an I-frame at the ending of the meeting for both. For PC-1 there are no I-frames at the end. The reason for sending I-frame at the end of the meeting is most likely in correlation with PC-1 leaving the meeting. When PC-1 has left the meeting the graphical interface of Webex starts shifting from a three box view to two boxes view side by side. Judging by the size of the I-frame each time a participant with video leaves forces the application to send a new I-frame.

The values FPS and jitter are interesting as these are the ones with the biggest change. The FPS is consistently high overall for all the PC's this is likely caused by all the clients using a lower resolution than their highest possible, making it easier to maintain a higher FPS. The jitter for video was higher for both PC-1 and PC-2, this could have been caused by an increase in congestion following the addition of another client into the meeting.

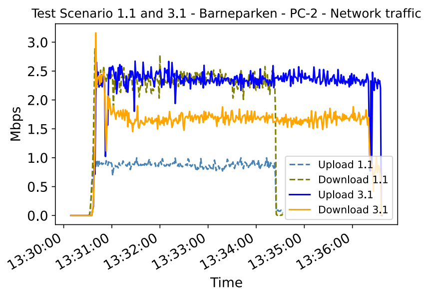
All of the values remained within the requirements by Webex stated in section 2.1.12.4. This meeting can therefore be qualified as good quality.

4.6.4 Results - Teams 3.1

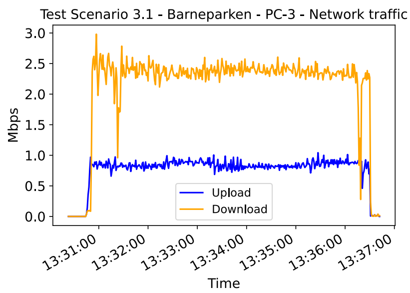
Server location for this meeting was Amsterdam. As seen in graphs 4.43 there are three clients with different bandwidth usage. PC-1 and PC-2's graph also has scenario 1.1's results plotted in for comparison. It is hard to see exactly where the line for test scenario 1.1 is in the graph as there is a lot of overlapping, therefore it can be helpful to see to the original results in 4.5.



(a) Throughtput for PC-1 (UDP)



(b) Throughtput for PC-2 (UDP)



(c) Throughtput for PC-3 (UDP)

Figure 4.43: Data connected to Teams meeting, *barneparken*: 3.1.4 [A.9]. License: Andreas Rømo, CC BY.

In table 4.16 streams for all clients is shown. A main difference in the table is the clients have two uploading streams for video instead of one.

Client	Media	Stream direction	Avg. RTT	Avg. packet loss	Avg. jitter
PC-1	Audio	Server to client	26ms → 25ms	0% → 0%	1ms → 2ms
PC-1	Audio	Client to server	26ms → 25ms	0% → 0%	1ms → 1ms
PC-2	Audio	Server to client	36ms → 34ms	0% → 0%	0ms → 1ms
PC-2	Audio	Client to server	36ms → 35ms	0% → 0%	1ms → 0ms
PC-3	Audio	Server to client	24ms	0%	1ms
PC-3	Audio	Client to server	24ms	0%	0ms
PC-1	Video	Server to client	26ms → 26ms	0% → 0%	0ms → 0ms
PC-1	Video	Server to client	26ms → 26ms	0% → 0%	0ms → 0ms
PC-1	Video	Client to server	26ms → 26ms	0% → 0%	0ms → 0ms
PC-2	Video	Server to client	36ms → 34ms	0% → 0%	0ms → 0ms
PC-2	Video	Server to client	36ms → 34ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	36ms → 35ms	0% → 0%	0ms → 0ms
PC-3	Video	Server to client	24ms	0%	0ms
PC-3	Video	Server to client	24ms	0%	0ms
PC-3	Video	Client to server	24ms	0%	3ms

Table 4.16: Data extracted using Power BI from a meeting on tenant barneparken: 3.1.4 [A.9].

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4.6.5 Discussion - Teams 3.1

Test scenario 1.1 and this test scenario is similar in parameters, as seen in table 4.16. There are no parameters to compare to PC-3 because test scenario 1.1 does not have a third client to compare with. The comparison between PC-1 from test scenario 3.1 and test scenario 1.1 shows no real deviations.

The video streams in table 4.16 has in this test scenario extra streams. Instead of the original two video streams seen from test scenario 1.1 there are three video streams. Each client now receives two streams from the server, one for each participant in the call, and sends one stream to the server. The voice streams do not have an any extra streams. When voice streams reached the server in Amsterdam from several clients it probably merges the voice streams into a single voice stream for each participant. On the other side when video streams is sent to the server in Amsterdam it is simply relayed instead of merged. Part of the reason for this is the possibility for a user to choose video layout within the Teams application. In the event a single user wants to enlarge a participants webcam only that single stream need to upscale its resolution. While in the case of merging the streams, the merged stream would send the upscaled resolution to every participant, even though some participants wouldn't need it.

The upload from client PC-2 is on average 2.5 Mbps and PC-3's upload averages at 1 Mbps. This means from PC-2 and PC-3 the server in Amsterdam received on average 3.5 Mbps from PC-2 and PC-3. PC-1 who is receiving the streams from the other clients however only receives 2.5 Mbps. One or both of the video streams has downgraded the video resolution. The reason for this downgrade is PC-1s screen size. For example it does not make sense to send a 4K video to a small phone. In the case of test scenario 1.1 PC-2s webcam took the whole screen for PC-1, but in scenario 3.1 PC-1 has two webcams showing on its screen. This means the resolution needed for showing PC-2s webcam has halved. In turn PC-1 does not need the full resolution sent to it, only the resolution required. This also is true for PC-2 and PC-3. PC-2 and PC-3 does not need to receive higher resolution than it can display on the screen.

From Teams quality document every parameter collected is within range. And the deviations from test scenario 1.1 did not impact the quality of the meeting.

4.7 Test scenario 4.1

Test scenario 4.1 simulates PC-2 going through a 4G cellular gateway, which connects to a Telenor cellular tower to join the meeting. This will try to showcase how a meeting from a cabin without a reliable network should be like. As for comparison purposes PC-1 would ideally have been the one to go trough a 4G cellular gateway, but due to problems replicating poor 4G signal it was decided to rather use PC-2.

4.7.1 Objective

Test scenario 4.1 will be compared against 1.1 and be put up against the quality documents for Webex and Teams. The objective is to see how a 4G router effects the quality of the meeting.

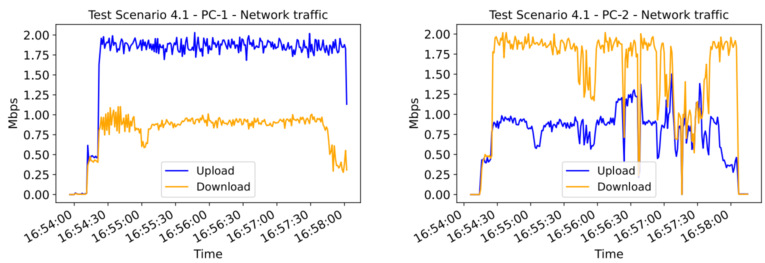
4.7.2 Results - Webex 4.1

Table 4.17 shows the Webex network test taken before the meeting. Comparing this to the table in test scenario 1.1 (table 4.1) there are some differences. The TCP delay for PC-2 has a difference of 21 ms. Cisco evaluates all delays <=100ms as good and 100 300 ms as fair. There is also a difference of 4.69 Mbps in TCP download speed and a difference of 9.99 Mbps in TCP upload speed. For PC-1 the only real difference for TCP is in TCP upload speed with a difference of 22.25 Mbps. The UDP download and upload speed for both PC's are quite comparable to test scenario with the UDP download speed of PC-1 having the largest difference of 1.74 Mbps. Cisco evaluates all bandwidth >=2 Mbps as good [81].

	PC-1	PC-2
TCP Delay	93.85 ms	117.65 ms
TCP Download Speed	92.58 Mbits/s	9.83 Mbits/s
TCP Upload Speed	66.83 Mbits/s	2.64 Mbits/s
UDP Delay	83.00 ms	105.00 ms
UDP Download LossRate	0.00%	0.00%
UDP Upload LossRate	0.00%	0.00%
UDP Download Speed	2.87 Mbits/s	3.56 Mbits/s
UDP Upload Speed	4.48 Mbits/s	3.43 Mbits/s

Table 4.17: Cisco Webex Network Test for PC-1 and PC-2.

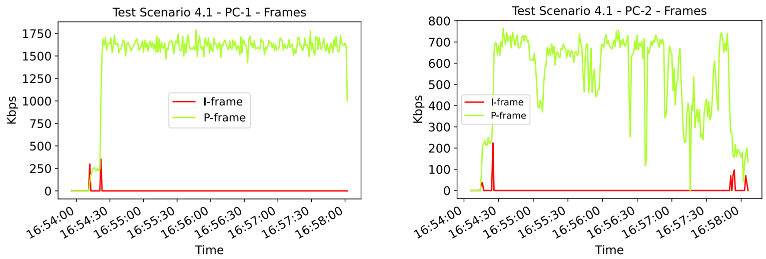
Looking at the graphs 4.44 for throughput, PC-1 had an average upload of 1.86 Mbps and an average download of 873 Kbps, compared to an average of 1.68 Mbps upload speed and 820 Kbps download speed in test scenario 1.1. PC-2's throughput had an average upload speed of 840 Kbps and an average download speed of 1.64 Mbps, compared to an average of 828 Kbps upload speed and an average of 1.68 Mbps download speed in test scenario 1.1.



(a) Throughput PC-1 (UDP) **(b)** Throughput PC-2 (UDP)

Figure 4.44: Data connected to Webex meeting: 4.1.1 [A.10]. License: Andreas Kilde Lien, CC BY.

The graphs 4.45a and 4.45b shows the upload dispersion of I- and P-frames for PC-1 and PC-2. For PC-1 it is mostly identically to test scenario 1.1. For PC-2 it starts off at 800 Kbps but after 30 seconds it falls to 400 Kbps and is fast to stabilise to around 600 to 750 Kbps for the rest of the meeting.

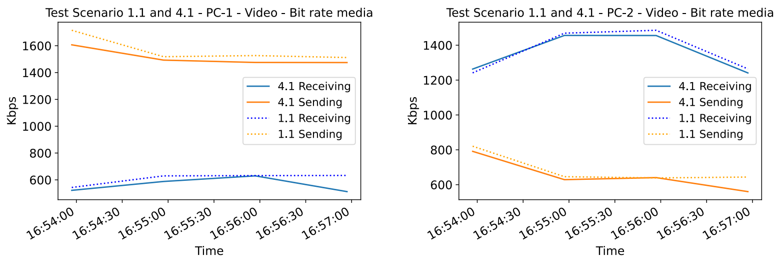


(a) The dispersion of I- and P-frames using the UDP protocol for PC-1. The graph is including L3 overhead.

(b) The dispersion of I- and P-frames for PC-2 from the AP. The graph is including L3 overhead.

Figure 4.45: Data from Webex meeting: 4.1.1 [A.10]. License: Andreas Kilde Lien, CC BY.

Values from Webex Control Hub shows that there is not much difference from test scenario 1.1 values in regards to bit rate for both PC-1 and PC-2, as shown in graphs 4.46a and 4.46b.

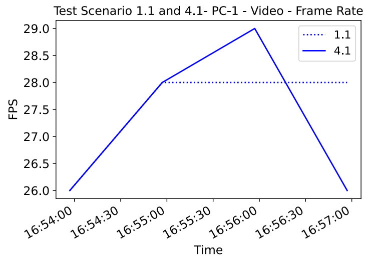


(a) Bit rate for PC-1.

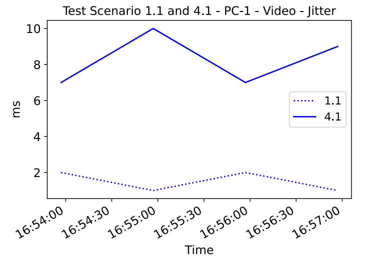
(b) Bit rate for PC-2.

Figure 4.46: Data from Webex meeting: 4.1.1 [A.10]. License: Andreas Kilde Lien, CC BY.

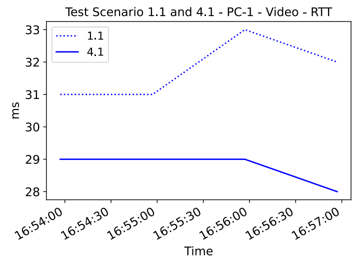
Looking at the graphs in 4.47 most of the values, with the exception of the FPS and the packet loss for voice and video, has a significant difference from test scenario 1.1. Both the jitter for voice and video is consistently higher than in test scenario 1.1. For video it is 10 ms at max, while for voice the max is 14 ms jitter. The RTT for video stays lower than test scenario 1.1 at 28 to 29 ms. RTT for voice stay higher than test scenario 1.1 throughout the meeting with 32 ms as max RTT.



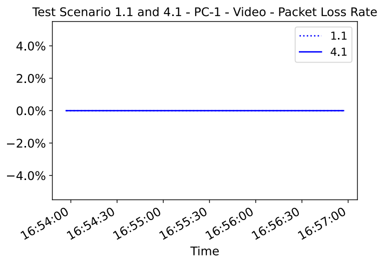
(a) Video - Frame Rate.



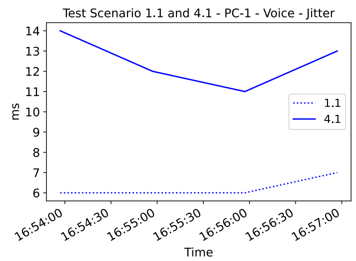
(b) Video - Jitter.



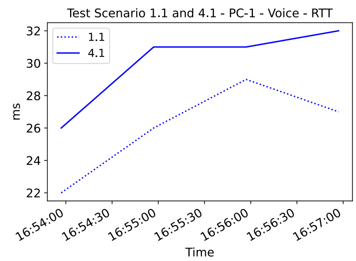
(c) Video - RTT.



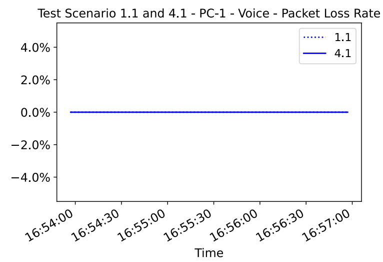
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



(f) Voice - RTT.

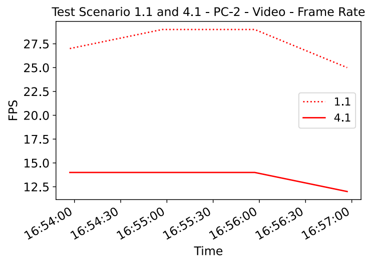


(g) Voice - Packet Loss Rate.

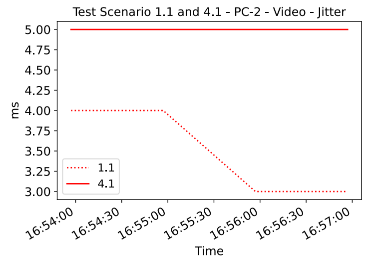
Figure 4.47: Data from Webex meeting: 1.1.6 and 4.1.1 [A.10]. License: Andreas Kilde Lien, CC BY.

Looking at the graphs in 4.48 all the values except for packet loss for voice and video has a change from test scenario 1.1. The FPS through the meeting is 12 FPS at minimum and 14 FPS at max. This is a drastic change from test scenario 1.1 where it stays around 23 to 28 FPS. The jitter for both voice and video stays around 1 to 3 ms higher than in test scenario 1.1. The RTT seen for video goes down to 96 ms minimum and up to 238 ms at max. Compared to test scenario

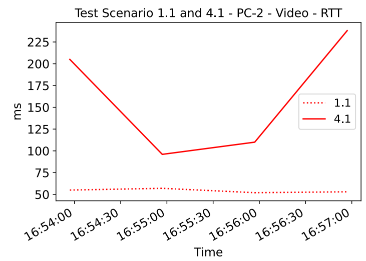
1.1 which was between 52 ms and 57 ms this is a large difference. The RTT seen for voice goes down to 92 ms minimum and up to 210 ms maximum. Compared to test scenario 1.1 which was between 42 and 48 which is also quite a large difference. The RTT with its high values does not meet the requirements stated by Webex in section 2.1.12.4, while the rest of the values do.



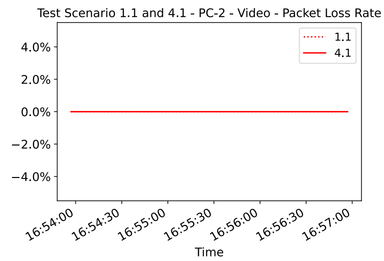
(a) Video - Frame Rate.



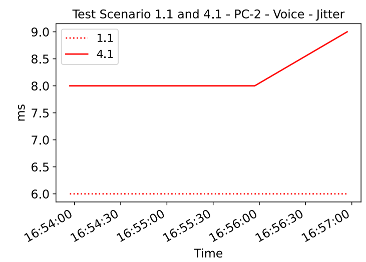
(b) Video - Jitter.



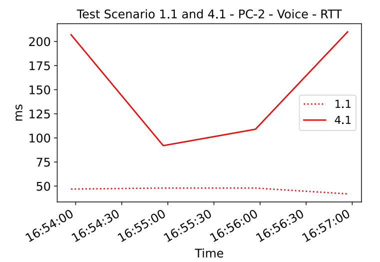
(c) Video - RTT.



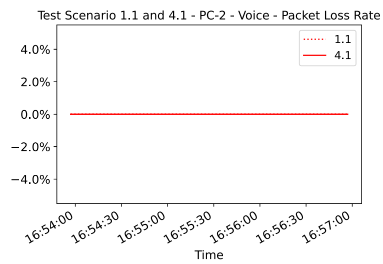
(d) Video - Packet Loss Rate.



(e) Voice - Jitter.



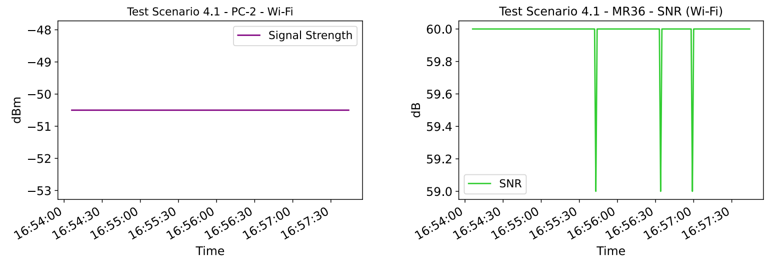
(f) Voice - RTT.



(g) Voice - Packet Loss Rate.

Figure 4.48: Data from Webex meeting: 1.1.6 and 4.1.1 [A.10]. License: Andreas Kilde Lien, CC BY.

The signal strength of PC-2 shown in graph 4.49a stays at -50.5 dBm throughout the whole meeting. The SNR shown in graph 4.49b stays at 60 dB through most of the meeting, but drops 1 dB around the times 16:55:42, 16:58:32 and 16:57:00.

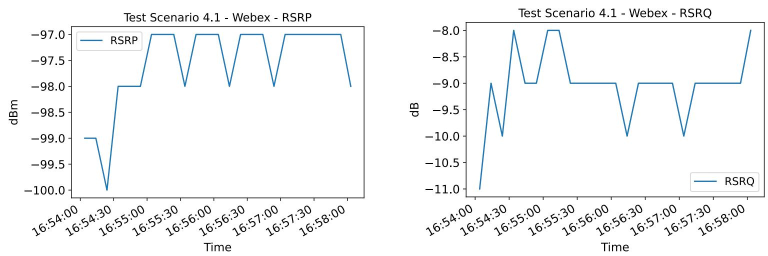


(a) Signal strength for PC-2's connection to AP.

(b) Signal-to-noise ratio for PC-2's connection to AP.

Figure 4.49: Data connected to Webex meeting: 4.1.1 [A.10]. License: Andreas Kilde Lien, CC BY.

The signal quality recorded in graph 4.50a shows the quality ranging from -100 to -97 dBm. Signal strength is shown in graph 4.50b it shows the strength ranging from -11 to -8 dB.



(a) Signal quality for 4G router connection.

(b) Signal strength for the 4G router connection.

Figure 4.50: Data connected to Webex meeting: 4.1.1 [A.10]. License: Andreas Kilde Lien, CC BY.

4.7.3 Discussion - Webex 4.1

Test scenario 4.1 changes from Ethernet connection on both clients to Ethernet on PC-1 and a 4G router connected to an AP which gives a Wi-Fi connection to PC-2. An interesting finding is the large difference on PC-2 from test scenario 1.1 seen in the throughput graph 4.44b and the frame dispersion graph 4.45b where the values in 4.1 are all over the place. A reason for this could be the connection to the server with the 4G router seen in the low values of TCP download and upload speed in the network test 4.17 compared to those of test scenario 1.1.

Looking at the RSRP and RSRQ and putting them in relation to the heatgraph

2.8 the meeting could be qualified as between a 4 and a 6 which is a decent to good connection to the 4G tower.

Even though the P-frames in the graph 4.45b are all over the place and even drops out one time, new I-frames are only sent at the start of the meeting and at the very end. An interesting discovery is that PC-2 has three I-frames close in time at the end of the meeting. The reason for this behavior could be found in figure 4.44b (extracted from Meraki) that experience a gain in download speed but in a shifting way.

The recorded FPS is very low compared to test scenario 1.1, this may have affected the meeting and caused stuttering in the video if there was a lot of movement. Another interesting value is the RTT for both voice and video which is at the start and the end of the meeting higher than what is recommended for a good meeting in the Webex requirements stated in section 2.1.12.4. Even with the high RTT it is strange that no packets were lost.

With the high values of RTT and the fluctuating throughput this meeting was likely experienced as poor in quality by the clients compared to test scenario 1.1.

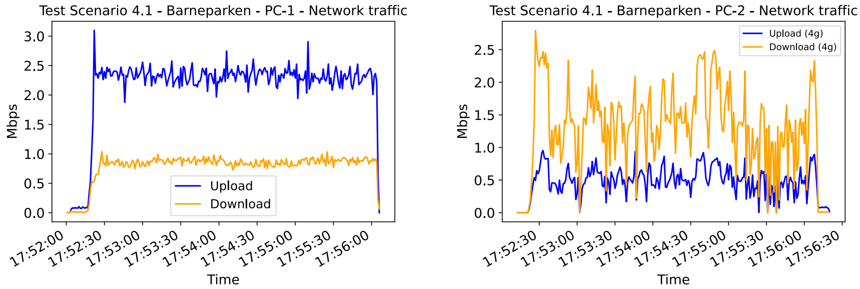
4.7.4 Results - Teams 4.1

Server location for this scenario was Amsterdam. Stream parameters from power BI can be seen in table 4.18. The only change from test scenario 1.1 for PC-1 is a 1 ms decrease for audio and video sent from the server. PC-2 in has increased in Avg RTT for all stream, 36 ms to 67 to 68 ms. Avg jitter for Audio PC-2, has also increased 8 to 12 ms.

Client	Media	Stream Direction	Avg. RTT	Avg. Packet Loss	Avg. Jitter
PC-1	Audio	Server to client	26ms → 25ms	0% → 0%	1ms → 1ms
PC-1	Audio	Client to server	26ms → 26ms	0% → 0%	1ms → 1ms
PC-2	Audio	Server to client	36ms → 68ms	0% → 0%	1ms → 8ms
PC-2	Audio	Client to server	36ms → 67ms	0% → 0%	0ms → 12ms
PC-1	Video	Server to client	26ms → 25ms	0% → 0%	0ms → 0ms
PC-1	Video	Client to server	26ms → 26ms	0% → 0%	0ms → 0ms
PC-2	Video	Server to client	36ms → 64ms	0% → 0%	0ms → 0ms
PC-2	Video	Client to server	36ms → 65ms	0% → 0%	0ms → 0ms

Table 4.18: Data connected to Teams meeting, *barneparken*: 4.1.1 [A.9].

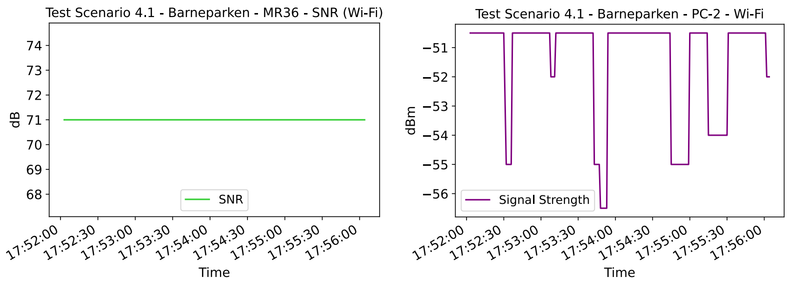
In figure 4.51 the bandwidth is shown for both clients.



(a) Throughtput for PC-1 (UDP). **(b)** Throughtput for PC-2 (UDP).

Figure 4.51: Data connected to Teams meeting, *barneparken*: 4.1.1 [A.9]. License: Andreas Rømo, CC BY.

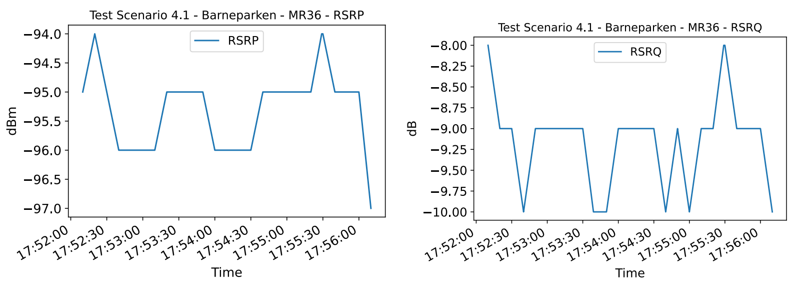
The SNR and signal strength is shown in figure 4.52. PC-2 has high oscillations through the whole meeting.



(a) SNR measured by the MR36 during Teams meeting 4.1. **(b)** Signal Strength between PC-2 and AP, during teams meeting.

Figure 4.52: Data connected to Teams meeting, *barneparken*: 4.1.1 [A.9]. License: Andreas Rømo, CC BY.

From the MR36, RSRP and RSRQ was recorded and is shown in figure 4.53.



(a) Signal quality for 4G router connection. **(b)** Signal strength for the 4G router connection.

Figure 4.53: Data connected to Teams meeting, *barneparken*: 4.1.1 [A.9]. License: Andreas Rømo, CC BY.

4.7.5 Discussion - Teams 4.1

From the power BI data PC-2 has higher RTT compared to test scenario 1.1 PC-2. There is also no packet loss, but voice stream jitter for PC-2 has increased from 0 - 1 ms to 8 - 12 ms. Despite these higher values the parameters from power BI are inside the required ranges set by MS Teams.

The used bandwidth for PC-1 has average of 0.84 Mbps for download and 2.27 Mbps for upload. This is inside the quality requirements. PC-2 has in comparison to PC-2 in test scenario 1.1 greater oscillations for both download and upload. The oscillations could be a result of the fact that 4G is a shared medium. If there are many users using the medium with different required bandwidth the available bandwidth for PC-2 could fluctuate as a consequence. The time of day could play a role in these fluctuations.

SNR for PC-2 is 46 dB above the requirement, which is enough for a voice application such as Teams. Since the signal strength shows similar results with test scenario 1.1 the communication between the client and the AP should not be a bottleneck. The connection between the client and AP can therefore be considered as "good".

The RSRP ranges from -97 to -94, and from the Meraki documentation equals to three out of five bars [89]. The RSRQ ranges from -10 to -8 and is four bars. Since the AP chooses the lowest of these two measurements the test was performed with three bars. From Technical Background the heat graph 2.8 depicts RSRP and RSRQ in relation to each other, and tells if the signal can be regarded as good or bad. From the graph the connection to the 4G tower is in the half yellow half green zone trough most of the meeting. This would be regarded as good connection, but not excellent.

Chapter 5

Main discussion

By having all the tests conducted and individually discussed there still remains a general discussion for the test scenarios quality and discussions on choices made in regards to procedures, methods and tools used.

5.1 General Discussion

In this section the scenarios are discussed in an overview for both Webex and Teams.

5.1.1 General discussion - Webex

The meeting quality on Webex was mostly good, but when poor connections were introduced into the meeting, the quality deteriorated as expected. When switching from one configuration to another e.g. UDP to TCP or Ethernet to Wi-Fi both clients experienced a long delay before the meeting went back to normal.

For every Webex meeting the Amsterdam media node was used, this has helped in keeping all the values comparable by not bringing in any additional values from going to another media node.

5.1.2 General discussion - Teams

Each and every test scenario was performed on both tenants. However, only test scenario 1.1 to test scenario 1.3 was discussed for tenant *skytjenester*. The reasoning for this is it was apparent from early on that tenants did not matter in meeting quality. Some investigation outside the test scenarios by asking Microsoft and Telenor directly gave information that this might not actually be entirely accurate. Because of the corona situation Teams probably decided to load-balance video calls on different data centers around Europe. Therefore it could be interesting to see if the results from this thesis would differ in the future.

In general Teams quality performed inside the required parameters set my MS Teams. Teams also showed resilience against different network deviations to keep the connection and quality of a call up. The times Teams struggled with quality was with bandwidth at 2 Mbps and below, when UDP was blocked

5.1.2.1 Teams - Network parameters

In MS Teams all the parameters from power BI, such as RTT, packet loss, jitter are all an average. This can induce false or misleading claims. As an example in test scenario 2.1 (WiFi) it is claimed that the reason for the higher RTT (compared to test scenario 1.1) is the Wi-Fi and the change from Ethernet to Wi-Fi. This is however a bold claim as the latency is never recorded in an interval, but rather just an average. There is no way of knowing if the RTT actually was equal to test scenario 1.1s RTT. It could all have been the transfer from Ethernet to Wi-Fi that increased the average.

