Bachelor's thesis

NTNU
Norwegian University of Science and Technology

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Webex and Networking Interference

Bachelor's thesis in Digital Infrastructure and Cyber Security
Supervisor: Ernst Gunnar Gran
Co-supervisor: Henning Elvestad and Bjørn Isene
May 2022

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Preface

A massive thanks goes out to the following invididuals and organizations that have been helpful during the span of this thesis:

NTNU - Ernst Gunnar Gran, for being an excellent supervisor, both during the pre-project planning, and the thesis itself. His advice and knowledge helped elevate this thesis.

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Cisco - Espen Berger, for his detailed insight into the Webex platform and general knowledge surrounding networking and videoconferencing related thematics.

Abstract

The purpose of this thesis was to gain insight into what happens to the Webex Meetings platform when it is affected by a poor network connection. Three different test scenarios containing a variable degree of network interference and latency were conducted, and these were correlated between using a baseline data set. Our analysis determined which networking concepts within this thesis become a contributing factor when conditions become subpar, and how they affect different video metrics within the Webex Meetings client.

Sammendrag

Hensikten med denne bacheloroppgaven var å få oversikt over hvordan forskjellige faktorer kan påvirke Webex Meetings plattformen og skape dårlig nettkvalitet. Oppgaven består av tre forskjellige scenarioer som omhandler forsinkelse og interferens, og disse ble sammenlignet med et datasett bestående av optimale testdata. Vår analyse fastslo hvorvidt de forskjellige nettverkskonseptene som denne oppgaven omhandler påvirker en dårlig nettforbindelse, samt hvordan de forskjellige videometrikkene i Webex Meetings-klienten ble påvirket av disse.

Contents

Figures

Tables

Acronyms

AM Ante meridiem. 32, 33, 69

Bps Bits per second. 9

COVID-19 Coronavirus Disease 2019. 1

dB decibel. 14, 45, 50

dBm decibel-milliwatts. 14, 35, 45, 46, 55, 60

FOM Fiber Optic Modem. 8

fps Frames per second. 21, 28, 45, 49, 64, 67

Gbps Gigabits per second. 19

GHz gigahertz. 17, 29

GSM Global System for Mobile Communications. 18

GUI Graphical User Interface. 23, 25

I-frames Intra-frame. 4, 16, 30, 33, 41, 42, 46, 48, 52, 59, 73, 74, 76, 77

IP Internet Protocol. 7, 8, 12, 13

ISP Internet service provider. 8, 19, 34

Kbps Kilobits per second. 21, 44, 46, 48, 52, 54, 65, 71, 72

kHz kilohertz. 17

LTE Long Term Evolution Network. 19

Mbps Megabits per second. 10, 19

MHz megahertz. 17–19, 29, 34

ms milliseconds. 9, 13, 35, 42, 43, 47–49, 51–54, 56, 58, 66, 74

NTNU Norwegian University of Science and Technology. 5, 28, 29, 33, 34, 36, 51

OSI Open Systems Interconnection. 23

p Pixels. 10, 21, 26, 28, 44, 45, 49, 50, 55, 59, 64, 67, 77

P-frames Predicted frame. 4, 16, 30, 41, 46, 48, 52, 59, 73, 74, 77

RSRP Reference Signals Received Power. 4, 15, 24, 35, 37–39, 45, 46, 50, 51, 55, 60, 69, 71

RSRQ Reference Signal Received Quality. 4, 15, 24, 37–39, 45, 50, 51, 55, 60, 69

RTP Real-time Transport Protocol. 23, 31, 77

RTT Round Trip Time. 9, 42, 47, 49, 51, 66

SIM Subscriber identity module. 34

SNR Signal-to-noise ratio. 4, 11, 13, 14, 23, 31, 33, 36, 38, 39, 50, 51, 68–70

Tbps Terabit per second. 7

TCP Transmission Control Protocol. 4, 9

UDP User datagram protocol. 4, 9, 25, 31

UHF Ultra-high frequency. 17, 18

UMTS Global System for Mobile Communications. 18

VoIP Voice over Internet Protocol. 9, 11, 21

Glossary

.pcap is a file containing packet data, often used to analyze and troubleshoot a network. 23, 25, 31, 74, 76

2G is a cellular network which is a second generation mobile network for mobile services. 17, 19

3G is a cellular network which is a 3rd generation mobile network for mobile services. 17–19

4G is a cellular network which is a 4th generation mobile network for mobile services. 1, 3, 17–19, 28, 29, 36, 41

4G+ offers the same features as 4G, at faster speeds and with even more frequencies. 19

5G is a 5th generation cellular network. 17, 19

artifacting refers to a noticeable distortion of images in a video stream. 63, 73

Bandwidth is a term used to measure the maximum amount of data transfer over a given connection. 3, 4, 7, 8, 10, 16–19, 21, 26, 30, 36, 44, 52, 61, 63–67, 70–73, 77, 78

Bitrate is a measurement to calculate the amount of data throughput used to transfer data. 10, 30, 48, 49, 54, 58, 64, 65, 71–73, 76–78

bottleneck is a network term used to describe a situation where a link receives more traffic than its own capacity. The network speeds will then slow down, resulting in delays or dropped packets. 30, 44, 59, 77

broadband is a term used for wide bandwidth data transmissions which tranports signals with a wide range of frequencies and Internet traffic types. 24

buffer is commonly connected to videos, and refers to a halt in the video transmitting to load data in to memory. 43

buffering is commonly connected to videos, and refers to a halt in the video transmitting to load data in to memory. 63

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.. xiii–xv, 2, 8, 12–14, 17, 18, 22–25, 30, 34, 35, 37, 42–60

Cisco is a multinational technology corporation that develops, manufactures and sells networking hardware, software and other technological products and services. 1, 21, 23, 27, 69, 74

coaxial cables are electrical cables used to carry high-frequency electrical signals with minimal loss. The cables are used for carrying telephone trunk lines, broadband internet networking cables, high-speed computer data buses, cable television signals, and connecting radio transmitters and receivers to their antennas. 7, 8

data center is usually a secure facility containing switches, storage systems, servers, routers and security devices. The data center is used to store data and to share applications and data across the Internet. 8

dead spot is a zone within the range of signal transmitters where little to no signal can be received. 13

dial-up is a connection that is established through the use of a modem. 18

downlink is the link from the cellular tower and down to the client, measured in MHz. 61, 64

Dropped frames is a term used in situations where a connection to a device is unstable, or one of the parties involved can not keep up with the bitrate level required for the connection. 11

EDGE is a cellular network for Enhanced Data rates. This version is up to three times faster than its predecessor 2G. 19

eduroam is an international WiFi internet access roaming service for users in research and higher education. 28, 29, 36, 39, 41

fiber optic cables are cables that contain wires in addition to optical fibers that are used to carry light. The purpose of these cables may vary, but they usually provide long-distance telecommunication or high-speed data connections. 7, 8

Fiber Optic Modem is a modem that converts optical data from a digital format into a format suitable for an analog transmission. 8

fiber optics are flexible, transparent fibers that transfers light with lower attenuation to electricity in electrical cables. 8

firewall is a security appliance used to stop or mitigate unauthorized access. 24

frame rate is the frequency at which consecutive images are displayed or captured on a device. Another word for frame rate is frames per second/FPS, and indicates how many frames are displayed every second. 4, 28, 46, 55, 59, 64, 67, 77

Global System for Mobile Communications is a second generation (2G) cellular network. 18

graphical user interface is a platform, system or software that uses buttons, icons and other menus making a visual interaction between the user and the system possible. 23

Information Systems is a degree in IT, economics and leadership. 5

Internet Protocol is a communication protocol used to transport data across the Internet. 7

Internet Service Provider is a company that provides access to the Internet for both personal and business users. 8

Jitter is a time delay causing variation when sending packets over the network, which may result in packets being received at unintended intervals. 2–4, 7, 12, 13, 35, 37, 43, 44, 48, 51, 53, 54, 58, 65, 66, 71, 74, 77

latency is best described as a time delay between senders and recipients, often as a result of network delay. 2–4, 7–9, 11, 15, 31, 32, 35, 37–39, 41, 42, 46, 47, 49, 51, 52, 56, 57, 61, 63, 65, 66, 68, 70, 71, 73, 74

Long Term Evolution Network is a 4th generation cellular network. 19

McKinsey Global Institute is a management consulting firm that advises on strategic management to corporations, governments, and other organizations. 1

media codecs is a service or program that encodes or decodes a data stream, typically a video stream or similar. 3

Microsoft Teams is a communication platform developed by Microsoft. The platform offers services such as chat functionality, videoconferencing, file storage, and application integration. 4

nodes are all the access points a data packet passes through to reach a destination. 9

noise power is term used for bandwidth at the input or output device when the signal is not present. 13, 14

Open Systems Interconnection is a model with seven layers detailing how computer systems communicate over a network for each layer. 23

OpenH264 codec is a codec library which supports H.264 encoding and decoding. 16

packet is term used in networking to describe a formatted unit of data. 7–9, 11–13, 23, 25–27, 29, 31

packet loss happens when one or more packets are dropped while in transit across a network. 2–4, 7, 9, 11, 35, 37, 42, 43, 48, 49, 51, 53, 57, 60, 64–66, 71, 74, 77

packet switched network is a type of computer communications network that groups and sends data in the form of small data packets. 7

ping is a process used for assessing the reachability of a host on a network. The process entails sending ICMP packets back and forth to measure the response time between the hosts. 37

Real-time Transport Protocol is a network protocol for delivering audio and video over IP networks. 31

Reference Signal Received Quality is an indicator of the quality of the received reference signal. RSRQ gives information on how much interference and noise there is.. 15, 24

Reference Signals Received Power is often seen in modern cellular networks, giving information of how strong a signal is from a cellular tower and the client. It's a key measurement used to making a cell selection for the client. 15, 24

resolution is the number of pixels in each dimension that can be displayed on a monitor. The pixel dimensions of the screen are separated in to height and length. 4, 10, 21, 28, 46, 50, 55, 59, 64, 67, 71, 77

scene complexity is a term used to describe resource consumption needed to maintain an image. 28–30, 39

signal noise is a term used to describe unwanted disturbances in electrical signals. 13, 69

Signal-to-noise ratio is a measurement used to determine how strong a signal is compared to its noise level. It is measured in decibels and if the value is above zero, the signal is stronger than the noise. The higher the value, the stronger the signal is. 13, 35

speedtest is a term used for measuring upload and download for an Internet connection. 36

spikes is a term used to describe bursts of irregular traffic in a network. 71

stuttering is a term used to describe irregular delays between data packets. 63

Throughput is the rate of which data is transferred through a system. 10, 11, 45, 57

thumbnail is a small sample image used to represent a file. 50

Transmission Control Protocol is a network protocol that requires an established connection before data can be sent. 9

Universal Telecommunications System is a third generation (3G) cellular network. 18

Uplink is the link from the client and up to the cellular tower, measured in MHz. 64

WiFi is a wireless networking technology. It can be used to make a connection between an access point and a device with WiFi capabilities to communicate. 8, 26, 28, 29, 36, 68

Chapter 1

Introduction

Videoconferencing and streaming video has experienced an increasing trend in businesses since the early 2000s, be it abroad or simply working from home. Meetings, presentations, and demos are easily presented from anywhere in the world - provided that the presenter and participants have an internet connection. Since the start of the COVID-19 pandemic, more and more people have started utilizing cellular networks such as 4G to work from home [1]. This has introduced a need for videoconferencing software such as Cisco Webex to work well over mobile networks, but how does Webex fare when network conditions are subpar?

After COVID-19 forced the hands of many businesses, working from home more or less became the norm. Businesses and educational institutions started using software such as Webex as a replacement for physical attendance and as a preventative measure for spreading the coronavirus. This led to a rapid transition from working on-site to working remotely for many, introducing a new style of collaboration [2]. This is where Webex slots in, as a useful application for remote collaboration. According to the McKinsey Global Institute, we will continue to see a prevalent use of remote collaboration post-pandemic [3].

Participating in online meetings where the network is experiencing disruptions will make it challenging to pay attention. By defining what poor network connectivity is, discovering how Webex Meetings is affected by this and how it compensates can be discussed.

1.1 Background

Telenor is one of the biggest digital service providers in Norway in regards to telecommunications and data services [4]. They are currently in collaboration with Cisco to offer an improved videoconferencing experience to users utilizing the Cisco Webex platform [5]. This bachelor thesis was requested by Telenor, and the aim is to see how 4G cellular networks affect the Webex Meetings platform under certain scenarios. Consequently, to pursue this aim, this thesis considers various experiments with testing configurations, as can be seen in figure 1.1, tailored to gather valuable data surrounding network interference such as packet loss and latency.

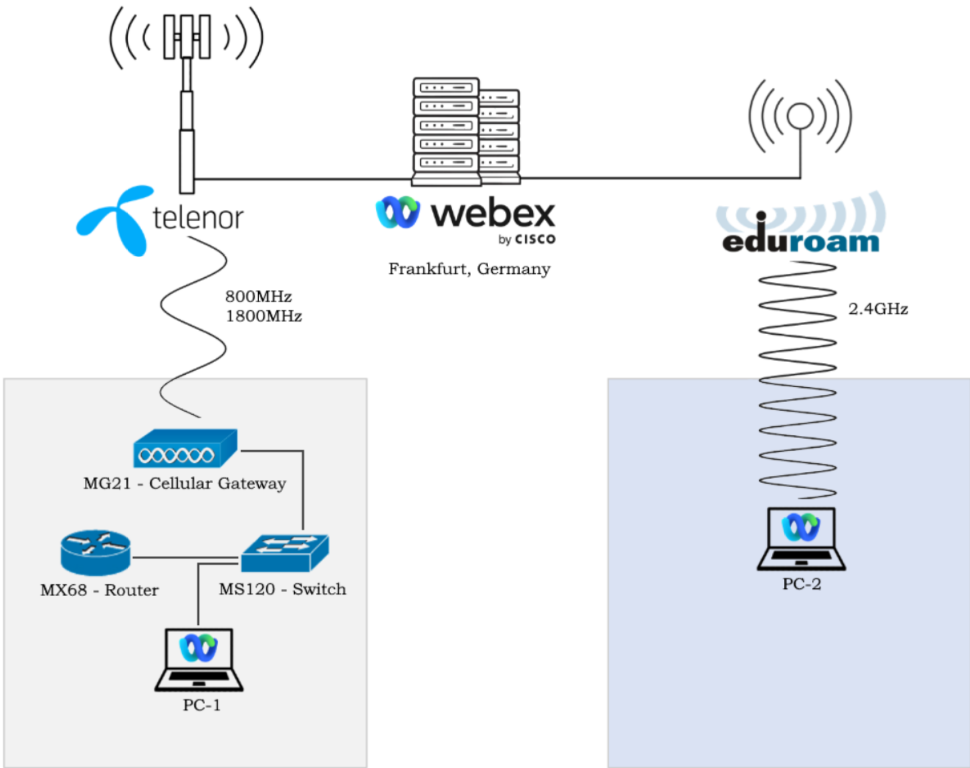

Figure 1.1: An example test setup from a test scenario. License: Dennis Jensen, CC BY

In order to test these scenarios that will later be outlined, a wide array of tools will be used. Webex will be utilized for videoconference testing. Webex allows users to collaborate with anyone through their application with built-in functionalities like screen sharing, video and audio calls, group creation and direct messaging [6].

Having a reliable connection to the videoconferencing application of your choice is not guaranteed if the network resources are insufficient. End users can be affected by poor internet connectivity due to jitter, packet loss and increased latency. These metrics will affect the Webex Meetings platform in various ways, depending on the severity of the metrics in question. A higher latency will results in a negative user experience and vice versa. This is what this thesis aims to uncover, how the Webex Meetings platform is affected by these metrics that make up a poor connection [7].

1.2 Problem Statement

This bachelor thesis will analyze how the variable quality of 4G networks affect the Webex Meetings platform. In order to research and analyze how videoconferencing is affected by poor cellular network connections, explaining what defines a poor network connection is crucial. This in turn means understanding how jitter, packet loss, latency and bandwidth correlate between each other, and how they affect Webex Meetings as a platform. As such, this thesis will be divided into three parts.

The first part of this thesis will define what constitutes a poor network connection. In order to properly define this, an introduction to all relevant technologies and services that are essential for cellular networking will be introduced in chapters 2 and 3, Theoretical and Technical Backgrounds respectively. This part is closely related to the third part of this thesis, which focuses on how Webex Meetings behaves during these conditions. This introduces the first hypothesis, *when the Webex Meetings platform has to compensate and lower the video metrics to be able to upkeep a continuous video stream, this can be used as a definition of a poor network connection.*

The second part of this thesis will explain the reasoning behind the specific test scenarios that will be conducted. One of the objectives is that test scenarios should be kept to a minimum, in order to create a greater discrepancy between test results. If the contrast between optimal and subpar test results can be distinguished with greater ease, the consequences for these metrics will be easier to ascertain. There is no direct hypothesis related to this part of the thesis, but rather an explanation of why the group has decided to choose these specific scenarios.

The third part of this thesis will focus on the results found in chapter 5, Results. Additionally, as mentioned in the previous paragraph regarding the first part of this thesis, detailing how the Webex Meetings platform is affected by subpar conditions will be essential in part three of this thesis. This part will cover how and why Webex Meetings adjusts video metrics in order to deal with the aforementioned subpar network conditions. This introduces the second hypothesis, *when Webex Meetings is affected by subpar networking conditions, video bandwidth is deemed more susceptible to reduction than audio, and a greater effort will therefore be put into maintaining a consistent audio stream.*

1.3 Scope

A bachelor thesis will traditionally be fairly open to interpretation of what the core issues are, and how to attempt to solve them. While it's entirely possible for bachelor assignments to put an emphasis on subjects related to media codecs, security around videoconferencing or developing additional features for videoconferencing applications, this thesis will not study these aspects. The core aim will be to understand and elaborate on networking aspects related to Webex Meetings as a platform. Similar videoconferencing platforms such as Microsoft Teams will likely have comparable results, but is not something this thesis aims to cover.

When on the topic of videoconferencing, networking becomes a key subject [8]. Transferring a video stream from one device to another is a complicated subject, but the nuances of networking topics and what problems may arise when attempting to create a fluid media experience, is what this thesis will explore. To delve deeper into how these networking topics affect the Webex Meetings video service, an introduction to each of them will be made in chapter 2. In no particular order, these are: jitter, packet loss, bandwidth, latency, SNR (RSRP and RSRQ), different types of frames (I-frames and P-frames) and various internet suite protocols (TCP/UDP). Due to the difficulty of separating the effect each of these metrics have, conclusions from eventual results will be considered en masse.

Furthermore, limiting external factors is an important aspect. This thesis aims to keep a focus on networking related subjects, and to limit the amount of external factors and their relevance. With this in mind, certain equipment and networks will have to be evaluated prior to testing. Identical webcams would be preferred, in order for resolution, frame rate and the quality of the image all be the same. If any of these factors differ among participants, readings will not be the same when performing test scenarios. The props used to simulate human behavior when meetings take place should also be similar for all video streams. To make sure the workload on individual computers involved in testing is not a limiting factor, similar specs and programs will be running on participant workstations. Should one computer be noticeably slower, this will have an effect on the performance and may result in worse perceived quality. These are the core issues that are likely to surface when testing has commenced, but additions may arise further down the line, and will be explained in detail if necessary.

1.4 Group Members

All the group members participating in this thesis have a similar basis of knowledge, as all are participants of the same bachelor program, Digital Infrastructure and Cyber Security[1]. This bachelor program includes subjects such as competence in the field of cyber security, robust data networks and modern IT operations with a focus on virtualization and cloud solutions. Additionally, there are several optional courses that candidates can specialize themselves in. Relevant to this bachelor thesis, all candidates have opted to enroll in the Campus Networks and

[1] `https://www.digsec.no`

Internet Architecture course [2]. This course was also a prerequisite subject to apply for this bachelor assignment.

Campus Networks and Internet Architecture is especially relevant for this thesis, since a lot of the topics from this course builds a foundation of knowledge that is beneficial for this bachelor assignment. This course builds on the previous obligatory networking courses, Data Communication and Networks, and Interconnected Networks and Network Security. The group members have found a keen interest in networking related subjects, and this was one of the main reasons the group pursued this bachelor assignment.

Prior to enrolling at this bachelor program, the group members have differing educational backgrounds. Dennis has gained experience from studying engineering and teaching, while Ole Morten has completed a bachelor degree in Information Systems at the University of South-Eastern Norway. Knut-Magnus has served in the Norwegian military prior to enrolling at NTNU. All these experiences provide a wide variety of knowledge, and can prove useful when working on a project together. Structure and discipline is an aspect where this is particularly helpful.

1.5 Thesis Structure

This bachelor thesis is written with the IMRaD[3] format as a guideline. IMRaD refers to the following four sections: Introduction, Method, Results, and Discussion.

Chapter 1, 2 and 3 of the thesis will serve as an introduction to the thesis, covering tools and theoretical background. Chapter 4 covers the method for which testing scenarios was conducted, the topology and what the testing procedures are. Chapter 5 of the thesis is the result section, where data from the testing scenarios will be presented and explained. Chapter 6 of the thesis is the main discussion section of the thesis, building up to the conclusion that is chapter 7.

[2] https://www.ntnu.no/studier/emner/IIKG3021#tab=omEmnet
[3] https://www.sokogskriv.no/en/writing/the-imrad-format.html

Chapter 2

Theoretical Background

In this chapter, a background into relevant terms and concepts will be introduced to build a theoretical foundation. This will guide readers through relevant technologies, how they work, and why they are required for this thesis. A fundamental theme surrounding networking related topics will be outlined, such as jitter, bandwidth, latency and packet loss. Additionally, topics regarding the flow of data packets across IP switched networks will be covered in detail, alongside the inner workings of cellular technologies used to transmit these data packets.

2.1 Internet Infrastructure

Throughout history, there have been several attempts to connect the world together to form a platform for seamless information sharing. One of the projects that was brought to fruition was the ARPANET in 1968, a type of packet switched network used for research purposes. This set out to be the foundation for the internet we use today. Further down the line, the internet infrastructure crossed continents and even oceans in order to connect computers from all around the world. The earliest examples of this utilized coaxial cables that were bolstered along the seabed [9]. Fast forward to today's technology, fiber optic cables have taken their place as seen in figure 2.1. Seeing speeds of up to 160 Tbps [10], these long-distance cables serve as the very backbone of the internet, exchanging internet traffic between continents.

The Internet Protocol (IP) is what makes this communication possible. This protocol is a set of rules and functionalities for routing data across a network from source to destination. The data transmitted is divided into packets containing information such as the destination IP address. This IP address information is how routers know where to forward the packets to, before eventually reaching its destination.

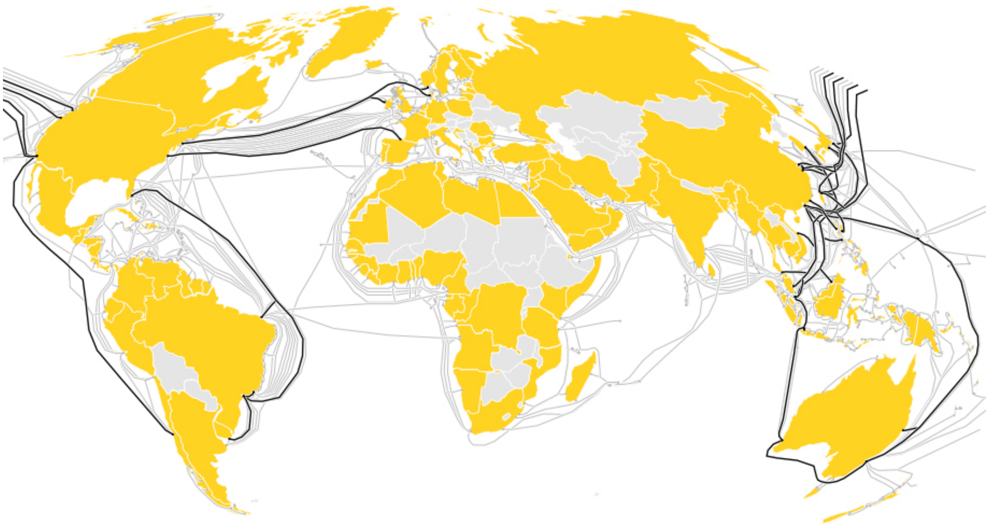

Figure 2.1: A representation of the global internet infrastructure cables running along the bottom of the world's oceans. Licence: Mozilla Public Licence, CC BY

2.1.1 Fiber Optics and Internet Connectivity

The fiber optic cables covered in the previous paragraph are connected to several data center in various locations around the world. When a client connects to an IP address, or its assigned domain name such as webex.com, the closest geographical data center that responds will send packets through coaxial cables, fiber optic cables or cellular networking equipment to reach your IP address provided to you by your Internet Service Provider (ISP). An Internet Protocol (IP) address works similarly to a home address, and the postal system will serve as the fiber optic cables in this instance, as a means of delivering the packets to their destination. The packets know which route they needs to take in order to reach their destination. Eventually, the packets reach a Fiber Optic Modem (FOM) which will convert signals to a readable signal to any given router. This home router may forward the signal through an ethernet cable to the receiving computer, or it will utilize its WiFi capabilities to send wireless signals to the recipient. Should the user instead be communicating with a cellular tower, most of these towers will also be connected via fiber optics, as high bandwidth is important for high-traffic cellular antennas. The end user will then receive radio signals in varying frequencies, depending on what technologies the respective antenna and receiving device support.

2.2 Latency

Latency is the time interval that data uses to travel between endpoints. Traditionally, latency is a metric you want to keep as low as possible, especially when dealing with real-time communication such as videoconferencing and VoIP. These types of technologies are heavily dependent on latency between participants, as longer delays would interrupt the flow of a traditional conversation.

2.2.1 Latency in Webex

Webex Meetings will primarily attempt to use User Datagram Protocol (UDP) as a means of transferring data between clients. This means the Webex platform will have less overhead than Transmission Control Protocol (TCP), which is why UDP is beneficial in real-time communications. When packet loss occurs, a natural assumption would be that TCP is better suited for videoconferencing, so that the missing data could be sent again. This is not the case, as real-time communication is unable to do so fast enough to display or replay a lost video and audio stream. In such a scenario, it is better to ignore the gap, and reestablish the frames as quickly as the video stream allows. For the purposes of latency within this thesis, latency will be referenced in regards to RTT when presenting results, as this is the type of latency that Webex Control Hub records. An acceptable amount of latency in Webex Meetings should be lower than 400ms RTT.

Webex will fall over to utilizing TCP if the UDP ports it normally uses are blocked [11]. As already discussed, this will result in a bigger overhead than UDP, since TCP will need to confirm that data packets have been received with acknowledgements. This in turn means higher latency, and the added latency to the meeting will be noticeable for meeting participants when communicating.