Another aspect which can be misleading is the way some of the scenarios was conducted with half in half of Ethernet and the new connection type. As an example in test scenario 1.2 (ADSL) 2 minutes of the meeting was exactly the same as test scenario 1.1, while the last 2 minutes had a simulated ADSL connection by applying a group-policy. All the parameters gathered by power BI are summarized, and as such the results shown in power BI will only show half the truth. As an example the average in the first half can be 30ms and the second half can be 60ms. From power BI it will show 45ms average, which is not untrue, just not the whole picture.

5.2 Testing procedure

In this section choices made for the testing procedure is touched upon.

5.2.1 Webcam and video replaying

A crucial decision made in the testing procedure was if to use a glitter lamp (RGB) instead of video replaying software on the Webcam. With the use of a video replaying software it would eliminate various uncertainties. It would be easier to argue that nothing unexpected comes from a replayed video.

A counter argument would be that it won't be realistic, which was important from this thesis. A glitter lamp is of course not 100% realistic, but it is realistic enough for the purpose of this paper. In addition to having realistic scenarios, video replaying software might influence the test results by eating resources from the client. This is unlikely, but contributed to rather using a lava lamp than software.

When running a minimum of 72 videoconference meetings it is important to also keep the audio equal, but realistic. Therefore the use of pre-recorded audio from a mobile device close to the computer microphone was used to imitate a human conversation. An alternative was to exclude the microphone of the computer and to convert the audio output to the microphone, then run a pre-recorded audio on the audio output. This was excluded for the same reasons as mentioned for video.

5.2.2 Averaging data-points

Another important decision was if all the scenarios should have an average of network parameters or if a single test was to be analysed. In this thesis only one test was shown for each scenario. As an example, test scenario 2.2 (Poor WiFi) was ran 5 times, but only the last one test-run is analysed in the report.

Making an average of tests could have flattened out errors, and be more representative as a whole. This was however not chosen because of the time limit and amount of work needed. As there is several gigabytes of data collected each test it would require several hours of either manually merging data or several hours of making scripts to automate the process.

5.2.3 Tools used

There are a lot of tools used to gather data in this thesis. One of them is Meraki cloud. Meraki was used to do the pcaps, but the pcaps could also have been captured locally at each client. The reason for using Meraki was because of a wish from Telenor. The same goes for data extracted from the tenants using Webex Control Hub and power BI. This however have not weakened the thesis. It was only a different way of gathering data.

A part of the data collection procedure is recovering analytical data from ntopng. Originally this was not part of the tools used, and only Meraki pcap, power BI and Webex Control Hub was to be used. It was however necessary to use ntopng as both a way to keep tabs on a meeting real-time and as a way to back up claims made by other tools. Webex Control Hub has been a good tool for gathering data, even though the export to JSON function loses the last minute of the meeting.

5.2.4 Rating

As part of the procedure there was a wish from Telenor to rate each meeting to see how the different test scenarios affected the end users experience. These ratings was carried out, but by discussing with Telenor the data was decided to be too biased. Everyone involved in the testing knew the test scenarios were being conducted and what to expect. All the iterations with rating can be found in appendix, see section A.11.

Chapter 6

Conclusion

In previous chapters it was pointed out that the aim of this thesis as a whole was to examine the quality of Webex and Teams meetings. To realise this, technical information regarding the platforms was presented to give a basis for understanding the platforms. Next in line was the method, in which included the baselines environment, how to perform the experiment and how to extract the data. The method laid the ground work for analysis and discussion for each test scenario.

With the technical understanding and the method set the data was presented, analysed and discussed in relation to quality. Now remains to conclude.

6.1 Future use of thesis

This thesis could be used for troubleshooting Webex and Teams for system administrators. By giving insight into the quality requirements and how the applications function a system administrator could more easily give users in an organization a high quality experience.

Further on a set of new test scenarios could be created using this thesis. The new scenarios could be a meeting where both participants block UDP, or a meeting where one participant is on a VPN. There are lots of scenarios that can be built upon the baseline and also combination of different test scenarios.

It could also be completed tests on different geographical locations. This would give insight into how a location for participants in a meeting affects network variables. The results could be compared to results presented in this paper.

6.2 Strengths and weaknesses

In this section the strength and weaknesses in the paper is described. Validity of the data and statements will also be put into question.

6.2.1 Strengths

During this thesis, information has been gathered from individuals working inside Cisco Meraki, Cisco Webex, MS Teams, Uninett and Telenor. This information includes Merakis NTP setup, Telenors peering partners, Uninetts peering partners, Teams internal infrastructure and Webex infrastructure.

The information gathered from these individuals was used to figure out bandwidth requirements for Webex, I- and P-frames utilization for Webex, FEC and RTX packets used in test scenario 3.1, centralized topology for Webex, UDP block affecting reconnecting time for Webex.

6.2.2 Weaknesses

For all the data gathered everything is interpreted by the writers of this thesis. Every statement from communicating with personnel, such as personnel from MS Teams or Webex, have not confirmed their statements in this thesis.

In a general view from all the scenarios present in the thesis, the quality was mostly regarded as good. The exceptions are test scenario 2.2 and 4.1, which both intentionally had quality reducing measures induced.

There is a discussion to be made about what is "good" quality. Teams has (seen in section 2.1.12.5) put forward a set of parameters to inspect for classifying a call as good or bad.

Teams - power BI data
As an example for Teams, if the packet-loss exceeds 1% a call is classified as bad. However, does Teams measure packet-loss from the average from the whole call, or short intervals? In the case that a Teams call is based on the average from the whole call a meeting can easily be perceived as bad from the individual watching the web camera feed. This can happen because if a meeting lasts 30 minutes, and only the first minute has 3% packet-loss, the meeting average will be below 1%.

With intervals however this can to a certain extent be avoided. By taking the average packet-loss from each 5 seconds and investigate each value if it goes above the recommended value of 1% it is easier to determine the real quality. In this thesis the average was used to determine call quality, as Teams does not provide packet-loss in intervals.

6.3 Improvements

In this section a set of general improvements are discussed, while more specific improvements for test scenarios are discussed alone.

6.3.1 General improvements

In the event that the tests would be recreated an area of improvement is the clients. All clients should have equal hardware as to make comparing more reliable and be able to tell differences from the actual meetings more easily. In the case of this paper the test scenarios had two different web cameras which influences the bandwidth utilization.

6.3.1.1 Averaging data

Another improvement would be to focus longer on the scenarios. With additional focus on one scenario a deeper analysis of the data could be performed. A test would then be run several times as to give consistent data-points which then again could be averaged. With the use of for example average bandwidth consumption shown in a graph an error from a single meeting would flatten out.

6.3.1.2 The human component

Through out this thesis there has been no mass testing using groups of people rating the meetings. Without mass testing it is really not known if the quality is "good" or "bad", as it is subjective to each individual. The documents indicating meeting quality for Teams and Webex helps to tell the quality. But only the end user can actually tell if the quality is sufficient or not. An improvement would therefore to perform mass-testing of the scenarios.

6.4 Closing Summary

Through analysing data, implications regarding quality in the two platforms Webex and Teams showed that a client is highly dependent on the used network environment. By introducing or removing different network elements for a meeting, quality differed in every scenario. By blocking a connection, lost packets and time for reconnecting caused declining meeting quality. The same was true for a poor connection, but did not have the complication of long waiting time for reconnecting to the service.

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Appendix A

Additional Material

A.1 Data collected

All the data collected during this thesis is found in google drive[1]. The only data missing is the data from power BI and Webex Control Hub.

[1]See `https://drive.google.com/drive/folders/1Ve4JAI11mRAZ8ZtSvvY-DCnWt7Ht7jI6?usp=sharing`

A.2 General appendix

NTNU

Kunnskap for en bedre verden

Project Plan; Quality and Networking Flow in Videoconferencing Infrastructure

Kristoffer Fagerbekk,
Andreas Kilde Lien,
Andreas Rømo,
Håkon Erstad

Contents

Figures

Acronyms

CQD Call Quality Dashboard. 5

IID Iterative and incrementally development. 10

MS Microsoft. 4, 5, 12, 13

NTNU Norges teknisk-naturvitenskapelige universitet. 14

Glossary

4G Fourth generation of broadband cellular network technology. 3

bandwidth Measures how much data can be sent over a specific connection in a given amount of time. 4, 5

barneparken A Microsoft teams tenant. 5

Cisco Control Hub An admin page for a private Webex tenant which provides network information such as a meetings download and upload, ping, jitter and packet loss. 5

end-to-end delay Time taken for a packet to be transmitted across a network from source to destination. 1

hoyskolestudent A Webex tenant. 5

jitter Intermittent delays during data transfers. 2, 4, 5

latency A measuring of the time it takes for data to get to its destination in the network. 2, 4, 5

packet loss One or more packets of data travelling across a computer network fail to reach their destination. 2, 4, 5

playbook A document that describes the most interesting elements from the bachelor thesis in order to allow re-creating scenarios. 2, 3

skytjenester A Microsoft teams tenant. 5

tenant An administrative domain that a costumer can control from the cloud. In a sense this is your "virtual data center in the cloud". v, 3–5

Wi-Fi A family of wireless network protocols. 3

Chapter 1

Goals and boundaries

1.1 Introduction

Videoconferencing is an important global application that enables people around the globe to interact when distance separates them. In the year 2020 the use of videotelephony software experienced an exponential growth of users due to the COVID-19 pandemic [1]. The extensive meeting schedules for businesses and private persons forced individuals to utilise videoconferencing for communication to limit physical contact and further spreading of the virus.

With the rise of users on different videotelephony software, which acts as a replacement for physical meetings, a huge workload is created for the videotelephony services and network usage [2][1]. This shows the importance of having a reliable network connection and a decent network speed to compete with a physical meeting. There has already been conducted research on videoconferencing software with focus on end-to-end delay like in Baldi and Ofek [3], where specific thresholds have been found for an acceptable meeting quality.

1.2 Background

This project is a part of the bachelor thesis with a task given by the Norwegian telecommunications company Telenor. Telenor is one of the world's largest mobile telecommunications companies with operations worldwide [4]. Telenor offers small to mid-size businesses a Cisco Meraki[1] network solution with everything from routers, firewalls, and access points to security cameras [5]. Alongside the different equipment, Telenor sells network plans that are optimized for videoconferencing, such as Microsoft Teams[2] and Cisco Webex[3]. Telenor also provides consultancy regarding network troubleshooting.

[1] see `https://meraki.cisco.com/` for Cisco Meraki's homepage.
[2] see `https://www.microsoft.com/en/microsoft-teams/` for Microsoft Teams homepage.
[3] see `https://www.webex.com/` for Cisco Webex's homepage.

For videoconferencing communication, Telenor has a collaboration with Microsoft and Cisco. This is to improve quality and performance for both videoconferencing software. The collaboration utilizes Peering. Peering is an agreement between organizations that creates a direct path between two networks for reducing latency and improving users experience [6]. As Telenor provides these different networking solutions towards Microsoft and Cisco, Telenor would like to run a set of different test scenarios. These test scenarios revolves around video, voice and screen sharing quality in videoconferencing.

Telenor also want a demo/playbook that includes interesting discoveries from the bachelor thesis to replicate some of the test scenarios. The playbook will be used by consultants working at Telenor for expanding knowledge regarding troubleshooting videoconferencing software and Meraki network equipment.

1.3 Task description

The first part of the task involves looking for relevant literature. This includes previous bachelors, research papers doing network testing or running network test scenarios. In parallel test scenarios will be created.

The test scenarios is supposed to find irregularities and/or new discoveries when provoking packet loss, latency and jitter. Furthermore, the thesis will investigate how these provocations affects an ongoing meeting. The data collected during the investigation will be compared to patterns and findings in existing resource papers.

After the test scenarios are completed there will be made a playbook that should be a step-by-step guide on how to reconstruct and run the tests. The guide is for Telenor's customers to use for simulating the different solutions mentioned in 1.2 and for Telenor to see the flow of traffic inside the network.

1.4 Project Objectives

1.4.1 Effects

The long-term goal of the project is for:

- Telenor to increase traction to their videoconferencing solutions.
- Telenor to increase their knowledge about Cisco Meraki and video meeting solutions for consultants.
- Partner development towards Cisco and Microsoft
- Playbook for Telenor Expo 2021
- Testing scenarios and results should be replicate-able

1.4.2 Results

The results from completing this project should be:

- A well documented bachelor thesis that can be easily adopted by anyone
- Presentation around the thesis to:
  - Document flow of traffic in network regarding videoconferencing
  - Document Telenor's Meraki solutions
- Creating a working playbook/demo

1.5 Boundaries

After close dialogue with Telenor the boundaries has been agreed upon. Below is said boundaries.

- The project should be strictly network oriented.
- The project will only look at Cisco Webex and Microsoft Teams as virtual meeting platforms.
- The project only revolves around a set of scenarios presented by said employer.
- When running tests the same physical locations should be used if possible.
- When conducting the different test scenarios it is important to keep the localization and equipment consistent.
- Tests will be run on Wi-Fi, fiber and 4G.
- Tests will be run on two tenant for Microsoft teams and one for Cisco Webex.

Chapter 2

Scope

This project has a fixed time frame and it is important to know the limitations it puts on the thesis. One of those limitations is to determine what parts of a problem that will be scoped. If a project scopes too narrowly into a problem, there will be nothing to solve. But if the scope is too far-reaching, the thesis will never arrive at a conclusion. This is why scoping is its own chapter, as it is of such importance.

At 2.1 the problem area is explained. This section is for readers of this thesis to understand which subject field the paper will tackle. In the subsequent chapter at 2.2 the limitation of the problem area will be be explained.

2.1 Problem area

A few of the problem areas in this thesis that were relevant are hardware, security and networking. However it was decided networking should be the main focused area in this bachelors thesis.

Some challenges related to networking is latency, jitter and packet loss in a videoconferencing call. Another challenge is how attendees of a meeting are located and which access type each and every attendee employs.

2.2 Problem delimitation

The test scenarios will only focus on networking. That means network parameters like latency, jitter and bandwidth are relevant to the issue at hand, while client side parameters like old drivers, low frequency CPU, old webcam and outdated network cards will be ignored in the bachelor thesis.
All of the test scenarios will be running on two virtual platforms; Webex and MS Teams. The tests run on these platforms will use tenants inside the European

Union. MS Teams tenant includes *barneparken* which lies in Europe and *skytjenester* that lies in Oslo, Norway. The Webex tenant *hoyskolestudent* resides in Amsterdam, Netherlands.

2.3 Issue at hand

As the world keeps getting more digitized and closed down due to the ongoing pandemic, videoconferencing has become an essential tool for workplaces, schools and private persons [1]. But also videoconferencing comes with its problems. And to pin down these problems the thesis paper will try to identify challenges concerning quality in video and collaboration solutions.

2.3.1 Identifying videoconferencing challenges

A way to uncover these challenges is by inspecting how collected data from virtual meetings are represented in analytics tools such as Cisco Meraki Dashboard, Cisco Control Hub and MS Teams Call Quality Dashboard (CQD). These analytics tools uses different thresholds for measuring acceptable quality such as latency, bandwidth, jitter, and packet loss. Therefore by measuring and monitoring MS Teams and Cisco Webex the results may vary based on the different parameters used in the thresholds.

2.3.2 Different access types

Videoconferencing video and voice quality will differ depending on the end users access type to the respective videoconferencing platform the user is connected to. Telenor wants to find out how their different solutions of Meraki setups perform in combination with their own network, Uninett's network and student home networks.

Chapter 3

Project organization

3.1 Group roles and rules

Roles in the group:
The team will consist of four members. Going alphabetically through the members, Andreas Kilde Lien will be the referent. A referent is supposed to take notes in key meetings for possible later review. Lien will also specialise in Meraki Insight.

Andreas Rømo will be the project leader and Scrum Master. As project leader Rømo will have the privilege to decide on matters that can not be resolved in a short enough time frame. As Rømo also is Scrum Master it is expected he will give the team positive encouragement and provide guidance throughout the whole project span.

Next is Håkon Holm Erstad, the teams deputy referent and Cisco Webex responsible. This means Erstad must be ready to take over Lien's role as referent in case Lien is unavailable. Erstad is going to specialise in Cisco Webex.

The last member is Kristoffer Fagerbekk, the deputy project leader and communications manager. The deputy project leader will be on standby to take over the role as project leader if Rømo suddenly becomes unavailable. As a second role Fagerbekk will also be named communications manager. This roles responsibility is to watch over all communication to the project owner. Fagerbekk will also specialize in Microsoft Teams.

Values, description and purpose of the contract.

Group-members must abide to the common rules that has been set, and work within the boundaries that has been decided.

The core values:

- Have respect for each other.
    1. Respect each other's ideas
    2. Don't interrupt each other
    3. Everyone's opinion should be heard
    4. Be honest with each other
    5. Help each other to understand all concepts
- Be open to compromise
    1. Be willing to cooperate with others on their ideas
    2. Keep an open mind
    3. Vote on disagreements
        - Equal votes would give project leader an extra vote.
- Effective communication
    1. Make sure everyone is able to be vocal about their ideas and problems
    2. Give ideas no matter how "off" you may think they are
    3. Listen effectively
    4. Don't be rude
- Time management
    1. Attend and arrive on time to all group meetings
    2. Be flexible about meeting times
    3. Keep on task (limit talk about non-related events)
    4. Minimum 25 hour's of weekly work

Meeting structure

1. Assign meeting referent
2. Work-through of previous assignments
3. Discussion and planning around new assignments and goals
4. Distribution of new tasks

Consequences
Routines in the case of serious violations of the rules:

- The problem(s) are addressed before/after a group meeting by the members.
- If the problem isn't solved, a written notice will be sent to the concerning member from the rest of the group (if necessary only the project leader) where:

    - the violation is addressed.
    - specific (deadlines, expected amount of work(effort), etc.) about what can be done to resolve the issue.
    - specifically about the consequences (dialog w/supervisor/advisor, exclusion from the group and such) of not being able to meet the required work.
    - If the issue still persists the project leader will notify the project supervisor Ernst Gunnar Gran.
- If nothing gets resolved the group member will be forcefully kicked from the group.

Absence/Illness

- A member must give notice to another group member that he/she is sick and cannot attend a group meeting, in advance if possible.

Andreas Rømo	Date
Kristoffer Fagerbekk	Date
Andreas Kilde Lien	Date
Håkon Holm Erstad	Date

Chapter 4

Planning, follow up and reporting

4.1 Main division of the project

4.1.1 Development model

This project is time-restricted and the team consists of a small group. It's therefore important to utilize every resource that is available, and to not get entangled into complex development models that require a lot of coordination and administration.

A big part of this project is that it won't be only about developing. This is also a literature study, which no single development model could work with directly, and this is the main reason to why this project needs the development model to be customized. In the following sections there is discussion about the different practices from the plan-driven and agile models that are chosen, and in the bottom subsection there is a description of the final model.

Incremental delivery

Incremental delivery is a plan-driven development method [7, p. 64]. The use of increments is used to create parts of the project first, and allow changes to the published modules. The project has a set deadline and it's therefore important to develop the most important part first. Determining basic system definitions that are used by different parts of the system is hard to plan in an incremental delivery. Therefore "daily review" from the agile method will be important, especially because the bachelor thesis team doesn't have much experience with bigger projects like this.

Scrum

Scrum is an agile development method [7, p. 73]. Most of the techniques from scrum would probably lead to a lot of overhead, with minimal benefit, given that the team is small. This project will use its agile features like user stories, frequent releases, customer feedback, daily meetings, and continuous integration. This is included because the project has a strict deadline that shows the importance of predictability. Agile also requires minimal planning at the beginning making it easy to get sidetracked delivering new, unexpected functionality. With the incremental delivery it is possible to always extend existing modules.

Usage of model

This bachelor thesis project uses a well known combination of developing iterative and incrementally (IID) model, taking the best parts of plan-driven and agile development method [8]. This method is to develop a system through repeated cycles (iterative) and in smaller portions at a time (incremental), allowing group members to take advantage of what was learned during development of earlier parts or versions of the system. It will make the team able to maintain the overall plan within cycles, so that it is possible to work individually.

In figure 4.1, the backlog contains a set of items. A given number of items is inserted into an iteration. While an iteration is ongoing there will be daily reviews about how the work is going and discussions around the items in the iteration. When an iteration is done those items will be released and delivered. At the end of a sprint there will be a feedback meeting discussing the items delivered with Scrum Master. For this bachelor project an item could be to write a section in the report or conduct a specific test scenario.

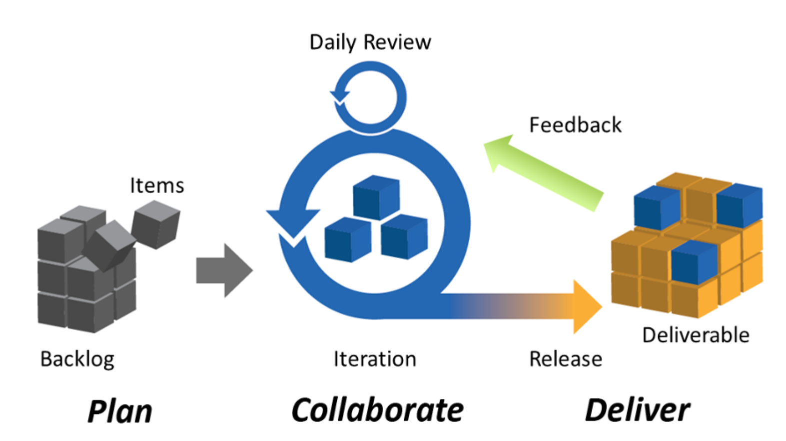


Figure 4.1: The figure shows an agile approach that is chosen for this bachelor thesis. License: CC BY-SA 3.0[9].

4.2 Plan for status meetings and decision points in the period

Status meetings
As requested by Telenor, an update/feedback meeting will happen on a weekly basis on each Thursday 09:00 AM. The group has also decided to keep weekly group-meetings Tuesdays at 0915 AM. Tuesday meetings is an iteration meeting to get reviews on our work, deliver items and put new items into the new iteration. This fits with the chosen development model and meets the employers wish to keep in close contact during the project.

Decision points

In projects it is important to have certain dates for different decision based progression and deadlines. Working with no key-decisions it would quickly cause issues, as there have been no thoughts put into time managing or how to escalate the project. Below is the most important deadlines and decision points (see Figure 4.1) for this thesis and how to approach this thesis.

Milestones	Date
Preliminary project	**01.02.2021**
Testing Equipment	**03.02.2021**
Testing Scenarios	**10.02.2021**
Approach for bottlenecking	**15.02.2021**
Thesis, first draw	**29.03.2021**
Playbook	**01.04.2021**
Bachelor Submission	**20.05.2021**

Table 4.1: Different dates for the project

Chapter 5

Organization of quality assurance

5.1 Documentation and standards

The documentation will be written in English. All the documentation will be done through Overleaf[1] as a LaTeX platform. Some information will be kept on private messaging channels until it can be included in the LaTeX document. Each meeting will be documented by a referent, the report will be stored in a private Google Drive[2]. The `ntnuthesis` document class is a customised version of the standard LaTeX `report` document class and will be the document layout that will be used for all the LaTeX documentation.

The Work-platforms that will be used:

- Trello [3]
- Meraki Dashboard [4]
- Cisco Webex Control Hub [5]
- Microsoft Teams Call Quality Dashboard [6]
- Webex
- MS Teams [7]

The Communication platforms that will be used:

[1]see `https://en.wikipedia.org/wiki/Overleaf` for Info on Overleaf

[2]see `https://www.google.com/drive/` for info on Google Drive

[3]see `https://blog.trello.com/beginner-tips-for-using-trello#what-is-trello?` for What is Trello.

[4]see `https://documentation.meraki.com/Getting_Started#Introduction_to_the_Meraki_Cloud_and_Dashboard` for Meraki Dashboard introduction.

[5]see `https://www.cisco.com/c/en/us/products/conferencing/webex-control-hub/index.html` for Quick summary of Webex Control Hub.

[6]see `https://docs.microsoft.com/en-us/microsoftteams/cqd-what-is-call-quality-dashboard` for what is CQD.

[7]see `https://support.microsoft.com/en-us/office/welcome-to-microsoft-teams-b98d533f-118e-4bae-bf44-3df2470` for Teams functonalities

- Webex
- MS Teams
- Mail
- Discord

5.2 Configuration management

To achieve the most reliable outputs to the actual test scenarios, configuration management is important. During the thesis the setup will be kept close to the original setup. This removes the differences added by the changing of location, hardware and ISP. This creates a core environment that is optimal for our testing scenarios.

5.3 Risk analysis

With every project comes different risks. Listed below are the most important risks associated with this project regarding equipment, human assets and the critical time period. See figure 5.1.

5.3.1 Risk factors

Risk 1: NTNU facilities close due to the pandemic
Risk 2: Student(s) leave bachelor group due to internal or external reasons
Risk 3: Student(s) becomes sick
Risk 4: Meraki test kit doesn't arrive in Gjøvik
Risk 5: Testing tenants become unavailable
Risk 6: Losing access to tools
Risk 7: Inconsistent testing environment
Risk 8: Receive faulty equipment

5.3.2 Risk reduction

Risk 1: Enforce NTNU & Government rules and be a role model for other students
Risk 2: Focus on good work environment and communication
Risk 3: Workload is divided equally between the remaining student(s)
Risk 4:

1. Delivery of Meraki test kit happens in person
2. Worst case, simulate Meraki test kit with equipment from NTNU if possible.

Risk 5: Communication with Telenor and platform providers
Risk 6: Communication with Telenor
Risk 7: Acquiring a dedicated room for the testing equipment
Risk 8: Return faulty equipment and wait for new equipment

	Not Significant	Minor	Moderate	Major	Severe
Almost Certain					
Likely					
Possible		R7	R3	R1	
Unlikely			R2		R4, R5
Rare		R8	R6		

Figure 5.1: Risk Matrix

Chapter 6

Plan for implementation

To follow through a bachelors program and write a good bachelor it is important to have a solid plan. This is because without a plan you plan to fail[10]. Planning is important to ensure the thesis keeps inside the time-frame and that it is possible to achieve the implementation that are set as a goal for the thesis. The gantt diagram shown below illustrates the time-frame and the planned progression of this thesis.

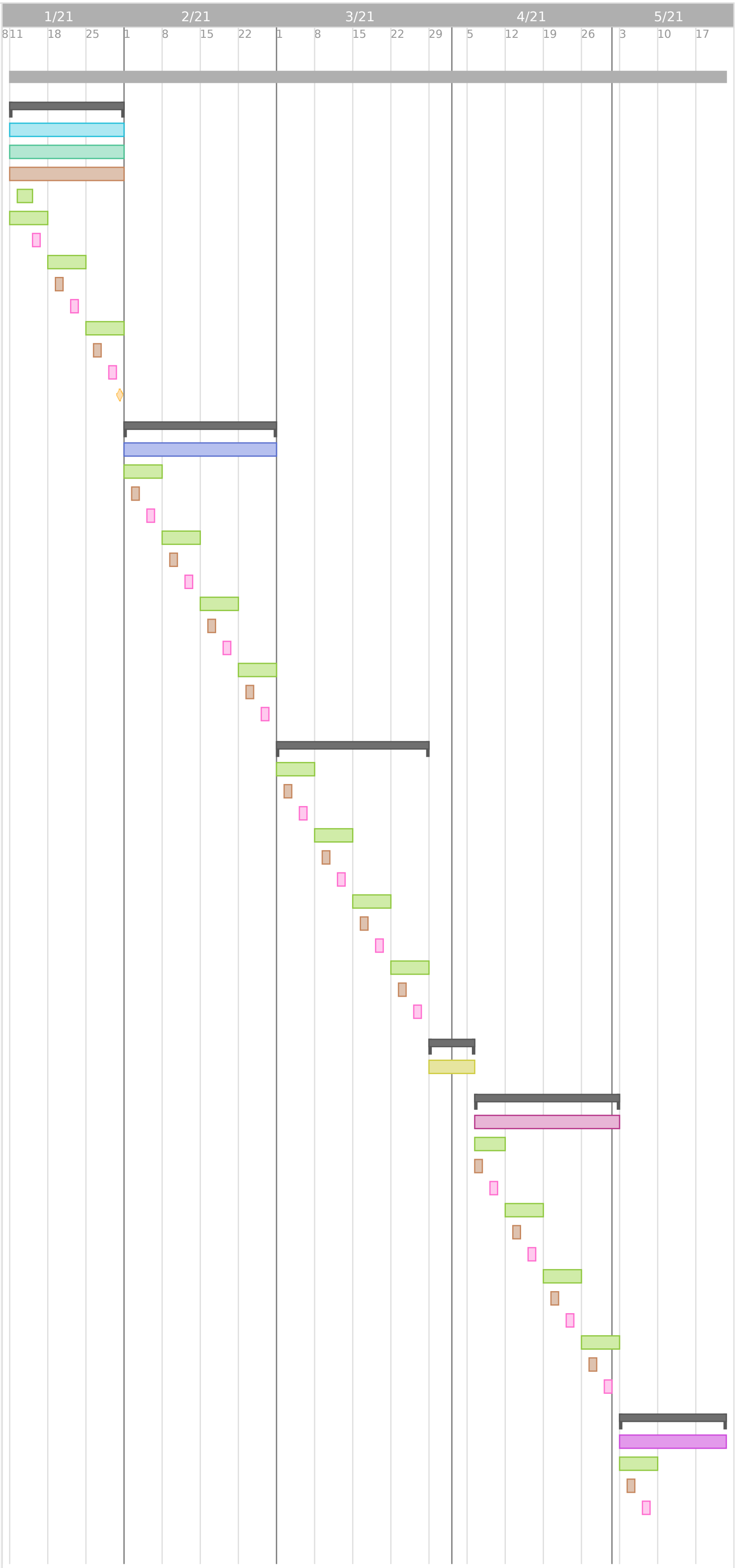
6.1 Gantt scheme

This gantt scheme is fairly detailed and some of the sections or lines might be unclear. This section is to clarify the phases in the gantt diagram.

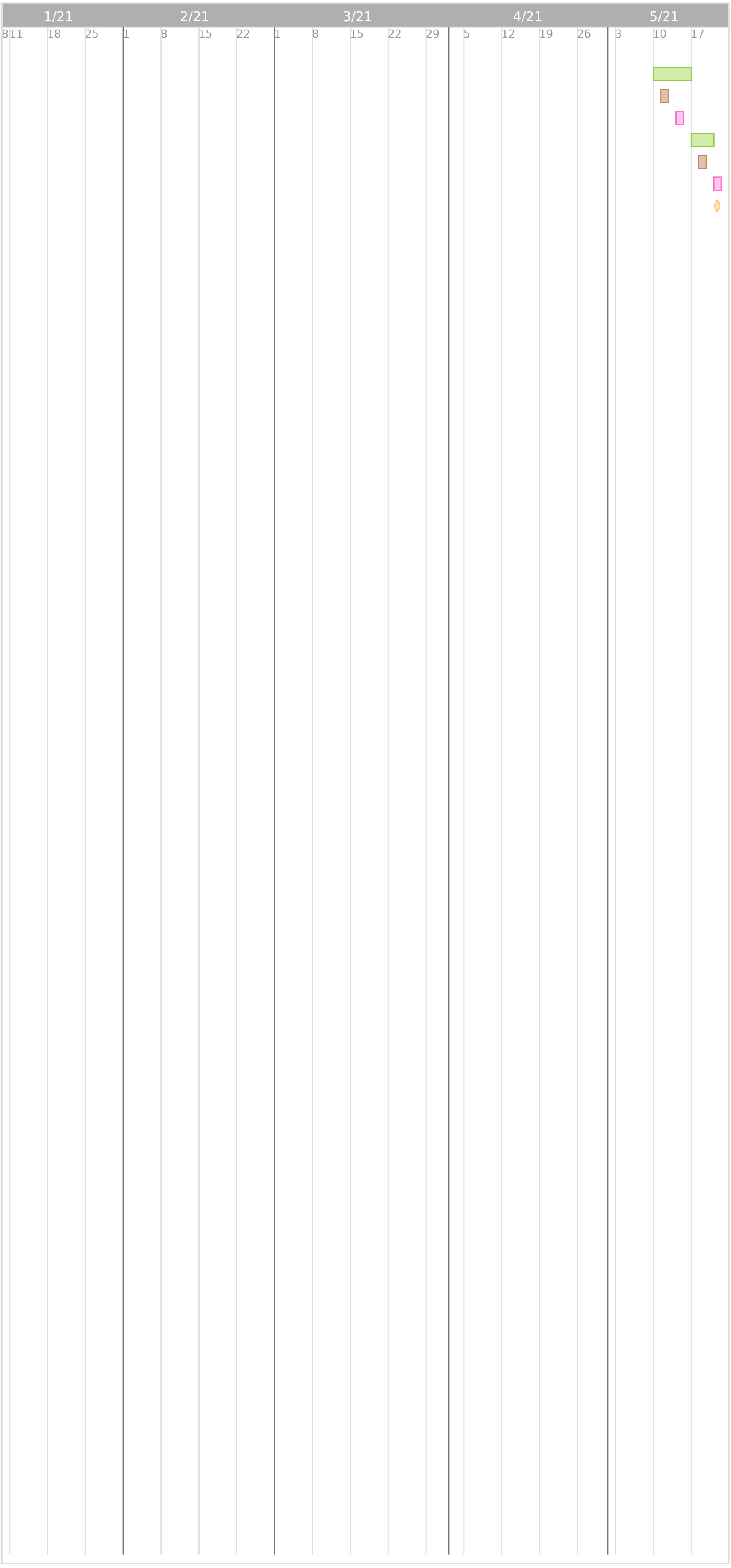
This Bachelor project is split up into five phases. Starting off with the project plan/learning phase that lasts a month like the other phases. The learning phase consists of a kick-off that is planned for a two day session. In this session the bachelor students are going to further expand their understanding of the given problem, tools needed and knowledge required to solve the task. The learning phase will be used to gather knowledge relevant to the test scenarios. This needs to be completed, so testing can begin in February.
Next is the testing phase that mainly revolves around work with testing scenarios and the Bachelor report. In March writing will be the major task. Working with finishing the "method" part of the report and to dig into the data collected from testing phase. The demo/playbook in April is planned to be used to create a demo/playbook that goes deeper into the most interesting findings from the test scenarios that we went through in the testing phase. At the end there is a buffer section. This section is like the name implies; a buffer for the bachelor group to use if there is more time needed on the project report.

teamgantt
Created with Free Edition

Bachelor 2021	**0h**	**0%**
Project plan/Learning phase	**0h**	**0%**
Project plan	0	0%
Learning phase	0	0%
Read other bachelors	0	0%
Kick-off	0	0%
One week sprint	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
Project plan delivery	0	0%
Testing	**0h**	**0%**
Testing of scenarios	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
Writing	**0h**	**0%**
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
Easter	**0h**	**0%**
Vacation	0	0%
Demo/Playbook	**0h**	**0%**
Testing of scenarios	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
Buffer	**0h**	**0%**
Buffer	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%



One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
One week sprint	0	0%
Meeting	0	0%
Meeting/Feedback - supervisor/Empl...	0	0%
Bachelor delivery	0	0%



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Playbook

Kristoffer Fagerbekk,
Andreas Kilde Lien,
Andreas Rømo,
Håkon Holm Erstad

Contents

Chapter 1

Introduction

This playbook will contain a guide on how to setup new Meraki devices, some key features related to Meraki cloud that the equipment is connected to and a specific testing scenario taken from the bachelor "Quality and Traffic Flow in Videoconferencing Infrastructures". It will also include how the group experienced the equipment. An example of a basic test scenario can be seen in figure 1.1 below.

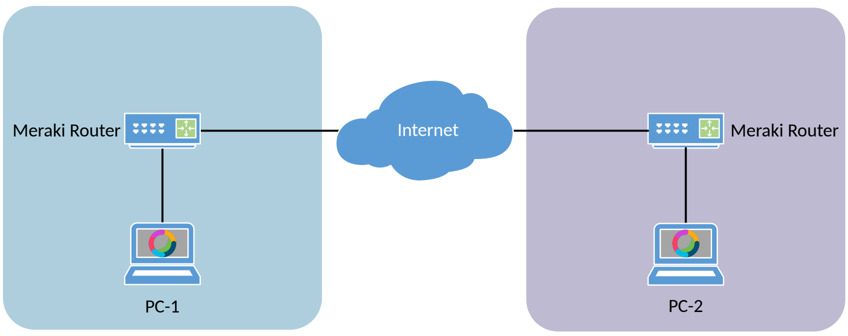


Figure 1.1: Basic test scenario using Microsoft Teams

Chapter 2

Getting started

This chapter will cover how the group experienced setting up the equipment and a step by step guide on how to set it up.

2.1 The first impression

In the beginning of March we received all the equipment needed for conducting the test scenarios Telenor had given to us. We started by setting up one router (MX68). The MX68 we received first were already registered on a network in the cloud, managed by Telenor. The MX68 only used a couple of minutes to connect to the cloud. As the device was in a existing network with other equipment we were not going to use, we requested to create our separate networks within the organization. We then created our own two networks, one for each MX68, as the two clients running test scenarios where on two different locations. This was purely to separate the information that could be observed for each client within Meraki Dashboard. All the other equipment that was not already registered, was added to the inventory in the cloud and moved to the respective network. Below is a more detailed guide, step by step.

2.2 Setting up equipment

To connect the device(s) to the cloud, the serial number of the devices are needed, found in the order confirmation email or on a sticker on the device. After this connect the device to a functional WAN and login to the Meraki Dahboard [1].

Once logged in to the dashboard, create your network in the cloud. This is done under ”Network”, upper left side. Fill in the information accordingly.

Next is adding the device into the cloud. This can done from two different places;
1: In the same place the "Network" information was filled in -> Add Device.
2: Left side ribbon -> Organization -> Inventory -> Add Device.

Once the device has been added, assign the added device to the ”Network” created earlier. After a couple minutes the device LED light should go from rainbow color to static white or green depending on the device [2].

To change the ”Network” for devices that are already assigned to a ”Network” or is already in the ”Inventory”, go to Organization -> Inventory -> Actions -> Change network Assignment.

[1] Access Meraki Dashboard `https://account.meraki.com/login/dashboard_login?go=%2F`

[2] See `https://documentation.meraki.com/Go/Meraki_Go_-_Decoding_the_LED_Light` for documentation regarding the color codes

Chapter 3

Control Hub

Cisco Webex Control Hub[1] is a management and analytics tool. It is used to monitor and manage different Cisco devices and workplaces created from the Control hub to simulate physical locations. The control hub has been used for collecting data from Webex test scenarios. The playbook will only cover what the thesis focused on during the test scenarios.

Two main areas of the Control Hub will be looked further into. ”Analytics” and ”Troubleshooting” tab, that is located on the left side under ”Monitoring” segment. The Analytics tab has an overall view of all the meetings held within the organization. Information that resides here are spread over four main categories: *Engagement*, *Participants*, *Quality* and *Audio*.

The Engagement tab shows total meetings over a set duration. It also contains information about top participants and meetings spaces. Below is a picture 3.1 of the meetings activity held during our thesis. Januar to May for *hoyskolestudent*, as this was the tenant used during the bachelor.

[1] see `https://www.cisco.com/c/en/us/products/conferencing/webex-control-hub/index.html#~for-partners` for more info on Webex Control Hub

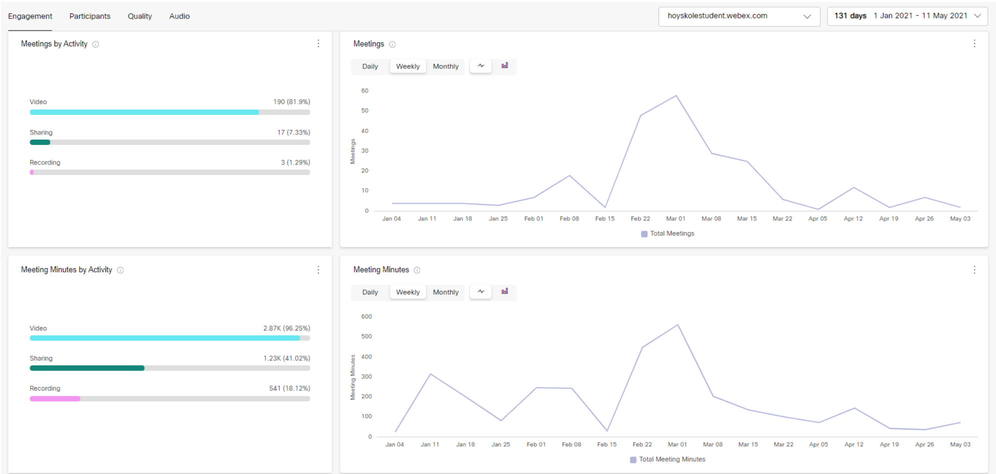


Figure 3.1: Picture taken from the "Engagement" tab in the Control Hub. License: Cisco Meraki, CC BY

"Participants" tab has information about how participants joined, roles, location and minutes. It also includes number of meetings registered to a specific users, and so on. Below is a picture showing "Participants by Join Method" and "Participants by Roles", see figure 3.2.

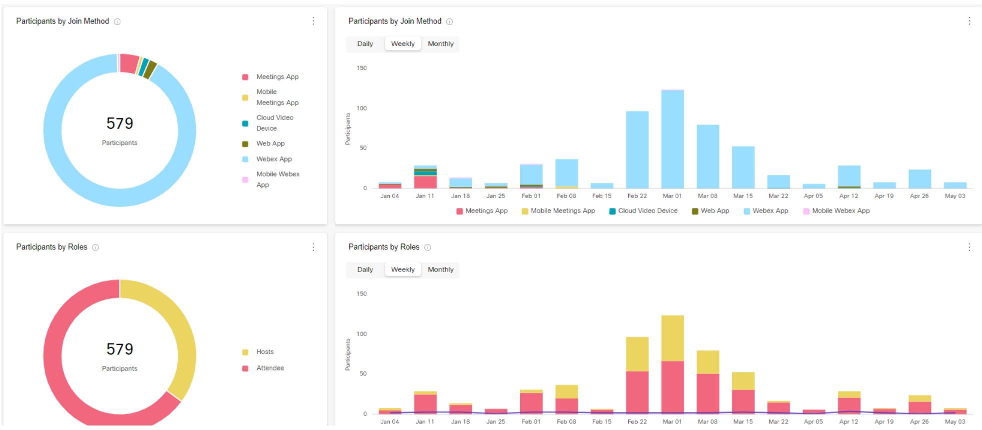


Figure 3.2: Picture taken from the "Participant" tab in the Control Hub. License: Cisco Meraki, CC BY

The "Quality" tab shows an overview of VoIP quality, quality difference among different OS (Operating System) among meetings and percentages for X minutes and Participants Webex classifies as "Good Media Quality". A picture of VoIP Quality can be seen below in figure 3.3.

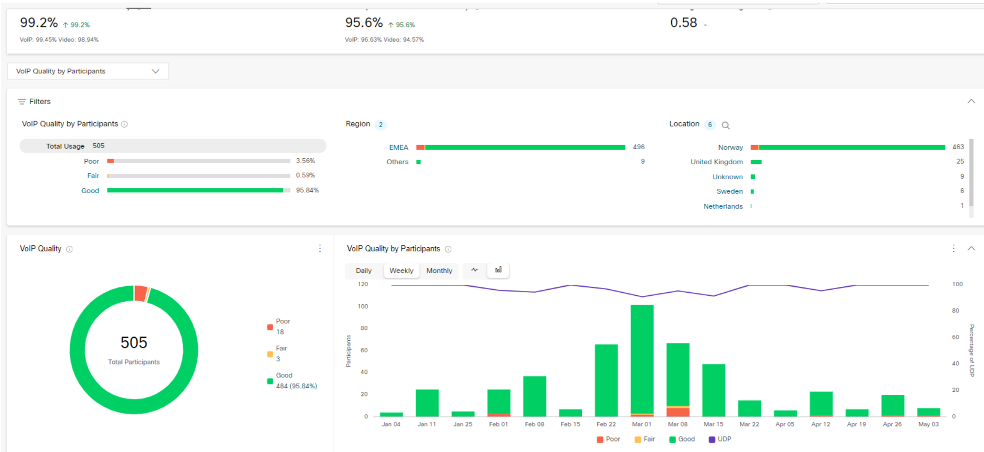


Figure 3.3: Picture taken from the "Quality" tab in the Control Hub. License: Cisco Meraki, CC BY

Audio tab shows the total audio, VoIP and Telephony usage over the time period selected for all meetings. Picture 3.4, below shows all the information residing here.

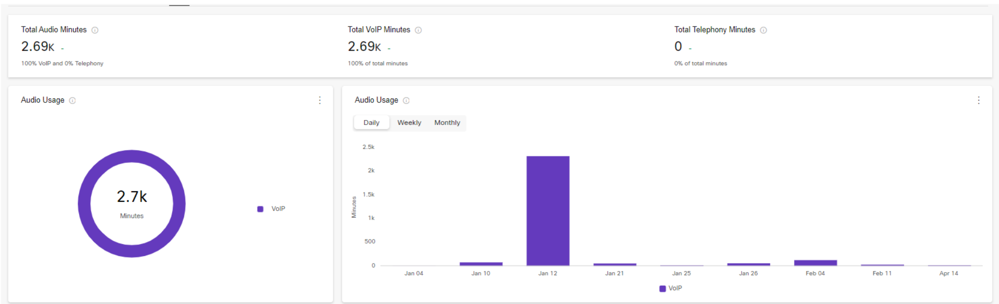


Figure 3.4: Picture taken from the "Audio" tab in the Control Hub. License: Cisco Meraki, CC BY

Next up is the ”Troubleshooting” tab. Here you can see more detailed information regarding specific meetings. This is done by providing either email of a participant, device name, meeting number or conference id. As seen in picture 3.5, we have found all the meetings the participant with the email "Email@of.participant" has either joined or hosted.

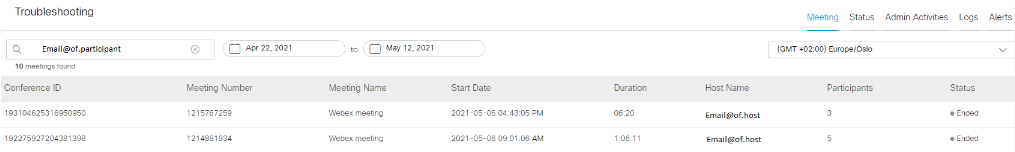


Figure 3.5: Picture taken from the ”Troubleshooting” tab in the Control Hub. License: Cisco Meraki, CC BY

After the meeting that is of interest has been identified, click on the meeting and a timeline from the meeting with regards of Audio quality will appear, see picture 3.6.

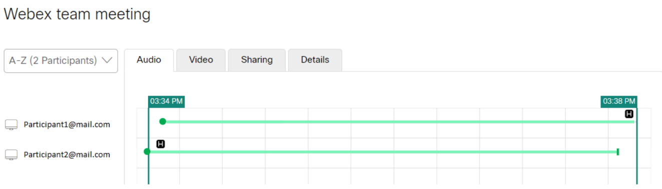


Figure 3.6: Timeline for a specific meeting, taken from ”Troubleshooting” tab. License: Cisco Meraki, CC BY

As seen in picture 3.6, there are three more ”tabs” of information that can be viewed. The ”Video” and ”Sharing” tab will also display a timeline, but instead of information regarding audio, it contains video and sharing information. The green timeline shown above shows that Webex classifies this meeting "good". Below is figure 3.7 showing all the different icons that can occur in the timeline.

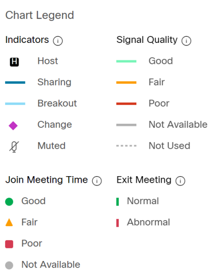


Figure 3.7: Icons used for the timelines. License: Cisco Meraki, CC BY

These timelines also provides more detailed information about the quality of video or audio, by hovering the mouse over the timeline. In the picture below 3.8, the first participant got increased delay and Webex classifies this as "Poor" signal quality 3.7.

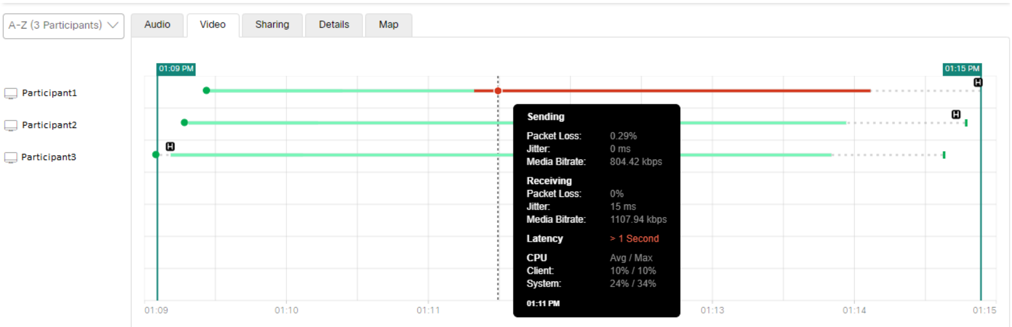


Figure 3.8: Timeline for a specific meeting with three participants, one participant had latency increased to more than 1000ms. License: Cisco Meraki, CC BY

While hovering a participants timeline, one can also see more information that could explain the causes of why Webex marked the quality the way it did. As shown in picture 3.8, the user experienced a sudden increase in latency.

In the top right corner, users can also choose to download the meeting information in either JSON or CSV format to display information relevant in other formats than Webex already does.

Chapter 4

Testing Scenario

In this chapter we will go through a specific test scenario taken from the bachelor ”Quality and Traffic Flow in Videoconferencing Infrastructures”. It will show the topology and settings configured in Meraki Dashboard.

4.1 Scenario 2.2

Test scenario 2.2 was conducted to see how low signal strength affected quality in videoconferencing. A few examples of where this scenario can take place is in a workplace were the coverage of access points are not optimal or when people are forced to work from home and does not have access points to cover the whole apartment etc.

As seen below in figure 4.1, there are two clients involved were PC-1 will be using an Ethernet from the beginning but swap to WiFi that is barely in range, in the middle of the meeting. PC-2 will have Ethernet connection throughout the whole meeting.

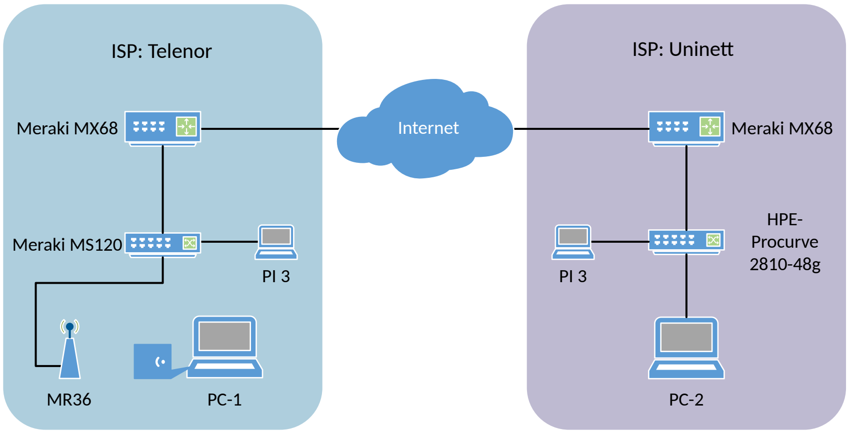


Figure 4.1: Test

How to configure: As we were running the test scenario from a small apartment, we had to artificially create low signal strength by adjusting the "Target Power(dBm)". In our environment we also had to place the access point (MR36) as far away as possible and wrapped it two times in aluminum foil to get the absolutely lowest signal strength achievable without losing connection entirely. Most of these steps can however be skipped if the building is big enough to just place it far enough away from the personal computer.

Adjusting the "Target Power" can be done as followed:
Choose the "Network" where the access point was added to. Next click on Wireless -> Radio Settings. Make sure the correct band, 2.4GHz or 5GHz is selected, our test scenario used 5GHz as it decreases the range of the signal compared to 2.4GHz. Click on the Box to the left of Status, then Edit Settings. Picture 4.2 below shows the process.

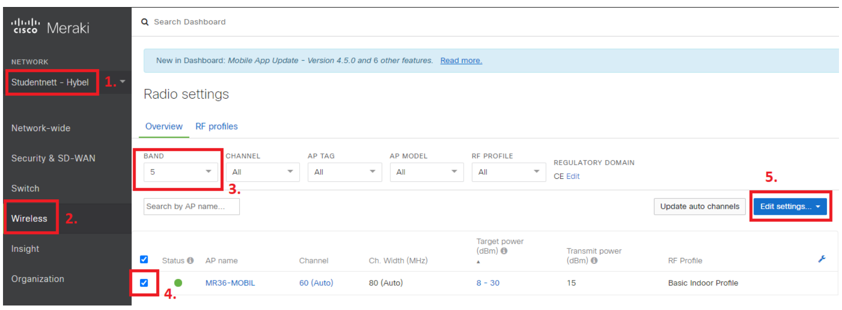


Figure 4.2: Procedure to change Target power: Cisco Meraki, CC BY

Next choose ”Manually override power settings and choose the desired Transmit Power as shown in pictures 4.3 below.

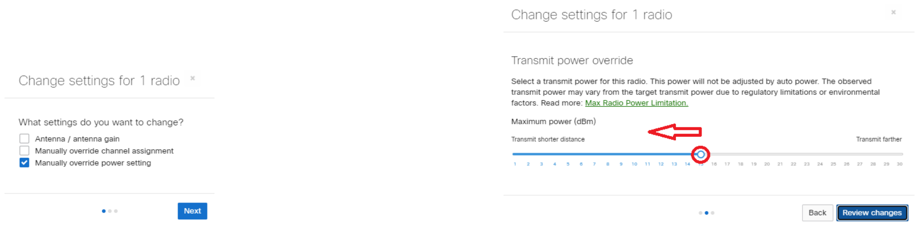


Figure 4.3: Procedure to change Target power: Cisco Meraki, CC BY

Once the desired transmit power is found and applied the end results should look the picture 4.4 below. Transmit power will auto adjust itself accordingly if the target power is achievable.

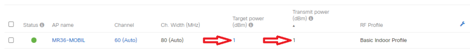


Figure 4.4: Low target power -> Low Transmit power. License: Cisco Meraki, CC BY

A.3 Meeting Notes

Group meeting 19. January 2021 - Referat

We are discussing the following subjects in the agenda:

- Project plan
- Agreement with Erns for WebEx meeting
- Documents for referats
- Choose roles in the group
- Group rules

We have chosen to use incremental sequential development model for this project work.

The following roles have been set:

Project leader: Andreas Røme
Deputy project leader: Kristoffer Fagerbekk
Referent: Andreas Kilde Lien
Deputy referent: Håkon Holm Erstad

Timetables will be logged on Clockify (testing for a week), it has to be done by each member individually. Might change to Microsoft Excel if the group is not happy with Clockify.

Supervisor, Ernst, is added as a member to «hoyskolestudent» tenant by Andreas Røme.

The group rules are chosen (and signed), and attached to the report.

The work on the project plan has started, and we have divided work between us.

Meeting with supervisor and contact person from Telenor – 21. January 2021
Henning, Bjørn, Ernst and Bachelor group meeting on WebEx.

What should be included in the project plan?
Ernst answers this question.

He mentions risk assessment. Like someone is leaving the group. Ernst also points out that the group has to make their own "solutions".

Henning (Telenor) talks about the plan he has thought of.
January – Get to know the equipment
February – Testing
March – Writing
April – Demo/playbook (from the most interesting testing scenario)

Henning wants to have a playbook for showing what has been made of the project

Ernst mentions the importance of always writing things down. Like from what you are reading, etc. The writing should be active all the time.

Should be a “første utkast” at the beginning of the easter of the project report. Henning agrees with this and will include that in his plan.

Henning talks about simulation of the tests and gather documentation/data from the systems. He also mentions the importance of “avvik” and …

Ernst suggest including the playbook in the presentation that happens after the report is done.

Henning shows the “test scenarios” on WebEx:
Test scenario 1 – Baseline (Meraki to Meraki over the internet)
Barrowing a MX of Meraki (Cisco)
Block UDP for the conference platform, forcing it to use TCP.
Test scenario 2 – Baseline + WiFi
Test scenario 3 – Home office fiber
Test scenario 4 – Home office fiber (VPN)
Test scenario 5 – 4G net. Telenor mobile network

Henning talks about how Telenor knows the best test scenarios for this task.

Henning talks about the importance of using a Telenor ISP, because of peering like to Microsoft’s Teams. To test if that has a big impact.

Document in the report that this has importance, with data would be nice.

Henning mentions “projectavtale”, he wants it to be filled, so they can sign it digitally.

Ernst mentions importance of scenarios and what does the net usage has of an importance of quality. Find a conclusion that gathers a conclusion and some new information.
Should open the report if there is extra time. Better than doing more then you have time for. Say Ernst

“The team should be working individually, but at the same time work together/collaborate” said Ernst.

Henning mentions a “teknologiansvarlig” in the report and to rotate to some of the technology area.

Henning talks about rating meetings and how it should be included. For a visual experience. Also known as “erfaringsblogg”. Ernst likes this and think it should be added as a “krydder”.
Teams < WebEx

Focus on the network part and not too much into devices.

Ernst leaves the meeting.

Questions:
Can we write in English? Yes

Teams vs. WebEx comparison? Some customers get WebEx, others get Teams. They have
their strength and weakness. The team should get experience from both and see the difference. Also look into Meraki with the platforms. Bjørn mention how both get affected by the network speed, etc. instead of directly Teams vs. WebEx.

Rating meeting that is not going through Meraki. The more metadata the more information for us. Bjørn mention an own chapter for "opplevd kvalitet" and compare it to the real data. Make some changes on the devices like wifi, vpn, etc. for some of the participants.
Bjørn has been on WiFi this meeting

How long should the sprints be? We could have a meeting every week. Open for suggestions.

Miro – tool for post-it notes

Group meeting January 26, 2021

Started with work on the project plan. Everyone had done their part, so the group worked on adjusting it.

Everyone signed the group rules and work agreement.

There were also some questions sent to Henning and Ernst about the project.

Lastly agreed on the next meeting January 27, 2021 after the meeting with Ernst and another one on February 2, 2021.

Meeting with supervisor – January 28, 2021

Meeting is on MS Teams (NTNU server).