2.3 Bandwidth, Throughput and Bitrate

Transferring data between two nodes is measured by the amount of data transferred, and the speed this data is travelling throughout the network. These factors are combined in the unit called Bits per second (Bps). Bps can be used in multiple ways, depending on the desired result. Some of these results could entail the capability of transferring data through the network at any given time. Other results might detail the actual amount of data transferred between two nodes. Details like these inform the users of limitations and upper limits concerning the network.

2.3.1 Bandwidth Measurement

One way of measuring a network's capability is to calculate something called bandwidth, which measures a medium's capability of transferring data. This is usually measured in bits per second, and gives a theoretical estimate of the upper limit for the respective medium. The reason why this number is simply a theoretical estimate, is because bandwidth only provides a best-effort scenario. It doesn't account for external factors that can have an effect on the medium. Examples of this can be a network lease that has a theoretical maximum bandwidth of 500 megabits per second (Mbps) download and upload. This is referred to as a symmetrical connection, as the upload and download have the same capacity. In theory, this network should be able to provide speeds equal to 500 Mbps.

2.3.2 Throughput Measurement

While bandwidth shows a best-case scenario, the reality usually does not always reflect these types of bandwidth speeds. Having bandwidth of 500 Mbps for upload and download, an end user might only receive 470 Mbps download and 350 Mbps upload. This data transfer rate is called throughput, and is an accurate representation of the data being transported through a network. Throughput is dependant on internal and external factors, and it changes regularly. Some factors include geographical locations, hardware variations, and current traffic load.

2.3.3 Bitrate

Bitrate is a measurement to calculate the amount of data throughput used to transfer packets. This data is often associated with features such as sending and receiving video and audio files. Both of these two features require a steady bitrate to provide desired results. The specific amount depends on the quality of the product. A 720p video will require a higher amount of bitrate than a 360p video. This is because the 720p video will require more data to detail all of its additional pixels, compared to the 360p video stream. Having lower bitrate than what is required to run a specific resolution, will then result in providing pixelated imagery. The same goes for audio, in regards to how high bitrate results in better quality audio. Having lower quality audio will result in stutters and muffled voices.

2.4 Packet Loss

In order to communicate across a computer network, packets have to travel from one device to another. When these packets fail to reach their destination, this is referred to as packet loss. This can be either a single packet, or multiple packets in quick succession. The root cause for packet loss mainly stems from two different aspects, errors in the transmission of data, and network congestion.

2.4.1 Errors in Data Transmission

When devices experience packet loss, there are two main aspects that cause this to happen. The first one is errors in the data transmission across a medium. This can be as a result of either software or hardware related faults. If a device is overloaded by its capability to route packets due to operating at a higher capacity than its software or hardware allows, packets cannot be processed and will be dropped as a result. Specific to this thesis, there are also specific occurrences of packet loss related to cellular and wireless infrastructure. Distance - or latency, physical obstructions and SNR are some to consider, and will be especially relevant when attempting to provoke packet loss manually.

2.4.2 Network Congestion

The second type of packet loss that can occur is called network congestion. This concept will have an effect on most networks, as it relates to throughput. For any given router, if the amount of data packets that is awaiting transmission is greater than the router's capabilities to route such packets, packet loss will be in effect. Packets may be dropped in order of importance, and this is something to be vary of when executing the testing scenarios. A lot of devices have implemented quality of service aspects in order to distinguish between the importance of various packets [12]. Traditionally, services such as VoIP and videoconferencing is topping this list, as such services rely on real-time communication. Dropped frames make audio and video feeds lag behind, and is detrimental for end users.

2.5 Jitter

Jitter is a phenomenon characterized by data packets being unable to reach their target destination in a sequential order. Packed switched networks were mentioned in chapter 2.1, and these types of networks make no guarantee that packets are sent and received using the same routes. For this reason, jitter can occur when data packets are not received in the correct order. In figure 2.2 below, an example of how data is sent over a network is displayed. Without jitter, packets are sent and received with the same amount of delay, and the stream of packets is steady throughout the transmission.

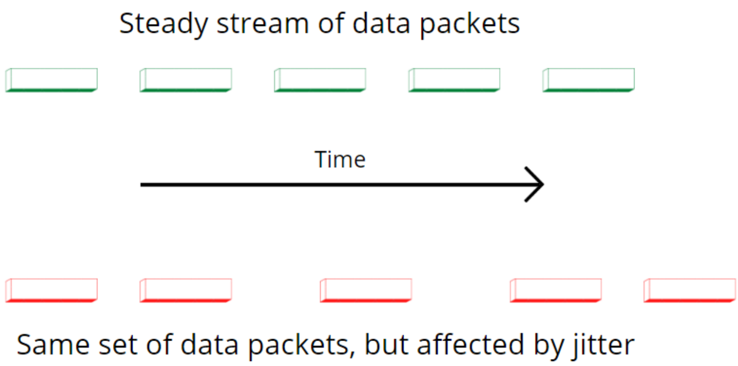

Figure 2.2: A depiction of time delay between data packets, caused by jitter. License: Dennis Jensen, CC BY

When jitter occurs, this stream of packet data will experience disruptions either because packets were sent through different routes, or because of network congestion, which was mentioned in the previous section. An example of data packets travelling through different routes can be seen in figure 2.3. With additional hops, the packets spend more time being routed, instead of being transmitted directly. This can cause a time delay on IP switched networks, as packets may not arrive by their intended sequential order.

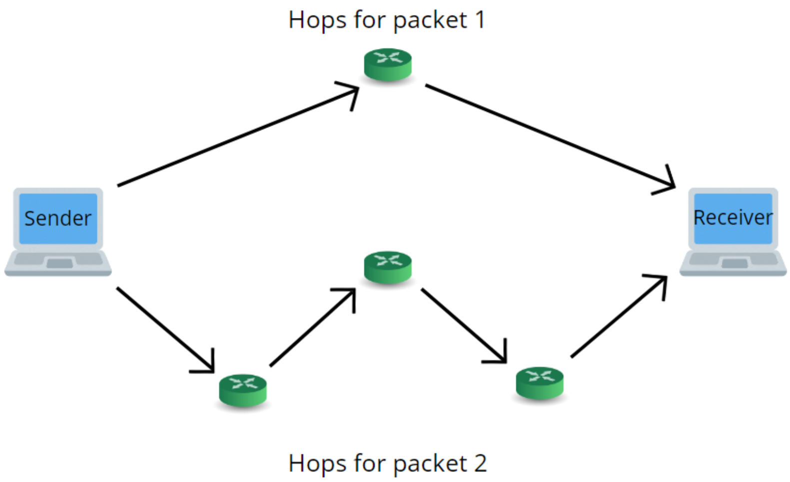

Figure 2.3: During transmission, if packets are sent through different routes, jitter can occur. License: Dennis Jensen, CC BY

Since every IP switched network experiences some form of jitter, applications traditionally have an accepted upper threshold of jitter. This threshold is called a jitter buffer, and can vary depending on the application. For most videoconferencing applications, a jitter buffer ranging from 20-100 ms is acceptable [13].

When referencing jitter in this thesis, a focus will be put on end-to-end transmissions. This will become evident when the test results are presented in chapter 5, where graphs will display the total amount of delay from when packets are sent and received.

2.6 Signal-to-Noise Ratio

Devices that are connected to a wireless or cellular network communicate with each other using radio waves. Quality and connectivity issues are more likely to arise when using these types of networks compared to cabled connections. There are several factors in play when determining what may affect the quality of a wireless connection. One of these factors are hindrances between the signal sender and recipient, such as concrete walls. Range is also an important metric to consider, and quite frequently there will be certain dead spots for any type of wireless communication. If there is a lot of signal noise in the nearby vicinity, such as other devices also communicating wirelessly, Signal-to-noise ratio (SNR) becomes a problem. But what is SNR, and how detrimental is it for a network?

Signal-to-noise ratio is the ratio between the received signal power and the noise power [14]. When using a cellular network on a device, the communication between the device and the cellular tower is realized by sending radio wave signals back and forth. These signals can be visualized as a linear wave and changes depending on the SNR. An example of this type of communication can be seen in figure 2.4, where the cellular tower sends a signal to a device with a receiving signal of -65 dBm and a noise power of -80 dBm. It's possible to determine the state of a signal based on the signal strength. The signal strength is determined by looking at the received signal's dBm and subtracting it from the noise power's dBm. In this case, it would result in a signal strength of -65 dBm minus -80 dBm, which would equal to 15 dB.

The higher the signal strength is, the easier it is to pick out *that* specific signal, resulting in better quality. Generally, a signal strength of 30 dBm or more is recommended for high speeds [15]. A signal strength of 15 dBm such as in figure 2.4 is considered in the range of poor connectivity.

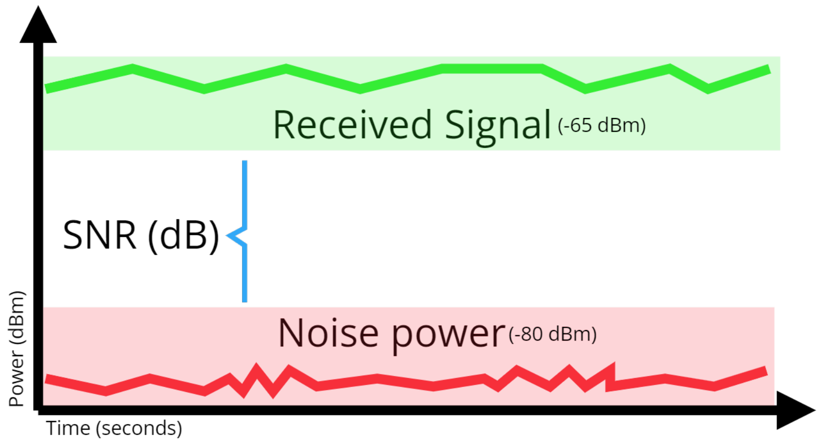

Figure 2.4: An illustration of Signal-to-noise ratio (SNR). License: Knut-Magnus Karlsen, CC BY

The noise power could be varying depending on the area of the device and the cellular tower it utilizes. As illustrated in figure 2.4, the gap between the noise power and the received signal results in a signal strength of 15 dBm. This is bound to change, and the SNR may make it difficult for the devices to pick out the signal in a busy area with a lot of wireless traffic. This will be tested in meticulous detail when carrying out the different test scenarios later in this thesis.

2.7 RSRP and RSRQ

When performing different scenarios at different locations, knowing which cellular tower is utilized will be useful for comparing results. When devices figure out which cellular tower to use, they will look at key measurements such as RSRP and RSRQ to determine which cellular tower will be best suited for a connection at that specific location. It is entirely possible that there are multiple cellular towers within range that may be utilized. This is also the case for an end user that is travelling, and the device needs to do a handover, which means exchanging the currently used cellular connection with another tower. This selection is based on signal strength and how good the connection is [16]. The actual guidelines for devices of RSRP and RSRQ and their perceived signal strength can be seen in figure 2.1.

Signal Bar	RSRP range (dBm)	RSRQ range (dB)
5	RSRP >= -83	RSRQ >= -7
4	-83 > RSRP >= -92	-7 > RSRQ >= -10
3	-92 > RSRP >= -102	-10 > RSRQ >= -13
2	-102 > RSRP >= -111	-13 > RSRQ >= -16
1	-111 > RSRP >= -140	-16 > RSRQ >= -20

Table 2.1: Reference table for RSRP and RSRQ values [17].

RSRP is short for Reference Signals Received Power [16], and is an indicator of how strong the signal is from one device to another. Looking at this in a cellular network perspective, it could be the signal strength from the cellular tower and the device the client is using. The stronger the signal is, the easier it becomes for devices to pick a signal out. This will result in a greater ease of communicating between devices [18]. This metric is measured in decibels, which means a signal is stronger if the RSRP value is closer to zero.

RSRQ is short for Reference Signal Received Quality [16], and it's an indicator of the quality of the received reference signal. RSRQ gives information on how much interference and noise there is [17]. RSRQ is also measured in decibels and as with RSRP, the closer the RSRQ signal value is to zero, the better the signal is. Generally RSRQ serves as a backup in a cell selection if the RSRP measurement is not enough for the device to make a sufficient cell selection based on one metric alone.

2.8 Videoconferencing

When on the topic of videoconferencing, it's easy to focus on the video aspect of a meeting. Images and video often communicate meaning and purpose in a better fashion than audio can, at least in a concise format. "A picture is worth a thousand words", is a common adage, but how important is the image or video portion in relation to videoconferencing? If a video feed were to freeze unexpectedly in the middle of a presentation, not a whole lot changes, unless the video feed is dropped entirely alongside the connection. Jitter and latency issues have already been covered in this chapter, and may result in a lesser quality video stream, but the text should still be readable in most cases.

Audio on the other hand, will vary wildly depending on the state of the connection issues at hand. Video will require more bandwidth and needs to be manipulated to a higher degree than audio does, but that doesn't mean it's any more important than audio. In fact, quite the contrary, loss of audio will make it more difficult for meeting participants to digest the information being communicated. This is something that will be discussed further in chapter 4, where testing scenarios and methodology is discussed in detail.

2.9 Image quality, I-frames and P-frames

Webex Meetings utilizes the OpenH264 codec for compressing video [19]. As video files can get quite huge in size it is important to reduce the size of the data by compressing it, so it is better suited to be transmitted over the internet. Webex Meetings does this by using something called I-frames and P-frames. An I-frames is short for Intra-frame and is the entire frame, meaning it transmits the entire image as shown in figure 2.5. A P-frame is short for Predicted Frame and builds upon the previous I-frames. For instance, if there is motion in the frame it will only transmit the changes in the frame at the time as shown in figure 2.5, compressing the video. This is done by using data from previously encoded frames. As seen in figure 2.5, the person makes a hand gesture waving; here the P-frames would use the data from the previous frame to determine how to compress the motion being captured. It does not have to send the entire frame, it simply builds upon the previously encoded frames. Webex Meetings does this by sending multiple I-frames throughout a meeting that P-frames can build upon.

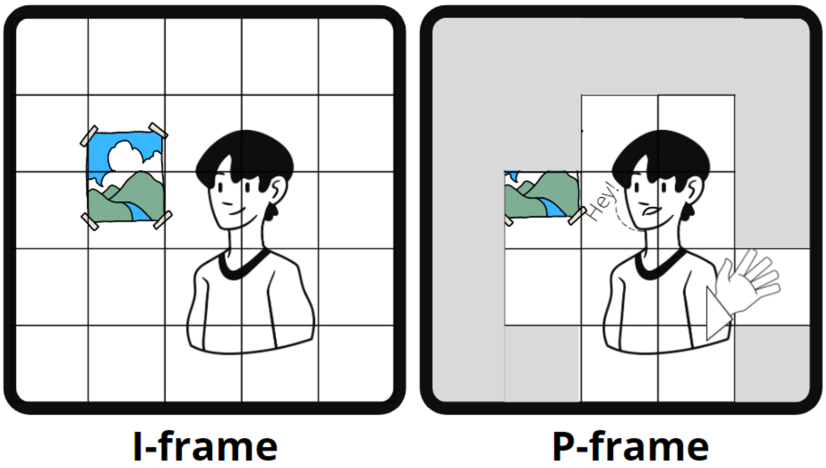

Figure 2.5: An illustration simulating I-frames and P-frames. License: Knut-Magnus Karlsen, CC BY

2.10 Frequencies

Cellular networks like 4G uses radio waves to communicate. The foundation of this communication is a broad spectrum of electromagnetic waves ranging from 3 kHz to 300 GHz. Because of this broad spectrum of electromagnetic waves there are a lot of frequencies or bands to choose from with different functionalities and purposes as shown in figure 2.6. Ultra-high frequency (UHF) is one of them and ranges from 300 MHz to 3 GHz; this band uses cellular networks such as 2G, 3G, 4G and 5G. The UHF band is capable of sending radio waves through buildings while maintaining a high bandwidth making it suitable for cellular networks such as 4G, where a signal has to reach devices through materials such as concrete and metal in an urban environments. This is determined by frequencies, as different frequencies have different capabilities.

Generally, frequencies may reach further but at the cost of bandwidth as the further the signal must travel the less bandwidth it is able to maintain. Frequencies below 1 GHz known as coverage bands will reach further in terms of distance but will have less bandwidth making it more suited for long distance travels, for example where the signal must reach beyond mountain tops or several buildings. Frequencies above 1 GHz known as capacity bands will have a shorter reach but have a higher bandwidth making it more suited for example in an indoor environment depending on the situation. Therefore, the UHF band is a necessity for cellular networks as the range of frequencies can maintain high bandwidth and distance suited for cellular communications.

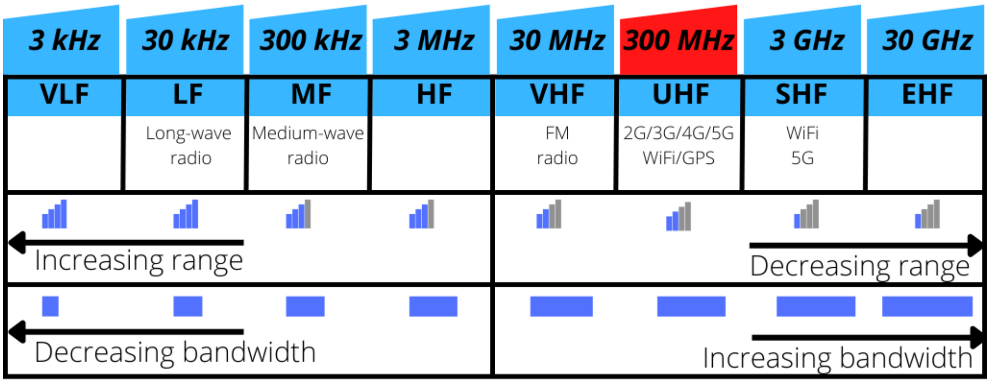

Figure 2.6: A representation of the radio spectrum. License: Knut-Magnus Karlsen, CC BY, Inspiration taken from: NKOM

Although the UHF band offers multiple frequencies to choose from, only the 800 MHz and 1800 MHz frequencies will be of importance during the testing. The reason for this is because Telenor's cellular towers available around campus Gjøvik only offers these two frequencies for 4G communications. Telenor's 800 MHz frequencies have a bandwidth of 20 MHz, uplink between 852-862 MHz and downlink of 811-821 MHz [20]. As previously mentioned, this is known as a coverage band and has the potential to reach further but at the cost of bandwidth. The other band that is available from Telenor's cellular towers around campus Gjøvik is the 1800 MHz frequencies. This is a capacity band has a bandwidth of 40 MHz, uplink between 1715-1745 MHz and a downlink between 1810-1840 MHz [20]. Both bands may be used during the testing scenarios, but it will depend on the environment that the cellular gateway is in. If the signal has a lot of blockages like concrete walls for example, the gateway might utilize the 800 MHz frequency band as it is able to receive the signal better. On the other hand, the 1800 MHz band will be better suited for more urban areas, where the amount of physical obstructions are less frequent.

2.11 Cellular Networks

Connecting to the internet has become trivial with the progression of current technologies. Gone are the days where the main way of connecting to another computer was through a landline connection, such as with dial-up. Nowadays, people are able to connect to the internet from almost anywhere, and one solution that makes this possible are cellular networks. The type of cellular networks we interact with today started taking shape in the early 2000s [21]. This saw the introduction of the Universal Telecommunications System (UMTS). UMTS was the third generation of mobile cellular system for networks, used to improve the standards set from the Global System for Mobile Communications (GSM). This third generation network was called a 3G network, which was an improvement from its predecessor, the 2G or EDGE networks.

2.11.1 Different Generations of Cellular Networks

With the introduction of 3G, the world saw a huge leap forward in connecting devices to the internet. A higher bandwidth resulted in features such as video phone calls and the ability to consume media from various platforms [21]. As the years have passed, progress within the realm of cellular networks has continued to flourish. During the past 15 years, Long Term Evolution Network (LTE, commonly referred to as 4G) and its different versions like 4G+, have been the most commonly used cellular networks in the world [22]. One of the main reasons for this widespread use, is the determination from ISPs to continuously expand the reach of their cellular networks. New base stations and cellular towers are built all the time, and this gives the general public additional zones they can connect to the internet, places that previously lacked access.

When using a 3G connection, a user could experience bandwidth speeds of up to 2 Mbps. In comparison, a user switching to a 4G connection could experience an increase of up to 1 Gbps if conditions were optimal. This means 4G networks had the potential of being up to 500 times faster than their predecessor, 3G networks [23]. With an evolution from 3G to 4G, and its massive increase in bandwidth, cellular networks started to see an increase in popularity, but also a massive increase in available infrastructure.

LTE and its evolutions have served as the fastest cellular network since it was created [23]. However, similarly to the transition from 3G to 4G, the world is about to see a transition from 4G to 5G. 5G networks are currently being implemented in numerous countries around the globe, and Norway is among them. In Norway, Telenor and Telia are some of the competing ISPs, and they are currently expanding their arsenal to include 5G cellular networking in medium-sized urban areas and cities, including Gjøvik.

2.11.2 Cellular Networks in Norway

Norway still sees a heavy use of 4G, and it's the most used cellular network by far [22]. Cellular networks need a wide array of frequencies in order to function, and a lot of the data surrounding available signals is found at `https://finnsenderen.no/`. Base stations across Norway use a mix between 800 MHz, 1800 MHz and 2600 MHz. That said, not all base stations provide the same selection of frequencies. By using the aforementioned website, greater control can be exerted on the various test scenarios that will be executed later in this thesis, by selecting locations that are best covered for the specific frequencies.

2.11.3 Cellular Reach and Stability

If the distance between cellular devices and towers becomes too substantial, communication between them becomes challenging. The most immediate ways to remedy this would be to either increase the range of the signal, or to build additional cellular towers in the surrounding area. By covering reach and stability for the most amount of users simultaneously, there are built a plethora of cellular towers within urban areas. Users will stay connected to the nearest cellular tower until another tower in the vicinity becomes closer than the previous. Signals will continuously switch between cellular towers if the need arises. This is helpful in giving end users the best possible signal strength they can receive at any given location, and also serves to balance the traffic of connected devices between the various cellular towers, offloading connections that are out of reach.

Chapter 3

Technical Background

Throughout this thesis, a number of tools, platforms and devices will be utilized to analyze network interference. In order to understand which of these will be applicable during the different testing scenarios, a rundown detailing all of these will be presented in this chapter.

3.1 Webex Meetings

Webex Meetings is a software application that is used for videoconferencing, and it is developed and maintained by Cisco [6]. The application offers meeting functionality such as screen sharing, VoIP, file sharing, live video feeds, chat and messaging, and group creation. All these services aim to aid the users in remote collaboration. It is also possible to schedule meetings with groups of people. The invited participants can enter and leave meetings at their own leisure from various clients, such as in browsers or on mobile devices. The specific functionality that this thesis aims to utilize is primarily live video and VoIP. Screen sharing will also be covered in brief. This is the main way data collection will happen, but gauging the different connections will also be done via Webex Control Hub.

Webex Meetings supports a variety of video resolution options, as seen in figure 3.1 below. These resolution options ranges from 90p to 1080p, and are dependant on the available bandwidth [24]. Higher amounts of bandwidth will provide a higher resolution. When the bandwidth reaches 120 Kbps, changing the fps is an option. The fps is able to reach values of 30, providing a smother transitions between every motion captured in the video feed.

Layer	Bandwidth Range
90p (each thumbnail)	60 - 100 kbps
180p main video	125 - 200 kbps
360p main video	470 - 640 kbps
720p main video	900 kbps - 1.5 mbps
Content sharing (1080p at 5 fps)	900 kbps - 2.5 mbps

Table 3.1: A table displaying the correlating bandwidth ranges for each resolution that Webex Meetings offers [24].

3.2 Webex Control Hub

Webex Control Hub is used to view and manage an organization's user base and devices [25], as can be seen in figure 3.1. This can be done through the user interface from Webex Control Hub, and also by viewing usage analytics that can be especially useful for verifying results amid testing. Webex Control Hub will see significant use during the span of this thesis, both in testing scenarios to gather additional data and for general insight.

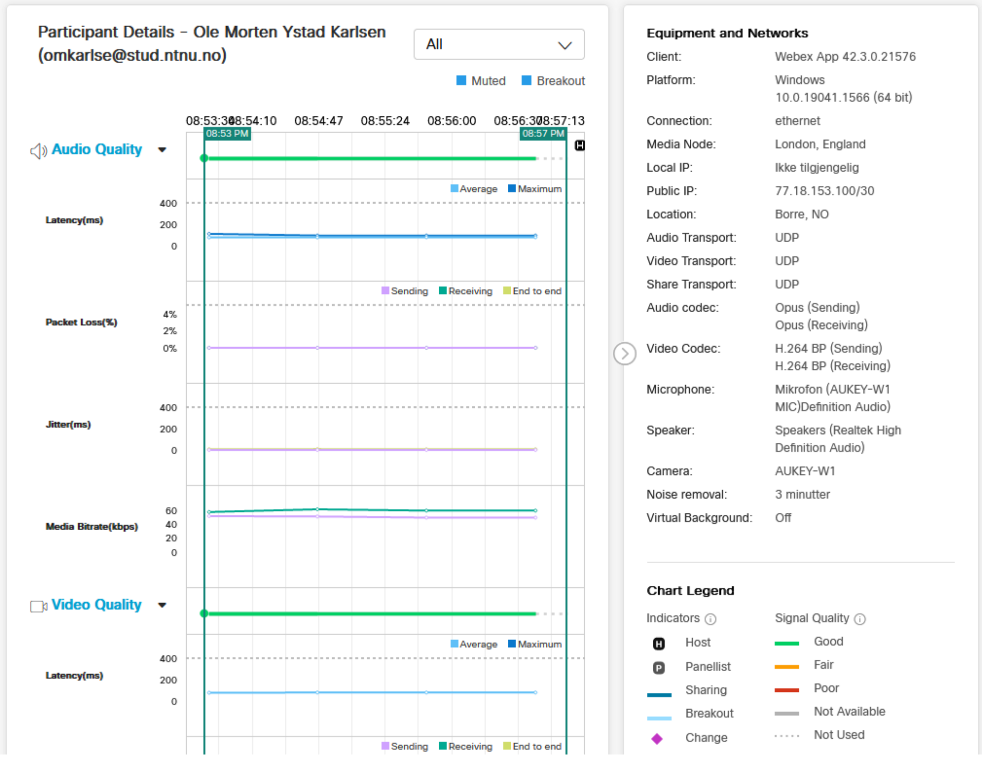

Figure 3.1: Collected data from a meeting represented in Webex Control Hub. License: Cisco Webex, CC BY

3.3 Wireshark

Wireshark is a free network protocol analyzer that provides detailed information about packet information transferred on a specific interface at a minuscule level [26]. The software runs in the background while users perform their usual tasks. Wireshark then tracks all the relevant protocols on a specific interface and captures these in a .pcap file format and presents it to the user. The software will keep running until stopped, but can still present data even if it is not actively capturing new packets. The graphical user interface (GUI) and the data accessibility are both reasons why Wireshark is one of the world's foremost and widely used network protocol analyzers. [27]

Wireshark's GUI, as seen in figure 3.2, is particularly useful because of its search function, which makes it trivial to find relevant data. The GUI is easy to traverse, and the different networking protocols are color coordinated in order to differentiate them from each other with greater ease. The filtering allows users to find specific protocols, addresses, and detailed information about specific Open Systems Interconnection (OSI) model layers. Creating templates for new protocols is possible, but this is not a built-in feature, and has to be added through external scripts. SNR data is not currently trivial to differentiate in Wireshark with the default protocol dissectors, and various scripts related to RTP will be utilized in this thesis. Due to confidentiality agreements with Telenor and Cisco, these scrips cannot be included as appendices.