Questions for Ernst:

Q: Mustad Næringspark and Telenor's office, did not work out. Henning has made some changes to the test scenarios. How should the group work with these changes in the project plan.

A: Write the "overordnet" in the project plan. And don't write what is done in the project plan. We don't need to specifically mention what the test scenarios are about. The project plan should be open but follow Telenor's recommendations.

Q: Does the project plan need to be accepted?

A: Yes, of Ernst and he will give feedback on some of it.

Q: Can bad language affect the result?
A: Nah, if it’s hard to understand the report, that gives a bad impression of the report. Could be bad for the report.

Q: How does the project plan affect the result
A: The report is the most important. But it should be consistent and is important for the group as a tool for the further work on the project.
“problemstillingen” is the most important part of the project plan, say Ernst

Q: Should we include the signed document?
A: Check the pages for what should be included in the report.

Meeting ends 08:22

Group meeting - February 2, 2021

Meeting starts at 09:15 in A162.

Project report
- Discussed with Ernst about how the structure of the project report

The work with the setup of the project report started structuring “.tex” files with folders.

We discussed other bachelor projects that have a similar style.

Add cards to Trello for dividing work between group members.

We have prepared some questions for meeting with Telenor February 4, 2021 and meeting with Ernst the same day.

Scheduling platform for meeting with Ernst on Webex.

Meeting ends 11:25.

Supervisor meeting – February 4, 2021

Meeting starts 08:00 on Webex (hoyskolestudent tenant).

We changed the platform because of problems with Webex for a student, to MS Teams (NTNU).

Q: Own room at NTNU Gjøvik?

A: Ernst will look into this when he is at Campus. But has some options.

Feedback on Project Plan
Ernst has added comments in the document.

Q: How should we structure the report?
A: The group has thought about using IMRaD. Ernst mentioned the importance of introducing the project. Also write some about how the project has been going in the report. IMRaD should be good, but we will find out about it at a later time.

Q: Test scenarios – Should they be included in the report or added as an appendix?
A: Mentions the scenarios in the report if you were to write a page about each test.

The topic “Related Work” is tightly connected with the chapter “Background”

Q: Is the report a playbook?
A: The report is probably bigger than what the playbook should be. Playbook could be a “sammendrag” of the report’s scenarios. Add playbook as an appendix in the report. Talk to Henning about this.

Q: Should it be a speed test for each time we do one of the test scenarios?
A: Could be interesting to mention in the report. It’s not much work to take that test each time the test scenarios are running. Something the group should discuss internally.

Q: Should we include spec. of the PC, etc.?
A: How relevant is that information. If you add too much information that extends the report, it’s easy for the reader to get lost in the report. Ernst wonders why we have chosen to not look at devices. This could be important for reproducing the report and it would be hard to determine jitter, etc. without looking at the networks card.

Q: The team will be split up with “ansvar” for each platform. Where each “split up team” will make use of different computers.
A: This is not a problem. But should add a section about this in the report. If there is a speed difference it should be included.

Ernst shared some thoughts about the project report.

Ernst mentions that figures should be explained in the text before the figure appears. Also there should be a good explanation below the figure.

Ernst also says that if there are very few acronyms in use you can consider to drop that page.

Something that Ernst said is very important is to “les teksten som om du ikke vet hva teksten handler om”. Also the use of “kilder og referanser” was very poor.

Meeting ends 08:55.

Group meeting February 9, 2021

Meeting starts at 12:30.

We have sent a message to Ernst about the room situation. Also sent an e-mail to a representative at Uninett for peering questions. Received an answer with info on their peering and transit.

Next was to move Meraki equipment into its own network in Meraki Dashboard.

Started the work with creating a procedure for testing scenarios.
https://docs.google.com/document/d/1fbZFJ6J5s8HubLUBI9TeeKm4VF4e3fyVkK57uEnmgP

And fixing the chapter “Introduction” in the report.

Discussed the different solutions for keeping a consistent video and audio in the video conferences, concluded with asking Ernst what he thinks and Henning.

Made questions for the next meeting with Ernst.

Made questions for Henning as well for the next meeting.

Meeting ended at 16:40

Supervisor meeting – February 11, 2021

Q: Thoughts on the project plan:
A:

- Seems good to use as a start to the bachelor.
- Still small comments on the references. Specific a network load reference.
- Comments on the Telenor reference, use another site than one made by Telenor itself.
- Unsure of using more time on it, but if we had anything we wanted Ernst to look at specifically, he could take a look.

Ernst introduced the idea of giving the first 3 pages of the bachelor to some acquaintances with some background knowledge in computers. And then they could explain what “they” thought we were going to do. The introduction should give a good explanation on what we should do.

Q: Looked through bachelors from previous years and found a page in the beginning of multiple. Asked if this was needed.
A:

- It was usual previously.
- Ernst doesn't think there is any requirement on it now.
- Ask about this in the next “lynkurs”.

Q: About writing on peering, codec, etc.. Does Ernst have any ideas on where this should be placed?
A:
- Usual to have it in a background chapter
- Can also put earlier works on these things here as well
- A little bit early to say yet where it will fit.
- Important to give the reader the knowledge it needs, where he/she is reading.

Q: Talked with Jon. About the dedicated line. Unsure if he or we didn't understand how it would impact. Could a lot of traffic create congestion in the network?
A:
- If enough students uses the network it is possible
- Ernst was unsure on the capacity of NTNU in Gjøvik going out.
- Less people on campus in these times
- Jon knows the campus network the best.
- But if still unsure, ask Jon again.

Q: Further on the dedicated line, if we don't get it. Should we still write on it in the bachelor?
A:
- This should be looked at when writing the test scenarios.
- A lot of details is good, but it can be too much, and then the reader might lose some of the meaning.

The bachelor group had no further questions.

Ernst had a comment that we were to let him know if we wanted him to check out anything in the ntnu/admin webex channel.

Meeting ended 08:17

Group meeting - February 16, 2021

Sent message to Espen about the procedure plan.(Still waiting for answer)(Got answer)
He said it seemed good, commented that they did similar in Cisco, came with a possible test scenario.

Room for meeting next week. Room A158

Discussion about why the traceroute of teams.microsoft.com, goes to America. Contemplating if to ask Andre the network infrastructure guy could answer on this. An email will be sent.

What thresholds do we need to look at after each meeting?

Webex have some thresholds stated, unsure if it is for okay or bad connection. Haven't found anything for Teams yet.

Room change from cisco lab to a260.

Procedure for showing and classifying data? What data should we look at? How do we classify the data?
Data to look at ping, jitter, packet loss, tcp and udp, throughput bitrate. Python script to sort through JSON files(?). Use the tools given by Henning at first and utilize them before adding on more advanced and more detailed tools.

Make a procedure for what data we need to look at after each test and what data we need to store. What data do we document for each test scenario and which can wait.

Plan to do a lot of work on Thursday and Friday when we get the MX.

Andreas had to leave.

Lunch Break around 12.

Discussion about where to place in the text, why we set up the “studentnett” network in meraki, possible rehaul of the procedure section, to have one general environment setup, one baseline test setup and one subsection for how and what data to gather after the tests.

Fixing meeting with Ernst on the “barneparken” tenant on teams. Meeting scheduled thursday at 8’o clock.

Andreas had to leave for another group meeting.
Meeting ended at 14:15.

Supervisor meeting – February 18, 2021

Meeting start at 08:00 on MS Teams (barneparken)

We gave a status update to Ernst.

- Waiting for the MX.
- Meeting with MS yesterday, about Teams (was a bit narrow what we could extract from the API).
- Discuss what data to be extracted from the different systems. Ernst says the values we have chosen are the ”standard ones” and suggests talking with Henning regarding further type of data.
- We are about three weeks behind in the project, because of a missing MX. How should we discuss this in the report? Ernst mentions that the best is to finnish the project, and it should be expressed as a “problemstilling” instead of an excuse if we do not manage to do all the tasks. In a way that does not describe the “problemstilling” to the assignment.

- Should we limit the time on the playbook, to make up for the lost time? Ernst says that we should look into doing the playbook parallel with other tasks.
- Don’t need to mention that we are behind in the work, if it won’t have an affect on the delivered product. If it does affect the project it should be mentioned.

Meeting ended at 08:23

In the meeting with Telenor there was some of the same question.

Q:Discuss what data to be extracted from the different systems.
A: Telenor agreed that the most important was in the loop.

Q: Should we limit the time on the playbook, to make up for the lost time?
A: Yes, focus on the writing.

Group meeting February 23, 2021

Meeting starts at 09:15.

We are going through Trello. Discussing workload and dividing tasks between the group members.

Everyone is working on getting procedure done and starting on test scenario 1. Because the MX will arrive today and the work on conducting test scenario 1 is scheduled tomorrow.

Meeting ends at 13:12

Supervisor meeting – February 25, 2021

Meeting start at 08:00 on MS Teams (Skytjenester tenant)

Q: Have we gone too far from the main scope/project?(with the hp switch, ntopng and raspberry pi) Henning has a plan set, but we want to see a little bit deeper about the data.
A:

- Important to be able to finish the main task
- But not wrong gathering more data
- Be mindful how much time it consumes, how much work is needed.

Q: Concern since we are 3 weeks behind schedule on testing. What are Ernst’s thoughts on the priorities?
A:

- Sensor won't notice if the delay doesn't affect the main result
- Try to switch around if stuck on something instead of waiting around

Ernst: Do you see yourself still completing?

A: At the moment if we can get the tests done we think so. Some details about yesterday's work.

Ernst: Comment on when he first saw the plan for being done with testing before february was done, he wasn't optimistic about us being able to, but he also commented on it being good to have a “strict” plan to push the group to work hard.

Q: Bytte Wireshark mot Meraki Insight da dette er buggy as fuck?
Q: Change software’s used in testing if
A:

- If we need something else than what was initially planned, then it is important to ask for it as soon as possible. To get issues fixed as soon as possible.

Q: What do you think about the new methods we have implemented? If Ernst had any feedback on it.
A:

- Ernst doesn't have any specific knowledge about all the tools we use, so he can't really comment.
- Had an idea that next meeting we could show it.(Raspberry pi/NtopNG)

Q: Thoughts on the test scenarios and our changes to it.
A:

- Doesn't think it is a bad idea to change around with different parameters
- But also have a continuous dialogue about it with the task giver

Q: About if we were to show the procedure next meeting?
A:

- Yes, if we could explain it as if Ernst was a completely beginner in the area.

Ernst comments on if we look at the supervisor meetings or “normal” meetings for data. (If they were a part of testing)
A:

- We have only used it to get a feel for the equipment and tools, but will not use it for data in the bachelor.

Meeting ended at 08:22.

Group meeting - March 2, 2021

Meeting start at 08:58.

Fagerbekk and Rømo went to try a fix to the HPe switch.

While the rest of the group went through some overleaf work and trello.

Fixed the HPe switch for the raspberry pi, but got double traffic on a wireshark session on a lab computer.

Talked with Jon on Teams in a call as well as in messages to discuss the need of another switch or if the fix to the HPe switch would suffice. Jon said that he thought the tool would be able to filter out any double packeting.

Discussion about the IPv4 compatibility on meraki MX where it works contrary to the traffic from IPv6. Because of problems on the MX on Saturday.

Going through Trello. Some comments on how the Trello was supposed to be used.
A bunch of things now done.

Question for Ernst about where to put the troubleshooting and issues that arrived in the test scenarios.

Meeting ending 14:45.

Meeting With supervisor – March 4, 2021

Meeting started at 08:00 on MS Teams (NTNU)

The group updates Ernst on the newest updates regarding the switch situation and test scenarios.

Fagerbekk is showing a demo of how the group conduct a typical test scenario

Ernst says that we should mention in the report why we make the decisions like with DNS.

Ernst says that the speed test part should be discussed in the report in regard to the number of times it should be run.

Q: Ernst: how often are you running speed test in regard to having gaps between meetings
A: We do it like this to save time

Ernst says that it is important to explain the images and use them in the text, so the reader gets an understanding of it.

Feedback on procedure
Looks like it is through tough out, say Ernst

Q: Should we have a Troubleshooting chapter?
A: Yes and no. Should mention problems that have occurred if it affects the result. The size of it depends on if the assignment gets all done. If not, it should be included.

Q: Should we include a “hypotese” of each test scenario?
A: ~~«ikke hva man tror, men hva som er tilfellet».~~ EDIT: “ forklare hvorfor man kjører de testene/scenarioene man gjør, og hva man ønsker å undersøke. Dette *kan* formuleres som

en hyoptese, men ikke skriv noe à la at scenario X test Y gjennomføres fordi dere tror at resultatet vil bli Z.”

Q: Should we have a Project Owner?
A: it depends. Is it important for the report? And check how other BC. reports have implemented this.

Group meeting - March 9, 2021

Meeting started at 09:10 (NTNU)

Going through the Trello board including feedback.

Meeting ends 13:00

Meeting with Supervisor – March 11, 2021

Meeting started at 08:00.

Ernst asks how the bachelor is going.
A: We’re missing only two test scenarios of those Telenor wanted, and will probably be finished with those to tomorrow.

Q: How much is too much text to explain something? Too many pictures?
A:

- “When do you think you need a picture to explain”, without any pictures at all, it can become (gørt?)
- In the context of procedure he didn't think it would need a lot of pictures and if it is explained enough in the text
- A picture every 3rd page isn't the end of the world, but a picture book is too much.

Q: How long do we need to remind the reader that this is about videoconferencing?
A:

- Use it where it is needed.
- If already explained this for a few pages, it is probably not needed

Q: “Why do we discuss”, where do we draw the line of discussing something?(e.g. why we used, what it is used for.)
A:

- It needs to have a reason for being there.
- Why we didn't choose a thing is only really important if it's obvious for the reader that we should have used that thing and then didn't.
- Not too much to draw away the attention of the reader.
- The discussion can be placed under the method where it is relevant.

- After discussion of the tests, could have a chapter of “what could be done different” to discuss this.

Q: Telenor has given us these test scenarios. We would’ve wanted to run another test for
A:

- This can be relevant for the bachelor, but needs to be phrased in a good way a not in a “just because of Telenor”.

Q: Should we write a hypothesis on the UDP test that we think it will fall over on TCP?
A:

- Theory and practicality are not always the same so even if the outcome is “known” it still will be something to discuss.

Meeting ended at 08:30.

Group meeting - 16/03-2021

Meeting started at 09:15.

Trello is examined to look at what has been done. The group then read through the “ready for feedback” section to acknowledge that it is ready to be delivered.

Meeting moved from A270 to “Atriet”.

Further work on the overleaf is done.

Discussion about what to look at in Wireshark, and where to look at, and how much time should be allotted to finding out what kind of information to extract from the pcap files.

- Comment on the formulas on latency, if they aren't used by us, should they be described in an earlier chapter still.
- A lot of information to sift through in the pcap files, hard to find exact values we want to find.
- can see packets sent per second and take out graphs.
- Is it worth spending time on it now before Easter where we want to have at least a template of how we want the test scenarios done.
- Should we ask Ernst about some questions of the pcap files what we chose to take out and if that is a valid reason. The decision was to ask Ernst.

Discussion on if we should use the three meetings on each app tenant or if one(the best) scenario is enough

- Microsoft Teams Power BI does not have a functional graph for showing the jitter and latency through every second/minute of the meeting only an average.
- Discussion on the different uses of a graph of the three meetings contrary to one meeting.
- Asked Ernst if an average of three meetings could be good enough or if a picked “best” scenario would be enough.

Further work for this iteration.

- Doing method for the rest of the test scenarios
- Writing the results for test scenario 1.1, using only the “best”/median test scenario.

Andreas Kilde Lien and Håkon Erstad will continue on the results for test 1.1 and follow the structure already given for the test scenario. May be changed by response from Ernst.

Kristoffer Fagerbekk and Andreas Rømo will look at power bi graphs and try to find good solutions for graphs to include in results. and then take it further in results.

Both two-man teams will try to get done the discussion as well for test 1.1.

Test scenario 1.2 will also be worked on to have something to compare to. This will be tried to be done by next tuesday.

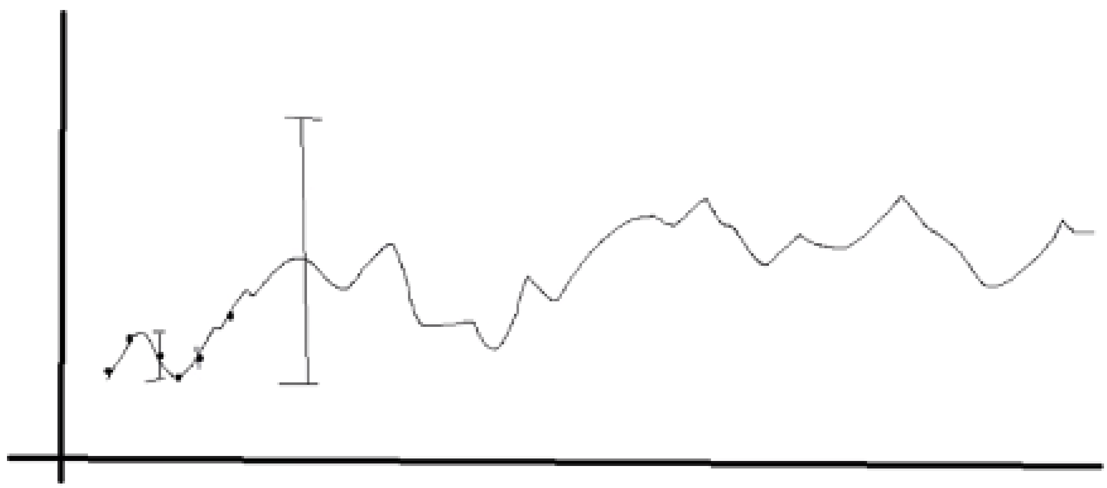
Meeting ended at 16:05.

Meeting with supervisor – March 18, 2021

Meeting started 08:00 at Teams (NTNU)

Rømo gives Ernst an update on what has been done since last Thursday.

Q: We have 3 test iterations on each tenant, how should we pick the correct one? Avg, the best one, ect.
A: What do you wish to show and present in the report. Why do you choose to do 3 iterations? (Telenor wanted 5). Avg and variations



Rømo mentions that a human factor is included in the math problem. That makes it tricky.

Should mention that there was run 3 test iteration and show one of them. But could include the iteration for where it could be interesting.

Mention in the report why there was no use of avg and/or variations.

Q: What should we compare all the test scenarios with baseline?
A: Depends, whether it gives any information.

Q: How to calculate End-to-end for UDP in Wireshark?
A: Important with NTP, for an accurate result.

Parse the data to find UDP packets and calculate the values and compare to platforms requirements.

Q: Where should we include the speed test tables?
A: Explain it thoroughly the first time and mention it just in the later test scenarios. Could also be added in the appendix.

Q: Both Teams' tenants choose a "random" server, not a specific one.
A: Talk with Telenor about this. But could be an interesting finding.

Send the report to Ernst BEFORE March 28, 2021. Else contact Ernst for a new agreement further into the easter.

Comment in the report on less important things for feedback.

Meeting ends 08:55

Group meeting - March 23, 2021

Meeting starts at 09:15 on NTNU.

The group starts by going through Trello.

Mainly discuss chapter "Results" in the report.

Fixing comments in the "Introduction" chapter.

Group meeting - April 6, 2021

Meeting started at 09:15 on Discord.

Starting with discussing feedback from supervisor, Ernst, on the report.

Ends with making changes in the report on the Introduction chapter.

Meeting ends at 14:15

Meeting with supervisor – April 8, 2021

Meeting starts at 08:00.

Q: About comments made in the text about sources and if the title is kinda "misleading" if it doesn't immediately show the point that is referenced.
A:
- "Sensor" will probably read the sources less than what Ernst did
- It's nice to have good sources.
- If it happens frequently with unrelated or bad sources, it will affect the thesis poorly.
- Source [4] as an example was suggested again that it wasn't super relevant and could be replaced.

Q: About comments made on the problem subsection. Change to a more text-based approach rather than a task-list? Not completely consistent information around the goals and tasks.
A:
- Reader hasn't got the insights we or Ernst has so will probably not understand the overall task at hand for the project by just reading the list.
- To find out if Meraki is "good or not" is more of "sales-based" and not academia.
- Chapter 1 should be small and consistent, but give a good overview of what the thesis is about and what we want to find out.
- Our report did not have a consistent red thread through the report. Reader could stop and think "why am I reading this?/why is this relevant?".

Q: Take out existing systems and tools and related works as well as split technical background or test scenarios into its own section? (IMRAD)
A:
- The IMRAD method is a framework for how it can be done, but it should not constrain the thesis to a poorer quality.
- Too long chapters is a problem, and should be addressed.
- Try to look at things in the report and find out if something is important to know for the thesis or just "nice-to-know" facts.
- Don't get too ambitious and write "too much", the thesis seems to be on the way to become long so important to cut away things that aren't relevant to the understanding of the tasks.

Q: Some of the existing systems and tools were a little too focused on the company and information about its things rather than relevant information to the task. But can we take a picture of the MS teams and Webex?
A:
- Teams and webex as companies are not relevant to the thesis task, but more the applications

- Ernst wasn't sure if we had permission to use pictures of the platforms, therefore we need to ask our contacts if we can.

Q: How high level should we base the report on the reader's knowledge being?
A:
- The reader should have at least basic computer knowledge.
- How the meeting is set-up should be more detailed. Write about peer-to-peer, “sentral”, etc.
- Missing some explanations in the earlier chapters of things we may understand, but the reader may not.
- What is a tenant? What is “barneparken”?
- Is the function of a tenant important for the problem statement? If so then maybe use a couple of sentences to explain this in the background.

Q: Progressively found that the tenant has become less and less important throughout the making of the report. Should this be removed?
A:
- If there were findings around the tenant, then it should not be removed
- Maybe introduce it very briefly with a line or two.
- The reader most likely won't know the words we think they “should” know.

Q: Avgrensing?
A:
- Make a subsection of the “avgrensing”/scope(?) of the task.
- Don't write in a story-based approach, but more in a way that shows what has been done and what the results we got from this was.
- Include the playbook in the problem statement.

Q: Want to make a playbook, but have a lot of other things to get into place to answer some of the things found through the report.
A:
- Don't include something that isn't included in the final report as a goal or a task in a problem statement.

Ernst recommends asking someone with a good english understanding and if lucky someone with some computer knowledge to read through and get an unbiased opinion on what is explained good enough. As both the group and Ernst already have somewhat of an understanding of the things explained and will therefore understand more than the average reader.

Q: Tenants and Teams. How we should approach the test scenarios as the choosing of tenants still doesn't lock a path through the network. Is there a point to compare the tenants if it doesn't really matter which is chosen?
A:
- This can be somewhat of a problem statement that we want to find out.
- The reader will want to know how the tests were done. The comparison may not be explicitly explained how it should be done. Or start with results and conclude with if it does or doesn't matter.

- Average of average as long as it gives a value that gives some data.

Q: Vector-based graphics?
A:
- Use if possible.
- If you can find a vector-based picture instead of a normal picture then it should be okay.

Meeting ended at 08:58.

Group meeting - April 13, 2021

Group started at 09:15.

Sent a message to Bjørn about meeting plans for the 15th, if any additional people were booked.

Went through last Thursday's report to see if everything was good before sendoff.

Went through Trello to see what needed to be done and what had been done.

Talk about what to prioritize of the different changes that need to be made. Conclusion was to use a week to “fix” changes and other things in already written text. And then next week make a new decision about the priorities.

Discussion about restructuring much of the method chapter, Ernst made a comment about the continuity and the understanding of a test scenario would be forgotten a little when you read chronologically through the report and therefore wanted the results of the test scenario and the discussion to be right after the method of the test scenario. The group then discusses how that would be done in theory and if it can be done in an IMRaD way.

Readthrough and fixing of more of the comments by Ernst on the overleaf report.

Discussion on where to place an introduction to a tenant and how to actually explain what a tenant is. Found an explanation for tenants which the group members were okay with and added it in.

Meeting ended at 16:15.

Meeting with supervisor – April 15, 2021

Meeting started at 08:00 on MS Teams (NTNU).

Fagerbekk suggests that we stop the meeting and move outside to feel the sunshine (joke).

Rømo gives Ernst an update on what has been done since last time.

Starting the meeting with some questions.

Q: Could you, Ernst, read the new problem statement?
A: The main goal or point of the objective for the thesis is the important part for the problem statement. Nothing wrong with having the list, but there was something that was missing.
Feedback on text: This was much better and/or good to include.

Q: What do you think about the section “structure of the test scenarios” (like differences between 1.1, 1.2, etc.)
A: This is good and is like an introduction for them all. Should not have too many repetitions.

Q: Should there be an image of the topology early?
A: Yeah, should include a simplified topology.

Q: How should we structure the section “structure of the thesis?
A: Some write a whole page, but that is kind of too much. A sentence for each chapter is prefered.

Done with all the predefined questions.

First time mentioning something like a tenant or explaining it, write it in italic.

Comment on the text: MS Teams and Webex are the application, not the one that sells.

Q: if we don’t mention something about video codec or/and audio codec. Should it be removed from Tech. back.
A: Yes, is distracting if not used. But you may include reasons for not including this. Like in the discussion chapter.

Q: We have made some adjustment to glossary with ref. and some adjustment.
A: That is good.

Q: Is the network topology alright?
A: It’s good.

Making your own figures is recommended.

Group meeting - April 20, 2021

Meeting started on NTNU at 09:15.

First on the agenda was going through Trello.

Worked on parts of the report that was finished since the last meeting. For the following sections: “videoconference”, “networking and protocols” and “network deviation”.

Lastly the group reviewed the sections about the test scenarios in method.

meeting ends at 14:30

Meeting with supervisor – April 22, 2021

Meeting starts at 08:00.

Q: Structure of the report and restructuring of the method. If the baseline was done well, we should be able to describe the changes for the next test scenarios in 2 sentences. We wonder if we have misunderstood since some test scenarios have a little bit more changes?
A:

- As long as this is changes this should be included
- But it should be short and precise
- Make it simple for the reader to understand, without getting lost
- Can make the introduction short as well, and be kept to a couple of sentences, but just to describe why the test is run(what is the point of the test)

Q: Baseline and test scenario 1.1-1.3 has mostly the same setup should we keep a picture to describe each?
A:

- Ernst thinks having pictures is fine, but the pictures should focus on the changes.
- The reader will more easily remember the topology if it is repeated

Q: Chosen a setup where we have a discussion after each test scenario and then a more general discussion. Do you have any thoughts on this?
A:

- Ernst thinks having just read the results it is smart to discuss those results right after
- Should have an overview of everything that will be presented in the results and discussion chapter and the structure of what should be expected moving forward

Q: Read the new goal subsection. Thoughts?
A:

- Some cites/references seems to be missing
- The text seems to be alright

Q: References to tables. Making tables of requirements from Teams and Webex since this may be changed in the website it is taken from.
A:

- Good to have the requirements in the report and not need to go to an outside source

Q: Having a technical background where the subject isn't talked about later but is relevant to the task.
A:

- If it is kept short and just introduces it and explains it simply there is no problem in doing this
- If it was a lot of text it would be more of an issue

Q: Is it relevant to describe the youtube video name we played during the meeting?
A:

- Completely fine to say which videos were used.

Q: Comments about the speed tests. They were placed in the introduction of the first test scenario, the comment was that this was more a result. But we think it is a preliminary to the meeting and should be placed earlier.
A:

- Wasn't explained well enough that this was just a control of the values and not something to use as a result.

Q: Where to place the tables for bandwidth requirements?
A:

- If they are talked about they should also be placed here
- When the table is referenced, try to describe the relevant information where you are in the text so the referenced values are understood

Q: The bandwidth tables are taken from somewhere else, how should this be referenced?
A:

- Should be either cited or put in a footnote

Ernst commented again on the licensing of each picture. Since some of it is made by the makers of the report and the report is made to become public. Should this need another licensing. If the licensing is the same for a lot of pictures, it could maybe be placed somewhere else. But it takes up alot of space. The license of the pictures is probably not an interest of the reader.

Q: The unknown chapter, what could it be named?
A:

- Could be put under technical background
- Or renamed the chapter technical background and related work

Q: Related works, should this be dropped if we don't find anything more in it?
A:

- If there is a lot of previous work that relates to the report then it should maybe be expanded on a bit more.
- Maybe try to search around a bit more and find some more related works if possible

Meeting ended at 09:01.

Introduction for why the test scenario was done and the goal of the scenario was moved to method.

Group meeting - April 27, 2021

Meeting started at 09:15 on NTNU.

The group went through the test scenarios 1.1 - 1.3 in chapter “Discussion and results”. What has been done and what should be fixed.

Discussing a section about “general discussion”.

Discussing the test 3.2 of what value it gives to the thesis by doing.

Discussion of the workflow continuing on if we should implement having another group meeting to have more time to discuss together as a group and decide on problems that may arise through the week.

- Conclusion: taking an extra meeting Saturdays at 11 on discord.

Closing in on the delivery date so priorities needed to be discussed again. Concluded with doing as we have done, but put more emphasis on if complete with the tasks look for new tasks to do.

Using a timestamp of 24h instead of 12h.

Own pictures need to be reformatted to svg file.

Meeting ended at 12:15.

Meeting with supervisor – April 29, 2021

Meeting started at 08:00 on MS Teams (NTNU)

Rømo gives a recap for what the group has done since last time.

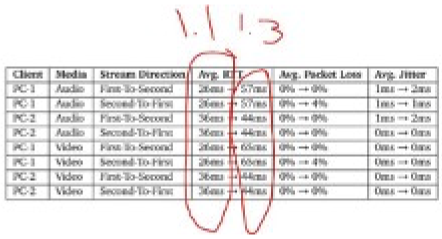
Questions:

Q: How should we structure the appendix?
A: Hard to say. Should be somewhat like the other part of the report in regard to figures and tables. Make sure that the other part is good first and then try to clean the appendix.

Q: Have a meeting ID for the figure, should it be removed?
A: Should use something else then the meeting ID as reference to the raw data in appendix.

Q: We have a long report at around 76 pages. How should we cope with it?
A: 50-80 pages is “normal”. Higher importance of the text being important, and less repetition.

Q: How deeply should we explain a table?
A: Shorten the explanation on the UDP vs. TCP. The percentage is important for the comparison. Could be a bit too many numbers and percentages that the reader doesn’t find the interesting data.



Client	Media	Stream Direction	Avg. RTT	Avg. Packet Loss	Avg. Jitter
PC-1	Audio	First-To-Second	26ms → 57ms	0% → 0%	1ms → 2ms
PC-1	Audio	Second-To-First	26ms → 57ms	0% → 4%	1ms → 1ms
PC-2	Audio	First-To-Second	36ms → 44ms	0% → 0%	1ms → 2ms
PC-2	Audio	Second-To-First	36ms → 44ms	0% → 0%	0ms → 0ms
PC-1	Video	First-To-Second	26ms → 65ms	0% → 0%	0ms → 0ms
PC-1	Video	Second-To-First	26ms → 65ms	0% → 4%	0ms → 0ms
PC-2	Video	First-To-Second	36ms → 44ms	0% → 0%	0ms → 0ms
PC-2	Video	Second-To-First	36ms → 44ms	0% → 0%	0ms → 0ms

The difference between 1.1 and 1.3 is that it is interrupted.

Q: Should we include site survey in the report’s result chapter or appendix?
A: If it is relevant for the reader, include it in the report’s result chapter.

Q: Lost data for SNR on test scenario 4.1. What should we do?
A: Is it important for the test scenario? If it could tell something about why the data act as it does. Had been nice to had a bit more data regarding changes in data, something that

Group meeting - May 4, 2021

Meeting started at 09:00 on Discord.

First off was a status update on what has been done since last time. Then we discussed the section “Traffic flow”. Lastly the figures were discussed and the following had to be changed: I- and P-frame graph, and Control Hub graphs in regards to title and setup of graph.

Meeting ended at 11:25

Meeting with supervisor – April 29, 2021

Meeting started at 08:00.

Q: Test scenario 3.1 and 3.2 changes: should we continue using the Telenor provided as we dont see it fits into the report or can we rather use our own?(Without many pictures)
A:

- Ernst agrees that it wouldn't be scientific

Q: Shows a graph from the 4.1 scenario where 4G was used, the graph is pretty all over the place, but should be stable? Unsure why this could be as the signal strength seemed to be good?

A:

- May be affected by surrounding frequencies(shared medium)
- But ask Henning for confirmation

Q: Shows graphs from 3.2 of throughput in Mbps. Unsure if the graphs are readable? A strange dip in the graph at the start?
A:

- Explain it shortly why we have a graph with 2 scenarios.
- Try to discuss possible explanations for why the dip happened.

Q: Discussion for 3.1: 2 streams for audio compared to 3 video streams, not quite sure why
A:

- Possible that the 2 webcams received needs a stream of their own while the audio can be combined(?).

Ernst wonders if we need next Thursday as a meeting as it is a “helligdag”.

- We may need it as it's the last chance for asking anything before delivering it.

As the thesis is very long already it is important to look at the information that is given and make sure it's not repeated, and that there’s a reason for showing this.

Meeting ended at 08:54.

Meeting with supervisor – May 13, 2021

Meeting started at 08:00 on Teams (NTNU)

Rømo gives an update on what has been done since last time.

Q: How should we handle RSSI?
A: Ernst recommends getting back to Jon to explain further what he means.

Q: On test scenario 1.3 there is a long drop for PC-2 while on PC-1 there is a much shorter, Why?
A: The server receives a stream of packages. The server will detect missing packages, so it will wait by doing something. Like now I will have to change to using TCP. Why it is longer on Webex compared to Teams is unknown. Contact someone within the organization for answers.

Q: Is it necessary with refs in the Playbook?
A: Depends on whom it the target users for the Playbook.

Q: Next week is it possible to get a new meeting?
A: yes, that could be scheduled if needed. Tuesday fits the best. 10 o’clock.

Q: QOS

A: Ask Jon about how this could be important for the report.

The meeting ends at 09:58

Meeting with supervisor – May 18, 2021

Meeting started at 10:00.

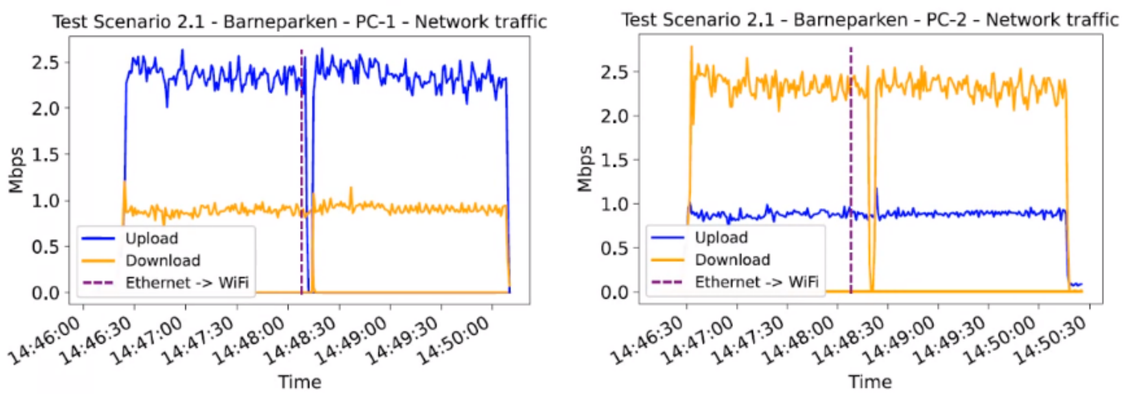
Q: What are the supervisors thoughts on adding in the google drive with data to showcase the work that has been put in and show that work has been done and not just trust us.
A:

- It should be systematic, so the reader won't get lost, and will find what is wanted to show
- Adding a “guide” in appendix on how to navigate the Drive
- Referenced in the text as well what and where to find

Q: Discussion, very much that can be discussed. How much is too much for discussing possible aspects with the thesis? Too specific or too little of possible causes?
A:

- The discussions needs to have a relevancy
- Not positive with much “synsing”

Showing a graph of 2.1.



(a) Throughput in Mbps for PC-1, towards Teams service using Ethernet and .

(b) Throughput in Mbps for PC-2, towards Teams service using Ethernet cable.

Weird that the drop for PC-2 is so much later than when it happened on PC-1. Maybe some things have happened around the merging of pcap files.

Q: Hypothesis vs Goal
A:

- The supervisor hasn't seen the use of hypothesis in a lot of the theses he has read.
- How we have done sounds reasonable
- The goals should be milestones to achieve an answer to the main problem statement

Meeting ended at 11:15.

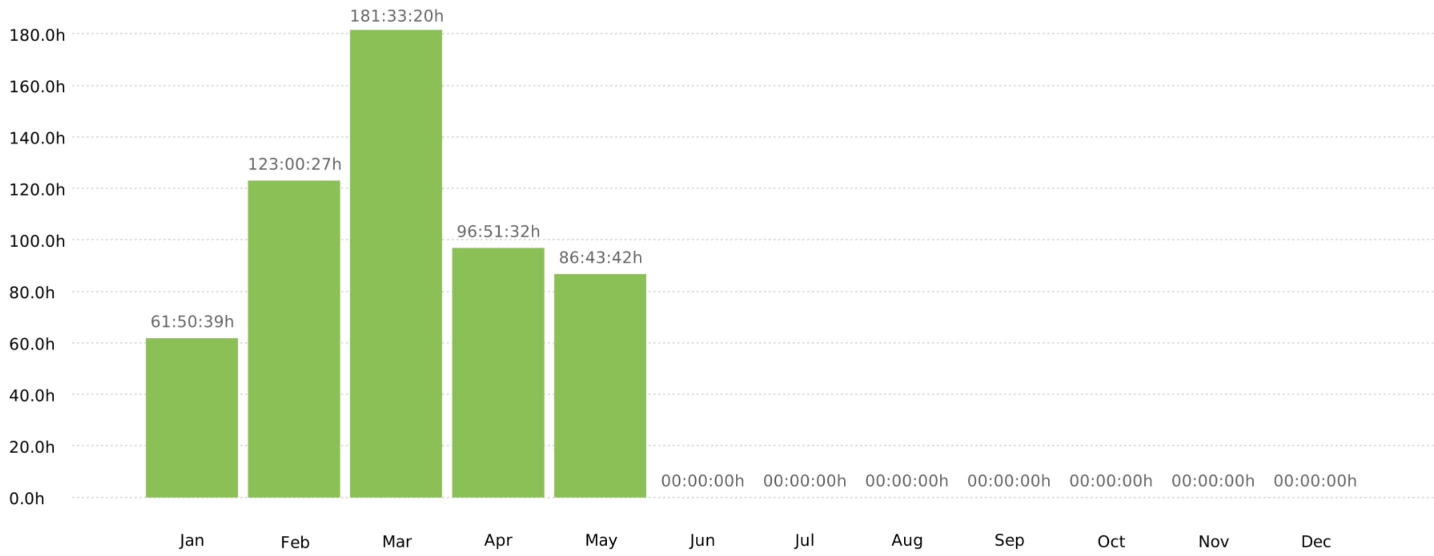
A.4 Work hours

A.4.1 Håkon

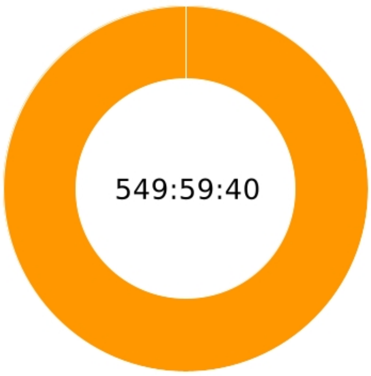
Summary report

01/01/2021 - 12/31/2021

Total: 549:59:40

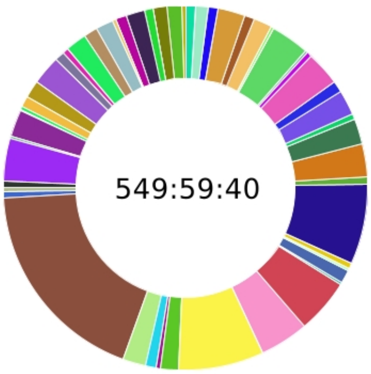


Project



Bachelor - Håkon Holm Erstad	549:59:40	100.00%

Description



Meeting with Bachelor Group + Telenor	01:34:10	0.29%
Testing Scenario 2.1v2	07:15:00	1.32%
Test Scenario Baseline Results + Discussion	06:45:00	1.23%
Bachelor group meeting	00:11:12	0.03%
Meraki Testkit Setup + Bachelor Meeting	04:09:53	0.76%

Meeting with Supervisor	09:14:18	1.68%
Meeting with Bachelor Group + Test scenarios	05:00:00	0.91%
Meeting with Supervisor and Taskgiver	01:45:00	0.32%
Overleaf Readthrough	08:29:06	1.54%
Meeting with Supervisor + Bachelor Group + Overleaf Work	06:50:06	1.24%
Testing scenarios 3.1 + 3.2	10:30:00	1.91%
Q&A Session	02:38:48	0.48%
Meeting with Telenor + Webex Meraki	04:45:00	0.86%
Prep for testing + Raspberry pi setup + Switch config	14:12:25	2.58%
Testing scenario 2.2	08:25:00	1.53%
Meeting with Telenor + Aftermeeting with Bachelor Group	05:13:00	0.95%
Meeting with Bachelor Group + Meeting with Telenor	01:52:48	0.34%
M365 teams rooms optimize network	00:14:55	0.05%
Prep for testing	13:00:00	2.36%
Configure your network for Microsoft 365	01:04:33	0.20%
Test Scenarios + Overleaf	21:10:00	3.85%
Meeting with bachelors group	03:05:20	0.56%
Reading previous Bachelors	02:00:00	0.36%
Meeting with Supervisor + Bachelor Group	02:41:23	0.49%
Overleaf Work	102:15:09	18.59%
Testing scenario 4.1	11:00:00	2.00%
Meeting with Bachelor Group + Readthrough of feedback	05:30:00	1.00%
Test Scenario 1.2 + Overleaf	02:00:00	0.36%
Setting up and doing Test Scenarios 1+2	08:50:00	1.61%
Meeting with Bachelor Group	41:36:58	7.57%
Gantt scheme	00:12:53	0.04%
Testing scenario 2	23:30:00	4.27%
Meeting with Bachelor Group + Trello	26:38:00	4.84%

Bachelor Thesis Writing	01:06:00	0.20%
Testing scenario 1	06:23:53	1.16%
Testing Scenario 2.1 Webex Results	01:30:00	0.27%
Meeting with Telenor & Bachelor Group	02:40:00	0.49%
Reading & Writing	38:56:19	7.08%
Meraki Testkit Setup	03:36:59	0.66%
Meeting	15:50:07	2.88%
Research	12:45:00	2.32%
Reading bachelor/master thesis	02:25:21	0.44%
Testing scenarios	12:10:00	2.21%
Test Scenarios Overleaf Structure	05:30:00	1.00%
Writing Session with Bachelor Group	16:53:41	3.07%
Overleaf Work with Bachelor Group	02:20:00	0.42%
Project plan work	01:30:00	0.27%
Meeting with Bachelor Group + Overleaf Work	18:22:11	3.34%
Gnatt scheme	00:24:09	0.07%
Research - Microsoft Docs	01:17:09	0.23%
Testing Wireless + Scenarios	08:45:00	1.59%
Testing Switches and MX	05:00:00	0.91%
Test Scenario Baseline	13:15:00	2.41%
Research and Writing	04:45:00	0.86%
Meeting with Telenor	06:13:54	1.13%
Meraki Tests + Meeting with Supervisor + Bachelor Writing	04:40:00	0.85%

Project / Description	Duration
Bachelor - Håkon Holm Erstad	549:59:40
Meeting with Bachelor Group + Telenor	01:34:10
Testing Scenario 2.1v2	07:15:00
Test Scenario Baseline Results + Discussion	06:45:00
Bachelor group meeting	00:11:12
Meraki Testkit Setup + Bachelor Meeting	04:09:53
Meeting with Supervisor	09:14:18
Meeting with Bachelor Group + Test scenarios	05:00:00
Meeting with Supervisor and Taskgiver	01:45:00
Overleaf Readthrough	08:29:06
Meeting with Supervisor + Bachelor Group + Overleaf Work	06:50:06
Testing scenarios 3.1 + 3.2	10:30:00
Q&A Session	02:38:48
Meeting with Telenor + Webex Meraki	04:45:00
Prep for testing + Raspberry pi setup + Switch config	14:12:25
Testing scenario 2.2	08:25:00
Meeting with Telenor + Aftermeeting with Bachelor Group	05:13:00
Meeting with Bachelor Group + Meeting with Telenor	01:52:48
M365 teams rooms optimize network	00:14:55
Prep for testing	13:00:00
Configure your network for Microsoft 365	01:04:33
Test Scenarios + Overleaf	21:10:00
Meeting with bachelors group	03:05:20

Reading previous Bachelors	02:00:00
Meeting with Supervisor + Bachelor Group	02:41:23
Overleaf Work	102:15:09
Testing scenario 4.1	11:00:00
Meeting with Bachelor Group + Readthrough of feedback	05:30:00
Test Scenario 1.2 + Overleaf	02:00:00
Setting up and doing Test Scenarios 1+2	08:50:00
Meeting with Bachelor Group	41:36:58
Gantt scheme	00:12:53
Testing scenario 2	23:30:00
Meeting with Bachelor Group + Trello	26:38:00
Bachelor Thesis Writing	01:06:00
Testing scenario 1	06:23:53
Testing Scenario 2.1 Webex Results	01:30:00
Meeting with Telenor & Bachelor Group	02:40:00
Reading & Writing	38:56:19
Meraki Testkit Setup	03:36:59
Meeting	15:50:07
Research	12:45:00
Reading bachelor/master thesis	02:25:21
Testing scenarios	12:10:00
Test Scenarios Overleaf Structure	05:30:00
Writing Session with Bachelor Group	16:53:41
Overleaf Work with Bachelor Group	02:20:00

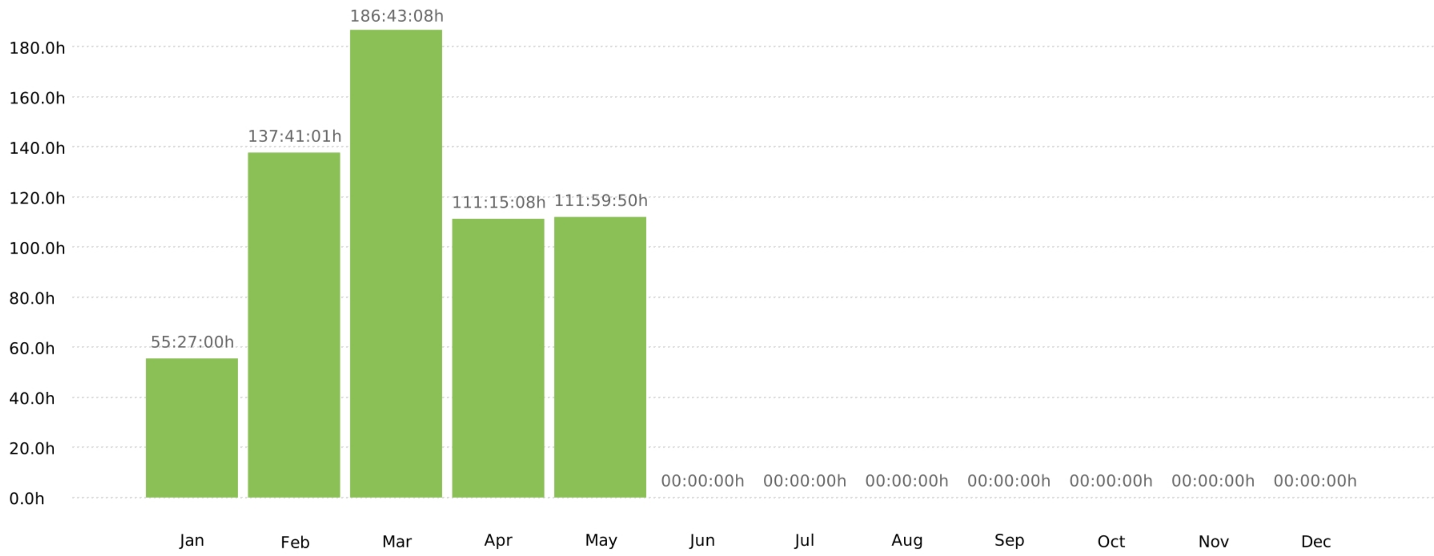
Project plan work	01:30:00
Meeting with Bachelor Group + Overleaf Work	18:22:11
Gnatt scheme	00:24:09
Research - Microsoft Docs	01:17:09
Testing Wireless + Scenarios	08:45:00
Testing Switches and MX	05:00:00
Test Scenario Baseline	13:15:00
Research and Writing	04:45:00
Meeting with Telenor	06:13:54
Meraki Tests + Meeting with Supervisor + Bachelor Writing	04:40:00

A.4.2 Andreas Rømo

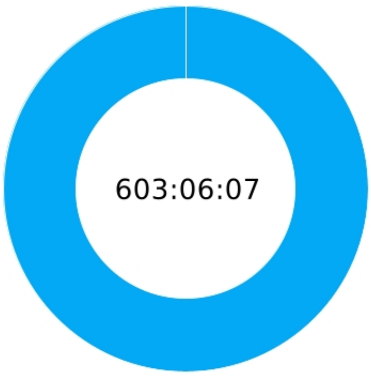
Summary report

01/01/2021 - 12/31/2021

Total: 603:06:07   Billable: 603:06:07   Amount: 0.00 USD

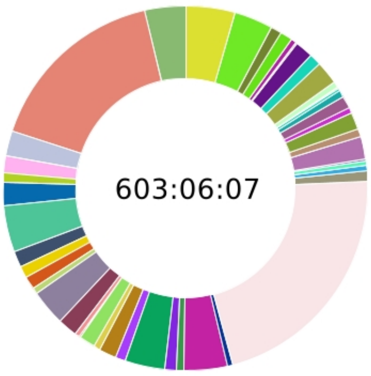


Project



Bachelor	603:06:07	100.00%

Description



Meeting - Supervisor	21:38:52	3.59%
Reading & Writing Thesis	98:31:58	16.34%
Creating images for test scenarios	13:38:25	2.26%
Test Scenario 2.2 (Wifi) ++	08:37:15	1.43%
Router & Switch | Troubleshooting	05:28:00	0.91%

Running Test Scenario 2	11:57:38	1.98%
Writing & Testing Scenarios	25:03:10	4.15%
Reading, DrPeering, Decoding & Microsoft	08:26:00	1.40%
Reading Bachelor's	06:25:09	1.06%
Test scenario 1.3 + 2.1 | Teams	06:48:04	1.13%
Test scenario 1.1 | Power BI	03:22:28	0.56%
Test scenario 1.1 + | Teams	18:57:32	3.14%
Testing / Running scenarios	10:28:07	1.74%
Restructuring Thesis	02:18:14	0.38%
Tools Exploring - Ad Azure, Tenants, WeBeX, Teams	01:01:47	0.17%
Scenario 1.1 (Teams)	08:17:43	1.38%
Preparing & discussing Scenario 4 (4g)	03:45:38	0.62%
Test scenarios	09:11:13	1.52%
Running Scenario 4 (4g)	05:29:19	0.91%
Writing - Projectplan	21:17:37	3.53%
Reading - Microsoft Doc	06:08:00	1.02%
Reading, DrPeering and Decoding	04:00:00	0.66%
Test Scenario 2 (WiFi)	22:51:25	3.79%
Exploring tools	03:00:00	0.50%
Meeting - BachelorGroup	129:14:05	21.43%
Test Scenario Structure/Discussing/Results	05:23:42	0.90%
Reading - Meraki & WeBeX	03:00:00	0.50%
Test scenario 2.2 | Teams	02:11:55	0.36%
Test scenario 1.3 | Teams	01:10:00	0.19%
Test scenario 3.1 & 3.2 (Improved)	12:05:27	2.00%
Test scenario 1.2 and 1.3 | Rerunning	04:15:00	0.70%
Running Test Scenario 1 & 2	09:15:00	1.53%
Testing of test scenarios - W/Group	02:37:00	0.43%

●	Testing Scenarios	06:57:06	1.15%
●	Meraki first time setup	03:45:00	0.62%
●	Router | Troubleshooting	01:14:00	0.20%
●	Reading - General	00:55:00	0.15%
●	Reading - DrPeering	03:04:00	0.51%
●	Test scenario 1.2 | Teams	12:06:41	2.01%
●	Running Test Scenario 1	06:35:00	1.09%
●	Preparing & Running Test 4.1 / 4.2	09:30:00	1.58%
●	Thesis - Animation drawing / Finding	00:59:35	0.16%
●	Researching - Peering	02:37:00	0.43%
●	Test scenario 1.1 to 2.1 | Teams	06:27:55	1.07%
●	Sorting data	06:05:20	1.01%
●	Playbook	20:43:00	3.44%
●	Meeting - Telenor	26:10:47	4.34%

Project / Description	Duration	Amount
Bachelor	603:06:07	0.00 USD
Meeting - Supervisor	21:38:52	0.00 USD
Reading & Writing Thesis	98:31:58	0.00 USD
Creating images for test scenarios	13:38:25	0.00 USD
Test Scenario 2.2 (Wifi) ++	08:37:15	0.00 USD
Router & Switch | Troubleshooting	05:28:00	0.00 USD
Running Test Scenario 2	11:57:38	0.00 USD
Writing & Testing Scenarios	25:03:10	0.00 USD
Reading, DrPeering, Decoding & Microsoft	08:26:00	0.00 USD
Reading Bachelor's	06:25:09	0.00 USD

Test scenario 1.3 + 2.1 | Teams	06:48:04	0.00 USD
Test scenario 1.1 | Power BI	03:22:28	0.00 USD
Test scenario 1.1 + | Teams	18:57:32	0.00 USD
Testing / Running scenarios	10:28:07	0.00 USD
Restructuring Thesis	02:18:14	0.00 USD
Tools Exploring - Ad Azure, Tenants, WeBeX, Teams	01:01:47	0.00 USD
Scenario 1.1 (Teams)	08:17:43	0.00 USD
Preparing & discussing Scenario 4 (4g)	03:45:38	0.00 USD
Test scenarios	09:11:13	0.00 USD
Running Scenario 4 (4g)	05:29:19	0.00 USD
Writing - Projectplan	21:17:37	0.00 USD
Reading - Microsoft Doc	06:08:00	0.00 USD
Reading, DrPeering and Decoding	04:00:00	0.00 USD
Test Scenario 2 (WiFi)	22:51:25	0.00 USD
Exploring tools	03:00:00	0.00 USD
Meeting - BachelorGroup	129:14:05	0.00 USD
Test Scenario Structure/Discussing/Results	05:23:42	0.00 USD
Reading - Meraki & WeBeX	03:00:00	0.00 USD
Test scenario 2.2 | Teams	02:11:55	0.00 USD
Test scenario 1.3 | Teams	01:10:00	0.00 USD
Test scenario 3.1 & 3.2 (Improved)	12:05:27	0.00 USD
Test scenario 1.2 and 1.3 | Rerunning	04:15:00	0.00 USD
Running Test Scenario 1 & 2	09:15:00	0.00 USD
Testing of test scenarios - W/Group	02:37:00	0.00 USD

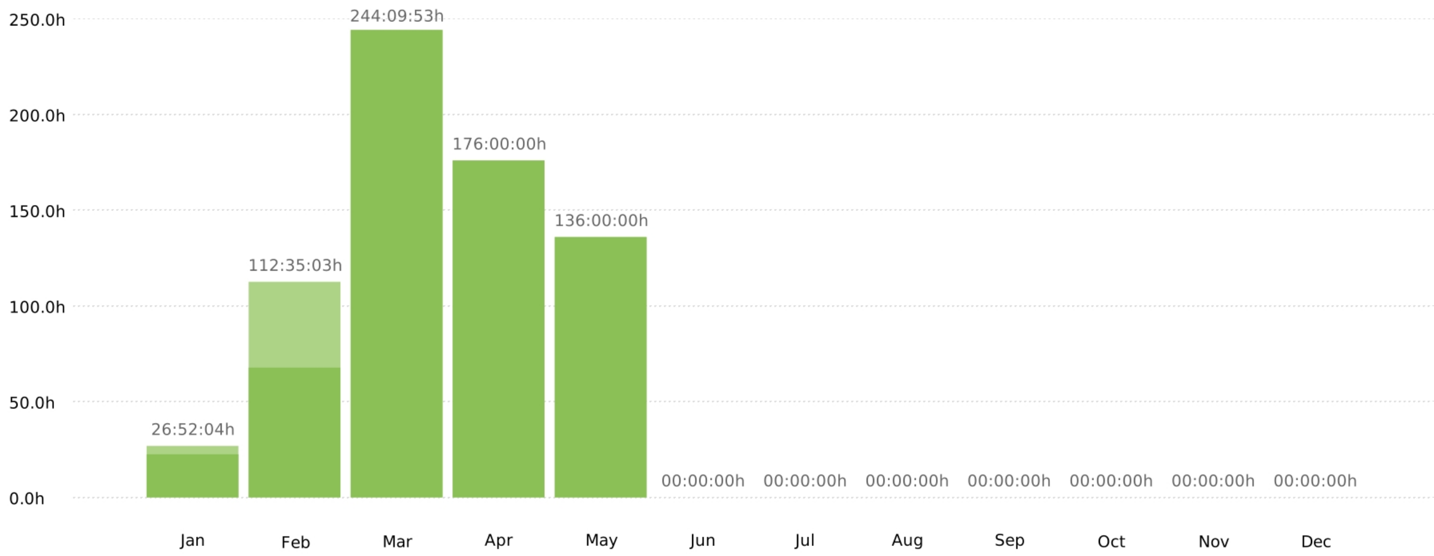
Testing Scenarios	06:57:06	0.00 USD
Meraki first time setup	03:45:00	0.00 USD
Router | Troubleshooting	01:14:00	0.00 USD
Reading - General	00:55:00	0.00 USD
Reading - DrPeering	03:04:00	0.00 USD
Test scenario 1.2 | Teams	12:06:41	0.00 USD
Running Test Scenario 1	06:35:00	0.00 USD
Preparing & Running Test 4.1 / 4.2	09:30:00	0.00 USD
Thesis - Animation drawing / Finding	00:59:35	0.00 USD
Researching - Peering	02:37:00	0.00 USD
Test scenario 1.1 to 2.1 | Teams	06:27:55	0.00 USD
Sorting data	06:05:20	0.00 USD
Playbook	20:43:00	0.00 USD
Meeting - Telenor	26:10:47	0.00 USD