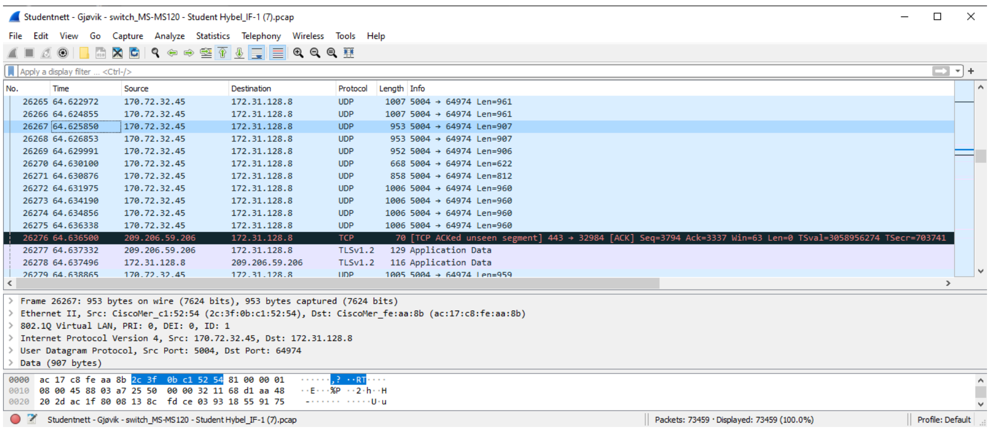

Figure 3.2: An example of a packet capture file loaded in Wireshark. License: Wireshark, CC BY

3.4 Meraki Dashboard

Meraki Dashboard is a centralized browser-based tool provided by Cisco [28]. The dashboard is used to monitor and configure Meraki devices and services. Some of these devices and services include cellular gateways, switches, teleworker gateways, and security appliances, such as firewalls. All devices need to be registered in the dashboard before use in order to provide data. As for cellular gateways, the dashboard tracks the location of the active network, and shows the locations on a built-in map.

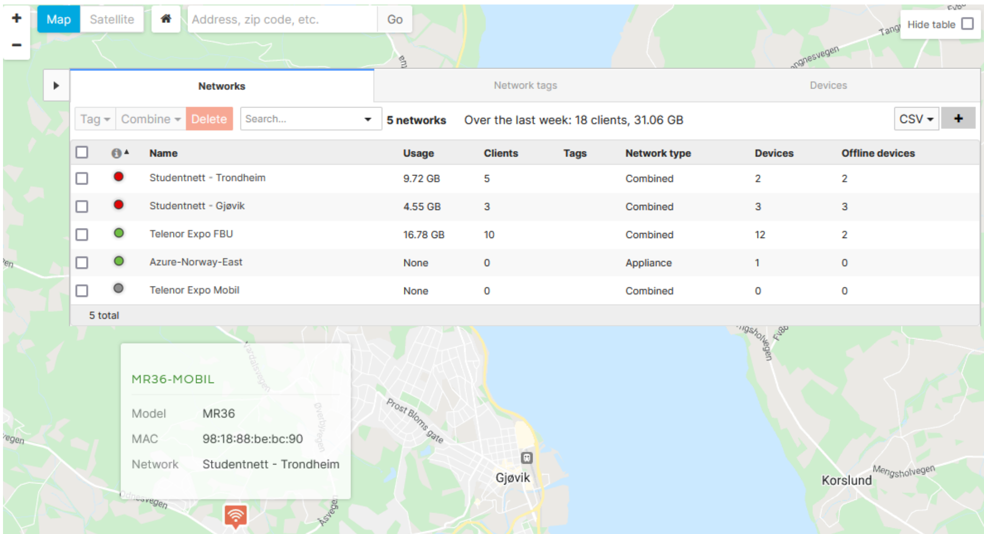

Figure 3.3: The map provided by the Cisco Meraki Dashboard, showing all registered devices and the location of active devices. License: Cisco Meraki, CC BY

By looking at the map in figure 3.3, there is a lot of information presented to users. Accessing one of the registered and connected cellular gateways shows information about the gateway in question. This can be information such as broadband frequencies offered by the gateway. It also tracks information such as traffic exiting the network, and also the amount of data used by each active node. This data will be particularly useful for testing purposes, and will be excellent for extracting key aspects of network information during the different test scenarios.

In addition to everything that is available from Meraki Dashboard, Meraki devices also have a local status page that can be accessed by being directly connected to a Meraki device. This page can be used to extract real-time information about metrics relating to the unit. For the purposes of this bachelor thesis, Reference Signals Received Power (RSRP) and Reference Signal Received Quality (RSRQ) data on this status page will be monitored during testing. Since this type of data is also recorded in Webex Control Hub, extracting this data is not necessary, but will serve as a helpful metric when conducting testing.

3.5 Meraki Packet Capture

Meraki Packet Capture is a data analytic tool provided through the Meraki Dashboard GUI and is used to capture and view live network traffic [29]. The network traffic can either be viewed as a live feed or downloaded through a .pcap file which can be opened and analyzed with Wireshark. This tool can be used to filter user datagram protocol (UDP) packets and also find which data center is used during Webex meetings from the clients on the network. Additionally, Meraki Packet Capture is also able to capture live network traffic on multiple devices and ports, but this will not be utilized during the test scenarios. The reasoning for this is because the traffic that is most interesting will be the network traffic on a specific client participating in a Webex meeting. Therefore, by limiting which port and device the network traffic gets captured on, the captured traffic will be isolated to capture traffic coming from a particular Webex application.

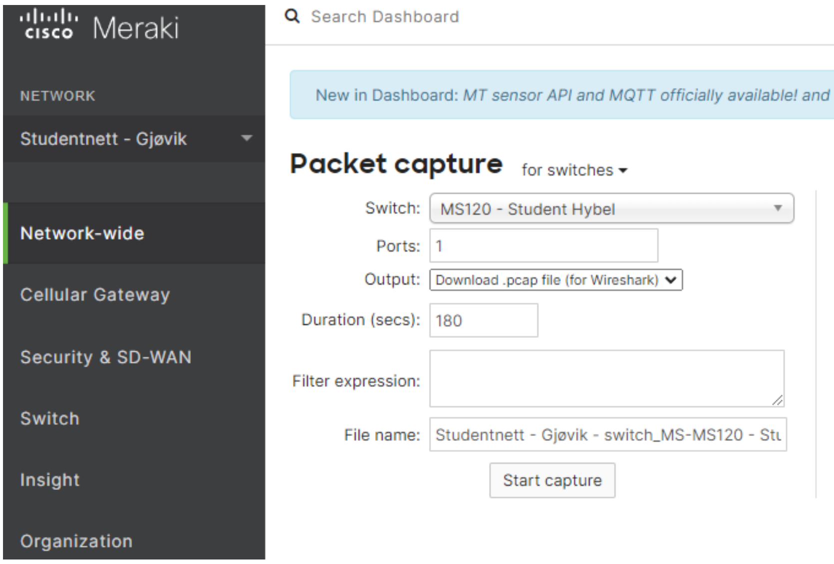

Figure 3.4: Meraki Packet Capture Graphical User Interface. License: Cisco Meraki, CC BY

As seen in figure 3.4, the network traffic will be captured on the switch 'MS120 - Student Hybel' on port 1, in order to avoid capturing excess traffic. Meraki Packet Capture will be used for every test scenario and the corresponding captures will be downloaded as a .pcap file upon completion. This way a thorough analysis of the network traffic can be done at a later point when reviewing the results, giving more insight into each meeting. There are some limitations with Meraki Packet Capture as it's only able to capture 100,000 packets [29]. This will cause problems for certain scenarios, where the amount of data packets captured will exceed this limit. Since there are tests that are performed during optimal conditions, video streams with sufficient enough bandwidth can stream 720p video. This results in more packets per meeting, which also means the quota of 100,000 packets fills up faster.

3.6 Network Cell Info Lite and WiFi

Network Cell Info Lite is a phone application for monitoring active cellular networks and WiFi [30]. The application offers features such as detailing the type, frequency and location of the connected cellular tower. The location feature is especially useful to track changes in the active cellular tower. This feature will be utilized in the Interference Scenarios in chapter 4.8. The application was recommended by Telenor as a substitute for not being able to connect to the MG21 cellular gateway directly, in order to figure out which cellular tower was utilized.

Chapter 4

Methodology

This chapter serves as a guide for understanding how testing was planned and conducted, with the knowledge and understanding of the technical and theoretical background introduced in the previous chapters. A detailed explanation surrounding the different test setups, locations and equipment will set the stage, before the relevant scenarios are detailed towards the end of this chapter.

4.1 Equipment

To conduct tests, there are multiple devices and general equipment that is required to execute these tests. The networking related devices are provided by Telenor, and the rest of the general use equipment such as computers and video hardware is provided by the group.

4.1.1 Networking Devices

There are three devices that are used to send and receive cellular connectivity between the test setup and cellular towers. The first one is the cellular gateway, Cisco Meraki MG21[1], in this thesis simply referred to as MG21. This device will send and receive the required radio signals required to transmit data over the network. The second device is an enterprise router, a Cisco Meraki MX68[2], simply referred to as MX68 in this thesis. This device routes all the data packets that are sent and received by the MG21 cellular gateway. The third and final device is

[1]The product used in this thesis does not have an external antenna. More information about the access point: `https://meraki.cisco.com/product/cellular/integrated-antenna/mg21/`

[2]More information about the router: `https://meraki.cisco.com/product/security-sd-wan/small-branch/mx68/`

an access switch, Cisco Meraki MS120-8LP[3], simply referred to as MS120 in this thesis. The main use of this switch is its Power over Ethernet capabilities, as the MG21 cellular gateway uses this as its power source.

4.1.2 Physical Equipment

In addition to networking equipment, the test setups require hardware to join videoconferencing on, and something to capture audio and video. For the computers, the relevant hardware specifications will be detailed in table 4.1 below. The two columns are differentiated between which network the computer will utilize for every test scenario, in this case PC-1 uses Telenor's 4G cellular network and, and PC-2 uses NTNU's WiFi, eduroam.

	PC-1 (4G Cellular Network)	PC-2 (eduroam)
Central Processing Unit	Intel Core i7-6700HQ	Intel Core i5-8350U
Graphics Processing Unit	NVIDIA GeForce GTX 1060	Intel HD Graphics 620
Random Access Memory	16GB	16GB
Operating System	Windows 10 Pro	Windows 10 Pro

Table 4.1: Physical hardware specifications of computers used for test scenarios.

As well as computers, there are two identical webcams named Aukey PC-W1. The reasoning behind picking two identical webcams is so they have the exact same video quality, and will perform similarly. This will also be a factor when talking about scene complexity in the subchapter 4.2.2. These cameras support up to 1920x1080 pixel resolutions, although 720p will be the highest resolution used for every scenario. In terms of frame rates, these support 30 fps. As far as audio goes, the webcams are outfitted with dual microphones, and these will be used for capturing audio during all scenarios. The type of audio that will be used are taken from podcasts to imitate real conversations. This is done to achieve a continuous audio stream, so that delays can be noted. For control tests, actual participation and testing of audio within meetings will be done to gauge the perceived audio delay. While the props used in these tests could have been covered in this section, an explanation into why the props that were chosen and where the idea originated from will be covered in chapter 4.4.

[3]More information about the switch: `https://meraki.cisco.com/product/switches/access-switches/ms120-8/`

4.2 Testing Setups and Procedures

In order to get an understanding of how the various test scenarios are conducted, an introduction into the testing procedures and setups will be covered in this chapter. From how the network devices are connected to each other, as well as the details surrounding scene complexity and testing procedures.

4.2.1 Network Overview

As mentioned in subchapter 4.1.1, there are three different networking devices that will be used to achieve cellular connectivity on PC-1 utilized in these tests. The explanation in this paragraph will use figure 4.1 below as a reference point. Starting from PC-1, the end goal is to reach PC-2. The first point of interest is PC-1, the setup that utilizes 4G connectivity, which can be seen on the left hand side in the. PC-1 is directly connected to the MS120 switch via an ethernet cable to provide network connectivity. Next, the switch is connected to the MX68 router, which is used to route packets between networks. When data packets are ready to leave the local network, packets are routed back via the MS120 switch, and towards the MG21 cellular gateway. This is connected in such a way, since the MS120 switch is the only device with Power over Ethernet capabilities, the MX68 router does not offer this functionality. The next order of business is to transmit data packets to Telenor's 4G antenna, using radio signals. Since the antennas within campus Gjøvik only supports bands of 800 MHz and 1800 MHz, these are the only available waves that can be used. The next stop is routing the required packets to one of the Cisco Webex data centers, in most cases this will be located in Frankfurt, Germany. From there, adjustments surrounding audio and video are made, before packets are routed back towards Gjøvik, where it finally arrives within the Uninett networks at NTNU campus Gjøvik. The data is directed towards the respective access point that PC-2 is connected to, and data then arrives at PC-2 by using NTNU's 2.4 GHz WiFi of choice, eduroam.

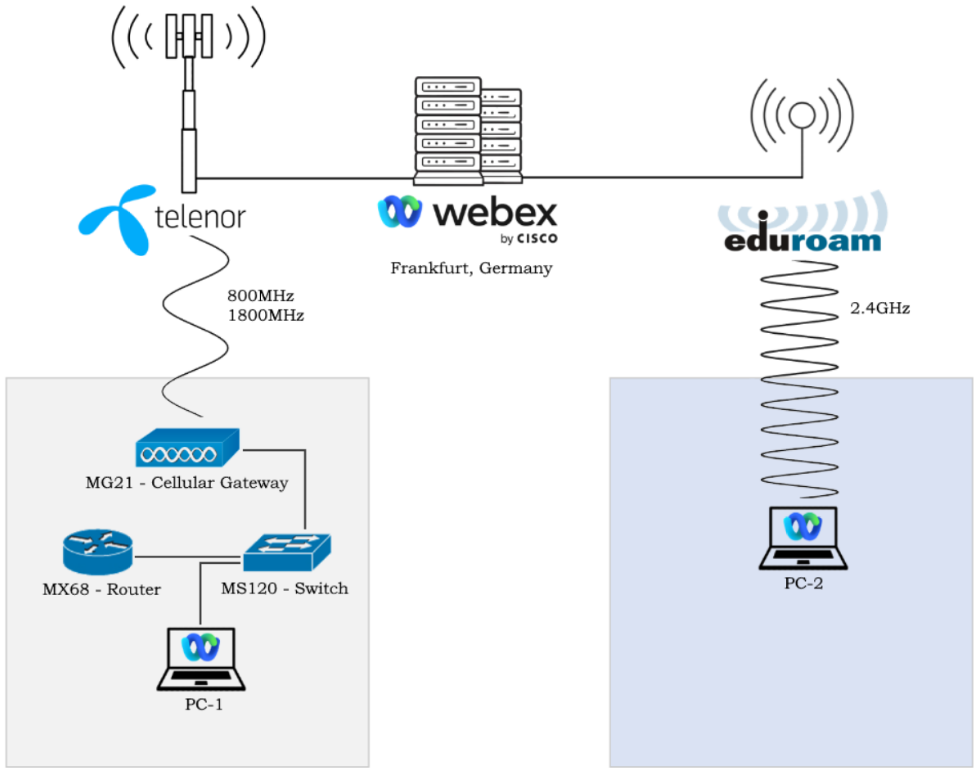

Figure 4.1: The cellular towers within a 500-meter radius of the NTNU campus. License: Dennis Jensen, CC BY

4.2.2 Scene Complexity

In a scene in regards to videoconferencing, there are two main subjects. The first is the motion created by the participant, and the second is background movement such as people or objects passing. The scene complexity is important to consider as it may cause bottlenecks during tests because of added bitrate requirements caused by background movement. This is closely related to I-frames and P-frames, as the latter will use less bitrate to update the video frames. If there is a lot of background movement, more I-frames are required, resulting in higher bitrate requirements as I-frames require more bandwidth than P-frames [31]. To make sure that this is not a factor for the test scenarios covered in this thesis, all tests will follow the same broad configurations to limit the effect of background motion causing additional I-frames. Webcams will be placed to capture props against walls that are unable to be disturbed by background motion.

4.2.3 Meeting Duration

The meeting duration of all meetings will have to be limited to a maximum of three minutes. Part of the testing is to capture packet data of the meetings that are held, and this will be achieved using Meraki's packet capture feature. The limiting factor of this functionality is the size of the files it creates. In order to keep file sizes to a minimum, meetings cannot exceed three minutes, and capture less than 100,000 packets each session. It's possible to filter the packets to UDP and Real-time Transport Protocol (RTP) since Webex mainly utilizes these protocols during a video conference [11], but this will also limit the analyzing section of the thesis. From the amount of data that is transmitted during ideal test scenarios, even three minutes will be too long to capture the required packets in a single file. Where applicable, meetings that have packet counts that exceed 100,000 will be split into two .pcap files.

4.2.4 Testing Structure

The structure of all the different test scenarios is important to cover in detail to avoid misunderstandings. There are four main testing locations that will be included in this thesis, more about these can be read about in subchapter 4.3. In terms of structure, there will be conducted a minimum of two recordings for each scenario. This is done in order to achieve a more even result, as tests can have a varying degree of unpredictability. There will be a baseline scenario that every test will be compared against. In addition to this, there will be conducted location specific baselines, but the data from these will not be recorded. This is done in order to figure out if a location has satisfactory results, and will serve as a point of reference. This is simply a retrospect, as some testing locations that have been conducted, are not a part of this thesis because the testing locations did not have satisfactory results.

Now that testing structure has been covered, the types of tests can be introduced. There are three main scenarios that will be recorded. The baseline scenario will serve as a meeting with ideal networking conditions and minimal latency, excellent audio and video metrics and stable connectivity for the entirety of the meeting. The second scenario is focused on latency, and will be referenced as the 'latency testing scenarios'. The point of these scenarios will be to provoke added latency, and typically utilize blockades such as bodies of water or concrete walls. The final type of scenario will focus on SNR. These types of scenarios will attempt to provoke a poor networking connectivity by interference of surrounding radio and cellular signals. These scenarios will be referred to as 'interference testing scenarios'.

4.3 Testing Locations

As mentioned in the previous paragraph, there are four main testing locations that have had satisfactory results in this thesis, and these are all detailed in this section. Additionally, there are tests that have to be conducted at a particular time during the day, and this will also be explained within this section.

4.3.1 Topas Building

This location will be used to conduct baseline test scenarios. Located on the fifth floor of the Topas building at NTNU campus Gjøvik, this location should have excellent cellular connectivity, no matter which cellular tower is utilized. Time of day will not be as important for the baseline tests, seeing as less people are present on the university campus during this period in time, but these scenarios will happen during the weekend to avoid most students and staff. From preliminary tests, before conducting the actual baselines, signals seemed marvelous, and by far the best witnessed at campus Gjøvik thus far.

4.3.2 Basement

The basement testing location is reserved for latency testing scenarios. Located in the basement of the Amethyst building, this location has severe obstructions as part of the basement's building materials. Concrete walls in every direction means cellular signals will have to be transmitted through these obstructions, and this will cause added latency for tests conducted within this area. With latency, higher amounts of traffic will only exacerbate the perceived latency clients have. For this reason, testing will primarily take place at 09-12 AM, as there are plenty of students present at this time.

4.3.3 Atrium

The atrium testing location is the second location that will be used for latency testing scenarios. This testing location will also focus on obstructions, and will use bodies of water to provoke latency in this regard. For most of these scenarios, this means blocking the cellular gateway by holding it close to one's body. Time of day is not essential, and will vary depending on the traffic in the Atrium. This testing location sees heavy traffic throughout the day, so a specific timestamp can therefore not be chosen.

4.3.4 Cafeteria

This is the final testing location, and it is conducted by the cafeteria of NTNU campus Gjøvik. This is a testing location reserved for interference testing scenarios, where the aim is to test SNR. As mobile devices within the surrounding area is a considerable factor for the interference testing scenario, time of day will have to be around peak lunch hour, which is within 11-12 AM on any given weekday.

4.4 Props

To avoid having to use people as subjects within conducted tests, props will be used instead. This will result in greater control exerted on the various test scenarios, as props will have more reliable and consistent movement for the duration of the meeting than any person can. Although the realism of a person versus any given prop will not be completely correlating, there is no requirement that tests have to include human subjects in this bachelor assignment.

In the initial stages of this assignment, one of the immediate problems was provoking movement within the scenes of video streams. To have a person wave throughout the meeting would no doubt become tiring after a couple of meeting, let alone a dozen. At the recommendation from Telenor, an idea of utilizing props for movement instead became relevant. In this instance, the suggestion was a rotating object, such as a Christmas tree, but a disco balls was chosen instead as it was easier to obtain. Disco balls are beneficial because they move at a constant pace, are very uniform objects and create enough movement to provoke the need for additional I-frames. The disco balls used during this thesis are same as the one in figure 4.2. The scene complexity matters a lot here, and in some sense disco balls have a negative aspect relating to the lights emitted behind itself. The light reflected upon the wall as part of the background will increase the amount of need for additional I-frames, but as long as this is consistent for every meeting, it should not impact results.

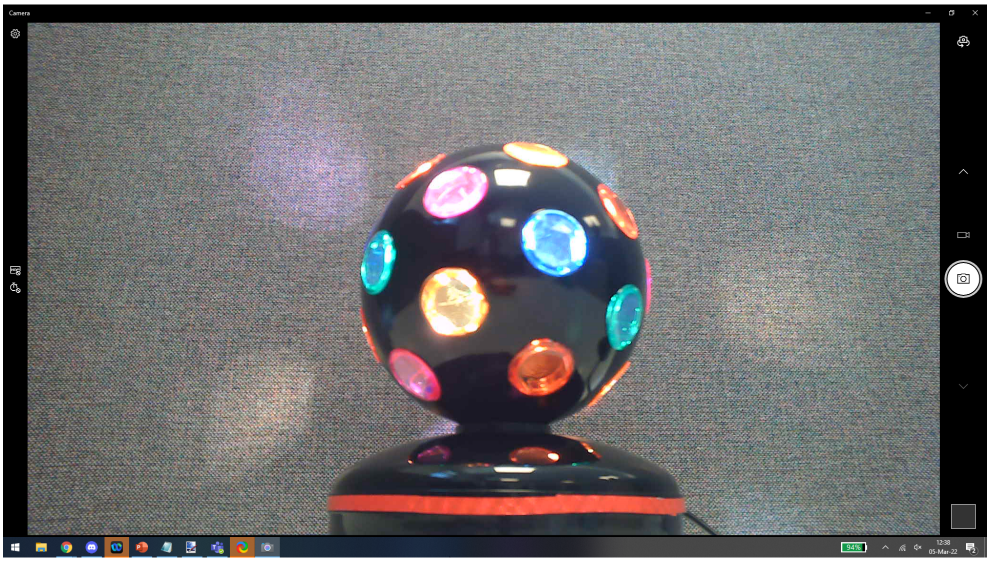

Figure 4.2: This is the same disco ball used for all of the test scenarios in this thesis. License: Ole Morten Ystad Karlsen, CC BY

4.5 Cellular Presence at Campus Gjøvik

Highly populated areas usually have multiple cellular towers located within the near vicinity of each other. This is done in order to achieve greater balance between connected devices, so that the amount of traffic any given cellular tower experiences doesn't bottleneck the user experience for these devices. There are three main ISPs in Norway that can provide cellular connectivity, as the bands for each ISP is heavily regulated [20].

For this thesis, all the test scenarios will take place at NTNU campus Gjøvik. This means that the cellular towers used during the testing phase are all located in the vicinity of the university campus. The MG21 cellular gateway utilized for network connectivity contains a SIM-card provided by Telenor, which only allows communication with Telenor operated cellular towers. As can be seen in figure 4.3, there are a total of three towers within a 500-meter radius of the university campus that can be utilized for the various scenarios, and there is no preference here. All three cellular towers offer the 800 MHz and 1800 MHz frequency bands.

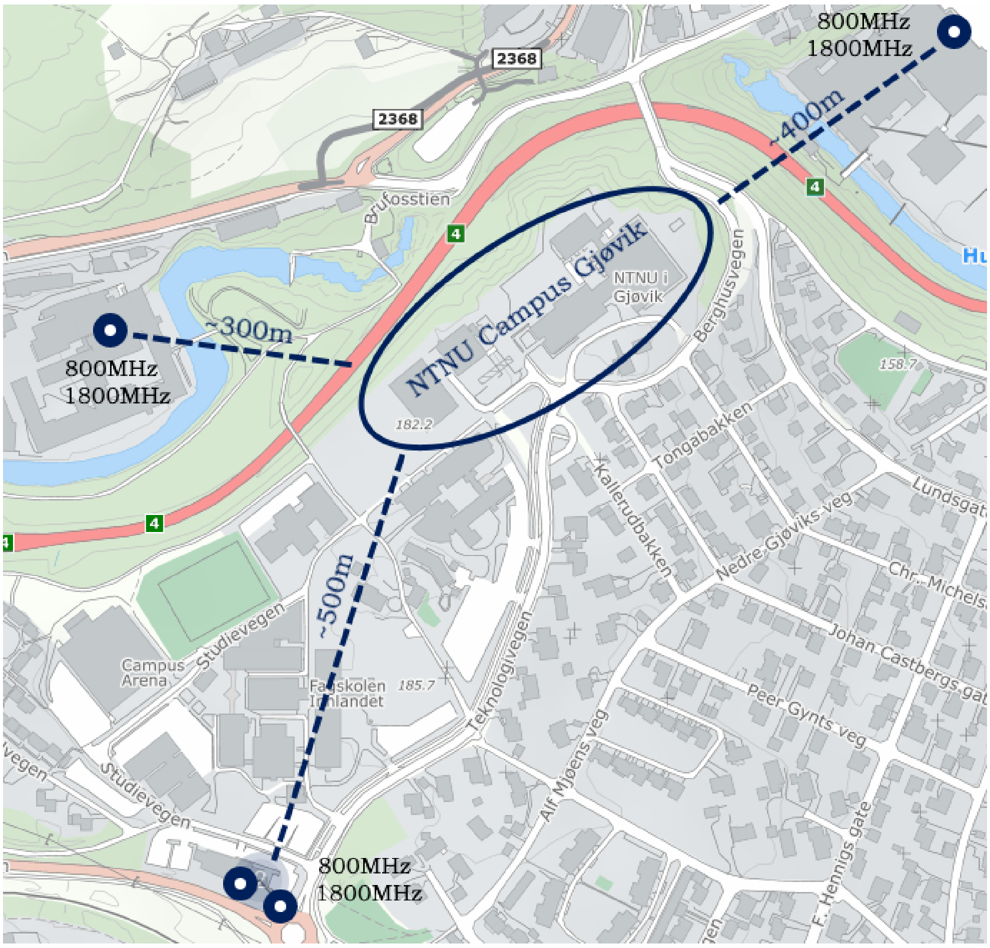

Figure 4.3: The cellular towers within a 500-meter radius of the NTNU campus. License: NKOM, CC BY

4.6 Baseline Scenarios

There are many different factors at play when conducting network tests like the ones performed in this thesis. Excessive load on a given cellular tower, location at which the receiving device is placed, surrounding Signal-to-noise ratio and which frequency the cellular tower offers. These are just some of the areas to consider. This scenario will function as a comparison to ideal results, and will be conducted until optimal results are obtained. The general metrics will be an average RSRP dBm of -85, latency of no more than 100 ms, jitter up to 30 ms, and packet loss should not be occurring at all.