A.4.3 Kristoffer

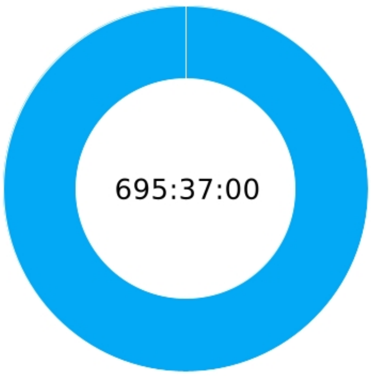
Summary report

01/01/2021 - 12/31/2021

Total: 695:37:00 Billable: 646:26:27 Amount: 0.00 USD

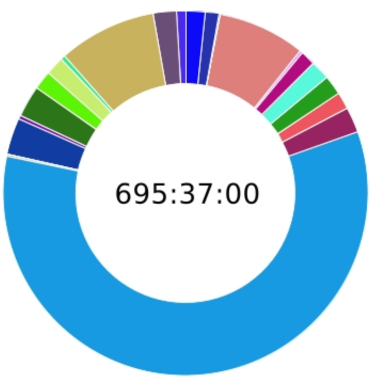


Project



Bachelor	695:37:00	100.00%

Description



Prosjektplan	05:20:49	0.77%
Søndag	14:42:36	2.11%
Bachelors meeting	60:01:14	8.63%
Reading bachelor/master thesis	02:50:00	0.41%
Fredag - Fant ut at pcap tullet med ntopng.	11:36:48	1.67%

●	Torsdag - siste test scenario 3.2 ferdig	11:30:00	1.65%
●	Onsdag	19:19:10	2.78%
●	Lørdag - Kjører første test scenario - meraki feilet på uninett side	02:09:29	0.31%
●	Fredag - testscenario 2	22:50:05	3.28%
●	Supervisor meeting	00:00:00	0.00%
●	Configure your network for Microsoft 365	01:05:00	0.16%
●	Writing and analyzing	408:00:00	58.65%
●	Tirsdag	16:00:00	2.30%
●	Torsdag	10:00:00	1.44%
●	Tirsdag - Fikset HP switch problem - Testscenario 1.1	12:00:00	1.73%
●	Onsdag - Testingscenario 1.2	12:11:02	1.75%
●	Mandag - Prøver å fikse HP switch	09:07:00	1.31%
●	Dr Peering	01:40:34	0.24%
●	(Without Description)	54:00:00	7.76%
●	Writing bachelor	00:03:13	0.01%
●	gantt scheme	00:25:00	0.06%
●	Employer meeting	00:00:00	0.00%
●	Fredag	08:30:00	1.22%
●	Torsdag - ernst - testscenario 2	12:00:00	1.73%
●	M365 teams rooms optimize network	00:15:00	0.04%

Project / Description	Duration	Amount
Bachelor	695:37:00	0.00 USD
Prosjektplan	05:20:49	0.00 USD
Søndag	14:42:36	0.00 USD
Bachelors meeting	60:01:14	0.00 USD
Reading bachelor/master thesis	02:50:00	0.00 USD

Fredag - Fant ut at pcap tullet med ntopng.	11:36:48	0.00 USD
Torsdag - siste test scenario 3.2 ferdig	11:30:00	0.00 USD
Onsdag	19:19:10	0.00 USD
Lørdag - Kjører første test scenario - meraki feilet på uninett side	02:09:29	0.00 USD
Fredag - testscenario 2	22:50:05	0.00 USD
Supervisor meeting	00:00:00	0.00 USD
Configure your network for Microsoft 365	01:05:00	0.00 USD
Writing and analyzing	408:00:00	0.00 USD
Tirsdag	16:00:00	0.00 USD
Torsdag	10:00:00	0.00 USD
Tirsdag - Fikset HP switch problem - Testscenario 1.1	12:00:00	0.00 USD
Onsdag - Testingscenario 1.2	12:11:02	0.00 USD
Mandag - Prøver å fikse HP switch	09:07:00	0.00 USD
Dr Peering	01:40:34	0.00 USD
(Without Description)	54:00:00	0.00 USD
Writing bachelor	00:03:13	0.00 USD
gantt scheme	00:25:00	0.00 USD
Employer meeting	00:00:00	0.00 USD
Fredag	08:30:00	0.00 USD
Torsdag - ernst - testscenario 2	12:00:00	0.00 USD
M365 teams rooms optimize network	00:15:00	0.00 USD

A.4.4 Andreas Lien

Date	Start	End
12. januar 2021	09:00	14:00
13. januar 2021	09:00	13:30
14. januar 2021	08:00	08:30
14. januar 2021	10:15	11:40
19. januar 2021	09:15	13:10
20. januar 2021	14:00	18:00
21. januar 2021	08:00	09:30
21. januar 2021	09:30	13:00
25. januar 2021	17:00	21:00
26. januar 2021	09:15	14:00
28. januar 2021	08:00	08:25
28. januar 2021	08:25	13:15
1. februar 2021	09:15	11:15
4. februar 2021	08:00	08:55
4. februar 2021	09:00	10:00
4. februar 2021	10:30	13:10
4. februar 2021	13:33	18:00
5. februar 2021	09:00	18:00
6. februar 2021	10:00	16:15
9. feburar 2021	12:30	16:00
10. feburar 2021	09:00	13:20
11. februar 2021	08:00	08:25
11. februar 2021	09:00	11:20
11. februar 2021	15:00	19:00
12. februar 2021	09:00	15:00
15. februar 2021	14:00	19:00
16. februar 2021	09:15	14:15
18. februar 2021	08:00	08:23
18. februar 2021	09:30	10:00
18. februar 2021	10:00	16:00
19. februar 2021	11:00	16:00
21. februar 2021	15:00	17:00
22. februar 2021	11:00	15:30
23. februar 2021	09:15	13:15
23. februar 2021	16:30	19:00
24. feburar 2021	08:00	22:12
25. februar 2021	08:00	08:24
25. februar 2021	09:00	10:00
25. februar 2021	10:00	15:00
26. februar 2021	09:00	20:00
27. februar 2021	09:00	20:00
28. februar 2021	12:00	16:00
1. March 2021	16:00	20:00
2. March 2021	09:15	21:22

3. March 2021	09:00	21:00
4. March 2021	08:00	08:24
4. March 2021	09:00	10:00
4. March 2021	11:30	19:20
5. March 2021	09:00	21:00
6. March 2021	09:00	20:36
7. March 2021	09:00	18:00
8. March 2021	09:00	18:00
9. March 2021	09:00	18:00
10. March 2021	09:00	18:00
11. March 2021	09:00	20:30
12. March 2021	09:15	18:30
14. March 2021	09:00	15:00
15-Mar-21	14:00	16:00
16-Mar-21	09:15	13:00
18-Mar-21	08:00	08:23
18-Mar-21	09:00	09:40
18-Mar-21	10:00	13:00
19-Mar-21	12:00	16:00
22-Mar-21	12:00	15:00
23-Mar-21	09:00	14:00
23-Mar-21	16:00	20:00
24-Mar-21	12:00	16:00
25-Mar-21	09:00	09:40
25-Mar-21	12:00	16:00
25-Mar-21	10:00	15:00
06-Apr-21	09:00	14:15
07-Apr-21	08:00	09:00
07-Apr-21	09:00	10:00
07-Apr-21	10:00	15:00
08-Apr-21	12:00	17:00
19-Apr-21	13:00	17:00
20-Apr-21	09:00	13:00
20-Apr-21	14:00	18:00
22-Apr-21	08:00	09:00
22-Apr-21	09:00	10:00
22-Apr-21	12:00	16:00
24-Apr-21	12:00	17:00
27-Apr-21	09:00	13:00
27-Apr-21	15:00	19:00
29-Apr-21	08:00	09:00
29-Apr-21	11:00	16:00
30-Apr-21	14:00	18:00
03-May-21	13:00	18:00
04-May-21	09:00	12:00
05-May-21	13:00	18:00
06-May-21	12:00	15:00
07-May-21	12:00	17:00

08-May-21	12:00	14:00
10-May-21	14:00	19:00
11-May-21	09:00	09:40
12-May-21	14:00	19:00
13-May-21	08:00	09:00
14-May-21	12:00	16:00
15-May-21	12:00	14:00
18-May-21	10:00	11:00
18-May-21	11:00	22:00
19-May-21	07:00	18:00

Timelog for Andreas Kilde Lien

Logg

Meeting with Telenor to get an insight into Cisco WebEx and Meraki. As well as discussion with the group ab
Meeting with Telenor to get an insight into Microsoft Teams. As well as discussion with the group about furt
Meeting with Ernst. Talked about the plan ahead and arrange a meeting with Telenor and Erns together to p
Tom gives intro courses. Talk about this in the group after.
First group meeting to start the project plan
Worked with project plan. Working with development model
Meeting with Ernst. Talk to Ernst and Telenor
Discuss with team member about project plan
Work with project plan
Group work with project plan
Meeting with Ernst
Work with project plan
Work with report and plan for the future
Meeting with Ernst
Meeting with Telenor
Telenor meeting with information about Webex and Meraki+F34
Work with project plan
Work with project plan
Work with project plan
Work with report
Work with report and talk with Ernst
Meeting with Ernst
Telenor meeting
Working on Meraki Insigth
Work on test scenarios
Working on Meraki Insigth
Group work with rapport
Meeting with Ernst
Telenor meeting about the project
Work with the report
Work with the report
Work with the report
Work with the report
Work with the report
Working on Meraki Insigth
Work on test scenarios
Meeting with Ernst
Telenor meeting about the project
Work with the report
Work on test scenarios
Work on test scenarios
Work with the report
Work with the report and test scenario
Work with the report

Work on test scenarios
Meeting with Ernst
Telenor meeting about the project
Work on test scenarios
Work on test scenarios
Work on test scenarios
Work on test scenarios
Work on test scenarios
Work on test scenarios
Work on test scenarios
Work on test scenarios
Work with the report and test scenario
Work with the report
Work with the report
Work with report
Meeting with Ernst
Telenor meeting about the project
Work with the report
Work with the report
Work with the report
Work with report
Work with report
Work with report
Telenor meeting about the project
Work with report
Work with report
Work with report
Meeting with Ernst
Telenor meeting about the project
Work with report
Work with report
Work with report
Work with report
Work with report
Meeting with Ernst
Telenor meeting about the project
Work with report
Work with report
Work with report
Work with report
Meeting with Ernst
Work with report
Work with report
Work with report
Work with report
Work with report
Work with report
Work with report

Work with report
Work with report
Work with report
Work with report
Meeting with Ernst
Work with report
Work with report
Meeting with Ernst
Work with report
Work with report

Category
Seminar
Seminar
supervisor meeting talk about project and report
Intro kurs
Group meeting

Veiledertime
Group meeting
Project plan
Project plan
supervisor meeting
Group meeting
group meeting
supervisor meeting talk about project plan
Telenor meeting
Meeting with Telenor
Project plan
Project plan
Project plan
Group meeting
Report
supervisor meeting talk about project plan
Telenor meeting
Meraki
Group work
Meraki
Repport
supervisor meeting talk about project and report
Telenor meeting
Report
Report
Report
Report
Group work
Meraki
Test scenario
supervisor meeting talk about project and report
Telenor meeting
Report
Test scenario
Test scenario
Report
Report & test scenario
Group work

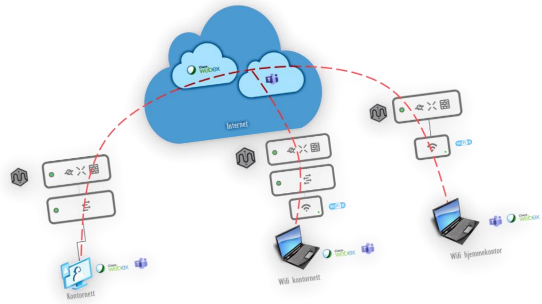
Test scenario
supervisor meeting talk about project and report
Telenor meeting
Test scenario
Test scenario
Test scenario
Test scenario
Test scenario
Test scenario
Test scenario
Test scenario
Report & test scenario
Report
Report
Group meeting
supervisor meeting talk about project and report
Telenor meeting
Report
Report
Report
Group meeting
Report
Report
Telenor meeting
Report
Report
Group meeting
supervisor meeting talk about project and report
Telenor meeting
Report
Report
Report
Group meeting
Report
supervisor meeting talk about project and report
Telenor meeting
Report
Report
Group meeting
Report
supervisor meeting talk about project and report
Report
Report
Report
Group meeting
Report
Report
Report

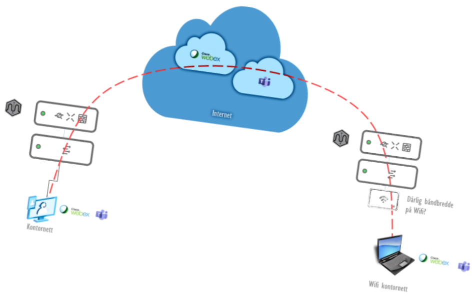
Report
Report
Group meeting
Report
supervisor meeting talk about project and report
Report
Report
supervisor meeting talk about project and report
Report
Report

A.5 Original test scenarios

Testscenarioer

MS Teams og Webex Teams





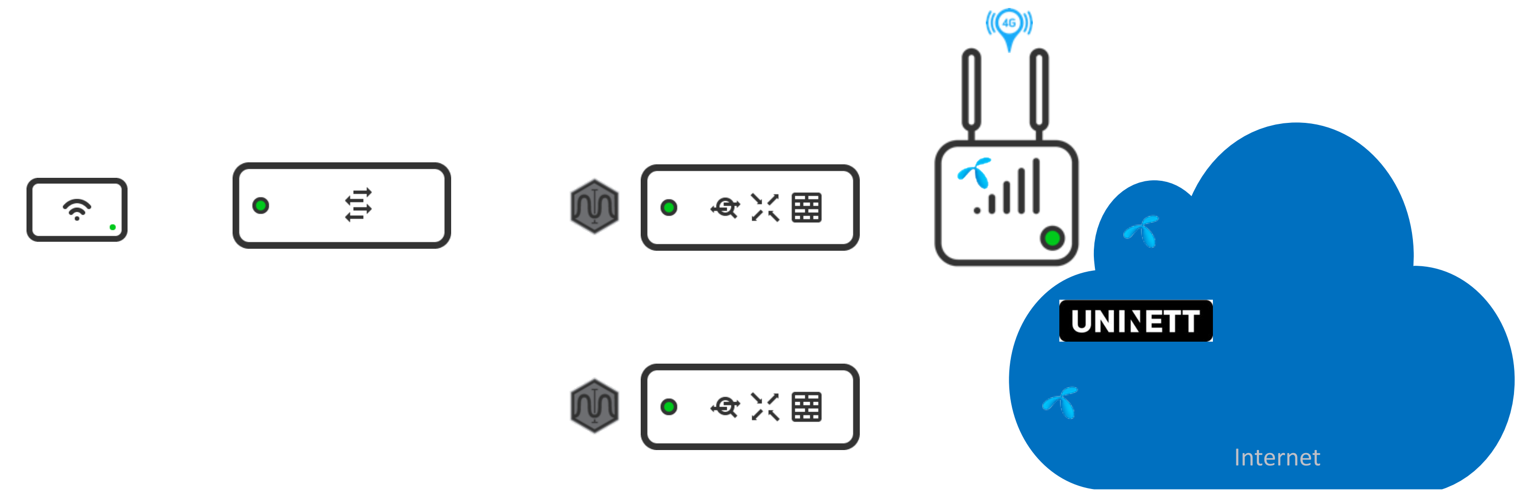
Meraki utstyr som sendes til Gjøvik

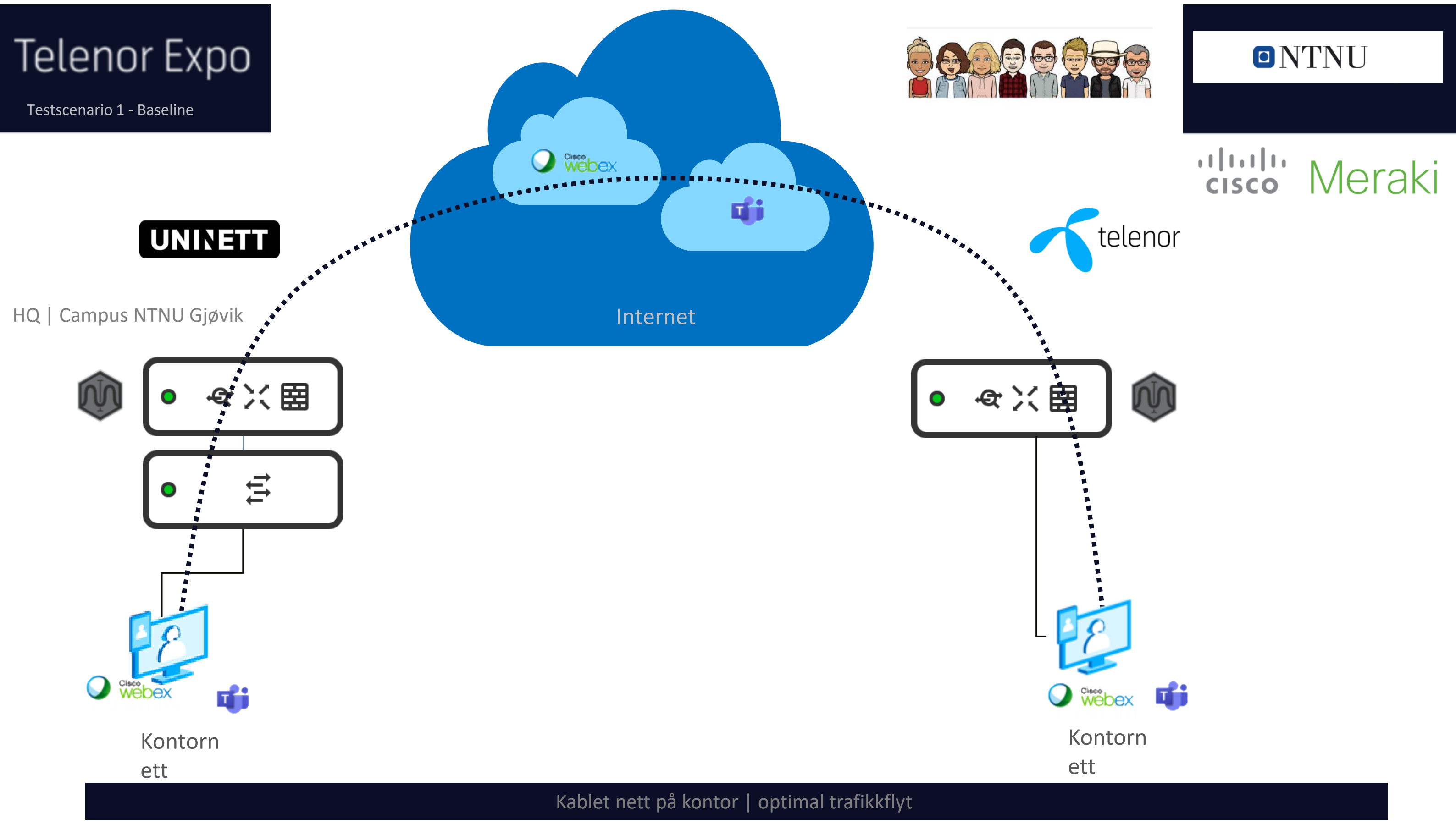
1 stk MR-36

1 stk MS-120 med PoE

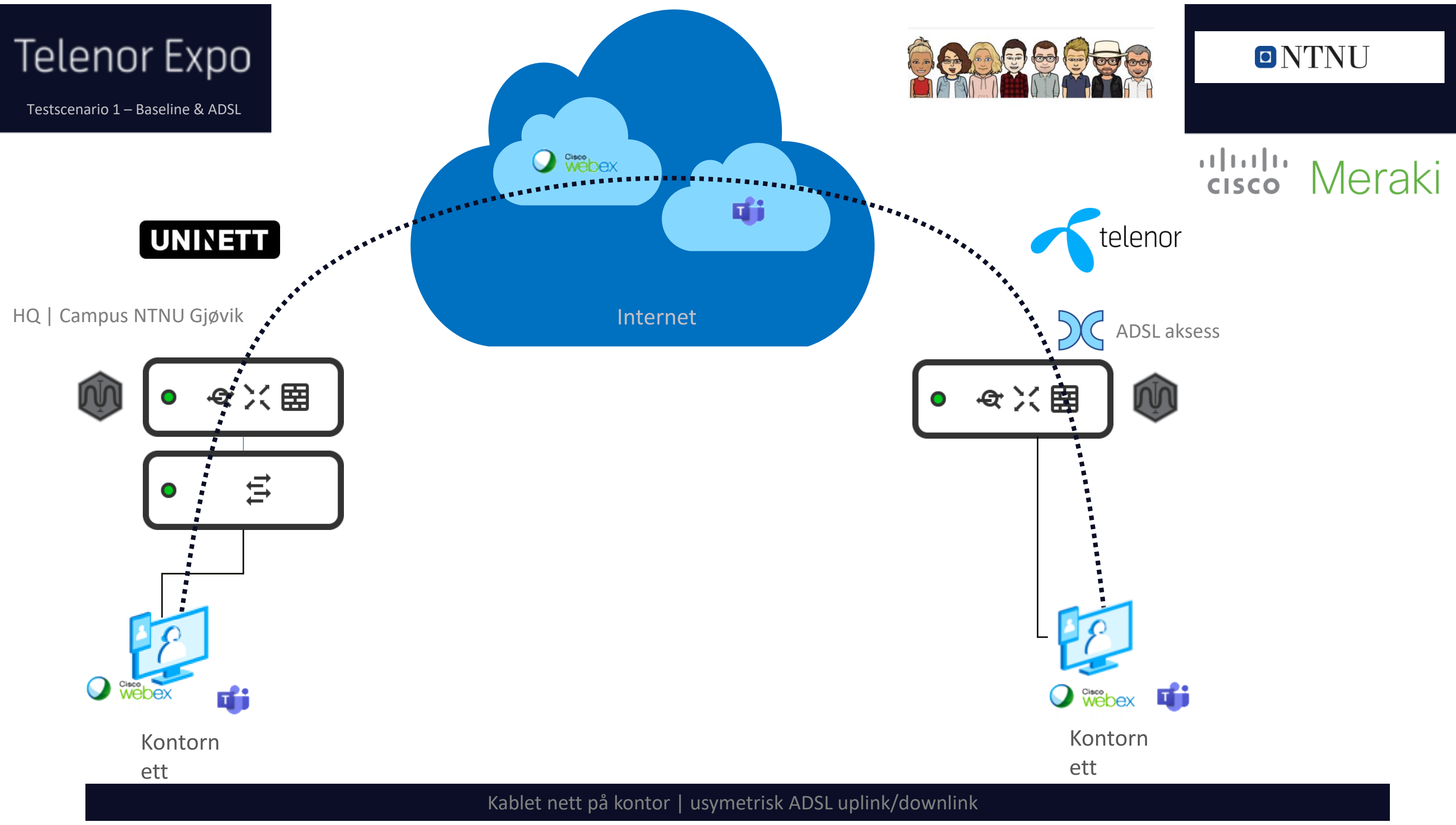
2 stk MX-68 med Insight

1 stk MG-21E

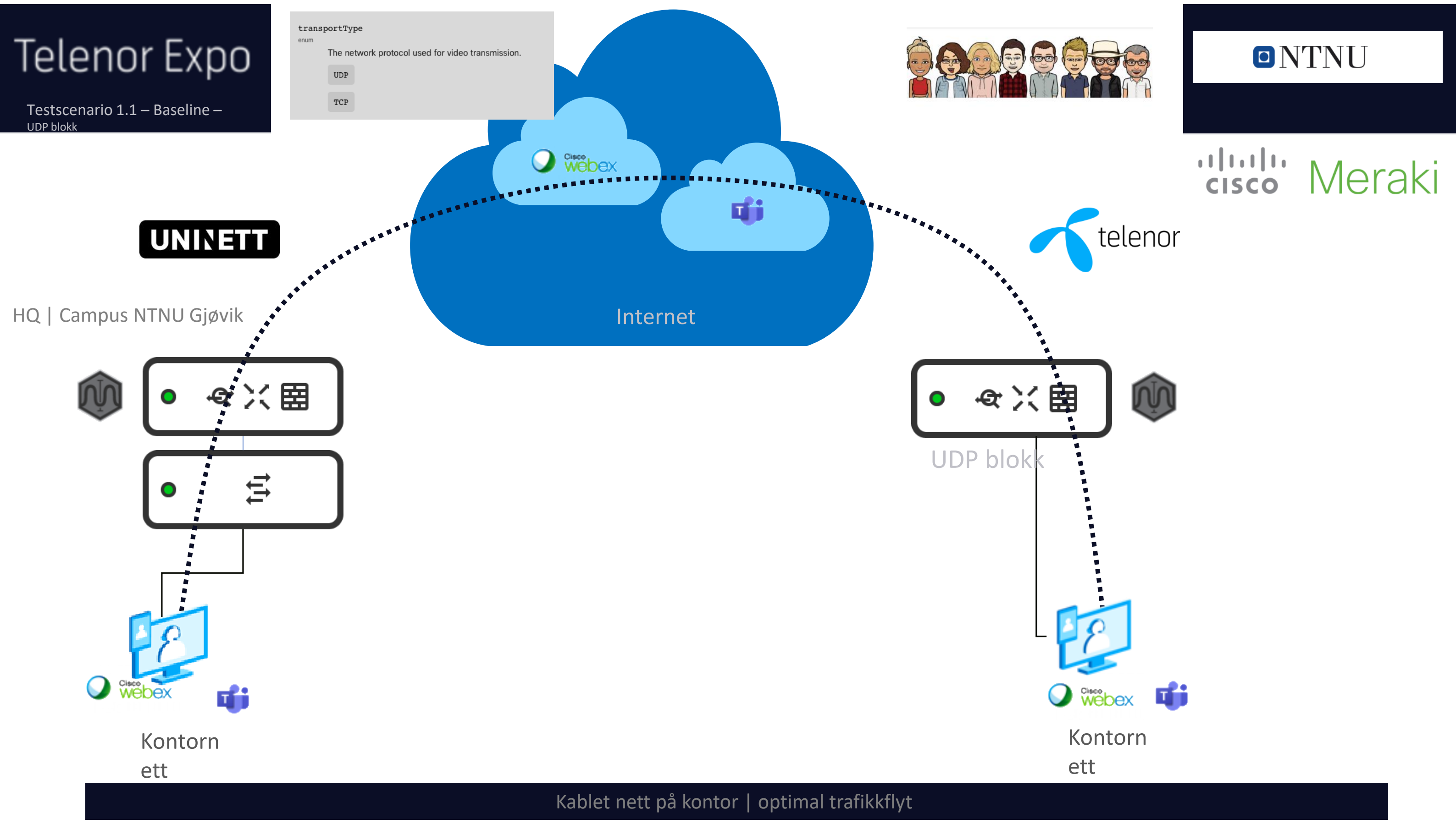




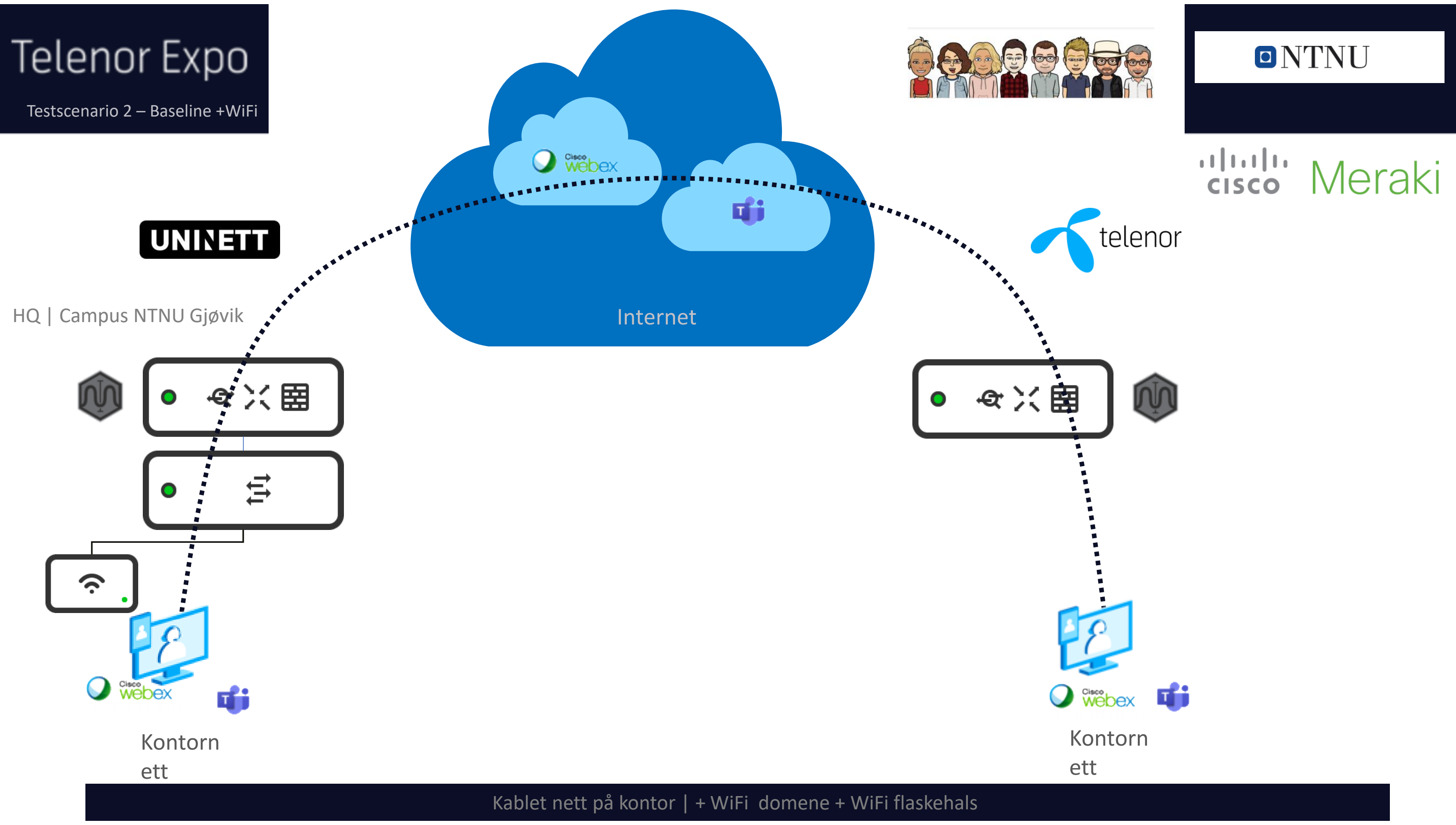
Telenor Expo
Testscenario 1 - Baseline
NTNU
cisco
Meraki
UNINETT
telenor
Internet
HQ | Campus NTNU Gjøvik
Kontornett
Kontornett
Kablet nett på kontor | optimal trafikkflyt