4.6.1 Objective of a Baseline Scenario

The aim of recording a baseline reading, is to compare ideal conditions to subpar ones. By having an excellent reference point of what constitutes perfect meeting conditions, metrics that have a major impact on meeting quality are more easily identifiable.With the massive amounts of data and tools accessible through Webex Control Hub and Meraki Insight, graphs and various tables give keen insight into connection metrics that can explain either good or poor network quality.

4.6.2 Baseline Confirmation

The baseline test is essential, and to confirm that a reading is accurate, this test will be conducted multiple times. To ensure traffic volume and SNR will not be a limiting factor, baseline tests will be performed during the weekends. At this time, fewer people visit the university campus, lecturers and university staff included. Less traffic will naturally result in more favorable readings for a baseline test. By executing the same test multiple times, at multiple times throughout the day, the overall quality can be determined by averaging the different results, or by choosing the baseline with the best result. Conditions are supposed to be ideal, and for that reason alone, it might make sense to pick the best result available. On the other hand, the best result will undeniably skew the interpretation of what an achievable connection looks like, so an average might be more suitable in this regard. To achieve a more consistent result, averages will be recorded for all test scenarios, including baselines.

4.6.3 Baseline Structure

One of the meeting participants will be connected to the 4G setup provided by Telenor, utilizing Telenor's 4G network. The second participant will be connected to NTNU's campus WiFi, eduroam. To ensure there is adequate bandwidth available, speedtests will be performed throughout the testing phase, both before and after baselines have been recorded. Immediate results are made available in Webex Control Hub after a meeting has ended, and this information will also be studied to make sure the connections were optimal before concluding baseline results.

4.7 Latency Scenarios

There are multiple ways of weakening the perceived capabilities of an internet connection, and one of them is to increase latency. The time it takes for a signal to reach its destination will have a considerable impact on applications that depend on real-time communications. The added latency to either of the participating clients will make the connection seem slow. This is precisely what this test scenario will achieve, as obstructions are introduced to provoke latency, as can be seen in figure 4.4 below.

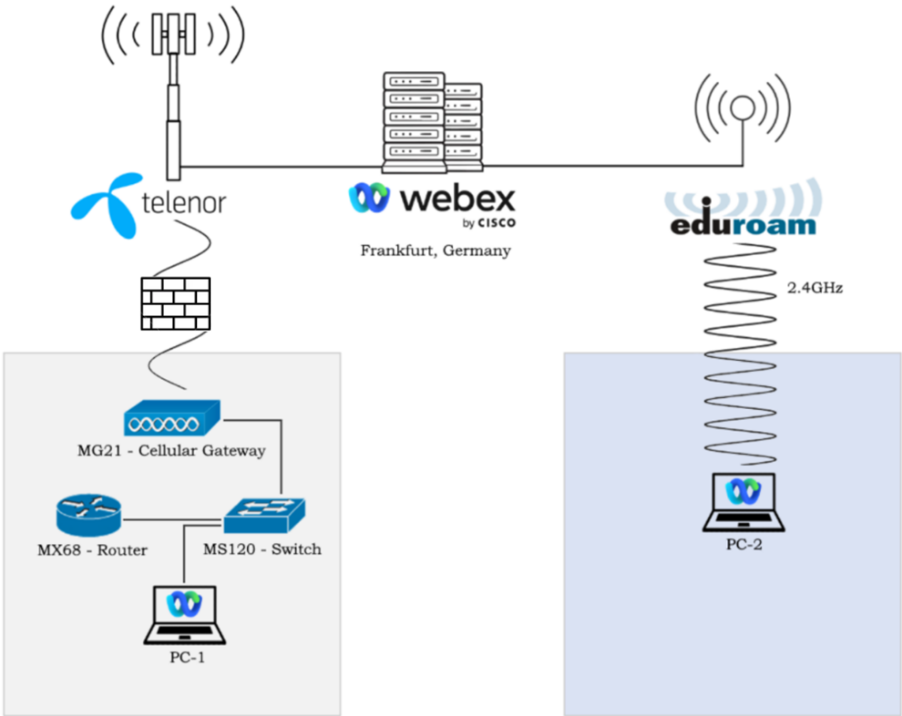

Figure 4.4: An overview of the intended obstructions between the 4G cellular tower and the MG21 cellular gateway. License: Dennis Jensen, CC BY

4.7.1 Purpose of Latency Scenarios

The ultimate goal of this test scenario is to highlight the importance that low latency has on videoconferencing, by provoking higher latency. Important metrics to note are ping, jitter and packet loss, but these scenarios will also factor in RSRP and RSRQ, even if these metrics will matter most in interference testing scenarios. From already having conducted multiple baseline readings, optimal values for all these metrics have been recorded. As such, as soon as the testing phase has concluded, information can be gleaned by correlating the baseline scenarios with the latency scenarios. If these correlations between latency and perceived quality of the video streams can be verified, the latency test scenarios will be deemed a success.

4.7.2 Confirming Results

As with baseline scenarios, to make sure the results from testing is not skewed in any way, tests are conducted multiple times. In addition to performing tests multiple times, they will also be performed at separate days, with timely intervals. Since the test meeting duration for each meeting is three minutes, performing several tests have little effect on time spent, but greatly improves test results as more data is collected.

4.7.3 Testing Structure

The testing setup will stay the same, and the only thing that will have changed from the baseline scenario is the location at which the test is conducted. The only external factor that is difficult to recreate is the scene complexity. The same backgrounds will not necessarily be available, but an attempt to recreate it will be made despite the colors being different. The most important aspect is that the percentage of the image that is covered stays the same. To mimic a real meeting, the percentage of the disco lamp that covers the camera view will be roughly 25%. An external tool called 'Meazure[4]' will be used to line up the lamp accordingly. The external meeting participant will continue to use eduroam.

4.8 Interference Scenarios

In addition to latency, a secondary type of scenario will be conducted to test interference. These scenarios will focus on SNR data, and executing tests where there are many people in a particular place. To determine the amount of SNR at any given time during a test scenario, RSRP and RSRQ values will be extracted from the local status page of the MG21 cellular gateway. As mentioned in subchapter 2.7, there are set values for RSRP and RSRQ that correlate between the displayed signal bar.

[4]More information about Meazure can be found on this page: `https://github.com/cthing/meazure`

4.8.1 Purpose

The purpose of this scenario is to research how the MG21 cellular gateway is affected by other wireless signals. Testing will include increasing the amount of nearby traffic, while watching RSRP and RSRQ data in real-time. This data is gathered from Meraki Dashbord. The goal of this scenario is to influence the SNR enough to have an effect on the meeting quality. Another desired result is having the MG21 cellular gateway change cellular tower based on the surrounding traffic being too high. Having one or more of these outcome occur, will be deemed as a success.

4.8.2 Confirming Results

From the already recorded baseline tests, normal RSRP and RSRQ values are known. If these metrics spike as the quality of the video and audio streams drop, conclusions can be drawn from graphs in Webex Control Hub. The most important factor is to provoke SNR enough to affect the meeting quality. Viewing the changes in connection to a cellular tower is also important. To track the active cellular tower, the use of an external phone application is required. The application is called Network Cell Info Lite, and was recommended by Telenor as a substitute for not being able to connect to the MG21 cellular gateway directly. Each test will be performed multiple times in order to capture the changes in traffic, and to study the effectiveness of the scenario.

4.8.3 Structure

As with the other scenarios, baseline and latency, the hardware used will not have changed. As with latency, however, scene complexity will also have to be taken into account. By using the equipment at different locations, lighting will be the biggest factor surrounding scene complexity. Attempts will be made to keep these as similar as possible to the baseline and latency testing scenarios. In this instance, the external participant will use eduroam.

Chapter 5

Results

This chapter will display and compare the results from the baseline, latency and interference testing scenarios. The data presented in this chapter will be used as a point of reference to the rest of the test scenarios, and is an introduction to the next chapter, where the main discussion and reflection will take place. As there have been several testing scenarios, and multiple recordings for each one, excess data is available as appendix A. Take note that all scenarios are referred to in their plural form, as redundancy means multiple tests are performed for each scenario. In some instances, clients will be differentiated between by their respective connectivity medium; PC-1 is connected to 4G and PC-2 is connected to eduroam. Unless stated otherwise, the point of view from these results in terms of which client is sending or receiving the data presented, will be from the 4G client, PC-1.
Due to the meetings only lasting for three minutes, there is not that much data covered in each test. The reason for this is because of a limitation in Webex Control Hub, that only updates points every minute. In addition to this, there are some inconsistent time frame measurements for capturing data in Webex. This leads to unreliable time frames that results in losing the last minute in some of the tests. The last minute then sometimes defaults to the last value captured. There are some examples of this in chapter 5.4.

5.1 Baseline

This section will display the results from the baseline scenarios covered in chapter 4.6. For each subsection, there will be an explanation into the general overview of the displayed results, and will serve as a first impression.

5.1.1 I- and P-frames

As seen in figure 5.1, there are no notable discrepancies in sent and received I-frames and P-frames, and the graphs are mostly stable throughout the tests. There are occasional dips, but this has no real consequence on video playback.

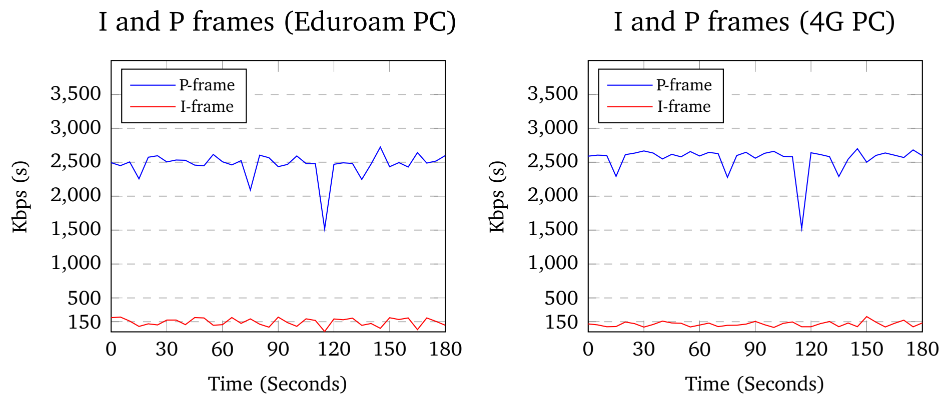

Figure 5.1: Sent and received I-frames / P-frames from the the PC using 4G. License: Knut-Magnus Karlsen, CC BY

5.1.2 Latency

As seen in figure 5.2, latency for audio and video are excellent, with no noticeable spikes. Even the initial I-frames for participants are not sufficiently demanding to lower the average delay for either video stream. Since the general rule of latency for these types of videoconferencing applications tries to stay below 150 ms, this meeting will have no negative results with an RTT of below 80 ms [13]. This test result went as expected, considering the low latency and the stability throughout the meeting.

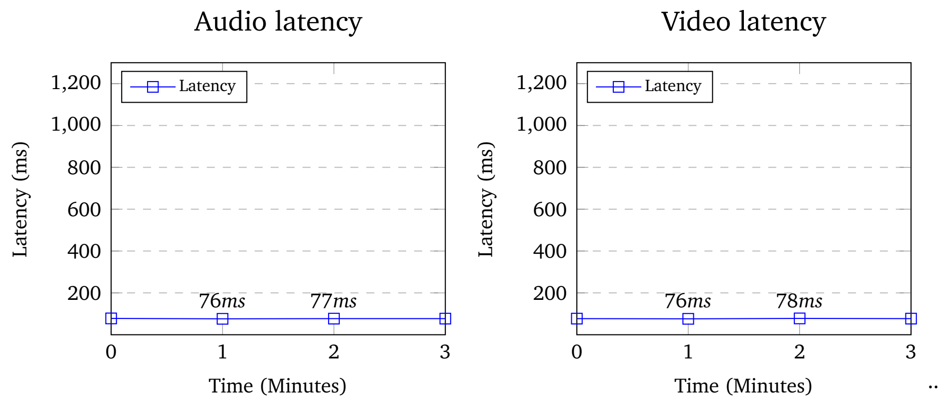

Figure 5.2: Latency graphs based on RTT data for baseline scenarios. License: Knut-Magnus Karlsen, CC BY

5.1.3 Packet Loss

With no disruptions and 0% packet loss overlapping both sending and receiving, this test result also went according to plan. As seen in figure 5.3, there are minimal obstructions between the MG21 cellular gateway and its receiving cellular tower, and as such this metric will not spike above 0%. As will become evident from upcoming test results, a packet loss percentage of less than 5% should still be adequate for most meetings, let alone 0%. This is certainly one of the metrics to monitor closely, as it's usually a good indicator of whether a Webex Meetings is struggling to upkeep the stream quality.

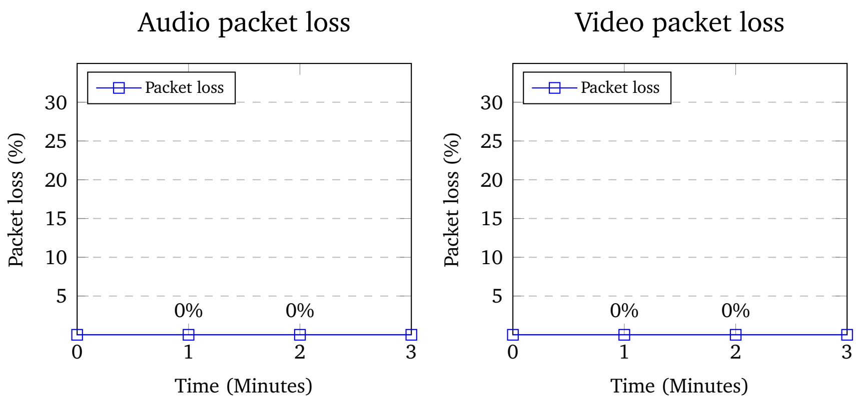

Figure 5.3: Average packet loss graphs for baseline scenarios. License: Knut-Magnus Karlsen, CC BY

5.1.4 Jitter

Jitter will traditionally be present when packets have to travel outside of a network, even if it does occur internally as well. As mentioned in chapter 2.5, video-conferencing applications will usually have a jitter buffer present, anywhere from 20-100 ms. As seen in figure 5.4 below, conditions for this meeting are excellent for both audio and video. With a jitter delay below 10 ms for both audio and video, jitter will not have an impact on the quality of this meeting.

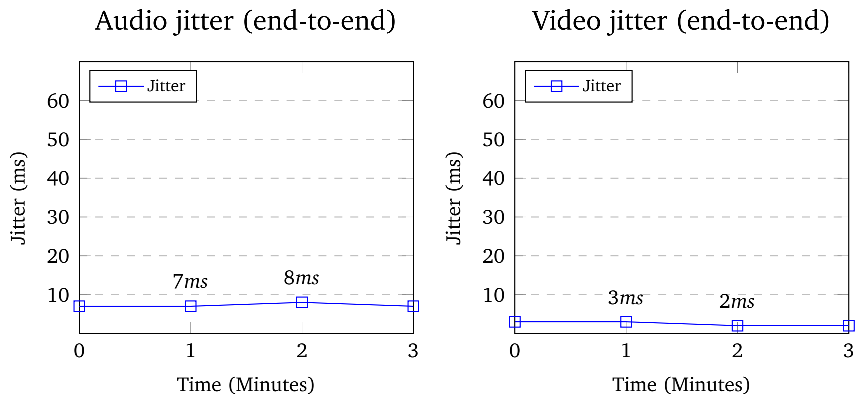

Figure 5.4: Jitter graphs for baseline scenarios. License: Knut-Magnus Karlsen, CC BY

5.1.5 Bitrate

As seen in figure 5.5, the graph contains the data being sent, and not received. While both could have been included, for all of the testing that was conducted, the upload capabilities were always the bottleneck. As such, for metrics where both options are available, sent data will be the main point of focus, except for with jitter, where end-to-end data is preferred. Evident by the two graphs, the bandwidth required for video will exceed audio many times over. This is self-evident, as more packets are required to transfer video as opposed to audio. After conducting several meetings, an average of around 2400 Kbps was found to be necessary for an ideal 720p video stream. As for audio, only 60 Kbps is required.

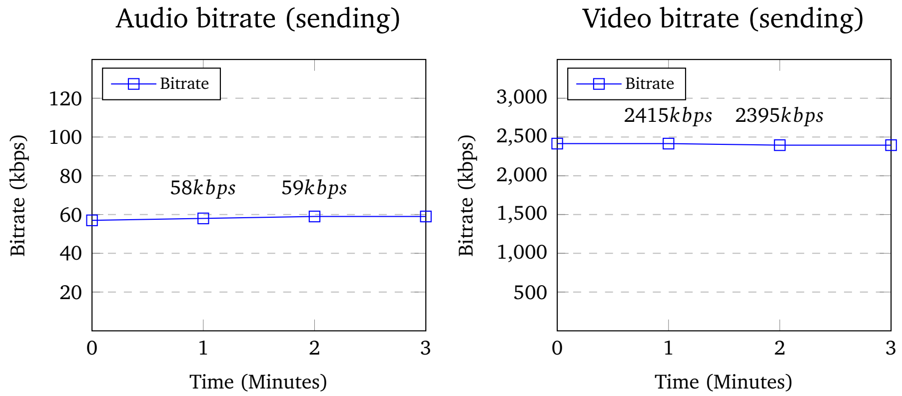

Figure 5.5: Bitrate graphs for baseline scenarios. License: Knut-Magnus Karlsen, CC BY

5.1.6 Video Quality

As mentioned in the previous paragraph, the video resolution for these meetings was stable at 720p, as can be seen in 5.6. While the received and sent frame rates differ sightly, at around 15 fps and 22 fps respectively, this occurs in every meeting, no matter how good the conditions are. Frames are mostly stable throughput the meeting, both for received and sent frames.

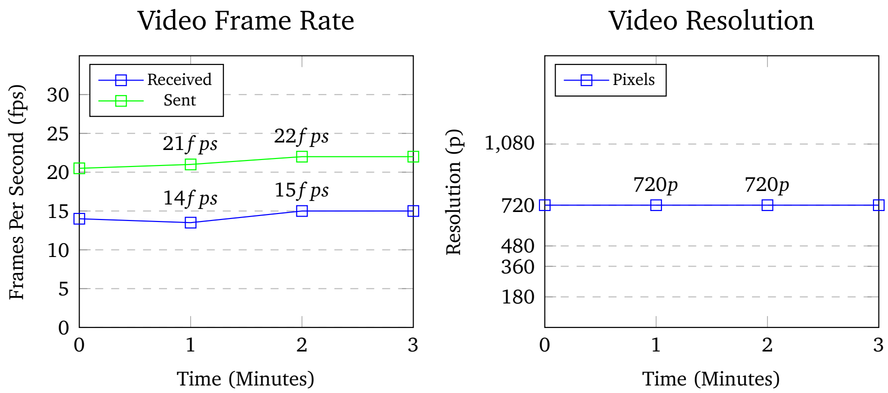

Figure 5.6: Video frame rate and resolution graphs for baseline scenarios. Knut-Magnus Karlsen, CC BY

5.1.7 SNR Data

The final metric that will be covered from the baseline is RSRP and RSRQ data, as shown in figure 5.7. As explained in chapter 2.7, RSRP and RSRQ values are better the closer they are to zero. That said, an RSRP average value of -90 dBm will be considered a good result. For all of the testing conducted, the lowest RSRP value thus far has been an average of -83 dBm. Not excellent, but good enough for regular meetings. An RSRQ average value of -7.5 dB is considered excellent, and is in line with the expected result.

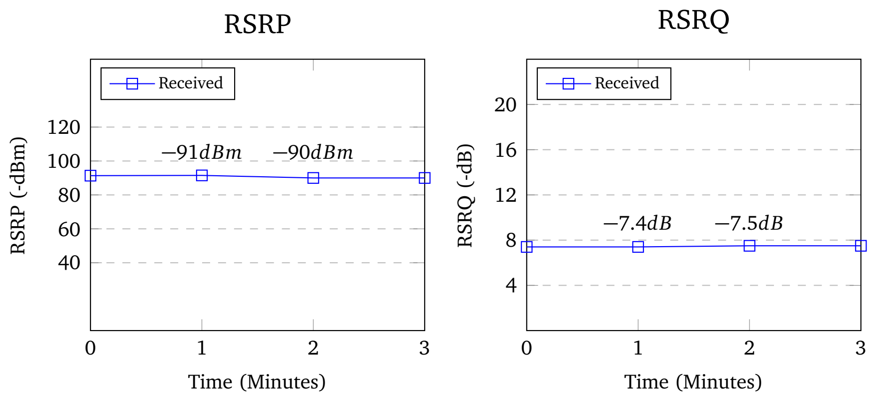

Figure 5.7: RSRP and RSRQ graphs for baseline scenarios. Knut-Magnus Karlsen, CC BY

5.1.8 Baseline Conclusion

Based on the data from the graphs provided in this section, the baseline scenario has went according to plan. Results have been satisfactory for all metrics, and this was also evident when conducting the actual meetings. High frame rates, high resolutions and negligible latency made for excellent meeting conditions, and is a good data set for later comparisons. The only metric that could have been improved was RSRP, but with a variance of around -5 dBm, this is negligible in the larger scope of all the tests. It's something to keep note of for the other scenarios, and fluctuations will be monitored accordingly.

5.2 Basement in the Atrium Building

The atrium basement scenario is a part of the latency testing scenarios, and will focus on obstructions to provoke poor connectivity. Being surrounded by concrete from all angles, this scenario will experience issues when transmitting signals through the mentioned obstacles. Materials such as concrete are known to be detrimental to cellular connectivity[1].

5.2.1 I- and P-frames

As seen in figure 5.8, the overall Kbps for the I-frames and P-frames has reduced when comparing it to the baseline. The sent and received I-frames and P-frames have also reduced drastically, but it is still able to maintain a video feed.

[1]According to this study that calculates the loss of signals from a Wireless Sensor Network passing through concrete, it proves that concrete has the ability to block wireless signals: [32]

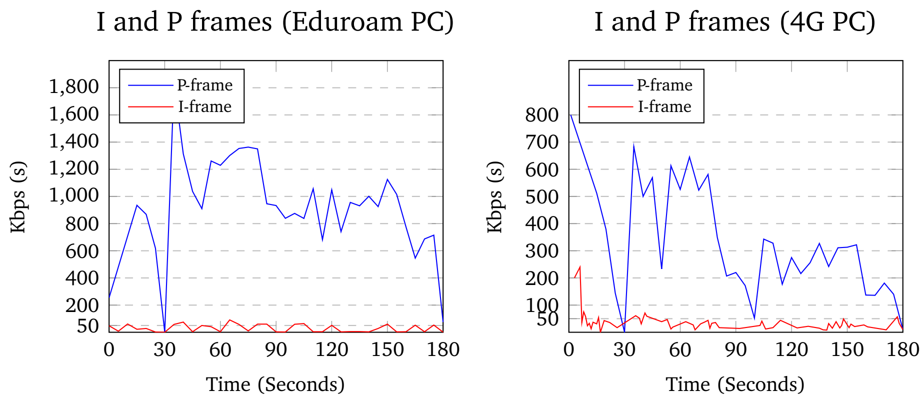

Figure 5.8: Sent and received I-frames / P-frames from the the PC using 4G. Knut-Magnus Karlsen, CC BY

5.2.2 Latency

This is the main metric that the latency testing scenarios attempts to provoke, and judging by the graphs seen in figure 5.9, the tests were successful. A latency RTT of above 400 ms will start to have major impact on the meeting, and obviously in a negative way. In fact, the latency from these tests is so major, that questions relating to the authenticity of this data can be raised. When attending these meetings, the quality seemed poor, but video was still being transmitted and displayed. While not as fluid as the baseline tests, it was still deemed at an acceptable level. This data does not seem to corroborate the perceived impressions from participating in these meetings, and this is something that will have to be looked at more closely in chapter 6, correlating Wireshark data to these metrics.

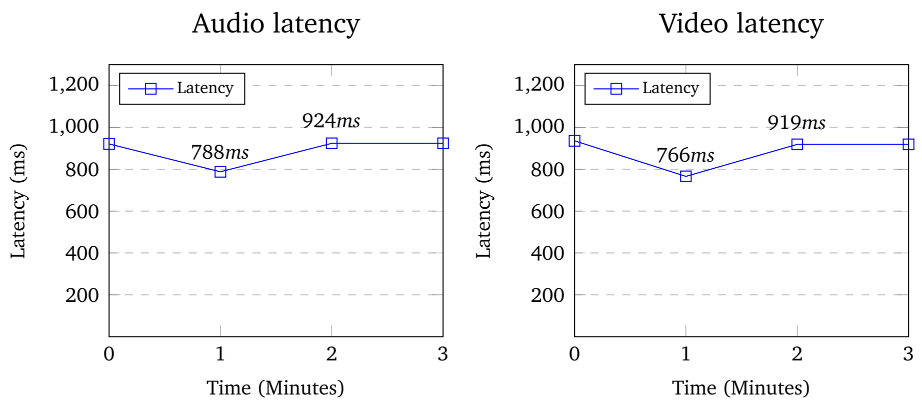

Figure 5.9: Levels of latency from the basement in the Atrium building. Knut-Magnus Karlsen, CC BY

5.2.3 Packet loss

The packet loss from the graphs seen in figure 5.10 is interesting, as there are major occurrences of packet loss in the early stages, but the graph flattens out approaching the midway point. This makes more sense upon looking at the bitrate results, found in subchapter 5.2.5. The bitrate initializes at around 500 Kbps, before it plumments to 150 Kbps. When a meeting participant first joins, there's an influx of I-frames, which require more data than P-frames, and this is likely one of the factors to the initial increase in packet loss of 12% for video.

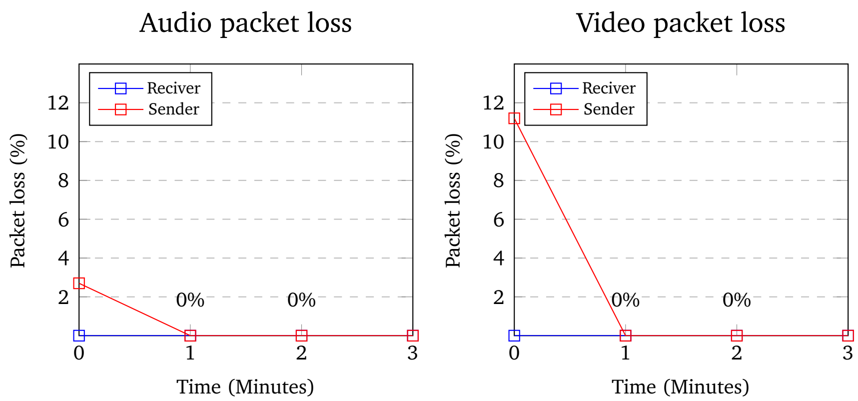

Figure 5.10: Percentage of packet loss for both audio and video from the basement in the Atrium building. Knut-Magnus Karlsen, CC BY

5.2.4 Jitter

As previously mentioned, a delay of 20-100 ms of jitter is an acceptable level in most cases. When looking at the graphs displayed in figure 5.11, the level of jitter present for both audio and video maintains at a stable level for the entire duration. Since neither of these graphs exceed 10 ms, jitter can be written off as a contributing metric for this meeting.

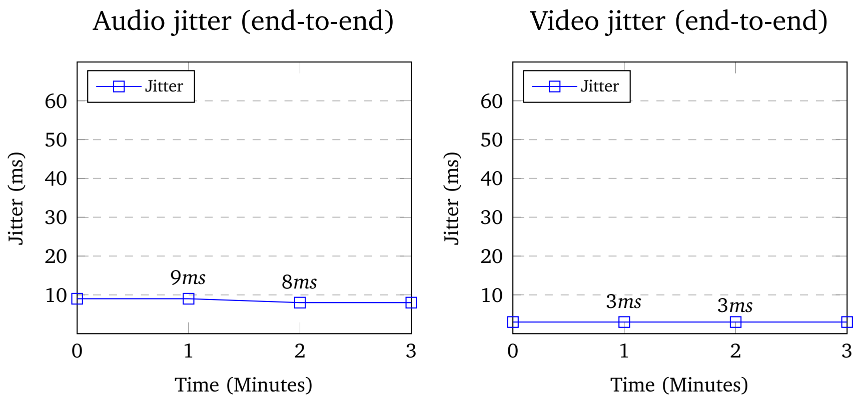

Figure 5.11: Jitter levels from the basement in the Atrium building. Knut-Magnus Karlsen, CC BY

5.2.5 Bitrate

The bitrate for these test scenarios differs somewhat to other test scenarios. Looking at the graphs in figure 5.12, the audio starts off lower than usual, and climbs within the first minute. For video, it's the other way around. The client attempts to set the bitrate at a higher value than what the cellular gateway is able to transmit. As a result, packet loss occurs, as referenced in the previous section.

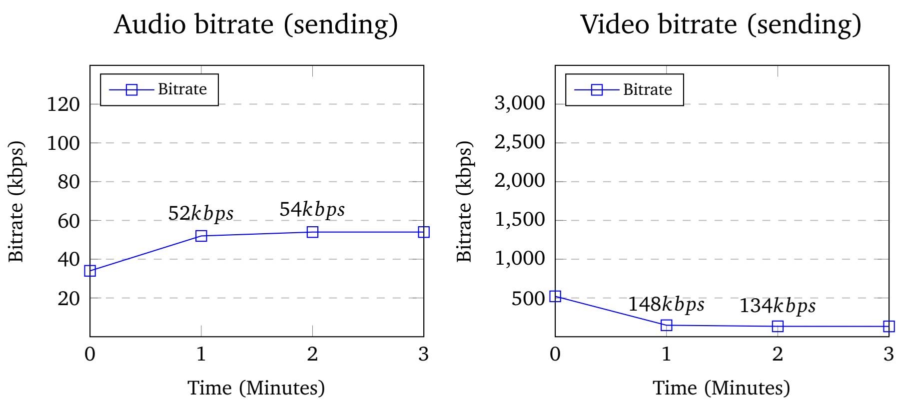

Figure 5.12: Bitrate levels from the basement in the Atrium building. Knut-Magnus Karlsen, CC BY

5.2.6 Video Quality

The video quality displayed in figure 5.13 means this meeting differs slightly from the baseline tests, but mostly in terms of sending data. While an fps averaging around 10 isn't fantastic, it's generally acceptable if there is no additional factors to consider. That said, with a latency as high as 900 ms RTT, video will no doubt have an easier time falling out of sync. The resolution is also poor, as 180p is the lowest resolution that Webex Meetings offers, except for thumbnails, which supports 90p. All in all, the video quality caused this meeting to have a fairly poor user experience.

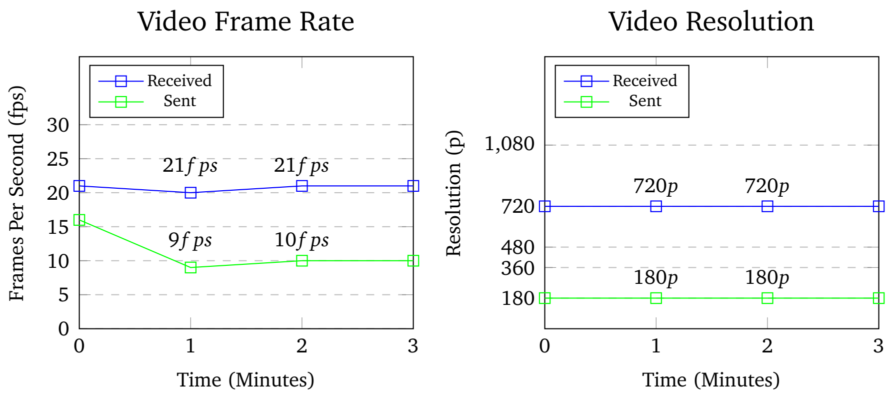

Figure 5.13: Video quality represented in video frame rate and resolution from the basement in the Atrium building. Knut-Magnus Karlsen, CC BY

5.2.7 SNR data

The final graphs from the atrium basement tests cover SNR between the cellular gateway and the respective cellular tower it's connected to. Looking at the graphs in figure 5.14, the values of RSRP are bordering fair to poor judging by the reference model introduced in chapter 2.7. Looking at the values displayed in the RSRQ graph however, shows that the quality of the signals is very good. Comparing the values seen in the baseline, there is a difference of about -0,5 dB. This means that the difference of using concrete to block signals has little effect when it comes to signal quality. Due to the fact that results are below 10 dB, the signal quality for this test can be deemed excellent. In testing, disruptions such as physical obstructions seem to have little effect on RSRQ, and RSRP seems to be more variable.

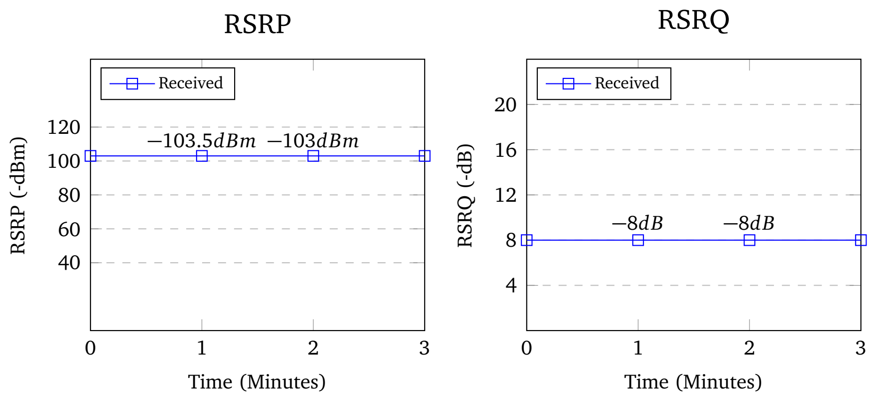

Figure 5.14: SNR data represented in RSRP and RSRQ from the basement in the Atrium building. Knut-Magnus Karlsen, CC BY

5.2.8 Test Conclusion

After looking at the data displayed in this subchapter, there is little doubt that concrete has an exceptional ability to disrupt cellular signals, and in turn affects the quality of videoconferencing. While the overall quality has diminished by a fair margin compared to the baseline scenarios, latency remains the most compelling metric from the atrium basement tests. In theory, averaging above 800 ms RTT should greatly affect the quality of a meeting, as many of the metrics will be affected by each other. If latency is high, packet loss and jitter are not far behind it in struggling. To corroborate these test results, Merkaki packet captures will be analyzed in Wireshark to eliminate the possibility of any discrepancies, but this will be discussed in chapter 6.

5.3 Campus Cafeteria

Although the previous subchapter regarding the atrium basement tests had increased latency, unanswered questions regarding SNR remain. The test results in this section will focus on SNR, and is a part of the interference testing scenarios. As SNR metrics can be provoked by how much network traffic and how many devices there are within close proximity, it can be difficult to test. The purpose of these results is to determine if SNR, and more specifically RSRP and RSRQ, will increase by having more devices and general traffic within the surrounding area. As detailed in chapter 4, these tests are conducted during the lunch period at NTNU's main cafeteria.

5.3.1 I-frames and P-frames

This scenario had a reduction in received I-frames and P-frames when comparing it to the previous basement scenario, while it had an increase in sent I-frames and P-frames. This can be seen in figure 5.15 below. The Kbps for both the sent and received I-frames and P-frames was lower than the previous scenario. Having a lower Kbps of the frames results in a worse image quality. However, the frequency and number of frames sent results in the video feed being more seamless, while increasing the bandwidth usage.

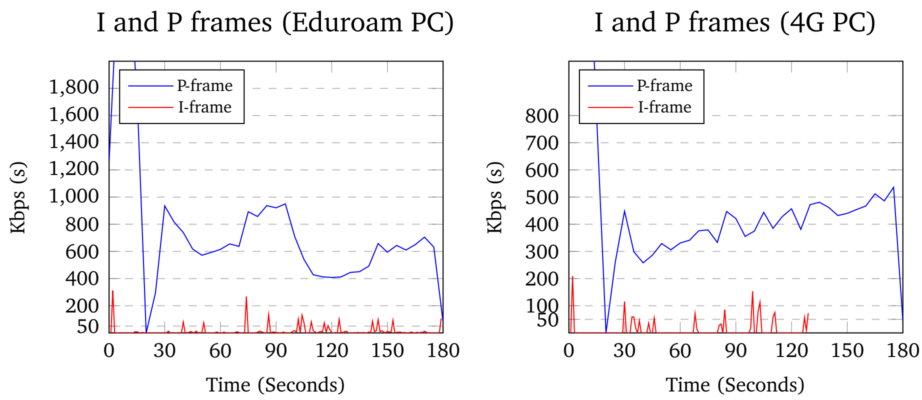

Figure 5.15: Sent and received I-frames / P-frames from the the PC using 4G. Knut-Magnus Karlsen, CC BY

5.3.2 Latency Effect

Looking at figure 5.16, the latency being between 300 ms and 600 ms is considered poor, but these are the lowest latency values observed apart from the basline tests. This can be explained by the surrounding materials, which consists of less concrete, and more windows. It clearly affects the tests, as the latency would not be this high otherwise. That said, these values are closer to the threshold of what is considered acceptable quality for videoconferencing, although they are on the poorer side. Similarly to the atrium basement tests results, latency is higher than the video stream would indicate.

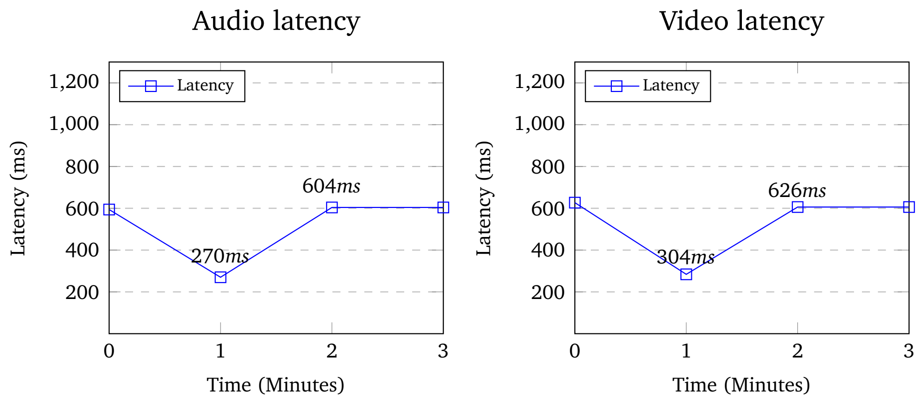

Figure 5.16: Levels of latency from the cafeteria at campus NTNU. Knut-Magnus Karlsen, CC BY

5.3.3 Packet Loss

Different from the atrium basement test results, the graphs in figure 5.17 indicate that there's a higher presence of packet loss at this location. While packet loss below 2% can generally be seen as negligible, it appears midway throughout the meeting, and not at the very start, as with the atrium basement results. The video from the cafeteria tests did seem to drop some frames, at least more than the reported 1% seems to indicate.

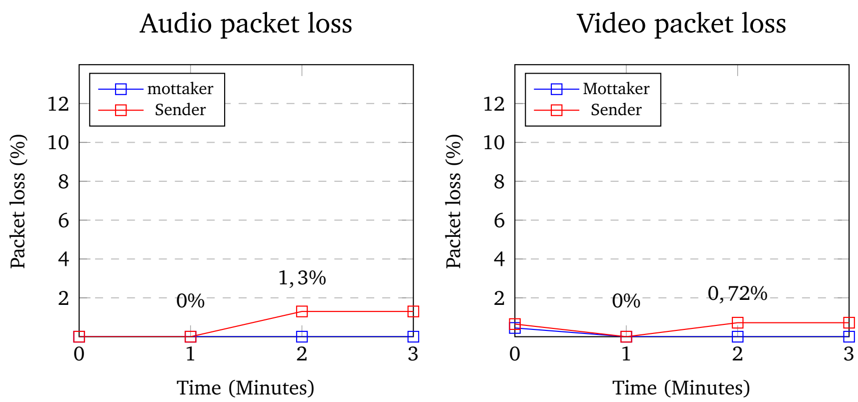

Figure 5.17: Levels of packet loss from the cafeteria at campus NTNU. Knut-Magnus Karlsen, CC BY

5.3.4 Jitter

The jitter is between 8 ms and 26 ms as displayed in figure 5.18. This is still considered good, as most videoconferencing applications will have a jitter buffer present; a range of 20-100 ms will be acceptable. The amount of jitter continues to rise throughout the meeting, ending on 26 ms, when latency reaches 600 ms.

While audio and video can start to see minor side effects at around 30 ms of jitter, this should still be too negligible to be considered a decrease in perceived meeting quality, and was not something that was witnessed for these tests.

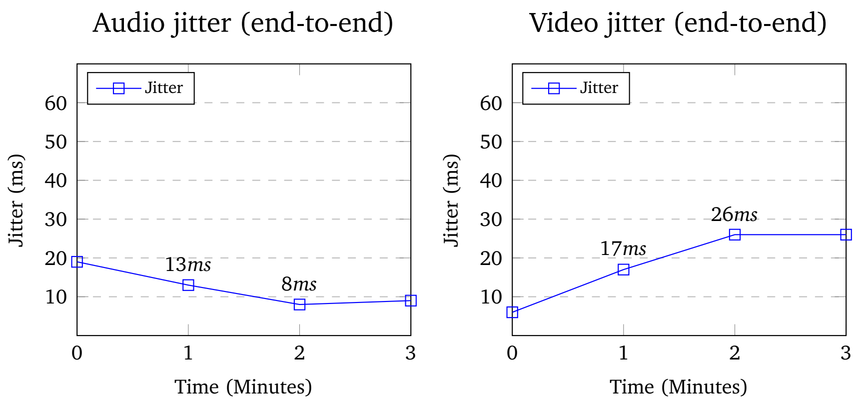

Figure 5.18: Levels of jitter from the cafeteria at campus NTNU. Knut-Magnus Karlsen, CC BY

5.3.5 Bitrate

By examining the graphs in figure 5.19, the overall bitrate for video is fairly steady around 270 Kbps following a slight correction. As the bitrate is lower, the overall quality of the video will decrease accordingly. For audio, however, the opposite happens. A plausible hypothesis is that the Webex Meetings application attempts to prioritize audio over video quality in this instance.

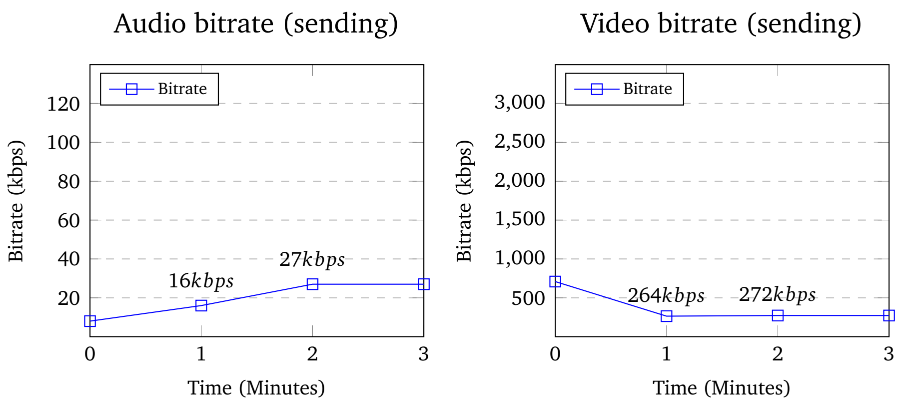

Figure 5.19: Levels of bitrate from the cafeteria at campus NTNU. Knut-Magnus Karlsen, CC BY

5.3.6 Video Quality

As seen in figure 5.20, the video metrics for these tests are fair to good, at least in terms of frame rate. The resolution is towards the lower end, dipping as low as 180p for the initial and final part of the tests. The hypothesis mentioned in the previous section is strengthened by the fact that the video resolution drops back down to 180p for the last portion of the tests.

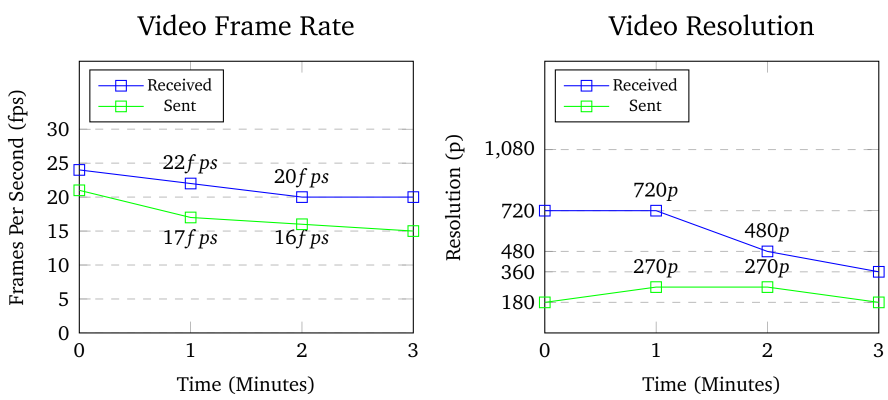

Figure 5.20: Levels of video quality from the cafeteria at campus NTNU. Knut-Magnus Karlsen, CC BY

5.3.7 SNR data

The RSRP and RSRQ metrics are what piques most interest from these results. It is far from what the expected result for this type of scenario. As shown in figure 5.21, there's not much of a difference compared to the results from subchapter 5.2.7, where the RSRP value from these tests are actually lower by half a point. As far as RSRQ goes, it differs by two points, but this is not substantial. As previously explained in chapter 2.7, the RSRQ value primarily acts as a backup if the device is not able to make a sufficient cell selection based purely on the RSRP metric alone. As pointed out in the basement scenario having an RSRP value of -103dBm is not optimal, and considered as a fairly poor connection.

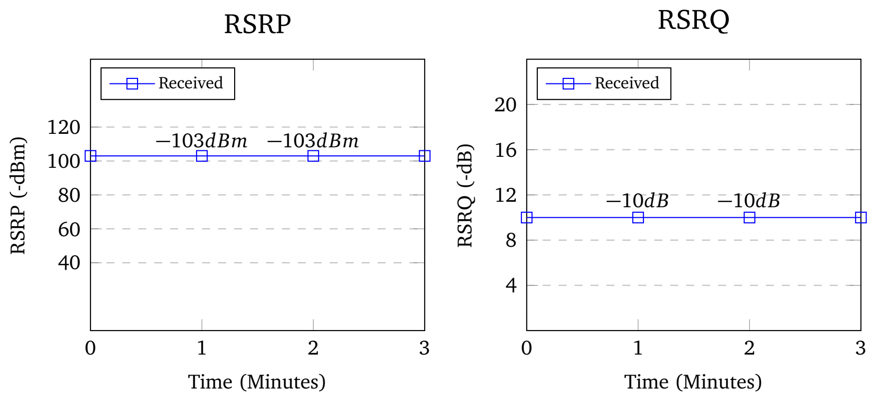

Figure 5.21: Levels of video quality from the cafeteria at campus NTNU. Knut-Magnus Karlsen, CC BY

5.3.8 Test Conclusion

The purpose of this test scenario was to see how the SNR metric affects a connection and a digital meeting. However, the test results did not give as big of a difference as intended in terms of SNR when compared to the other testing scenarios. A reason for why the SNR does not differentiate by much is because it needs a lot of devices and noise to provoke the metric. As there are 3 cellular towers nearby Campus Gjøvik providing 800 and 1800 bands it proved troublesome to single out SNR and to increase the metric with the resources that were available. This will be further explained in chapter 6.

5.4 Atrium With Obstructions

Upon realizing that some of the previous testing locations were not giving the intended results, new ideas were necessary. From earlier test scenarios, the cafeteria and the atrium basement were medium to poor at best, and with additional obstructions, even worse signals could be achieved. One of the very first test scenarios that were conducted, was to provoke poor connectivity by using a person to obstruct the signal. Functioning mostly as a body of water, signals were indeed exceedingly difficult to send and receive when blocked by a person. This test scenario was therefore conducted to recreate some of the connectivity problems with this method, essentially body blocking the cellular gateway.

5.4.1 Latency

As is evident in figure 5.22, test results are even worse than suspected. With a latency of up to five figures, the latency during these tests was tremendous. This is by far some of the worst user experience from all the testing that has been conducted. With a latency of this magnitude, averaging around 15,000ms for the duration of the meeting, even one-way communication ceased. While the connectivity was technically still there, conditions such as these can not be recommended for any type of meeting. The video stream proved to have a difficult time keeping up, but latency is not the only factor here.

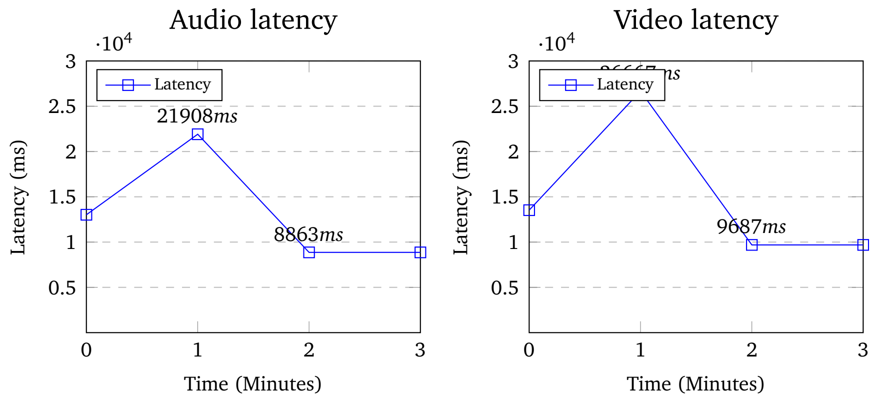

Figure 5.22: Latency graphs for Atrium With Obstructions scenarios. Knut-Magnus Karlsen, CC BY

5.4.2 Packet Loss

From the earlier test locations and scenarios, the highest packet loss percentage barely broke 10%. This was the main reasoning behind carrying out an additional scenario, where the main focus would be to provoke a higher percentage of packet loss. In some sense, the mission was successful, but in another sense, maybe by too much. With the amount of packet loss experienced in this test scenario, as can be seen in figure 5.23, it was a miracle the connection didn't cease entirely, at least for the first half of the meeting. When the throughput gets lowered, as will be evident in the next couple of graphs, the packet loss drops alongside it.

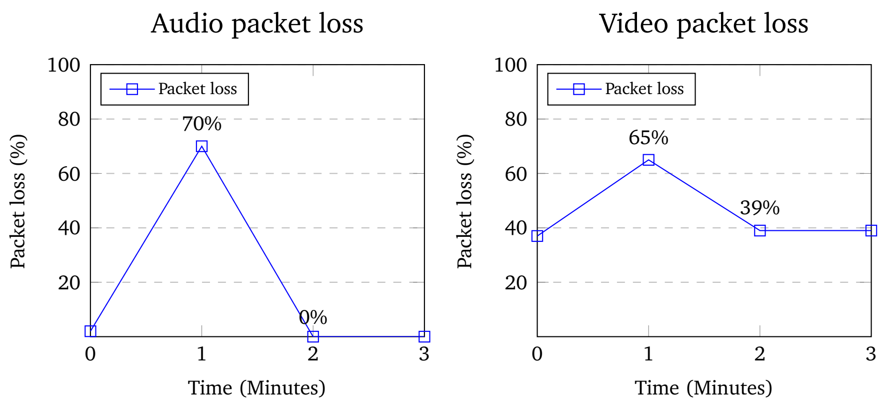

Figure 5.23: Packet loss graphs for atrium scenarios. Knut-Magnus Karlsen, CC BY

5.4.3 Jitter

As can be seen in figure 5.24, with jitter as high as 200ms, Webex is surpassing its maximum acceptable jitter threshold, and by quite a margin. As explained in chapter 2.5, on the characteristics of jitter, applications usually have an acceptable upper threshold of around 20-100ms. Almost doubling the normal threshold, jitter will have a big impact on streaming quality. To what effect, will be discussed in the next chapter, where test results are reflected upon, alongside all the other metrics.