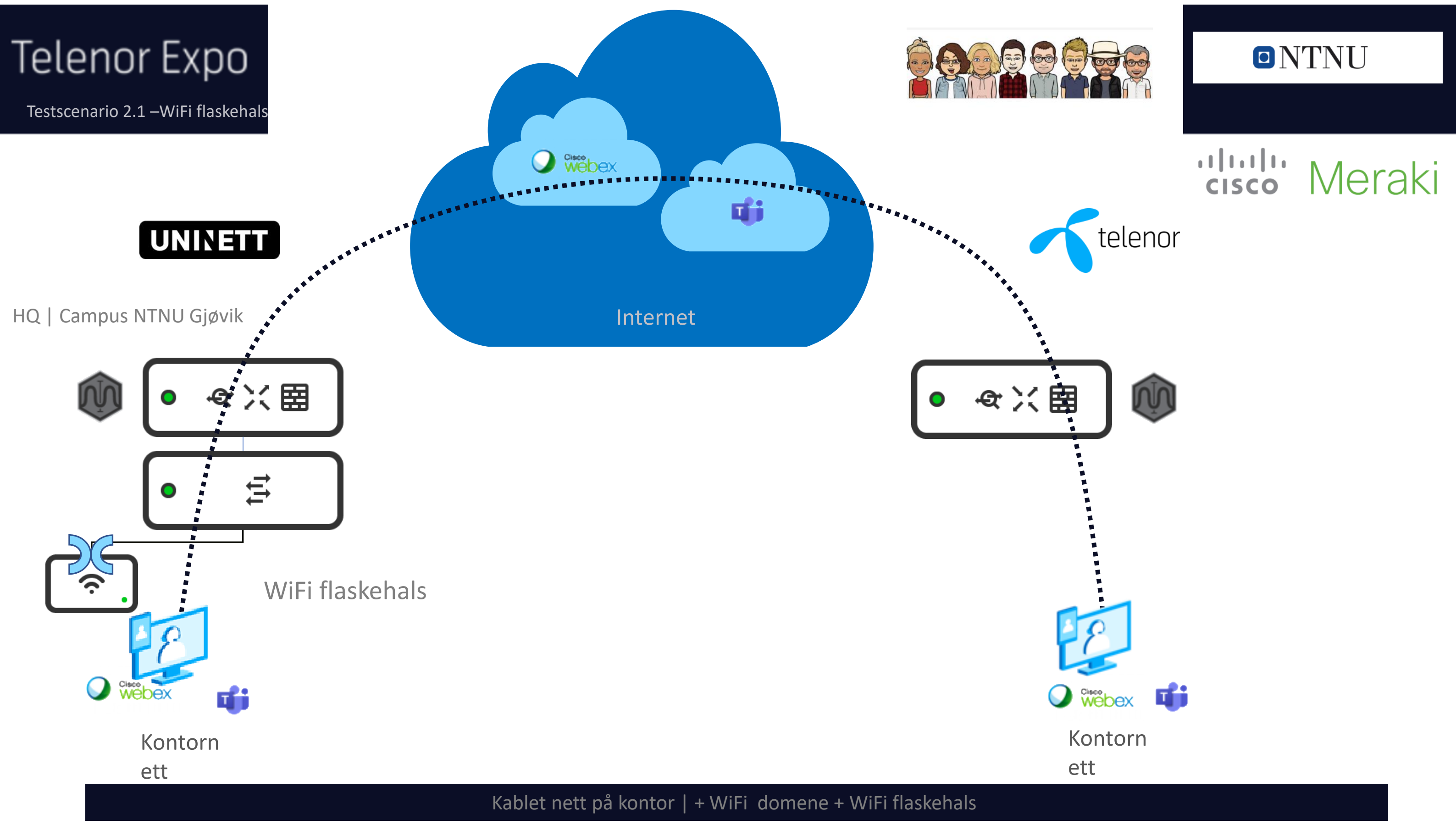


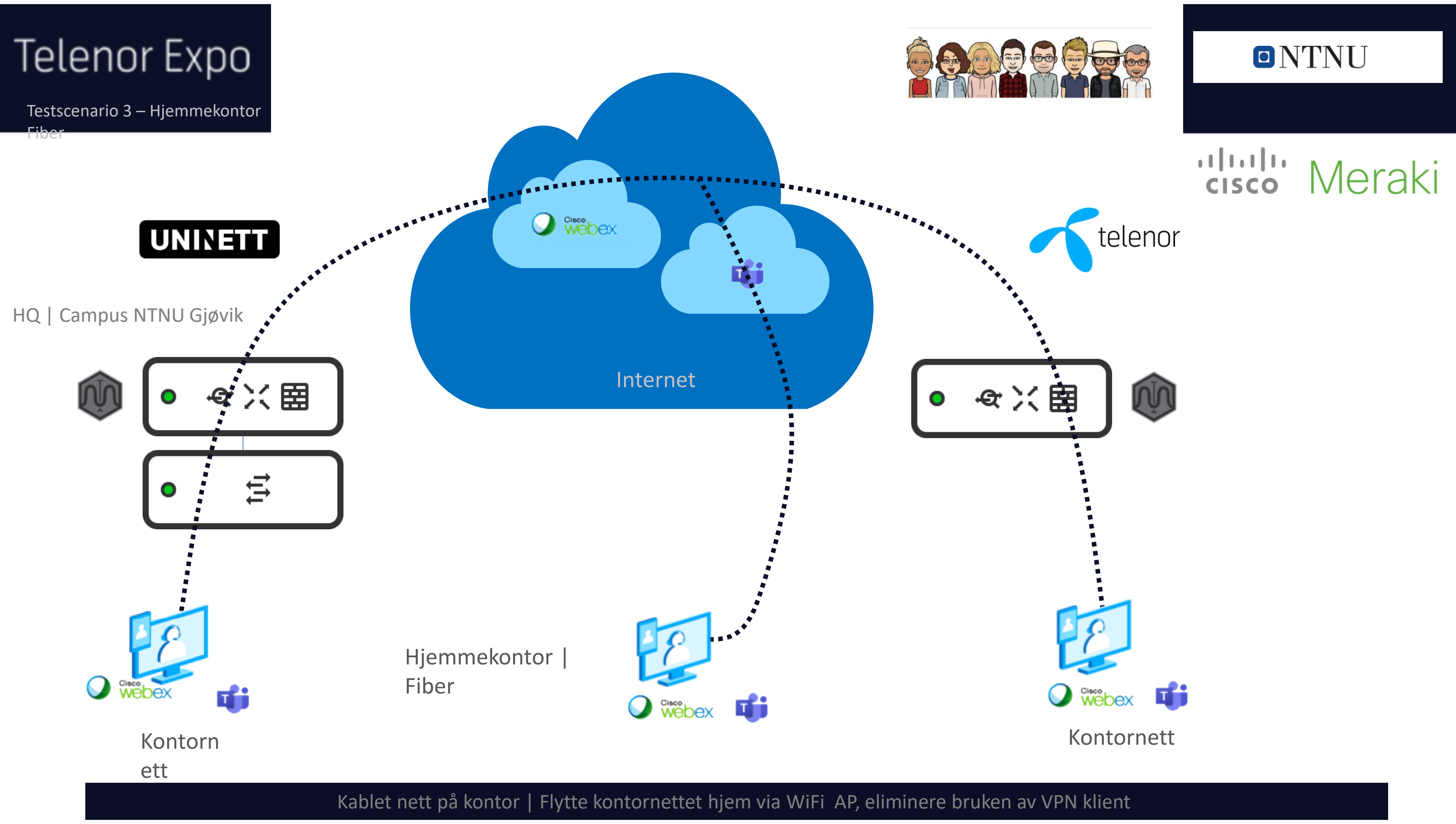
Telenor Expo
Testscenario 1 – Baseline & ADSL
NTNU
cisco
Meraki
telenor
UNINETT
Internet
HQ | Campus NTNU Gjøvik
ADSL aksess
Kontorn
ett
Kontorn
ett
Kablet nett på kontor | usymetrisk ADSL uplink/downlink

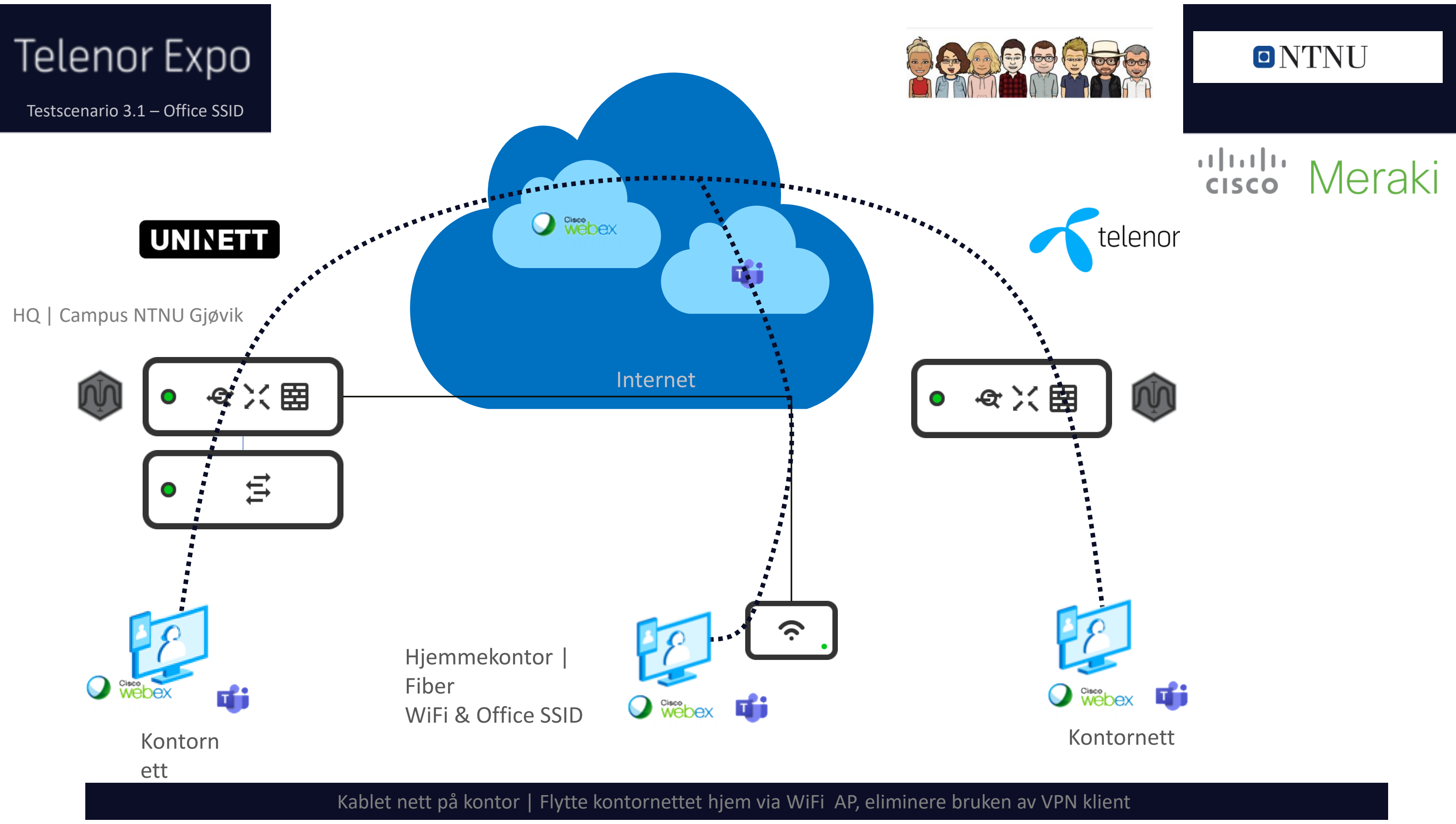


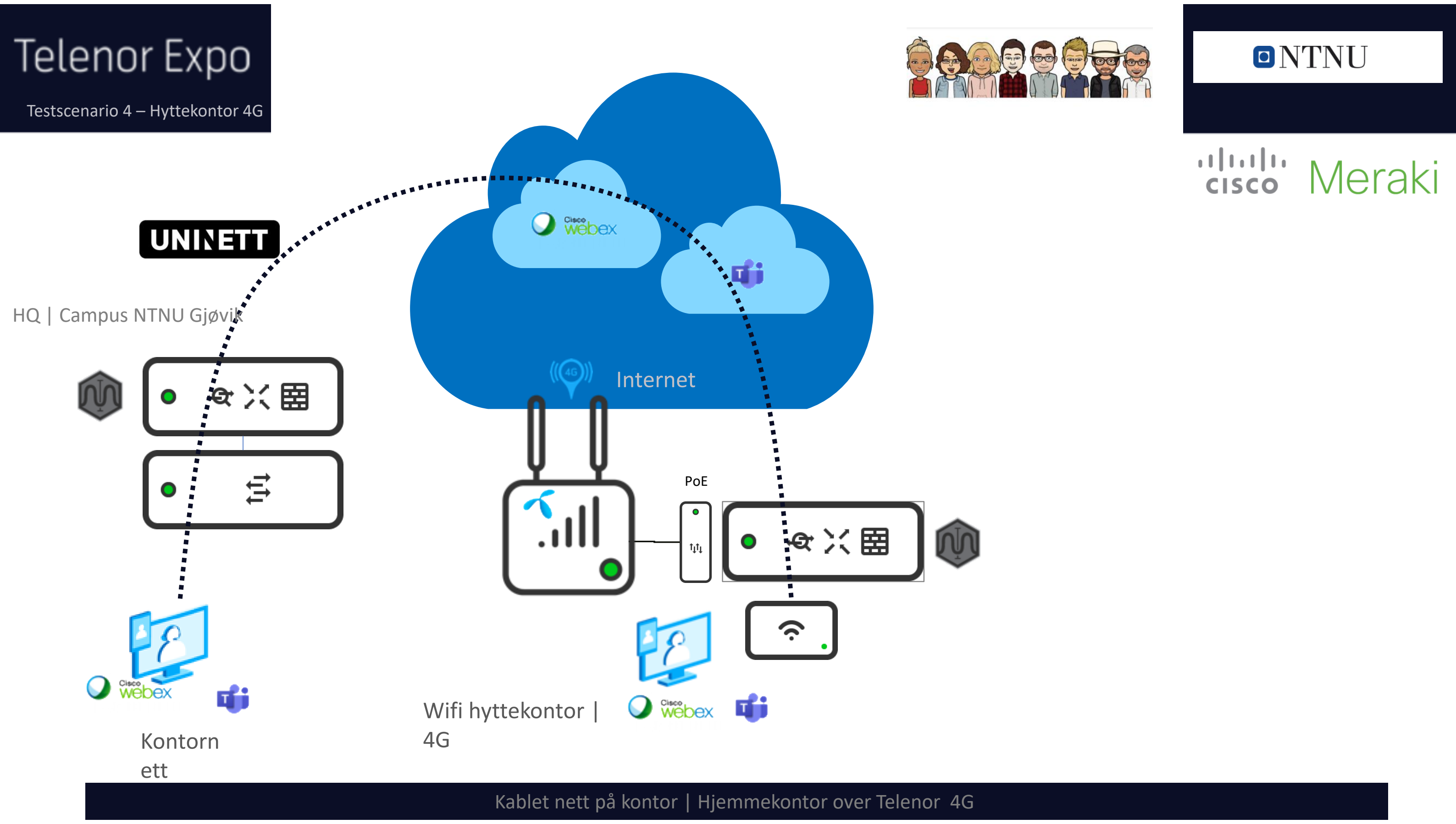
Telenor Expo
Testscenario 1.1 – Baseline –
UDP blokk
transportType
enum
The network protocol used for video transmission.
UDP
TCP
Cisco webex
Internet
NTNU
cisco
Meraki
UNINETT
telenor
HQ | Campus NTNU Gjøvik
UDP blokk
Cisco webex
Kontorn
ett
Cisco webex
Kontorn
ett
Kablet nett på kontor | optimal trafikkflyt

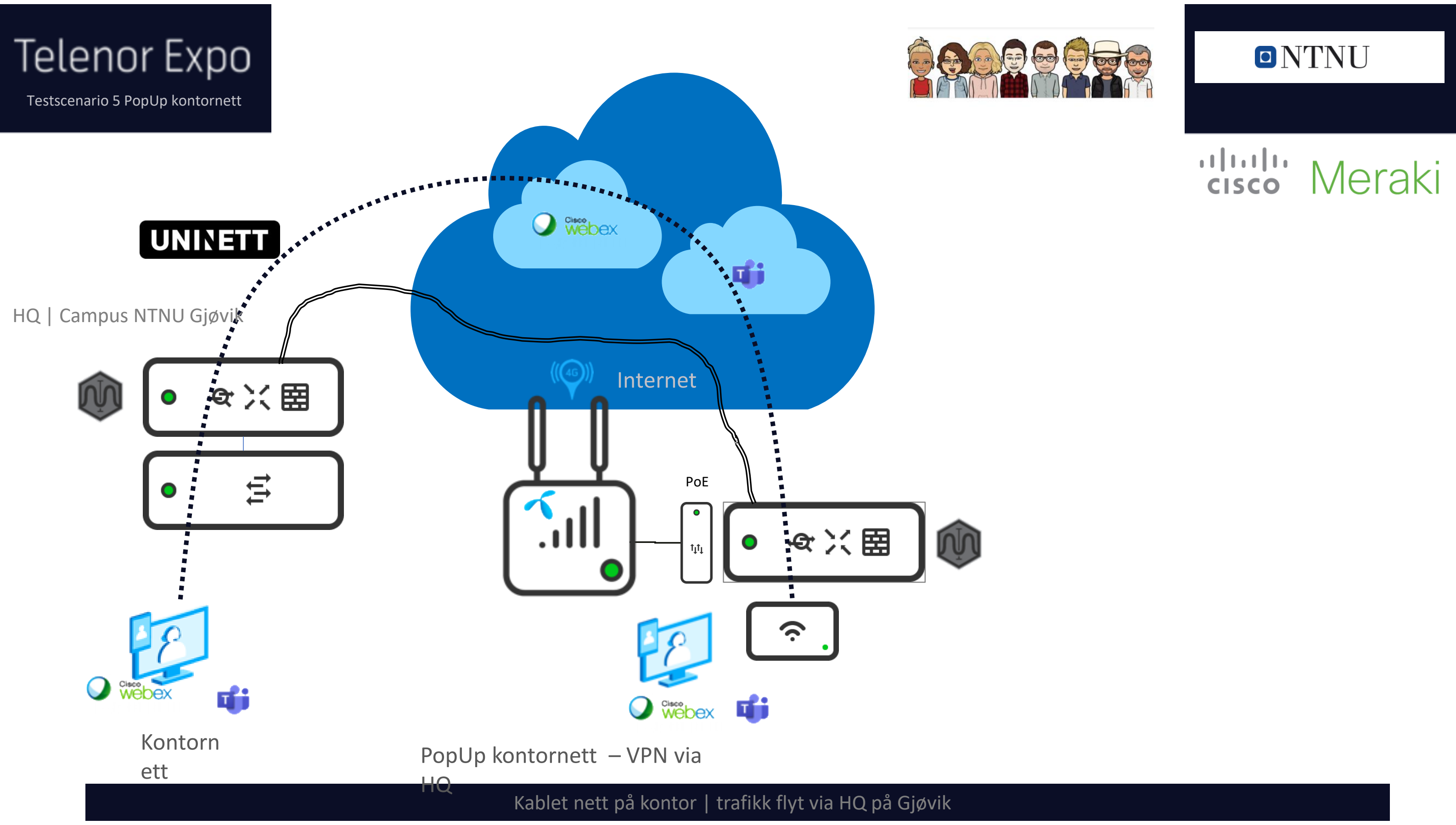

Telenor Expo
Testscenario 2 – Baseline +WiFi
Internet
NTNU
cisco
Meraki
UNINETT
telenor
HQ | Campus NTNU Gjøvik
Kontorn
ett
Kontorn
ett
Kablet nett på kontor | + WiFi domene + WiFi flaskehals


Telenor Expo
Testscenario 2.1 –WiFi flaskehals
NTNU
cisco Meraki
UNINETT
telenor
Cisco webex
Internet
HQ | Campus NTNU Gjøvik
WiFi flaskehals
Kontorn ett
Kontorn ett
Kablet nett på kontor | + WiFi domene + WiFi flaskehals


Telenor Expo
Testscenario 3 – Hjemmekontor Fiber
NTNU
cisco Meraki
telenor
UNINETT
HQ | Campus NTNU Gjøvik
Internet
Cisco webex
Hjemmekontor | Fiber
Kontornett
Kontornett
Kablet nett på kontor | Flytte kontornettet hjem via WiFi AP, eliminere bruken av VPN klient


Telenor Expo
Testscenario 3.1 – Office SSID
NTNU
cisco Meraki
UNINETT
telenor
HQ | Campus NTNU Gjøvik
Cisco webex
Internet
Hjemmekontor | Fiber
WiFi & Office SSID
Kontornett
Kontornett
Kablet nett på kontor | Flytte kontornettet hjem via WiFi AP, eliminere bruken av VPN klient


Telenor Expo
Testscenario 4 – Hyttekontor 4G
NTNU
cisco Meraki
UNINETT
HQ | Campus NTNU Gjøvik
Cisco webex
Internet
4G
PoE
Cisco webex
Kontornett
Wifi hyttekontor | 4G
Cisco webex
Kablet nett på kontor | Hjemmekontor over Telenor 4G


Telenor Expo
Testscenario 5 PopUp kontornett
NTNU
cisco
Meraki
UNINETT
HQ | Campus NTNU Gjøvik
Cisco webex
Internet
4G
PoE
Kontorn
ett
PopUp kontornett – VPN via
HQ
Kablet nett på kontor | trafikk flyt via HQ på Gjøvik

A.6 Scripts

Code listing A.1: Bash file to clean ntopng data

```
#!/bin/bash
systemctl stop ntopng
redis-cli FLUSHALL
redis-cli FLUSHDB
if [[ -d /var/tmp/ntopng]]; then
    rm -rf /var/tmp/ntopng
    systemctl restart ntopng
else
    echo "Folder␣doesn’t␣exist."
fi
```

Code listing A.2: Bash file to merge wireshark files

```
#!/bin/bash

# remove spaces all subfolders
./removeSpaces.sh

# get all relevant filenames
find . -name "Studentnett_-_*" | grep -v "_del" | grep -v "pcapDone" > paths.txt

# Go through all the files and merge with eachother. It is expected that
#the files are in order where the first and second file found is to be merged.
# The third and fourth to be merged, and so on...
exec 2< paths.txt
while read line1 <&2; do
        read line2 <&2
        # remove folder path
        newName=$(echo "$line1" | sed "s/.*\///")
        echo $newName
        ./mergeFiles.sh $line1 $line2 "pcapDone-${newName}"
done
exec 2<&-
```

Code listing A.3: Bash file to remove all spaces in subfolders and sub-files

```
#!/bin/bash
find -name "*␣*" -print0 | sort -rz | \
while read -d $'\0' f; do mv -v "$f" "$(dirname␣"$f")/$(basename␣"${f// /_}")"; done
```

Code listing A.4: Merges two wireshark files and gives a new name

```
#!/bin/bash
# this script does not check for the pcap ending. It is expected to give a name,
# not the name and the suffix.
if [[ "$#" -ne 3 ]]; then
    echo "Three␣arguments␣needed:␣mergeFiles␣<firstPcapFile>␣<secondPcapFile>␣<newFileName>"
else
    echo "Merging␣files.."
    firstPcap=$1
    secondPcap=$2
    newName=$3
    tempName="${newName}temp.pcap"
    if [[ $(mergecap ${firstPcap} ${secondPcap} -w ${tempName}) -eq 0 \
    && $(editcap -d ${tempName} ${newName}.pcap ) -eq 0 ]];then
        rm $tempName
        echo "Files␣merged␣and␣duplicate␣packets␣removed.␣New␣filename␣is␣${newName}"
    else
        echo "Something␣went␣wrong"
    fi
fi
```

Code listing A.5: Starts Wireshark for ethernet and wireless

```
param(
    [string]$wName, #wireless name
    [string]$eName  #ethernet name
)
if([string]::IsNullOrEmpty($wName) -and [string]::IsNullOrEmpty($eName)){
    echo "Usage: .\start-wireless-test.ps1 -wName <wireless-file-name.pcap> -eName <ethernet-file-name.pcap>"
} else {
    Wireshark -i 4 -a duration:5 -w $eName -k
    Wireshark -i 7 -a duration:5 -w $wName -k
}
```

Code listing A.6: The confirmation file of HPE Procurve 2810-48g

```
Running configuration:

; J9022A Configuration Editor; Created on release #N.11.15

hostname "Uninett"
mirror-port 2
interface 3
   name "PC-2"
exit
ip default-gateway 192.168.128.1
snmp-server community "public" Unrestricted
vlan 1
   name "DEFAULT_VLAN"
   untagged 1-48
   ip address 192.168.128.240 255.255.255.0
   exit
interface 1,3
   monitor
   exit
password manager
password operator
```

Code listing A.7: The confirmation file of ntopng (/etc/ntopng.conf)

```
 This configuration file is similar to the command line, with the exception
# that an equal sign '=' must be used between key and value. Example: -i=p1p2
# or --interface=p1p2 For options with no value (e.g. -v) the equal is also
# necessary. Example: "-v=" must be used.
#
# DO NOT REMOVE the following option, required for daemonization.
-e=
-i=eth0
--local-networks="192.168.128.0/24"
--packet-filter="not␣(host␣192.168.128.4)"
# * Interfaces to sniff on: one interface per line, prefix with -i=
# E.g.
#-i=eth0
#-i=wlan0
# If none is specified, ntopng will try to auto-detect the best interface.
#
# * Port on which ntopng will listen for the web-UI.
-w=3000
```

Code listing A.8: Logs signal strength received by an AP

```
param(
    [string]$fileName, # file name to save data to
    [int]$time = 240 # timespan to log value
)
$date = Get-Date -Format "hh-mm-ss"


if([string]::IsNullOrEmpty($fileName)){
    echo "Filename empty. Usage: .\log-signalStrength -f <filename> -t <time>"
} elseif (Test-Path -Path .\$fileName$date.dat){
    echo "Filename already exists."
} else {
    $signalPercent = ((netsh wlan show interfaces) -Match '^\s+Signal' -Replace '^\s+Signal\s+:\s+','' -replac
    $signal = ($signalPercent[0]/2) - 100 # calculate signal strength in dBm
    [int] $counter = 0;
    # run through loop in $time seconds
    do {
        $counter++
        echo $signal | Add-Content .\$fileName$date.dat
        sleep -s 1
    } while ($time -gt $counter)
}
```

Code listing A.9: Python script to extract signal quality from 4G router

```
#!/usr/bin/python
# -*- coding: utf-8 -*-

import requests
import datetime
import time
import json

IP_OF_MG = '%IP_OF_MG%'


def getData(url):
    r = requests.get(url)
    if r.status_code == 200:
        data = r.json()
        return data['connection_state']['cellular_tower_uplink_status'] \
            or ''

    return ''


def max_seconds(max_seconds, interval=1):
    interval = int(interval)
    start_time = time.time()
    end_time = start_time + max_seconds
    yield 0
    while time.time() < end_time:
        if interval > 0:
            next_time = start_time
            while next_time < time.time():
                next_time += interval
            time.sleep(int(round(next_time - time.time())))
        yield int(round(time.time() - start_time))
        if int(round(time.time() + interval)) > int(round(end_time)):
            return


def main(args):
    outputData = []

    for sec in max_seconds(245, interval=15):
        try:
            r = getData('http://{}/index.json'.format(IP_OF_MG))
            outputData.append({'time': datetime.datetime.now().strftim
            ('%d-%m-%Y␣%H:%M:%S'), 'status': r})
        except:
            continue

    with open('cellular_data_{}.json'.format(datetime.datetime.now().strftime(
        '%d-%m-%Y-%H:%M:%S' )), 'w') as fOut:
        json.dump(outputData, fOut)
    return 0


if __name__ == '__main__':
    import sys
    sys.exit(main(sys.argv))
```

A.7 Data

Barneparken	Media	Stream Direction	Transport	Avg RTT	Avg Jitter	Avg Packet Loss	Classified Poor Call	Avg Video Frame Rate	Avg Healer FEC	Avg Healer Packet Drop Ratio
PC-1	Audio	First-To-Second	UDP	26	1	0	False		0	0
PC-1	Audio	Second-To-First	UDP	26	1	0	False		0	0
PC-2	Audio	First-To-Second	UDP	36	1	0	False		0	0
PC-2	Audio	Second-To-First	UDP	36	0	0	False		0	0
PC-1	Video	First-To-Second	UDP	26	0	0	False	29,5 - 30 FPS	NaN	0
PC-1	Video	Second-To-First	UDP	26	0	0	False	29,5 - 30 FPS	NaN	0
PC-2	Video	First-To-Second	UDP	36	0	0	False	29,5 - 30 FPS	NaN	0
PC-2	Video	Second-To-First	UDP	36	0	0	False	29,5 - 30 FPS	NaN	0

Table A.1: Data from a teams meeting on barneparken using power BI.