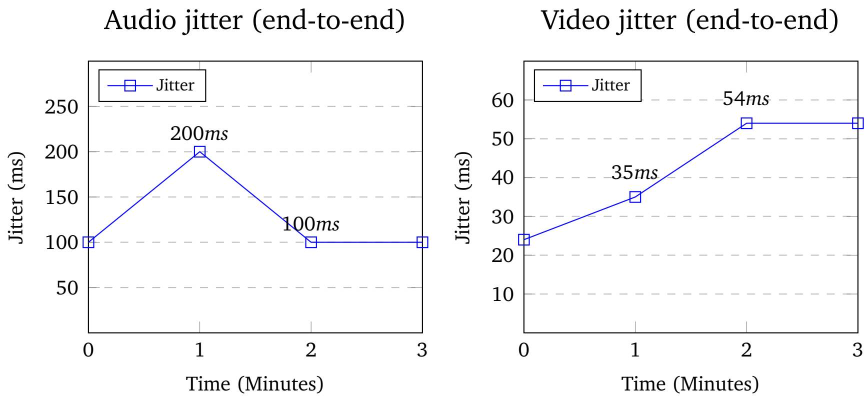

Figure 5.24: Jitter graphs for atrium scenarios. Knut-Magnus Karlsen, CC BY

5.4.4 Bitrate

For the entirety of the meeting, Webex was struggling to present data to the participants, on both ends. As can be seen in figure 5.25, the bitrate was already low from the very beginning of the meeting. Around the midway point, the bitrate plummets, which makes more sense when looking at the next section, where resolution is displayed.

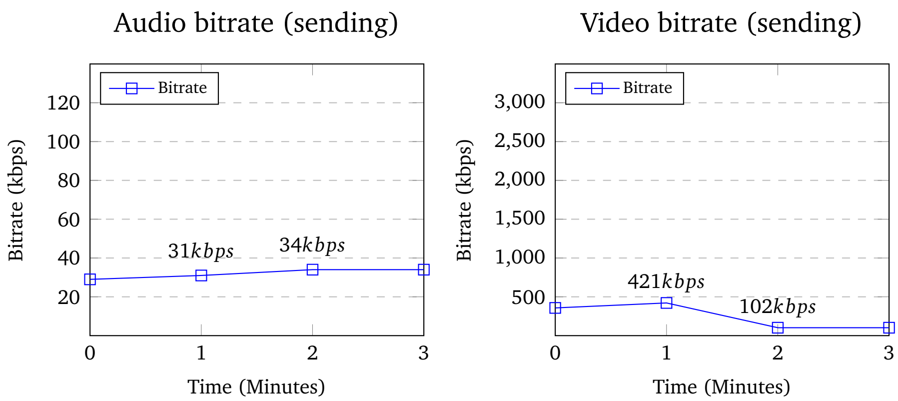

Figure 5.25: Bitrate graphs for atrium scenarios. Knut-Magnus Karlsen, CC BY

5.4.5 Video Quality

As far as video metrics go, the results presented in figure 5.26 are surprisingly adequate. With an average frame rate (receiving) of around 27 for the duration of the tests, video should be perceived as relatively smooth, at least on paper. The reality, however, was quite different for the participants. During the meetings, frame rates were so low that participants were able to count the amount of frames that were actively refreshing. This is evident in the data, where the frame rate drop as low as a single frame per second. The resolution is also struggling to maintain throughout the meeting, and drops to the lowest resolution that Webex Meetings offers, 180p, around the midway point.

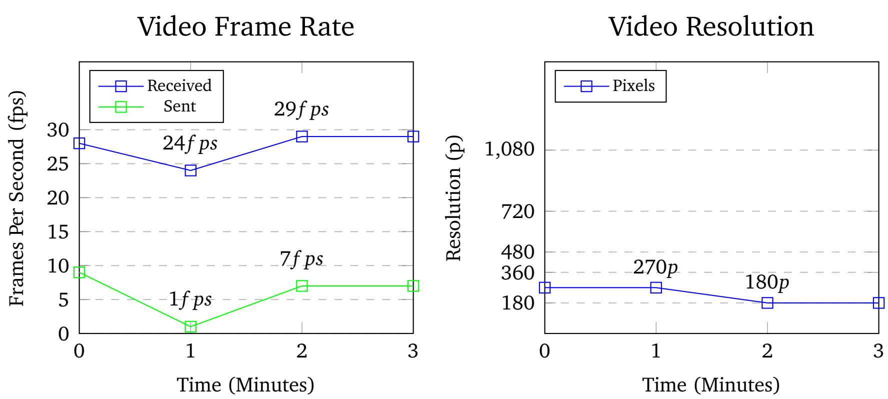

Figure 5.26: Video frame rate and resolution graphs for atrium scenarios. Knut-Magnus Karlsen, CC BY

5.4.6 I-frames and P-frames

As seen in figure 5.27, there are only a few spikes of sent and received I-frames and P-frames. During the entirety of the meeting there were only sent 82 I-frames packets. This resulted in the video having more artifacting than the other scenarios and was perceived as being choppy. Having less I-frames means that the whole video image updates less frequently, which causes bottlenecks as explained above. A screenshot of how this looks on a video feed is shown later in chapter 6.

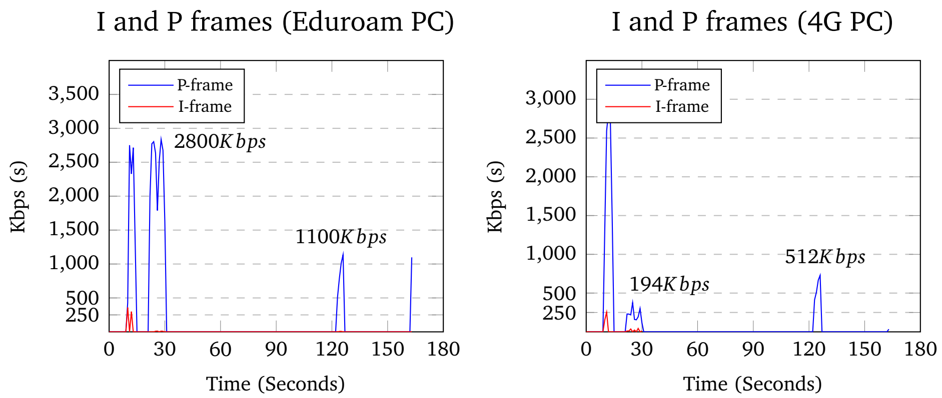

Figure 5.27: Sent and received I-frames / P-frames from the the PC using 4G. Knut-Magnus Karlsen, CC BY

5.4.7 RSRP and RSRQ

As seen in figure 5.28 below, the RSRQ value is mostly unchanged from baseline readings, but RSRP's value has increased dramatically. With an average dBm of around -85 for baselines, a value of -108 dBm from this scenario is a striking comparison. An RSRP value of >-100 will be considered fair to poor, and is the lowest from all the different scenarios covered.

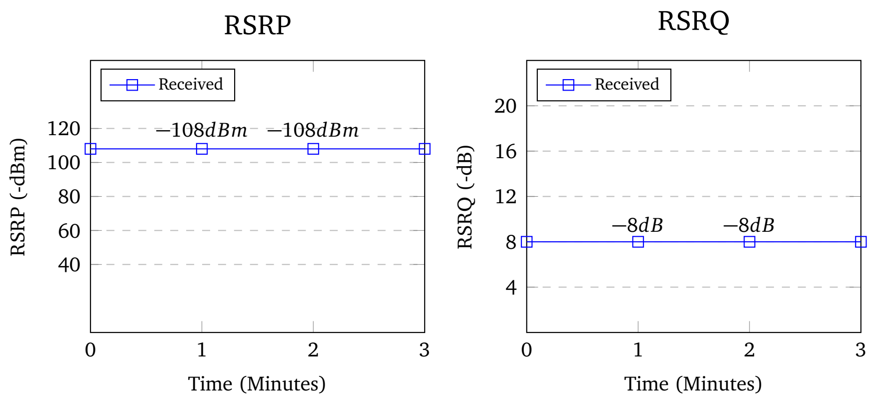

Figure 5.28: SNR graphs for atrium scenarios. Knut-Magnus Karlsen, CC BY

5.5 Summary and Result Comparisons

While most of the test scenarios went about as expected, there are still some aspects of certain tests that need to be investigated. More specifically, the latency seems unreasonably high for both of the atrium scenarios. There is also a rather lacking amount of packet loss up until the final test scenario in subchapter 5.4. That said, the data has been a good guideline to how meetings have went, and for the most part, has lined up with the expected results prior to conducting these tests. The Network Cell Info Lite application turned out to not provide much useful information, due to inaccurate data capture.

One of the biggest factors to consider from all the data contained within this chapter, is the difference between uplink and downlink capabilities. The upstream connection has always had a fraction of the available bandwidth, compared to downstream connection. This has influenced how the video streams originating from PC-1 looks, being fairly bad for most of the latency and interference testing scenarios. From the point of view of PC-1, however, the stream originating from PC-2 has been mostly fine as the downstream connectivity has been plentiful.

Chapter 6

Discussion

In the first chapter of this thesis, two hypotheses were introduced, and in this chapter these will both be discussed. The first one is as follows, *when the Webex Meetings platform has to compensate and lower the video metrics to be able to upkeep a continuous video stream, this can be used as a definition of a poor network connection.* After having detailed which technologies and services are important to explore this statement, this chapter will focus on the results obtained in chapter 5 to back up reflections made in relation to this hypothesis. Secondly, the reasoning behind the layout and procedures for each type of test scenario will be covered in this chapter, which ultimately led to the results seen in chapter 5. When on the topic of chapter 5, the second hypothesis is also relevant for this chapter, *when Webex Meetings is affected by subpar networking conditions, video bandwidth is deemed more susceptible to reduction than audio, and a greater effort will therefore be put into maintaining a consistent audio stream.* This section will discuss how the Webex Meetings platform behaves on an underlying level, and why the platform makes the adjustments it does. It will be supported by the results extracted in chapter 5, and also have ties to the first section of this chapter, regarding the first hypothesis.

6.1 Part 1: Poor Network Connection Characteristics

When contemplating a statement such as "what does it mean to have a poor network connection?", the answer will be highly subjective based on a multitude of factors. In relation to services hosted on the world wide web, one of the most common ways people will answer this is by pointing to latency. Examples here can be websites taking too long to load, or video streams buffering amid playback. In relation to videoconferencing however, expected answers would be inconsistency relating to both audio and video. A sure sign of poor network performance would be signal loss for a prolonged period of time, or severe stuttering and artifacting. These are telltale signs that something is amiss, but what are other signs that a network connection would be at fault?

6.1.1 Video Data

The difference between video resolutions can be readily apparent to most users viewing a video medium, based on the variation in pixels before and after the resolution has changed. For instance, if a video stream went from 720p to 360p in a moment, most users will likely be fairly observant of such an event. Since there are exactly four times as many pixels in 720p as opposed to 360p, the change will be quite striking based on the pixel count alone. Such a change will plummet the sending or receiving bitrate, depending on which aspect of the network is affected, the downlink or Uplink. While this change would be quite obvious for many, there are also occasions where the signal starts at a lower video resolution than 720p. If the jump from one resolution to another is minuscule enough, by the amount of pixels, the participants may have more issues differentiating between the two resolutions. The correlations between rough bandwidth estimations and resolutions can be seen in table 6.1. While a transition from 270p to 180p will nearly halve the amount of pixels in a video stream, the change will be less noticeable than the change from 720p to 360p, based on the portion of pixels altered. Even so, video resolution is not the only metric that can be noticed when the connectivity becomes poor.

Layer	Bandwidth Range
90p (each thumbnail)	60 - 100 kbps
180p main video	125 - 200 kbps
360p main video	470 - 640 kbps
720p main video	900 kbps - 1.5 mbps
Content sharing (1080p at 5 fps)	900 kbps - 2.5 mbps

Table 6.1: A table displaying the correlating bandwidth ranges for each resolution that Webex Meetings offers [24].

While packet loss was indirectly referenced when on the topic of signal loss for prolonged periods of time, another metric that is visually noticeable for participants is video frame rates. During a meeting, frame rates can fluctuate from as high as 30 fps, to as low as 1 fps. Experiencing frame rates as low as 1 is undoubtedly rare, but does happen if the connection is struggling enough. From the frame rate results in subchapter 5.4, as can be seen in figure 6.1, the frame rate does indeed drop to 1 fps for one interval of the meeting in question. This drop in frame rate will be very obvious to meeting participants, as the normal frame rate in most cases will be above 15 fps, and this is another sign the network connection is struggling to transmit enough data.

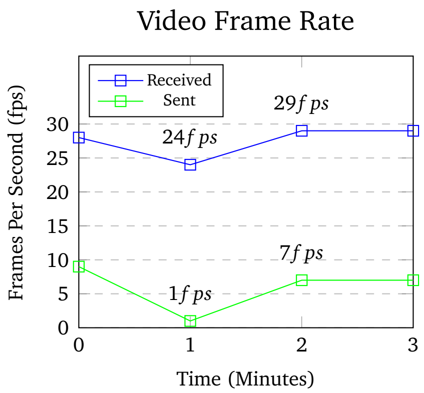

Figure 6.1: Video frame rate graph from subchapter 5.4.

It is easy to point to video statistics in this section, as images and video streams often convey a substantial amount of information at once, making it a good medium for relaying data. That said, these test scenarios also have audio to account for. Would the audio aspect of videoconferencing be much different from video in regards to this hypothesis, and if so, to what degree?

6.1.2 Audio Data

There are three main aspects of audio that can be studied in relation to this hypothesis. One of them is the audio quality of the given audio stream, or the bitrate per second, which will have an effect on the perceived quality. The next aspect is latency, and how difficult it is to follow and interpret the receiving audio stream based on this latency. The third and final aspect can be closely related to latency, which is the amount of gaps that are caused by external factors such as jitter or packet loss.

While video streams require a significant amount of bandwidth to be transmitted, audio only requires a fraction in comparison. The results of the baseline test scenario in subchapter 5.1 demonstrates this clearly, where the video stream is stable around 2400 Kbps, and the audio stream is stable at 60 Kbps. These are under ideal conditions, and is the upper limit of the expected quality of meetings that meet these requirements. However, when the network connection is struggling in terms of available bandwidth, the gap becomes much smaller. As can be seen in figure 6.2, which is an excerpt from subchapter 5.2, the audio portion of the meeting is responsible for $\frac{1}{4}$ of the available bandwidth. This speaks volumes about how much emphasis Webex Meetings puts on the audio portion of a meeting, even when conditions are subpar. For this reason alone, the connection has to be tremendously unstable for audio quality to see a diminishing result, at least to a noticeable degree.

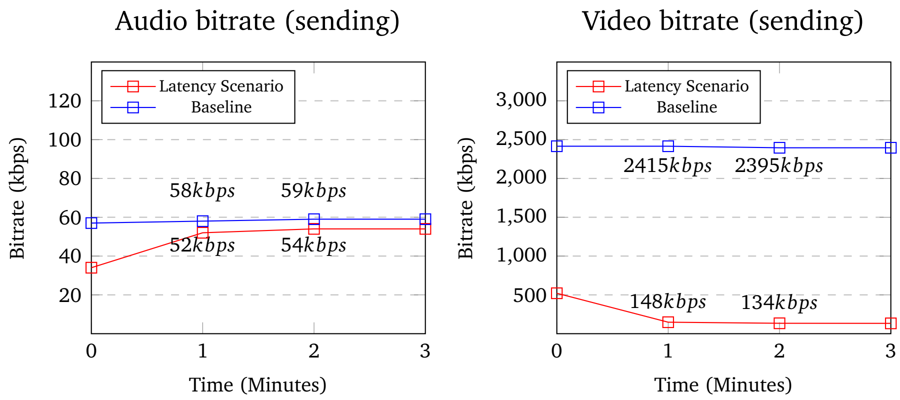

Figure 6.2: Video and audio bitrate from subchapter 5.2.

The second aspect of audio is how latency within the Webex Meetings platform affects participants. While audio has a massive advantage going for it, where the amount of bandwidth required is exceptionally low compared to video, latency will affect both mediums. From graphs throughout chapter 5, audio and video latency traditionally correlate fairly evenly between graphs. While the RTT for both audio and video mainly stay the same, the mediums are affected differently, at least to participants. A delayed audio response will traditionally have more of a negative impact than a delayed video stream. This is because verbal communication has a bigger emphasis on instantaneous feedback than video does, and if audio streams lag behind, it will be more detrimental to the overall user experience than if video frames were delayed by an equal amount. This is one of the reasons why it's exceedingly crucial to keep latency in videoconferencing to a minimum.

The final aspect of audio is how metrics such as jitter and packet loss affect it, causing gaps in audio feeds. As with video, audio also has an acceptable threshold of jitter for most videoconferencing applications. The exact value is difficult to determine, and should of course be kept to a minimum, but generally jitter below 30ms is considered adequate [13]. packet loss, on the other hand, is much more difficult to gauge. A packet loss of as low as 10% will already have massive impact on an audio feed, and will make the user experience poor as a result. These last two aspects are most important when considering which metrics affect audio the most, seeing as the first aspect is a non-issue for the reasons covered in the earlier paragraph.

6.1.3 Part 1: Conclusion

All of these sections, both audio and video, rely on the reactive nature of the Webex Meetings client to continually deliver the best quality based on the available bandwidth. When the network connectivity becomes worse, several metrics surrounding the media aspects of a meeting have to be adjusted accordingly. When the current resolution that Webex Meetings attempts to utilize is 720p at 25 fps, but the bandwidth only allows for 480p at 15 fps, the platform has to tune these settings in real-time. If the platform attempts to rectify the high resolution at 720p by lowering the frame rate accordingly, participants will have a worse user experience than if the aforementioned adjustments were made. In terms of the hypothesis from this section, can the compensation of these metrics be a used as a basis to define poor network quality? From the results and reasoning in this chapter, the answer is yes. There are distinctions to be made between different types of poor network connections however. Naturally there is a leap from 270p at 10fps to 480p at 25fps, but since Webex Meetings will always attempt to use the highest available video quality settings, adjusting these metrics to a lower setting means conditions are not ideal.

6.2 Part 2: Reasoning Behind Test Scenarios

In the introduction to this assignment, an overview of a similar bachelor's thesis from 2021 was covered. The bachelor's thesis had the title "Quality and Traffic Flow in Videoconferencing Infrastructures". Fast forward to 2022, Telenor made adjustments to the assignment specification for this thesis, noting that the amount of test scenarios conducted should be kept to a minimum. By having so many different test scenarios in the 2021 thesis, too much time was spent analyzing similar data sets. It therefore became of paramount importance to be rather picky about which test scenarios were to be conducted. While there is no direct hypothesis for this part, this section ties together the necessity for well structured test scenarios, and the reasoning behind them. The following section aims to explain and reflect upon why the different test scenarios were chosen, and what outcome they had on the thesis as a whole.

6.2.1 Initial Test Scenario Planning

From the early stages of this thesis, planning the various test scenarios remained a considerable obstacle. Not knowing which test scenarios would achieve the most optimal results, tests had to commence in order to gauge the actual capability and potential of the MG21 cellular gateway. A part of the assignment for this bachelor thesis focuses on the realistic utilization of cellular devices, such as working or participating in videoconferencing from remote areas - working from home included. After studying `https://www.finnsenderen.no/`, no realistic locations were feasible within the city center of Gjøvik. This meant that if a test of this caliber were to take place, travelling outside of urban areas was required. Ultimately, this was decided against, as the logistics behind finding suitable locations and powering devices on the move would prove difficult. As a result, scenarios were divided into two different types.

Following the project plan, a total of three test scenarios were planned. One baseline to be completed within the Topas building, one latency scenario to be conducted by wrapping the cellular gateway in aluminium wire netting, and one interference scenario to be conducted in the atrium. All these would serve as the total amount of data required to gain insight into the hypotheses that are contained within this chapter. There was an expected time interval where the all the required testing would be conducted, as well as a time buffer should any problems arise unexpectedly. Around halfway through the testing scenarios, problems did indeed begin to appear, as testing data proved too lacking in results. With the amount of cellular coverage within urban areas, and the university campus especially, favorable results were difficult to attain. The need for a struggling network connection became dire, and additional measurements had to be implemented should the data be significant enough to make comparisons against. By limiting the amount of total test scenarios conducted, the main objective was to create major discrepancies in the few scenarios that did take place.

To create a substantial gap between the optimal baseline and the subpar latency and interference scenarios, which locations the testing was conducted at had to change. Both the location and the time of day had to be carefully considered, so that results would become easily distinguishable between each other. While time of day wasn't crucial for the baseline, as all baseline tests were conducted during the weekends when limited students and staff were present on campus, this mattered for the remaining scenarios. To keep SNR data to a maximum, the amount of students present at the university campus at any given time of day should be the peak for that day. Although WiFi radio signals and cellular radio signals does not overlap, devices such as mobile phones, smartwatches and laptops with the correct functionality will be in proximity of local cellular towers, in theory impacting the amount of SNR within the immediate surrounding area. Having distinguished between the two scenarios, and the reasoning behind the attempts made to exacerbate poor network signals, what were the final results?

6.2.2 Latency Scenarios

For the latency test scenarios, there were two main locations that were utilized. The first one was in the atrium basement, an area surrounded by concrete in every direction. The cellular gateway would have to transmit signals through these obstructions, and as a result, would have an increased latency. The second testing location was conducted in the atrium, an area originally planned for SNR data, but depending on the cellular tower utilized, this area was also surrounded by concrete in most directions. Both of these areas have an emphasis on physical obstructions, such as concrete or water-based bodies. The results from these scenarios turned out to be some of the best scenarios for provoking a poor cellular networking connection, and helped tremendously in gathering exactly the type of data that the hypotheses in this chapter required.

6.2.3 Interference Scenarios

Due to the limitation of active cellular devices at any one location, the amount of places that interference scenarios could be conducted was scarce. By testing SNR data, cellular devices or cellular radio signals are required to create enough signal noise, which in turn means being in the vicinity of such devices. On a university campus, one of the busiest places is the cafeteria during the main lunch period, which ranges from 11-12 AM.

The aim of testing SNR is to achieve enough cellular radio signals within the immediate proximity of the cellular gateway, so that SNR will start to negatively impact the cellular connection. By creating a lot of signal noise, surrounding signals become difficult to distinguish between each other, and reaching the *correct* cellular device becomes problematic. As previously mentioned, the busiest period is during lunch hour between 11-12 AM, and this is when tests were executed. By conducting the test scenarios within this time frame, SNR should be a contributing factor for the received signals. The reality, however, turned out to be much different to what was expected. While the RSRP values did indeed increase during these scenarios, comparisons were not easy to distinguish, which was the entire point of these tests. SNR could not be seen as having a contributing factor to poor cellular connectivity, as the baselines conducted in the same locations to gauge the success of these scenarios, resulted in lacking data. The interference test scenarios ended up resulting in inadequate data, at least compared to the success of the latency test scenarios. The data can still be used, and it has been, but the plan was to provoke even larger discrepancies between the baseline and interference testing scenarios. That said, the interference tests were not a complete waste of time, as the only way of discovering the results was to conduct the tests in the first place. From conversations with Cisco employee Espen Berger, a conclusion was made that these types of scenarios are notoriously difficult to attain useful data from. There's also the fact of how Meraki devices report SNR data, as only RSRP and RSRQ are included. To make matters worse, only one of these metrics seemed to have meaningful results, this being RSRP.

6.2.4 Part 2: Conclusion

Since the results of the first couple of test scenarios were not satisfactory, adjustments had to be made. By switching locations for both the latency and interference testing scenarios, greater discrepancies between the baseline and suboptimal test results could be achieved. While the latency test scenarios ended up being more helpful in terms of test correlation, both the latency and interference scenarios were comprised of useful data to reflect upon in this chapter.

The test scenarios that have been conducted have served their purposed and provided a good indication of how a poor cellular network may affect a video conference in Webex. As a result of this, it has been easier to discuss the possible outcomes of the different metrics and how it affects the meeting quality. However, there is one test scenario that could have been done differently, the interference scenario. While the test scenario gave a deeper understanding of how SNR works and how it requires a lot of devices and signals to affect it, it did not have a significant impact. Cellular towers are built and designed to enable for a lot of devices at the same time. On a regular day to day basis, having more devices and signals won't have a noticeable effect on the SNR metric in terms of noise and the signal strength. When conferring with a Cisco employee that works with the Webex platform about SNR, and the findings of the conducted SNR test, he confirmed that it won't noticeably affect the SNR. The only way that it would have been a noticeable difference is if there was a extremely significant increase in traffic and devices. While SNR is difficult to test, theoretical estimates can be achieved in this instance. By calculating the amount of available bandwidth, and correlating between the amount of average bandwidth any given user needs, a theoretical upper threshold can be established.

6.3 Part 3: Using Webex As A Platform

When Webex Meetings decides which metrics to utilize for any given participant, the respective metrics need to align with the available bandwidth. Thankfully, Webex Control Hub has made data collection trivial throughout this thesis. This means studying how Webex acts when it has to adjust these metrics becomes easier. Coming back to the hypothesis for this section, *when Webex Meetings is affected by subpar networking conditions, video bandwidth is deemed more susceptible to reduction than audio, and a greater effort will therefore be put into maintaining a consistent audio stream*, does this hold true? By looking at the data collected in chapter 5, and correlating this with the discoveries made in chapter 6.1, this will be discussed in the following section.

6.3.1 Aspects and Features

While the data collection of metrics in Webex Control Hub has been trivial, distinguishing between the impact they have is another story. Every metric has seen recurring use throughout the different test scenarios. Separating and singling out one metric from the rest is not an easy thing to achieve, seeing as the different metrics usually arrive in a domino effect, or as a chain of events. When packet loss spikes, latency, jitter and RSRP will do the same.

From conducting various test scenarios, and obtaining a wide range of different test results, an insight into expected outcomes has been gathered. The correlation between bandwidth and receiving or sending resolution is not set in stone, but rough guidelines have been verified. These numbers were covered in chapter 6.1. Traditionally, the first thing that gets adjusted is the video quality. From graphs supplied in chapter 5, audio bitrate only dropped below 30 Kbps for a single scenario, even if the video bitrate was a fraction of what it was previously during the same meeting. The meeting in question is displayed below in figure 6.3, and as seen, towards the end of the meeting the quality starts to pick up (∼27 Kbps), even if the video quality stays the same (∼270 Kbps). From the baseline scenario, the amount of bandwidth of video versus audio is a ratio of 40 to 1, meaning that video will require 40 times more bandwidth during ideal conditions. In several graphs from chapter 5 however, this ratio is much lower, and in figure 6.3 below, around 9 to 1. This is a clear indication that a bigger emphasis is put on audio rather than using a tiny smidge amount more bandwidth on video quality. Upon reaching out to the Webex Engineering team, this was confirmed, although there is no official documentation stating this. In terms of bandwidth priority, the list from highest to lowest goes: Audio, content audio, content video, main video. Logically, this makes sense, seeing as video has to give up a small margin of its available bandwidth, at the benefit of an audio stream that requires much less.

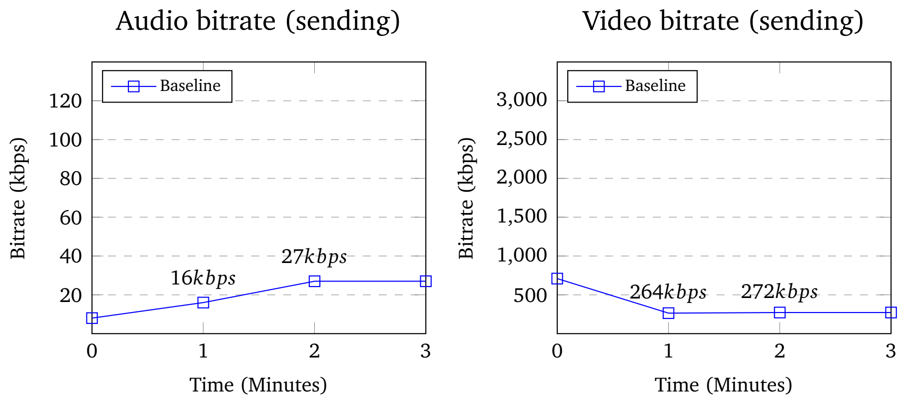

Figure 6.3: Video and audio bitrate from subchapter 5.2.

6.3.2 Audio Quality versus Video Quality

While arguments can be made depending on the type of meeting, audio and video functionality traditionally make up for the most important aspect of videoconferencing. Screen sharing is also a commonly used feature, but would likely be included under video functionality. The question then becomes, which is more important, audio or video? Chapter 6.1 discussed this briefly, coming to the conclusion that when Webex Meetings has to make adjustments due to poor network quality, audio is given a larger ratio of available bandwidth than video is. To clarify the previous statement, the amount of bandwidth that audio uses is not greater than what video uses, at least from the provided testing, but the ratio between them favors audio in such scenarios. Another detail to mention regarding the importance of audio, is the fact that Webex Meetings will automatically disable the video functionality of a participant, should the connection be unstable enough. In order to continue sharing video, the participant has to manually enable this feature again. This happened several times while conducting the various test scenarios, notably only when the network connection was struggling.

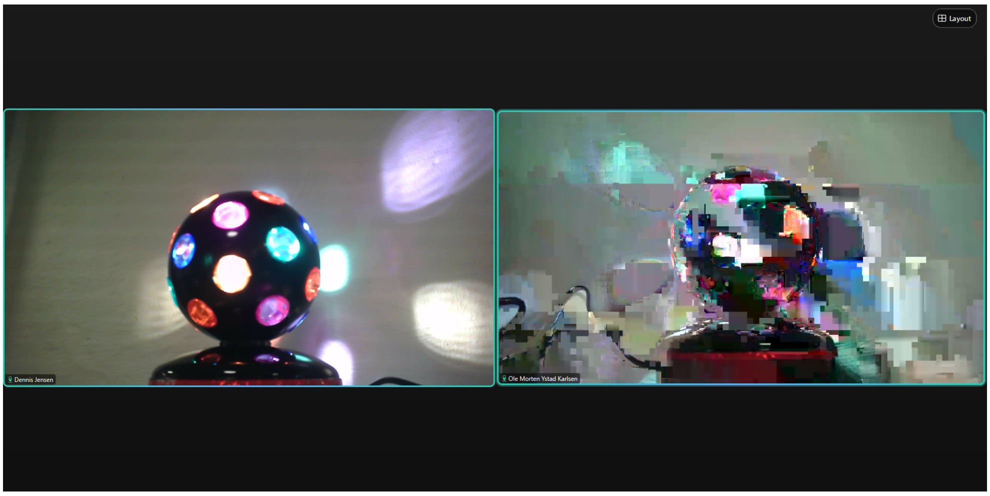

Figure 6.4: Comparing ideal conditions on the left to suboptimal conditions on the right. The right hand video feed has a bitrate of 130 kbps.

As mentioned in the previous paragraph, the ratio for required bandwidth of video compared to audio is quite significant. To make an illustration of how poor the video quality can become, an excerpt from one of the basement test scenarios has been included, as seen in figure 6.4. In this instance, the reported bitrate in Webex Control Hub shows an average bandwidth of 130 Kbps. When considering the upper threshold of audio bitrate during ideal conditions is ∼60 Kbps, this is still less than half of what this video stream requires to display the feed as in figure 6.4. Video functionality is beneficial, and is an extra input recipients can use to interact with a medium or other meeting participants. That said, when the video quality is this poor, transmitting just audio will elevate the overall user experience. If you were asked to join a videoconferencing meeting, and could only choose one of the two, would you rather choose audio or video? For lectures and other learning scenarios, two-way audio may not be necessary at all, but for meetings where interacting and communicating is key, audio would be the obvious answer.

6.3.3 Conclusion

This section led to the discovery of how important latency was, and how higher latency levels would affect other network related metrics. By contrasting the amount of bandwidth used, and the ratio between audio and video bitrate, the importance of how Webex handles these became evident. Any given meeting would provide a satisfactory user experience as long as the audio quality was preserved at a high enough level. Video quality was deemed less important, due to the immense bandwidth requirement that video streams need in order to run smoothly. With this information in mind, the conclusion is that audio quality is more important than video quality.

6.4 Part 4: Reflection

As covered in chapter 4, the chosen props for all the various test scenarios was a disco ball. This prop rotates and creates motion within the scene, imitating the motion of a human. One problem that this solution experienced, was the disco ball's ability to light up the backdrop of every scene for each scenario. This means the scene complexity increases accordingly, as the same backdrop for regular humans would not be similar, because the background is no longer static. A comparison here could be made, where the lights that shine towards the backdrop could be represented as bypassing cars in a scene filmed through a window. This would create equal motion within the backdrop, if not to the exact same degree. Although not confirmed, there is currently a theory that this could have an impact on the amount of I-frames and P-frames that are sent. For that reason alone, it's an aspect that should have been looked at more closely.

6.4.1 Impact of I-frames and P-frames

There were clear differences between the send and received I-frames and P-frames from the results covered in chapter 5. In the scenarios where a limited amount of I-frames packets were transmitted, the video feed struggled the most. Problems such as video artifacting would occur and be visible to all meeting participants. An example of how this looks was previously shown in figure 6.4, but it can also be observed in figure 6.5 below. Transmitting a small amount of I-frames results in images not updating from a clean sheet for a longer period of time, which is problematic. As P-frames transmit changes in the image such as motion, and constantly builds upon the previous frame, video artifacts are more likely to occur than if there are less I-frames [33].

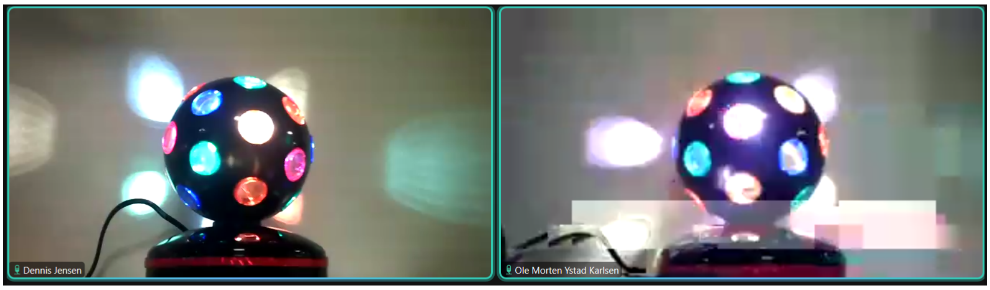

Figure 6.5: A screenshot of a video feed with block artifacts on the right, taken during latency testing scenarios.

6.4.2 Differentiating Between Layer 2 and 3

After analyzing the various graphs in chapter 5, latency was unreasonably high for many of the test scenarios. Having a latency of over 700ms RTT would mean that the connection is so delayed, that it is questionable if this is sufficient enough for digital meetings. As such, a hypothesis formed that the latency that was reported in Webex Control Hub was inaccurate. Espen Berger at Cisco mentioned that a latency surpassing 600ms would severely impact meetings in a negative fashion. By manually testing the audio feedback for this type of delay, and by the description received from Espen Berger, this hypothesis was further strengthened. If the latency actually saw numbers this high, packet loss and especially jitter would surely see a swift increase as well. But how could the packet loss reported in Webex Control Hub be slim to none, even when the latency increased towards an average of 1000ms?

It's plausible that this could have something to do with how Webex Control Hub measures packet loss. When analyzing the .pcap files obtained from Meraki Packet Capture, packet analysis in Wireshark showed exponentially more packet loss than what Webex Control Hub had reported for the same exact meetings. As seen in tables 6.2, 6.3, and 6.4, the packet loss from the respective testing scenarios is noticeably higher. The running theory for this is that there's a failure to deliver packets on layer 2, and since packet captures are done locally on the MS120 switch, layer 2 is able to be inspected. If Webex Control Hub only reports packet loss on layer 3, packet loss will not be reported correctly for the layer below it.

Source Client	Packets	Lost	Mean Delta (ms)	Duration (s)
PC-1	889	1612 (64.5%)	211ms	169
PC-1	1227	1736 (58.6%)	148ms	169
PC-1	271	220 (44.8%)	54ms	14
PC-1	1440	26 (1.8%)	2ms	4
PC-1	2241	3516 (61.1%)	79ms	176
PC-1	2270	3489 (60.6%)	79ms	177

Table 6.2: Basement Scenario; Packet loss and Mean Delta (ms) displayed with RTP Streams in Wireshark.

Source Client	Packets	Lost	Mean Delta (ms)	Duration (s)
PC-1	1915	162 (7.8%)	169ms	148
PC-1	2752	608 (18.1%)	68ms	149
PC-1	513	109 (17.5%)	24ms	12
PC-1	176	0 (0.0%)	6ms	2
PC-1	3610	464 (11.4%)	50ms	180
PC-1	3650	426 (10.5%)	24ms	180

Table 6.3: Campus Cafeteria Scenario; Packet loss and Mean Delta (ms) displayed with RTP Streams in Wireshark.

Source Client	Packets	Lost	Mean Delta (ms)	Duration (s)
PC-1	63	951 (93.8%)	634ms	40
PC-1	61	952 (94%)	655ms	40
PC-1	263	1210 (82.1%)	153ms	40
PC-1	183	30 (14.1%)	110ms	20
PC-1	189	24 (11.3%)	20ms	20
PC-1	145	15 (9.4%)	50ms	8

Table 6.4: Atriet Scenario; Packet loss and Mean Delta (ms) displayed with RTP Streams in Wireshark.

6.5 Part 5: Further Work

One of these subjects that can be looked at further revolves around finding out which cellular towers were utilized during teach test scenario. At the recommendation of Telenor, a third-party device should be used to glean information about which cellular towers are in use. The thought was that the MG21 cellular gateway and a mobile phone using the Telenor cellular network, would both be connected to the same cellular tower - as long as they were within the same approximate area. Without definitively knowing which tower the MG21 cellular gateway connects to, pinpointing which frequencies and the distance between cellular devices and their connection points, becomes problematic. Since the MG21 cellular gateway has the option to use multi-banding, knowing which towers are in use is beneficial knowledge to have.

Another subject that could be researched further is the impact that distance has on the MG21 cellular gateway. The testing environment utilized during the scenarios in this thesis were all within the city center of Gjøvik. By being surrounded by multiple cellular towers at close range, creating a distance between the cellular gateway and its connected tower, becomes impossible. Having the ability to select the distance carefully, by only having a limited amount of cellular towers available, better results can be achieved. As already mentioned, this would also give great detail about how the cellular gateway interacts with varying frequencies, as these are bound to change upon distance fluctuations.

The final subject that could see continuation is bitrate, and more specifically what causes sudden changes to this metric. So far, little is known about why this inconsistency occurs. As seen in figure 6.6, the .pcap file shows two separate drops in bitrate for short time periods. At first glance, the initial hypothesis was that it had something to do with I-frames or I-frames. However, after studying the recorded packets during several meetings in Wireshark, no correlations could be made. A further examination surrounding this phenomenon could provide information regarding the reasoning behind this occurrence, if it affects the meeting in any way, and how to avoid it.

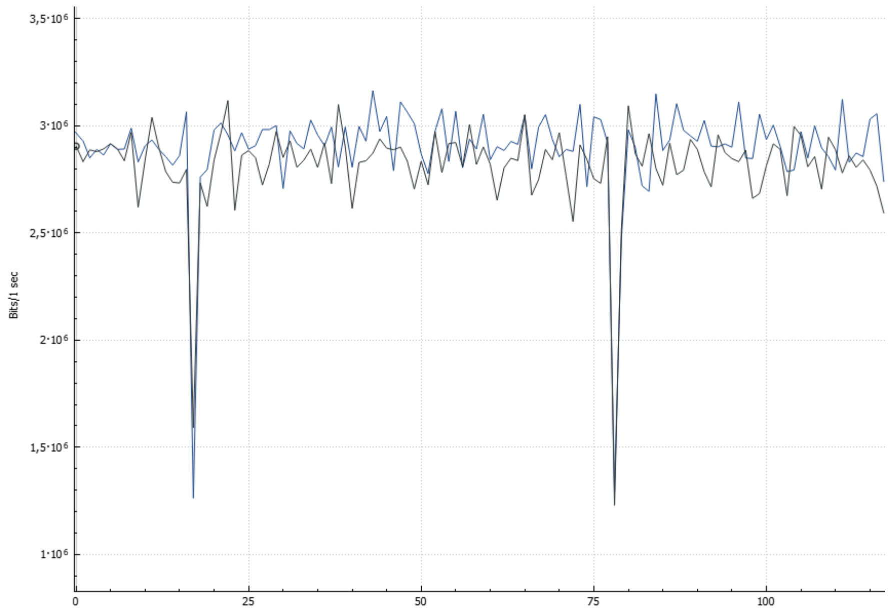

Figure 6.6: Graph tool in Wireshark displaying the throughput drops during a baseline test.

Chapter 7

Conclusion

This chapter will conclude the most essential findings from the results gathered in this thesis. An overview of the different aspects surrounding how Webex Meetings reacts to various connection issues will be described. Additionally, the validity of the hypotheses introduced in chapter 1 will be wrapped up.

7.1 Key Moments

When Webex Meetings is affected by poor networking conditions, the platform has to resolve this issue by lowering the video quality, and should the connection be unstable enough, the audio quality gets lowered as well.

7.1.1 Video Quality

Since the client will always attempt to choose a resolution of 720p for the initial moments of a meeting, this quickly becomes the first metric to downscale if bandwidth is insufficient. Additionally, the frame rates are often lowered depending on how severe the connection issues are. The video feed will also be affected by packet loss and jitter, even if this isn't necessarily the case from the data extracted from Webex Control Hub. One of the easiest ways to determine if the network connection is struggling, is to keep an eye on metrics such as bitrate, resolution and frame rates. That said, there are also other ways of identifying poor connectivity.

In addition to bitrate, resolution and frame rates, specific frames called I-frames and P-frames are affected. As Webex Meetings utilizes H.264 video compression, it is important to have a stable RTP stream of I-frames and P-frames to preserve the video quality during a meeting. If the amount and frequency of I-frames sent is low, it will negatively affect the video quality, causing bottlenecks such as video artifacts appearing in the video feed.

If Webex Meetings determines that the connection is too unstable to send video, the platform disables the video functionality altogether. This will not be prompted, and the video feed will simply turn off. If the participant wants to enable video again, this has to be done manually.

If all of the previous measures have taken effect, the next thing that Webex Meetings alters is the audio bitrate, lowering the bitrate according to the available bandwidth. Considering the small amount of bandwidth audio demands, if the connection is lowering the audio bitrate, the communication will likely be at the verge of ceasing altogether.

7.2 Results

At the start of this thesis, there were two hypotheses introduced in chapter 1, one regarding what constitutes a poor network connection, and the other focused on how Webex Meetings prioritizes audio and video. The latter has briefly been discussed in subchapter 7.1.1, as the different video metrics will be altered according to their priority. The first hypothesis was also deemed to be strengthened by the results found in chapter 6, defining a poor network connection as a result of Webex Meetings compensating and lowering video metrics. While the main discussion part of this thesis concludes that these hypotheses are strengthened by the discussed results, there are still discrepancies between what Webex Control Hub reports and Wireshark reports.

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Appendix A

Additional Material

A.1 Project plan

Project Plan; Webex and Networking Interference

Dennis Jensen,
Knut-Magnus Karlsen,
Ole Morten Ystad Karlsen

Contents

Acronyms

COVID-19 Coronavirus disease 2019.

IMRaD Introduction, Method, Results and Discussion.

NIPH Norwegian Institute of Public Health.

NTNU Norwegian University of Science and Technology.

SNR signal-to-noise ratio.

Glossary

4G mobile network is a 4th generation mobile network for mobile services.

bandwidth is a therm used to measure the maximum amount of data transfer over a given path.

frame rate is the frequency at which consecutive images are captured or displayed.

incident response plan is a plan combination between people, technology and processes that are used in order to deal with incidents. The entire plan is well documented, tested and practiced before the incident occurs.

jitter is the variation in the time delay between sender and receiver, often caused by latency and packet loss.

Kanban is a framework for workflow management focusing on visually representing work items on a kanban board to maximize efficiency.

latency is best described as a time delay between senders and recipients, often as a result of network delay.

Lean development is a part of the agile framework for software development. The main premise is a streamlined approach to delivery and updates.

media codecs is a service or program that encodes or decodes a data stream, typically a video stream or similar.

packet loss happens when one or more packets is dropped while in transit across a network.

playbook contains methods to replicate a common situation or activity to identify an issue.

Scrum is a framework for project or group management. Then main premise follows work in 'sprints', focusing on dynamic planning and meetings.

Signal-to-Noise ratio is the ratio between the desired signal and the noise signal known as background signals.

Chapter 1

Goals and Boundaries

1.1 Introduction

Videoconferencing and streaming video has experienced a surging trend in businesses since the introduction of the internet, be it abroad or simply working from home. Meetings, presentations, and demos are easily presented from anywhere in the world - provided the presenter has an internet connection. Since the start of the COVID-19 pandemic, more and more people have started utilizing mobile networks such as 4G to work from home [1]. This in turn, has introduced a need for videoconferencing software such as Cisco Webex[1], but how does Webex fare when conditions are subpar?

Participating in meetings where the network is experiencing disruptions will make it challenging to pay attention. If there is an acceptable lower threshold for network connectivity that is able to stream video, this thesis aims to find that threshold.

1.2 Background

Telenor is one of the biggest digital service providers in Norway in regards to telecommunications and data services [2]. They are currently in collaboration with Cisco to offer an improved videoconferencing experience to users utilizing the Cisco Webex[2] platform. This bachelor thesis is based on the assignment given by Telenor where we will analyze how 4G mobile networks affect the Webex video service during meetings in certain scenarios. The testing configurations will be tailored to gather valuable data surrounding network interference such as packet loss and latency. A wide array of tools will be used to test these scenarios, such as Wireshark, Webex Control Hub and Meraki Packet Capture.

In order to test the scenarios that are outlined, specific software is needed to

[1] `https://www.webex.com/`

[2] `https://www.webex.com/`

do so. As already mentioned, Webex will be utilized for videoconference testing. Webex allows you to collaborate with anyone through their application with built-in functionalities like screen sharing, video calls over the internet, group creation and direct messaging [3].

After COVID-19 forced the hands of many businesses, working from home more or less became the norm. Businesses and educational institutions started using software such as Webex as a replacement for physical attendance and as a preventative measure for spreading the coronavirus. This led to a rapid transition from working on-site to working remotely for many, introducing a new style of collaboration [4]. This is where Webex slots in, as a useful application for remote collaboration. According to the McKinsey Global Institute, we will continue to see a prevalent use of remote collaboration post-pandemic [5].

Having a reliable connection to the videoconferencing application of your choice is not guaranteed if the bandwidth is insufficient. End users can be affected by poor internet connectivity due to jitter, packet loss and increased latency. This will in turn make the video conference difficult to follow properly, and this is a scenario where a lower threshold of quality could be utilized [6]. This is one of the problems this thesis aims to investigate.

1.3 Task Description

In order to research and analyze how videoconferencing is affected by poor cellular network connections, explaining what defines a poor connection is crucial. This in turn means understanding how jitter, packet loss and latency correlate between each other, and how they affect Webex as a platform. By simulating poor or excellent working conditions, baseline scenarios can be created that mimic a travelling or remotely connected end user.

The first part of the task is to gather relevant information and literature about jitter, latency and packet loss. This will serve as a foundation for this thesis, and have an important role in understanding what determines a poor connection. This section will also have an important role when comparing results, and building towards a conclusion.

The second part of the task is to conduct the different testing scenarios that will simulate a poor cellular network connection. Currently, we are aiming to have three different test scenarios, each with multiple baselines to compare against. Poor, mediocre, and excellent working conditions are mainly the scenarios that will take place, but flexibility will still be a factor until time can be allocated between all three. From earlier exercises, too many scenarios has meant too much time has been allocated to the testing phase of the project, and this is something to avoid.

After gathering all the necessary data in the test scenarios, a light can hopefully be shed on how jitter, latency and packet loss affect Webex. Uplink and downlink problems may also arise when taking SNR into account. This will be the final part of this thesis, comparing and discussing the different results. The test scenarios will no doubt have different outcomes, and this thesis seeks to analyze, compare and conclude the most essential findings.

1.4 Project Goals

The main goal of this project is to determine how different types of network connectivity affects services such as streaming, video calls, or meetings in general across mobile networks. Multiple tests will be performed in order to collect data from the different scenarios. These scenarios will all differ from each other in an attempt to achieve varying amounts of interference. By having three different test scenarios, it should be possible to gather enough data in order to determine the cause of the difference in quality, and how the difference affects the user experience in general.

After concluding and interpreting all the different results, descriptions and guidelines can be created so that an incident response plan can be made. This is also called a playbook. By creating this playbook, we want to make it a useful tool in order to create a better user experience for people using mobile networks for similarly demanding tasks that also require considerable bandwidth.

1.5 Boundaries

For this project, the main focus will be on network connectivity across mobile networks. In this case, this will be a 4G network provided by Telenor. With this network, different network connectivity levels can be simulated, in order to collect data about how working from different locations with varying signal strength, latency and bandwidth can affect the Webex platform.

The tools utilized during this project are mainly for collecting data from the test scenarios, and this includes software such as Meraki Insight and Wireshark. With these tools, tracking the status of each scenario becomes easier, and they help in pinpointing the timestamps when loss of quality occurs.

In order to compare results, a baseline reading for the different scenarios will have to be recorded. This will be executed multiple times, so that the baseline is consistent. This is also the case for subsequent test scenario readings.

Time of day, location, signal strength and network throughput all have to be taken into account when performing tests. This is to achieve comparable and fair results for all the different scenarios. External factors have been discussed, such as webcam resolution, frame rate, and bitrate. These are important factors to consider. Hardware is no doubt something to think about, but it's too early to say anything about which factors that need an additional focus. This will hopefully be discussed thoroughly in the results section of the main report.

Chapter 2

Scope

Since this thesis has a set deadline, planning ahead and setting deadlines will be important. In order to properly evaluate the areas of interest, the scope of this thesis has to be defined beforehand. Knowing which subjects to focus on, where to allocate time and resources, and setting milestones will help us achieve this.

To further achieve this, we will have to define three areas regarding scope. Problem area will discuss various problems this thesis will attempt to answer, and within which areas we will spend most time. Problem delimitation will discuss which technical aspects we want to cover, and to what degree. Finally, when discussing the issue at hand, we will reflect on the main theme this thesis will cover.

2.1 Problem Area

When on the topic of videoconferencing, networking becomes a key subject.

Transferring a video stream from one device to another is a complicated subject, but the nuances of networking topics and what problems may arise when performing such an action, is what this thesis will explore. This can be end-to-end communications, which is communication where devices are directly connected to each other. If there are servers involved, data that is exchanged will have to pass through a varying amount of nodes along the way, and will often result in added latency compared to end-to-end communication.

In order to properly define which areas and subjects we will look at, which is mainly networking, we will attempt to limit the external factors, so that we can mainly focus on networking aspects. With this in mind, certain equipment and networks will have to be evaluated. Similar or equal webcams would be preferred, so that resolution, frame rate and quality of the image is not a differing factor between participants. Workload on the computer is also something to consider. Should one computer be noticeable slower, this will have to be taken into account. These are just some of the issues we will have to consider when performing testing.

2.2 Problem Delimitation

A bachelor thesis will traditionally be fairly open to interpretation of what the core issues are, and how to attempt to solve them. While it's entirely possible to put an emphasis on subjects related to media codecs, security around videoconferencing or developing additional software for testing and verifying a connection, our thesis will focus on networking.

While we want to focus on the technical aspects of networking such as jitter, latency and bandwidth, equipment is also an important part of achieving adequate results. Should webcams differ in resolution or frame rate, the results might change quite drastically based on those factors alone. Therefore, we will attempt to iron out as many points of failure as possible. By limiting the testing equipment to be of equal or similar makes, variance in quality should not be an issue.

2.3 Issue at Hand

When on the topic of videoconferencing, all of the factors already mentioned are important subjects. Networking is the foundation for sending and receiving video streams, and knowing the limitations is key to understanding how to perfect and utilize the platforms properly. With an ever growing demand, even as the COVID-19 pandemic starts to die down and people are returning to offices for work, videoconferencing is still on an upward trend.

2.3.1 Correlations Between Videoconferencing and Networking

As with most services, there is a wide array of technologies, hardware, software and utilities that make ***that*** specific service run smoothly. Our job in this thesis is to define what areas we want to focus on, and attempt to find correlations between these areas. Terms like jitter, latency and packet loss are well known terms within the world of networking, but how do they actually affect videoconferencing?

2.3.2 Thresholds for Network Quality

With videoconferencing in mind, and what tools we will be utilizing already defined, the issue at hand will be to correlate results and discover if any metrics are more important than others. What's the lowest threshold of network quality that can still uphold the desired video stream quality?