Skytjenester	Media	Stream Direction	Transport	Avg RTT	Avg Jitter	Avg Packet Loss	Classified Poor Call	Avg Video Framerate	Avg Healer FEC	Avg Healer Packet Drop Ratio
PC-1	Audio	First-To-Second	UDP	40	1	0	False		0	0
PC-1	Audio	Second-To-First	UDP	40	1	0	False		0	0
PC-2	Audio	First-To-Second	UDP	50	1	0	False		0	0
PC-2	Audio	Second-To-First	UDP	50	0	0	False		0	0
PC-1	Video	First-To-Second	UDP	42	0	0	False	29,5 - 30 FPS	NaN	NaN
PC-1	Video	Second-To-First	UDP	42	0	0	False	14,5 - 15 FPS	NaN	NaN
PC-2	Video	First-To-Second	UDP	50	0	0	False	14,5 - 15 FPS	NaN	NaN
PC-2	Video	Second-To-First	UDP	50	0	0	False	29,5 - 30 FPS	NaN	NaN

Table A.2: Data extracted from tenant skytjenester using power BI.

Client	Media	Stream direction	Transport	Avg RTT	Avg Packet loss	Avg Jitter	Classified poor call	Avg Frame Rate	Avg Healer FEC	Avg Healer Packet Drop Ratio
PC-1	Audio	First-to-Second	UDP	41	0	1	False		0	0
PC-1	Audio	Second-to-First	UDP	41	0	2	False		0	0
PC-2	Audio	First-to-Second	UDP	55	0	1	False		0	0
PC-2	Audio	Second-to-First	UDP	52	0	0	False		0	0
PC-1	Video	First-to-Second	UDP	98	0	0	False	29.5-30	NaN	NaN
PC-1	Video	Second-to-First	UDP	63	0.01	0	False	29.5-30	NaN	NaN
PC-2	Video	First-to-Second	UDP	52	0	0	False	29.5-30	NaN	NaN
PC-2	Video	Second-to-First	UDP	52	0	0	False	29.5-30	NaN	NaN

Table A.3: Data from a teams testscenario 1.2 on barneparken using power BI.

Client	Protocol	Sent	Recieved	Total
PC-1	TCP	250.15 KB	124.9 KB	365.05 KB(0.41%)
	UDP	52.45 MB	23.55 MB	86 MB(99.59%)
PC-2	TCP	413.45 KB	1.06 KB	1.46 MB(1.8%)
	UDP	23.47 MB	56.14 MB	76.61 MB(98.2%)

Table A.4: Protocols used during test scenario 1.2, for barneparken, from ntopng.

Client	Media	Stream Direction	Transport	Avg RTT	Avg Jitter	Avg Packet Loss	Classified Poor Call	Avg Video Framerate	Avg Healer FEC	Avg Healer Packet Drop Ratio
PC-1	Audio	First-To-Second	UDP	25	3	0	False		0	0
PC-1	Audio	Second-To-First	UDP	25	1	0	False		0	0
PC-2	Audio	First-To-Second	UDP	34	3	0	False		0	0
PC-2	Audio	Second-To-First	UDP	34	1	0	False		0	0
PC-1	Video	First-To-Second	UDP	37	0	0	False	29.5 - 30 FPS	NaN	NaN
PC-1	Video	Second-To-First	UDP	37	0	0.01 (1%)	False	29.5 - 30 FPS	NaN	NaN
PC-2	Video	First-To-Second	UDP	35	0	0	False	29.5 - 30 FPS	NaN	NaN
PC-2	Video	Second-To-First	UDP	35	0	0	False	29.5 - 30 FPS	NaN	NaN

Table A.5: Stuff

Client	Protocol	Sent	Received	Total
PC-1	TCP	358.18 KB	1.02 MB	1.37 MB (1.58%)
	UDP	61.71 MB	23.76 MB	85.47 MB (98.42%)
PC-2	TCP	383.57 KB	102.19 KB	385.76 KB (0.49%)
	UDP	22.33 MB	54.16 MB	76.49 MB (99.51%)

Table A.6: Protocols used during test scenario 1.2, for skytjenester, from ntopng.

A.8 Test scenario 1.2 - Webex

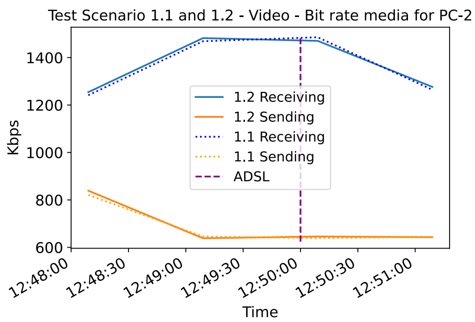


Figure A.1: Bit rate for PC-2. Data from Webex meeting: 1.1.6 and 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.

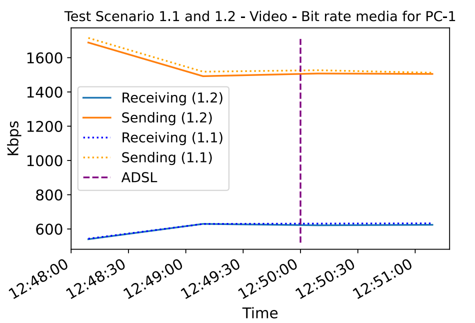
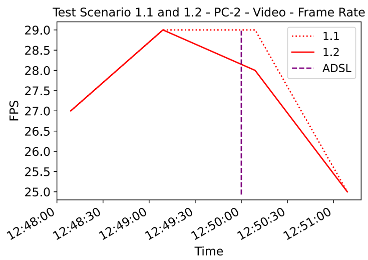
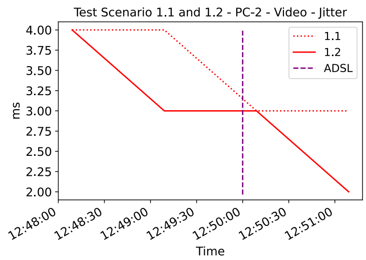


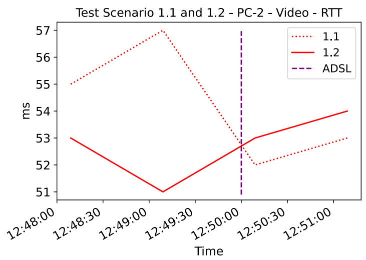
Figure A.2: Bit rate for PC-1. Data from Webex meeting: 1.1.6 and 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.



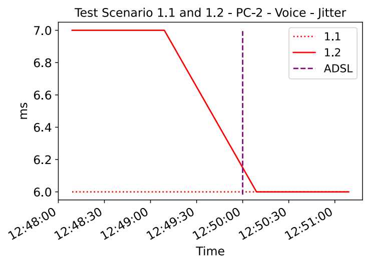
(a) Video - Frame Rate.



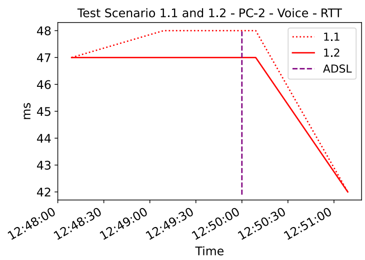
(b) Video - Jitter.



(c) Video - RTT.

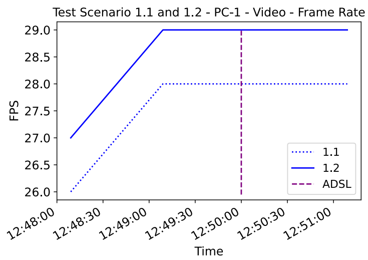


(d) Voice - Jitter.

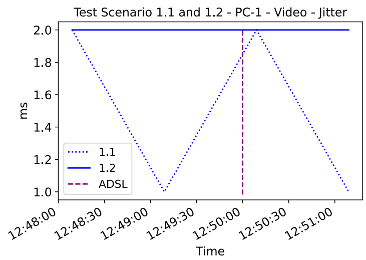


(e) Voice - RTT.

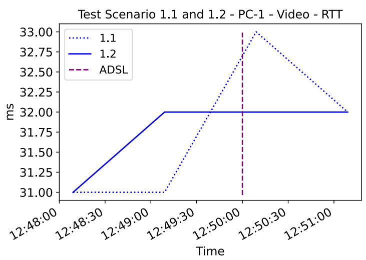
Figure A.3: Data from Webex meeting: 1.1.6 and 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.



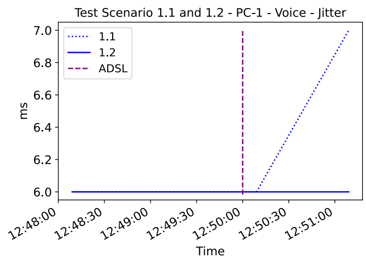
(a) Video - Frame Rate.



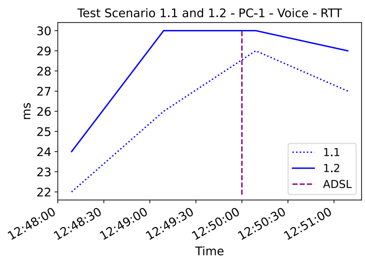
(b) Video - Jitter.



(c) Video - RTT.



(d) Voice - Jitter.



(e) Voice - RTT.

Figure A.4: Data from Webex meeting: 1.1.6 and 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.

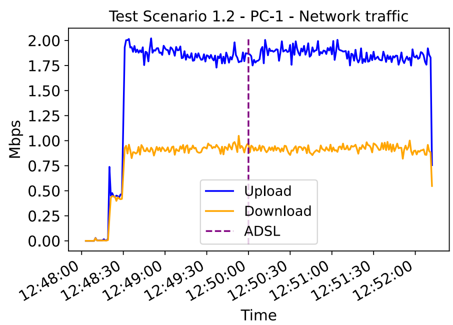


Figure A.5: Bandwidth usage for PC-1 of packets between PC-1 and Webex service using the UDP protocol. Data from Webex meeting: 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.

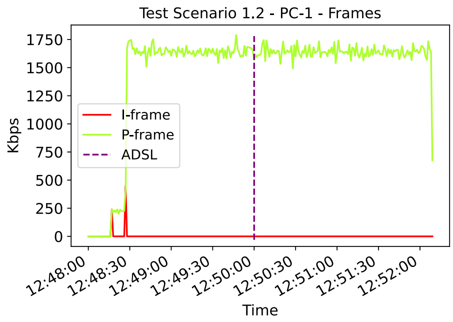


Figure A.6: The dispersion of I- and P-frames for PC-2. The graph is including L3 overhead. The pcap was missing the reverse video stream so, the graph has only the sent/upload stream. Data from Webex meeting: 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.

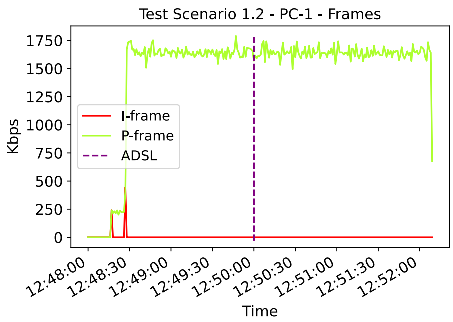


Figure A.7: The dispersion of I- and P-frames for PC-1. The graph is including L3 overhead. Data from Webex meeting: 1.2.5 [A.10]. License: Andreas Kilde Lien, CC BY.

A.9 Webex data for test scenario 2.1

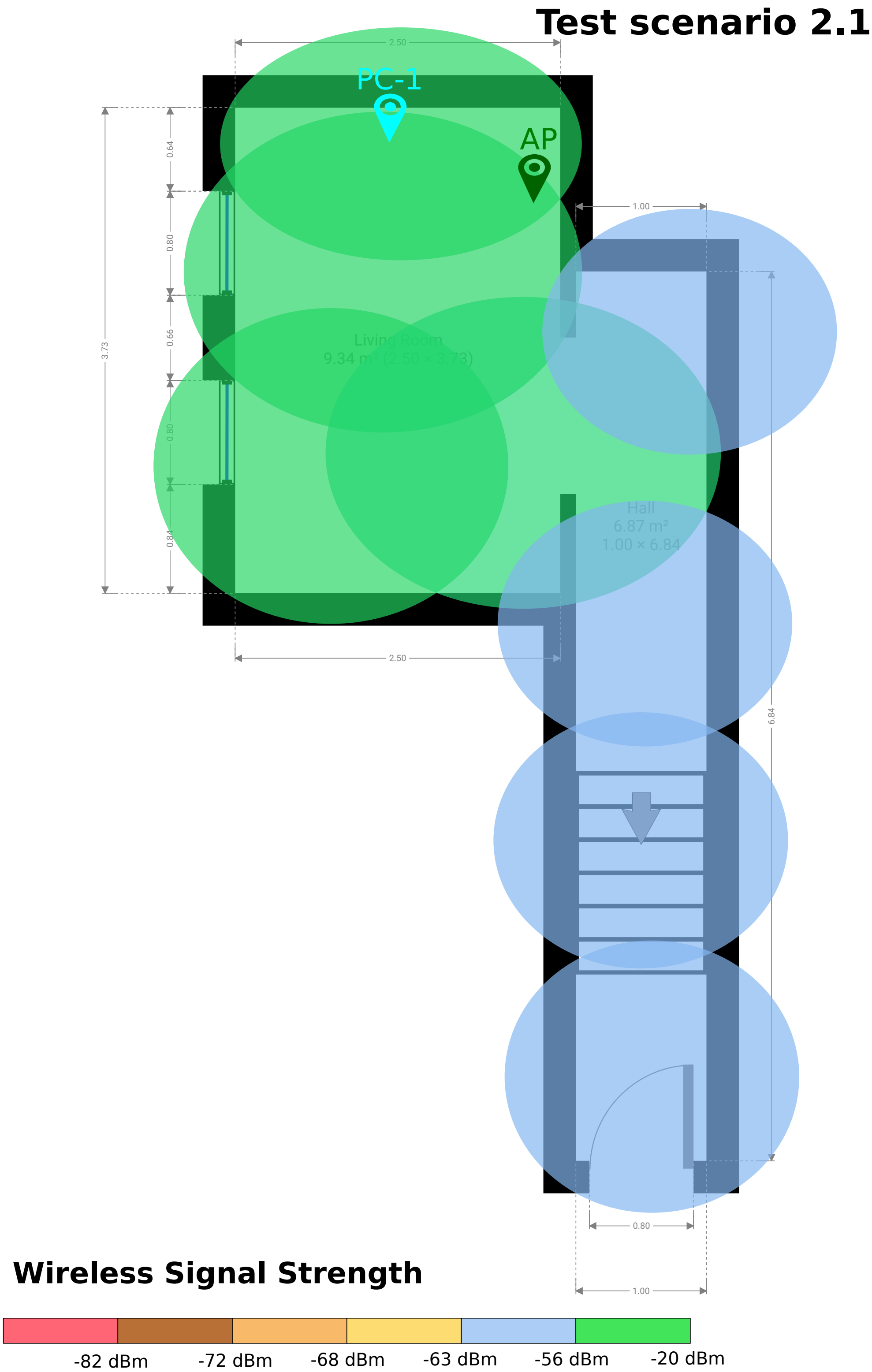


Figure A.8: Site survey of PC-1 location where lower values are worse and higher values are better. License: Andreas Kilde Lien, CC BY.

A.10 Teams Iterations and ID

Test iterations (Barneparken)	ID	Chosen		Test iterations (Skytjenester)	ID	Chosen
1.1.1	4ea22ce9			1.1.1	63f38510	
1.1.2	7dadf4			1.1.2	75d9d4de	
1.1.3	45ac46			1.1.3	fb4c1752	
1.1.4	798ada1a			1.1.4	4992f506	X
1.1.5	0a450bed	X		1.1.5	eaa656c3	
1.1.6	bbf16537			1.1.6	35de7daf	
1.2.1	b64d16e4			1.2.1	99bd81	
1.2.2	2e6d60cf			1.2.2	ff443645	
1.2.3	9fcc8ff7			1.2.3	84e67213	
1.2.4	fa632963			1.2.4	363356b0	X
1.2.5	5425f66b	X		1.2.5	e667ba6b	
1.3.1	fa768faf			1.3.1	504156a7	
1.3.2	da6d35ec			1.3.2	a17d790b	
1.3.3	bf5833f2			1.3.3	b834d2fb	
1.3.4	c02c0db	X		1.3.4	9c7567	X
1.3.5	4472b704			1.3.5	No data reg.	
2.1.1	71db20ca			2.1.1	9b090a88	Excluded
2.1.2	d32fbfda			2.1.2	8b20c777	Excluded
2.1.3	99bb831e			2.1.3	49dfb75a	Excluded
2.1.4	042e43a9	X		2.1.4	b3544f4e	Excluded
2.2.1	7f42bbf3			2.2.1	986e7b53	Excluded
2.2.2	0b17ec0b			2.2.2	eba711b9	Excluded
2.2.3	18ad750a	X		2.2.3	38ba6fc0	Excluded
3.1.1	876cfd86			3.1.1	9f20b33b	Excluded
3.1.2	310e2ae4			3.1.2	1295bf44	Excluded
3.1.3	08cfe088			3.1.3	c735ad1d	Excluded
3.1.4	ced2708a	X		3.1.4	No data reg.	Excluded
4.1.1	61e3f1fb	X		4.1.1	029ac831	Excluded
4.1.2	f60cdff5			4.1.2	60bfe88a	Excluded
4.1.3	86aa3197			4.1.3	53805b1d	Excluded

Table A.7: Test iterations and ID for barneparken and skytjenester

A.11 Webex Iterations and ID

Test iterations (Webex)	ID	Chosen
1.1.1	187391637777826648	
1.1.2	187687627399381170	
1.1.3	187688679190315285	
1.1.4	189023684646682850	
1.1.5	189026623923699896	
1.1.6	189027346881200161	X
1.2.1	187750904386044863	
1.2.2	187753604676862825	
1.2.3	187754310534185544	
1.2.4	189210058826990017	
1.2.5	189210790314656693	X
1.3.1	187779129509167189	
1.3.2	187781102374502736	
1.3.3	187782353868832856	
1.3.4	189211281895473209	X
1.3.5	189213483286414310	
2.1.1	187936565880701447	
2.1.2	187937495762092337	
2.1.3	187938357738750895	
2.1.4	188308663867294239	X
2.2.1	188062112708313767	
2.2.2	188112710910624560	
2.2.3	188114062858535533	X
3.1.1	188484443510423964	
3.1.2	188484985479512621	
3.1.3	188486251127521922	
3.1.4	193103242571313786	X
4.1.1	188410544937517440	X
4.1.2	188411372807143939	
4.1.3	188412866490024617	

Table A.8: Test iterations and ID for Webex

A.12 Rating - Webex

Webex (Hoyskolestudent) - PC-1						Webex (Hoyskolestudent) - PC-2			
Test iteration	**Rating**	**Video**	**Sound**	**Seamlessness**		**Rating**	**Video**	**Sound**	**Seamlessne**
1.1.1	6	6	6	6		6	6	6	6
1.1.2	6	6	6	6		6	6	6	6
1.1.3	6	6	6	6		5.6	6	5	6
1.1.4	6	6	6	6		5.6	6	5	6
1.1.5	6	6	6	6		6	6	6	6
1.1.6	6	6	6	6		5.6	6	5	6
1.2.1	6	6	6	6		6	6	6	6
1.2.2	6	6	6	6		6	6	6	6
1.2.3	6	6	6	6		6	6	6	6
1.2.4	5.7	5	6	6		5.3	6	4	6
1.2.5	5.7	5	6	6		5	5	5	5
1.3.1	5	3	6	6		5.7	5	6	6
1.3.2	4.6	3	5	6		5	5	5	5
1.3.3	5	3	6	6		4	4	4	4
1.3.4	5.3	5	6	5		3.7	3	4	4
1.3.5	4.7	4	5	5		3.7	3	4	4
2.1.1	3.3	3	3	4		4	4	4	4
2.1.2	3.3	3	3	4		4	4	4	4
2.1.3	3.3	3	3	4		4	4	4	4
2.1.4	3.3	4	3	3		4.6	5	4	5
2.2.1	3	3	3	3		2.3	2	2	3
2.2.2	3	3	3	3		2.7	2	3	3
2.2.3	3	3	3	3		2.0	2	2	2
3.1.1	5	5	5	5		5	5	5	5
3.1.2	5	5	5	5		5	5	5	5
3.1.3	5	5	5	5		5	5	5	5
3.1.4	5	5	5	5		6	6	6	6
4.1.1	3.3	3	4	3		5	5	5	5
4.1.2	4	4	4	4		4	4	4	4
4.1.3	2.6	2	3	3		5	5	5	5

Table A.9

A.13 Rating - Barneparken

Barneparken - PC-1						Barneparken - PC-2			
Test iteration	**Rating**	**Video**	**Sound**	**Seamlessness**		**Rating**	**Video**	**Sound**	**Seamlessness**
1.1.1	6	6	6	6		6	6	6	6
1.1.2	6	6	6	6		6	6	6	6
1.1.3	6	6	6	6		6	6	6	6
1.1.4	6	6	6	6		6	6	6	6
1.1.5	6	6	6	6		6	6	6	6
1.1.6	6	6	6	6		6	6	6	6
1.2.1	6	6	6	6		5.6	5	6	6
1.2.2	5.6	5	6	6		4.6	3	6	5
1.2.3	5.6	5	6	6		5.6	5	6	6
1.2.4	5	4	6	5		5	4	6	5
1.2.5	5	4	6	5		5	4	6	5
1.3.1	5.6	5	6	6		5	5	5	5
1.3.2	5.6	5	6	6		5	5	5	5
1.3.3	5.6	5	6	6		5	5	5	5
1.3.4	4.7	4	5	5		5	5	5	5
1.3.5	4.3	4	5	4		4.7	5	4	5
2.1.1	4.6	4	5	5		5	5	5	5
2.1.2	4.6	4	5	5		5	5	5	5
2.1.3	5	5	5	5		5	5	5	5
2.1.4	5	5	5	5		5	5	5	5
2.2.1	1	1	1	1		2.3	2	2	3
2.2.2	4	4	4	4		3.0	3	3	3
2.2.3	2	2	2	2		3.0	3	3	3
3.1.1	5	5	5	5		5	5	5	5
3.1.2	5	5	5	5		5	5	5	5
3.1.3	5	5	5	5		5	5	5	5
3.1.4	5	5	5	5		5	5	5	5
4.1.1	4	4	4	4		5	5	5	5
4.1.2	3.3	3	4	3		5	5	5	5
4.1.3	2.3	2	3	2		4.6	5	4	5

Table A.10

A.14 Rating - Skytjenester

Skytjenester - PC-1						Skytjenester - PC-2			
Test iteration	**Rating**	**Video**	**Sound**	**Seamlessness**		**Rating**	**Video**	**Sound**	**Seamlessness**
1.1.1	6	6	6	6		6	6	6	6
1.1.2	6	6	6	6		6	6	6	6
1.1.3	6	6	6	6		6	6	6	6
1.1.4	6	6	6	6		6	6	6	6
1.1.5	6	6	6	6		6	6	6	6
1.1.6	6	6	6	6		6	6	6	6
1.2.1	5.3	4	6	6		5.3	4	6	6
1.2.2	5.6	5	6	6		4.3	2	6	5
1.2.3	5.6	5	6	6		4.3	2	6	5
1.2.4	5	5	5	5		4.6	4	5	5
1.2.5	5	5	5	5		4	4	4	4
1.3.1	5.3	4	6	6		5	5	5	5
1.3.2	5.3	4	6	6		5	5	5	5
1.3.3	4	5	1	6		5	5	5	5
1.3.4	4.6	4	5	5		4.6	4	5	5
1.3.5	4.6	4	5	5		5	5	5	5
2.1.1	5	5	5	5		5	5	5	5
2.1.2	4	4	4	4		5	5	5	5
2.1.3	4	4	4	4		4.6	5	4	5
2.1.4	6	6	6	6		5	5	5	5
2.2.1	4.3	4	5	4		3	3	3	3
2.2.2	1	1	1	1		2	1	2	3
2.2.3	1.3	1	2	1		2	2	2	2
3.1.1	5	5	5	5		6	6	6	6
3.1.2	5	5	5	5		5	5	5	5
3.1.3	5	5	5	5		5	5	5	5
3.1.4	5	5	5	5		5	5	5	5
4.1.1	2	1	3	2		5	5	5	5
4.1.2	2	1	3	2		5	5	5	5
4.1.3	4	4	4	4		5	5	5	5

Table A.11

A.15 AP vs Ethernet

```
PS C:\Users\fager> ping vg.no

Pinging vg.no [195.88.54.16] with 32 bytes of data:
Reply from 195.88.54.16: bytes=32 time=4ms TTL=246
Reply from 195.88.54.16: bytes=32 time=4ms TTL=246
Reply from 195.88.54.16: bytes=32 time=4ms TTL=246
Reply from 195.88.54.16: bytes=32 time=4ms TTL=246

Ping statistics for 195.88.54.16:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
    Minimum = 4ms, Maximum = 4ms, Average = 4ms
PS C:\Users\fager> ping vg.no

Pinging vg.no [195.88.55.16] with 32 bytes of data:
Reply from 195.88.55.16: bytes=32 time=6ms TTL=246
Reply from 195.88.55.16: bytes=32 time=7ms TTL=246
Reply from 195.88.55.16: bytes=32 time=7ms TTL=246
Reply from 195.88.55.16: bytes=32 time=8ms TTL=246

Ping statistics for 195.88.55.16:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
Approximate round trip times in milli-seconds:
    Minimum = 6ms, Maximum = 8ms, Average = 7ms
PS C:\Users\fager>
```

Fra: Havard Eidnes
Sendt: tirsdag 9. februar 2021 kl. 14:47
Til: fagerbekk@hotmail.com
Kopi: andreasromo@protonmail.com; andreas@vfiles.no; haakoer@stud.ntnu.no
Emne: Re: Peering mot Telenor

> Vi er en bachelorgruppe fra Gjøvik som har i oppgave å teste og
> inspisere nettverksflyten under virtuelle møter. For å teste og
> inspisere gjøres et sett med tester. Disse testene går gjennom
> Uninett og Telenor som ISP. I kontakt med Gunnar Hole, Peering
> Kordinatoren for Telenor, har vi fått vite hvordan Telenor sender
> trafikk fram og tilbake til Microsoft sine datasentere. Nå skulle vi
> ønske å vite hvordan trafikken går fra Uninett sin side.

Fra Uninett sin side til Microsoft sine datasentre?

Uninett har en lokal sammekobling med AS8075 (ett av ASene til Microsoft) i Oslo der Uninett annonserer adresserommet tilhørende Uninett sine kunder, og der Microsoft annonserer "det de finner for godt" til oss. Jeg ser at vi mottar i nærheten av 350 individuelle ruter fra Microsoft over disse lokale sammenkoblinene i Oslo.

I sin alminnelighet, om vi i Uninett ikke har en eksplisitt rute mottatt over peering (som med Microsoft) så seder vi trafikken til NORDUnet. NORDUnet er det selskapt som er eid av de nordiske fosknings-nettene og som knytter oss sammen og gir konnektivitet til omverdenen på generell basis. NORDUnet har en relativt velutbygd peering-infrastruktur, og kjøper også transitt-trafikk "i bulk" for å dekke de destinasjonene NORDUnet av ymse årsaker ikke kan nå med egen peering. NORDUnet har sammenkobling (peering) med Microsoft på flere steder rundt omkring, men ser ut til å motta omtrent det samme antallet ruter fra Microsoft som vi i Uninett gjør lokalt i Oslo. Jeg har ikke ettergått listene i detalj.

Så... Det er en viss grad av sannsynlighet for at Uninett vil sende trafikken over til Microsoft over de lokale sammenkoblingene i Oslo og ikke bruke NORDUnet på veien ut til Microsoft, selv om trafikken skal til andre "regioner" internt i Microsoft sitt nettverk, og at retur-trafikken vil flyte mer eller mindre symmetrisk motsatt vei.

> Om vi kunne fått kontakt med noen som har kunnskap om slik
> informasjon hadde det vært supert.

Du har vel forsåvidt kommet rett...

Vennligst,

- Håvard

Quality and Traffic Flow in Videoconferencing Infrastructures

NTNU
Norwegian University of
Science and Technology