Chapter 3

Project Organization

3.1 Responsibilities and Roles

Being a group consisting of three members, it is important that the group divide responsibilities and roles evenly so that each workload is comparable. The roles the group will have for this project is a group leader, communications liaison and a referent.

The group leader will be responsible for solving conflicts within the group. Additionally, the leader will be able to veto a decision if there is a disagreement during the project period. This however, only applies in situations where the group members, other than the leader, have a disagreement. Democracy will still be the main decision making tool throughout this project. The leader will also be responsible for deadlines and that the group is on track with the current milestones.

The communications liaison will prepare and deliver material for meetings as well as written communication. Additionally, the liaison will keep an eye on the different communication platforms used by the group, so that no messages are left unread. The communications liaison will also report back to the group accordingly.

The referent will take notes during meetings and deliver them to both the group and the supervisor after the meeting. Providing material such as this gives both the group and the supervisor a reminder of what kind of subjects was discussed during the meeting. This also helps every stakeholder know the progression of the project, and also what to prepare for the next meeting.

The group has also decided that every member will specialize in certain tools that will be used during the project period for analyzing the data. The primary tools used in this project will be WireShark, Webex Control Hub and Meraki Packet Capture.

3.2 Routines and Guidelines

Working on a project can cause all types of situations. Disagreements are common occurrences during a project, and can both result in positive and negative consequences. Not knowing how to deal with situations and occurrences such as this, can result in grudges and a bad work environment. Therefore, it is important to establish rules and regulations for how to deal with such dilemmas. A work contract is a fine example of such a guideline tool. The work contract is created together between all group members and includes the interest of all parties involved. In order for the work contract to be valid, all the group members has to sign the work contract in agreement.

3.2.1 Work Contract

Team Members
Dennis Jensen, Knut-Magnus Karlsen and Ole Morten Ystad Karlsen

Introduction
This work contract is a foundation for how we want to perform and operate as a team, and will function as a guideline throughout this project. We will establish a good work environment, by having clear goals and rules in place to make sure everyone knows how to act in different situations. It's based on a template provided to us in one of our courses, Cybersecurity and Teamwork (DCSG1002) [7].

Effective Goals
Our goal with this work contract is to make a structured project, with clear instructions and roles for each team member. By creating a well organized work contract, we make sure everyone knows their own roles and how to act accordingly to those roles. In order to reach this goal we will:

- Utilize the strengths of each group member, and pick roles most suited for their set of skills.
- Attend scheduled meetings and come prepared in order to maximize the efficiency of each meeting.
- Create and maintain a good relationship between all group members and the employer.
- Use the supervisor assigned to us from NTNU, in order to keep the project on track and clarify the way forward.

End Goals

- An increased knowledge on the subject and on cellular networks.
- Provide useful insight to Telenor regarding our findings from the analysis.

- Create a thesis that can be used to gain attention from the job market, more specifically Telenor and Cisco.

Roles
Our group consists of three members, and each of us have specific roles we have to fulfill. Each member has been selected to fulfill one main role throughout the project. The selection of these roles is based on each of the individual group member's preferences and personality. In addition to this, we have also based the decision on their background and skills, in order to make sure they are up to the task.

All group members will specialize in certain tools, in order to reduce the workload for each group member. Note that even if we will specialize in certain technologies, all members are free to work on differing tools not related to their specialization. This is even encouraged.

Dennis Jensen: Primary role: Group Leader. Will specialize in: Wireshark and ThousandEyes.

Knut-Magnus Karlsen: Primary role: Communications Liaison. Will specialize in: Webex Control Hub.

Ole Morten Ystad Karlsen: Primary role: Referent. Will specialize in: Meraki Insight.

Attendance
There are a few activities where all group members are expected to participate. Meetings with Telenor and our thesis advisor are among such activities. In addition to this, the group also have frequent meetings throughout every week. These meetings usually consist of discussing what each group member has been working on since the last meeting, and also the progress of the project in general. There are also times where the group will be collaborating to finish subjects that require more attention and discussing than others. If a member cannot participate, ample notice is expected, considering these types of meetings are traditionally planned far in advance.

Absence
Should one of the members be absent for any reason, notice is expected. No specific timeline will be required, but not meeting or participating when otherwise agreed upon will be brought up at the next convention. Team members are expected to finish their tasks within a timely fashion. If unable to do so, notice will have to be given, so that other members are able to complete the work in due time.

Decisions
Should the group not be in agreement, the group leader will traditionally be able to veto the decision. As mentioned, this will only be the case if there is a disagreement between the two other group members, not including the group leader. This

serves as a purpose of not escalating altercations within the group.

Conflict Management
In most cases where a decision cannot be made by the group leader, a third-party will be queried. Our supervisor, Ernst, or our primary Telenor contact, Henning, will serve as external counsel in such matters.

Discipline and Resolution
Should a member not finish their work in time, or be late for official meetings, they are expected to compensate for the respective event. The time spent for the duration of the project should be fairly similar for each member. Should there be a necessity make deliberations, the group will have a discussion for each event where this occurs.

Dennis Jensen	Date
Knut-Magnus Karlsen	Date
Ole Morten Ystad Karlsen	Date

Chapter 4

Planning, Follow-up and Reporting

4.1 Development Model

For this thesis, we wanted to utilize a Lean development model. From talking internally within the group about which models we already knew about, we figured Scrum[1] or Kanban[2] would be a good fit. There is more structure and planning required when utilizing a Scrum development model, but this is not necessarily a good thing in our thesis. Not knowing many of the factors for a large portion of this project, keeping rough timelines rather than strict ones might be a good idea. This is one of the reasons we ended up picking Kanban.

From previous experience, we have found that putting tasks on hold and continuing at a later point in time works great for us. This is because tasks can be put aside if it requires more insight from more than one person to complete, or missing relevant information making it difficult to complete at the time being. With scrum, and more specifically its sprints, this is more difficult to achieve. Sprints are planned ahead, and changing the sprint either during or after the sprint if it was unsuccessful, is rather troublesome. It would skew the entire workload's timeline, and for this reason, Kanban seemed more fitting for our approach to this project.

4.2 How We Are Using Kanban

In this project we are going to use Kanban as a development model in order to structure our workflow, and to provide an overview for all the tasks planed for the foreseeable future. The process of using Kanban will mostly focus on the usage of a Kanban board. On this Kanban board, each member will first add ideas and

[1] `https://www.scrum.org/`

[2] `https://kanbanize.com/kanban-resources/getting-started/what-is-kanban`

suggestions to a to-do list or backlog section.

The group will then discuss the different suggestions and ideas in order to determine which tasks are suitable for the project. All of the agreed upon tasks will then move on to a "To do"-section, where each group member will select what kind of task they will be working on. The tasks are not restricted to have only one member working on them, but will be open for all group members to cooperate to finish the task. This is useful if there are tasks that require insight from more than one person, but also tasks that are more demanding.

When a task is selected, it moves on to the "In Progress"-section as shown in figure 4.1. This section will only contain three sections at a time, and if another one is to be started, the respective task already in motion will be put on hold. The reason why we limit the work in progress to three tasks is to not focus on too many tasks at once. By structuring the workload like this, the Kanban board will be easier to follow at any given point in time.

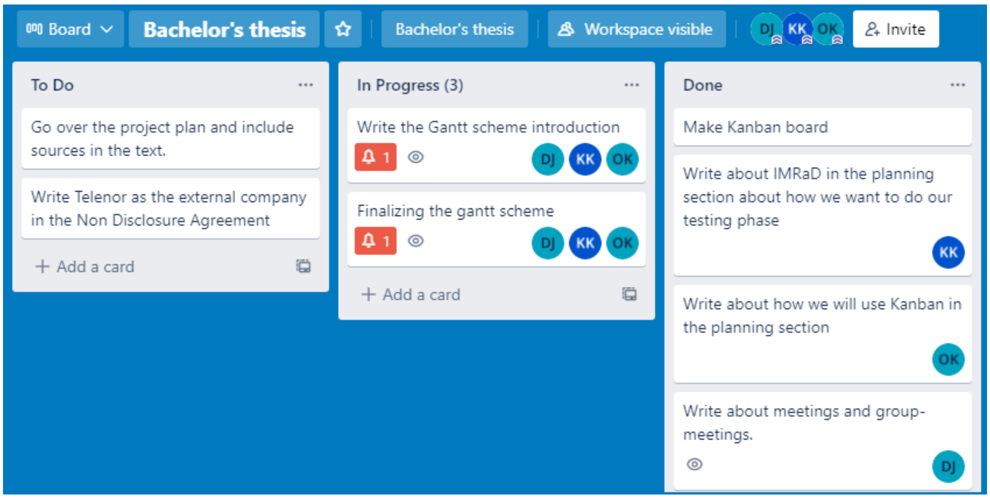

Figure 4.1: An extract from our current (27.01) Kanban Board. Not all sections are displayed in this image.

If a task should require more time or is missing relevant information, it can be put on hold. By doing this, the group members working on the task are able to move to working on a different tasks. The tasks in the "On hold"-section can then be picked back up again at a later date by the same member that was working on the task previously, or have another group member step up to take their place.

When a task is completed, the task is moved to the "Done"-section. Having a Kanban board like this creates an overview for the entire group in order to map the progress for the project. Additionally, it provides a status update for all the tasks committed to the board, and also the availability of each group member.

4.3 Plan for Status Meetings and Decision Points for This Term

4.3.1 Status Meetings with Telenor and our Supervisor

From talks with our supervisor at NTNU, Ernst, we are aiming to have status meetings every week. The preliminary date is agreed upon to be every Friday, but this is open to change should it be required later in the semester. Telenor has also specified that they want weekly updates, but status meetings will take place bi-weekly. This is also open to change later in the semester, should it be necessary.

4.3.2 Decision Points

In order to have control over what milestones we will have during our thesis, a Gantt chart has been made in chapter 6. All milestones are displayed in the chart by orange diamonds, and signify a finished section or deadline for the respective task.

Chapter 5

Organization and Quality Assurance

5.1 Documentation, Standards and Tools

All documentation will be written in Overleaf which is a LaTeX editor. The `ntnuthesis` class is also implemented as a standard for writing the thesis. However, some documentation is kept in our private Discord server until further implementation into the main report. The referent of the group will be taking notes from meetings and sharing them through Microsoft Teams to our supervisor.

When writing this bachelor thesis, it is important to have a good structural foundation to ensure the overall quality of the end result. To achieve this, we will utilize IMRaD[1] as a guideline for writing our thesis. This acronym refers to the following four sections: Introduction, Method, Results, and Discussion. This format will be useful when presenting our findings after executing the different test scenarios and as a general guideline for the thesis as a whole to be academically oriented.

Additionally, there will be used a number of tools and platforms for when the group is collaborating or communicating.
List of platforms that will be used:

- Webex
- Meraki Dashboard
- Discord
- Trello
- Facebook Messenger
- Microsoft Teams
- Overleaf
- Google Docs
- Google Sheets

[1] `https://sokogskriv.no/en/writing/the-imrad-format.html`

List of tools that will be used:

- WireShark
- Webex Control Hub
- Meraki Packet Capture
- ThousandEyes

5.2 Plan for Testing

We are planning to do three test scenarios and a baseline reading to compare the different outcomes of the scenarios. The scenarios configurations themselves will differ greatly. This is done in order to achieve a variation of data that can affect the videoconferencing. This will prove useful in the discussion section of the thesis, where we discuss our findings and how it affects the videoconferencing as a whole. Additionally, we will try to replicate inadequate cellular network conditions so we can see clear differences between the test scenarios and the recorded baseline. To achieve this, we plan to cover up the cellular network receiver in a metal mesh of wire. This will increase the Signal-to-Noise ratio, and interrupt the signal. Should the metal mesh also prove inadequate, glass is also a powerful disruptor.

5.3 Risk Analysis

Having a risk analysis is crucial for a project of this magnitude. Unforeseen events such as illness within the group could potentially set back deadlines and the overall quality of the thesis. Being prepared prior such events taking place and having set measures in place will massively decrease the overall risk during the project.

There are 13 different risks associated with this thesis. They have a varying degree of severity and possibility of occurring, refer to the list or figure 5.1 below.

- **R1**: Member(s) of the group get(s) affected by illness.
- **R2**: Member(s) of the group drop(s) out.
- **R3**: Equipment doesn't arrive in Gjøvik.
- **R4**: Equipment breaks during testing.
- **R5**: Equipment gets stolen.
- **R6**: Various platform unavailability.
- **R7**: Thesis not finished by deadline.
- **R8**: Network outage/instability during testing.
- **R9**: NTNU closes its facilities.
- **R10**: Losing access to the Overleaf Editor.
- **R11**: Communication with Telenor deteriorates.
- **R12**: Communication with supervisor deteriorates.
- **R13**: No private room available for testing.

	Insignificant	Moderate	Major	Severe
Almost certain	R1			
Possible	R6	R13		
Unlikely	R8	R9	R4, R5	R7
Rare			R3, R11, R12	R2, R10

Table 5.1: Risk matrix for this thesis.

5.4 Measures

As with the previous list of risks it has also been conducted a list of measures in place to decrease the drawbacks of said risks. The list consists of measures ranged with a number from one to thirteen, each number representing measures for the corresponding risk.

- **M1**: With the current ongoing pandemic of COVID-19 every group member should follow Norwegian Institute of Public Health's updates and guidelines.
- **M2**: Since there is only one semester left of the bachelor degree program, everyone is determined to complete the bachelor thesis. The only measure we can provide in order to make this a certainty, is to support each other and try to create a good work environment for everyone involved.
- **M3**: Would require new equipment to be sent by Telenor.
- **M4**: Depending on what equipment breaks, replacements would have to be made.
- **M5**: Equipment should be under supervision at all times. Upon leaving the testing station, equipment has to be brought along. However, if the equipment gets stolen, we will notify Telenor. Replacements would have to be made either way.
- **M6**: Utilize different platforms until the outage has passed.
- **M7**: Stick to the Gantt scheme timeline. Create a time buffer if problems should arise during the thesis.
- **M8**: Determine what time periods during the day are best suited for testing.
- **M9**: NTNU closing its facilities might happen due to the uncertainty of the ongoing pandemic. This will reduce the amount of testing we will be able to perform in a optimal environments, with a high amount of active users connected at the same time. However, we don't expect this will affect the project negatively. Since the group is able to work from home, we can use backup solutions that affect the signal strength. This can be either glass or wire mesh.
- **M10**: Weekly backups of our thesis will be made in the event that loss of access to the Overleaf editor should occur. We can continue our thesis work on other platforms if/until the issue is resolved.
- **M11**: Write down questions during the week and bring them up in the weekly meetings should our Telenor contacts be too busy. If this turns out

to be a lasting issue, this might need to be brought up during meetings with our supervisor at NTNU or with Telenor.

- **M12**: Talk with degree coordinator, alternatively get assigned a new supervisor.
- **M13**: Conduct testing elsewhere on campus, preferably in the same location for every scenario.

Chapter 6

Plan for Implementation

6.1 Reasoning Behind Planning

Having a good plan is essential for a bachelor thesis, so the group has clear instructions on when to do what or what work needs to be done to reach a milestone or deadline. Without a plan, you don't have a time frame to relate to or an overview of the relevant tasks. For that reason, we have chosen to include a Gantt scheme from start to finish as a guideline for the entirety of the project.

6.2 Gantt Scheme

The Gantt scheme seen in figure 6.1 is a rough timeline of the differing events throughout our thesis. We have omitted meetings from this Gantt scheme, both with our supervisor and with Telenor. While it is related to our thesis, meetings are held nearly every week. As a result, we felt it was pointless to include these in our diagram, as it doesn't fall under any of the existing categories.

There is also a degree of uncertainty relating to the final presentations, and when they'll take place. Initially, it looks like NTNU wants to coordinate this during the start of June, around the 7-10th.

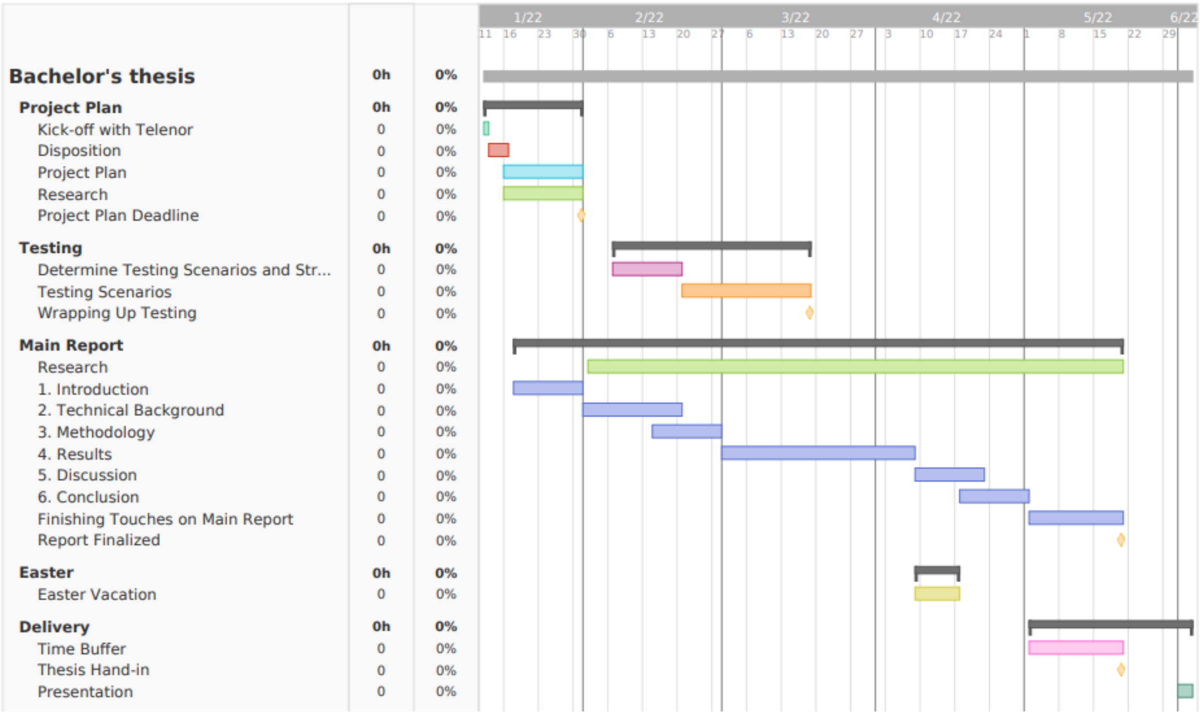

Figure 6.1: Final version of our time table for this project, represented in a Gantt scheme.

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Appendix A

Additional Material

We used a template to create the work contract. This template was provided to us by NTNU, and was given to us during our first semester in the course "DCSG1002 Cybersecurity and Teamwork". We also took some inspiration from our own work contract created in the same course. That work contract was also created by the same template.

Et eksempel på Arbeidskontrakt for Team X

Teammedlemmer:

Grethe Sandstrak, Bjørn Klefstad, Kurt Hansen, Bodil Bjerk

Mål:

Vi skal arbeide aktivt for at teamet skal bli godt kjent med hverandre og forsøke å nå fase 4 i utviklingsfasene for team beskrevet i teorien. Som et resultat av dette arbeidet skal vi være blant de tre beste i IIE-rallyet, gjennomføre et prosjekt som gir oss utfordringer både innen programmering og teamutvikling og bestå kurset prosjektteknikk

For å oppnå dette målet skal vi gjennomføre følgende aksjonsliste:

- Hvert teammedlem skal finne ut hvilke roller (Belbin) de selv dekker
- Alle teammedlemmer skal prøve seg i alle roller (møteleder, sekretær, legobygging, programmering), med unntak av lederrollen, i løpet av kurset
- Vi skal praktisere godfotteorien. Dvs. vi skal gjøre de andre på teamet gode. Vi skal utnytte hverandres sterke sider og utvikle vårt team til å bli det perfekte individ
- Vi ønsker å utnytte gruppens ressurser på en best mulig effektiv måte slik at alle er i aktivitet til enhver tid og er kjent med den interne arbeidsfordelingen
- Gjennom praktisk erfaring skal vi øve inn skriving av programkode, kompileringsprosessen, uttesting av programvare og forbedringer av koden
- Gjennom mengdetrening skal vi utvikle oss til å bli bedre programmerere i Java
- Utover i kurset skal vi fokusere mer og mer på objektorientering av programkoden
- Vårt team skal være blant de tre beste i IIE-rallyet pga. enkle, stabile løsninger både byggeteknisk og programteknisk
- Når vi skal definere prosjektoppgaven på slutten av kurset skal vi utfordre den kreative delen i hvert enkelt teammedlem, slik at vi på bakgrunn av 4 ulike forslag kan diskutere oss frem til en spennende og morsom oppgave

Roller:

Vi vurderer det som hensiktsmessig å ha en fast leder slik at det alltid er klart hvem som sitter med hovedansvaret og hvem det er som skal ta tak i ulike former for problemer. Lederen skal sette i verk nødvendige tiltak for å rette opp eventuelle feilskjær uansett hvilken type feil det er snakk om. Teamet har ubegrenset med tillit til lederen og forventer at denne styrer aktivt for å sikre at teamet når de målsettinger som er skissert. Bodil utnevnes til teamets leder.

Vi har identifisert følgende andre roller: møteleder, sekretær, byggeteknisk sjef, programteknisk sjef. Disse rollene vil gå på omgang etter følgende mønster.

	Møteleder	Sekretær	Byggeteknisk sjef	Programteknisk sjef
Uke 34/35	Grethe	Bjørn	Knut	Bodil
Uke 36/37	Bjørn	Knut	Bodil	Grethe
Uke 38/39	Knut	Bodil	Grethe	Bjørn
Uke 40/41	Bodil	Grethe	Bjørn	Knut

Videre trenger vi kreative personer når vi kommer til den egendefinerte prosjektoppgaven. En spennende prosjektoppgave vil være langt mer tilfredsstillende å arbeide med for alle. Dette blir en utfordring for alle teammedlemmene.

Prosedyrer:

Oppmøte
Alle teammøter med faglærer innebærer obligatorisk oppmøte, møteinnkalling, møteleder, sekretær og møtereferat. Vi ønsker å organisere alle interne teammøter uten faglærer på samme måte en gang hver 14.dag. På de interne teammøtene vil det bli fokusert på status pr i dag, hva skal gjøres videre, fordeling av arbeidet og teamsamarbeidet.
I første omgang skal teamet møtes alle onsdager klokken 09.00 og arbeide med dette faget på skolen i fellesskap frem til klokken 13.00. Etter klokken 13.00 er det opp til det enkelte teammedlem hva og hvor man arbeider så lenge man følger opp de arbeidsoppgaver som er fordelt. Ved spesielle milepæler som IIE-rally og prosjektinnlevering vil vi nok måtte påregne lengre arbeidsdager. Dette avtales underveis.

Fravær
Dersom man blir forhindret fra å møte på avtalte tidspunkter og steder skal det gis beskjed til lederen på forhånd slik at hun har full kontroll på teamet til enhver tid. Dette gjelder både akutt og planlagt fravær. Dessuten skal fravær arbeides inn på et senere tidspunkt etter avtale med lederen/teamet. Dersom lederen er fraværende er det denne ukes møteleder som overtar lederansvaret.

Møter
Alle møteinnkallinger skal sendes ut senest 2 dager før møtet. Møtereferater skal være på plass på it's learning senest neste dag etter at et møte er avholdt. Det skal komme klart frem i referatet hvem som skal gjøre hva, og når det skal være ferdig.
Alle arbeidsoppgaver skal være planlagt løst med 2 dager slakke i forhold til fristen.

Beslutninger
All beslutninger skal fattes i fellesskap der minst 3 teammedlemmer må være tilstede og lederen har dobbeltstemme. Ved stemmelikhet er det lederens stemme som gjelder.

Konflikthåndtering:

Alle uregelmessigheter skal i utgangpunktet diskuteres på teammøter (både de med faglærer og interne møter). Alle teammedlemmene kan ta opp ting de mener avviker fra de avtaler som er inngått. Den personen avviket gjelder skal først kunne komme med forslag til løsning, deretter er det opp til lederen for teamet å løse problemet. Dersom det er lederen problemet gjelder er det denne ukes møteleder som overtar lederansvaret. Og dersom dette ikke er nok til at ting bringes i orden sendes saken videre til faglærer, som da får ansvaret for at vedkommende problembarn igjen blir en ressurs for teamet eller permitteres fra kurset.

Trondheim, 17. august 2016

Grethe Sandstrak	Bjørn Klefstad
____________________________	____________________________
Kurt Hansen	Bodil Bjerk
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A.2 Time Management

A.2.1 Dennis

Timelogg for Dennis Jensen

Uke 2	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer		2 timer	6 timer	2.5 timer			2 timer	12,5
Aktivitet		Lynkurs	Telenor kickoff	Disposisjon			Møte, forprosjekt	
Uke 3	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	3 timer	2.5 timer	4 timer		2 timer		0.5	12
Aktivitet	Møte, forprosjekt	Forprosjekt	Forprosjekt		Teknologi møte		Research SU	
Uke 4	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	4 timer	3.5 timer	6 timer	4 timer	2 timer	5 timer	6 timer	30,5
Aktivitet	Forprosjekt	Møte og kontrakt	Forprosjekt	Forprosjekt	Møte og planlegging	Forprosjekt	Ferdigstille forprosjekt	
Uke 5	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	3 timer	2 timer		4 timer	2 timer	3 timer	5 timer	19
Aktivitet	Levere forprosjektet og gjøre research	Kildesøk og introduksjon		Planlegging og valg av temaer	Oppretting av dokument, møte	Reasearch av mulige temaer	Hovedrapportenen	
Uke 6	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	4 time	4 timer		4 timer	3 timer	5 timer	25
Aktivitet	Hovedrapporten	Motta utstyr fra Telenor og undersøke dette, samt skrive i Hovedrapporten	testing & innkjøp		Møte med veileder og internt med gruppa	Hovedrapporten	Hovedrapporten	
Uke 7	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	6 timer	2 timer	2 timer	4 timer	5 timer	9 timer	28
Aktivitet	Forprosjekt finpuss	Hovedrapporten og møte	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten og finpuss av første utkast til telenor	
Uke 8	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	
Antall timer	4 timer	4 timer	4 timer	3 timer	4 timer	5 timer	2 timer	26
Aktivitet	Første test og baseline	Samle data fra testene og lage presentasjon	Møte med Telenor og jobbe med tilbakemeldinger	Hovedrapportenen	Møte med Ernst og møte med gruppa	Hovedrapportenen	Hovedrapportenen	
Uke 9	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	4 time	4 timer	2 timer	5 timer	5 timer	4 timer	31
Aktivitet	Studere resultater og jobbe med Hovedrapportenen	Researche temaer	Hovedrapportenen	Hovedrapportenen	Møte med veileder diskusjon rundt videre arbeid og planlegge testing	Gjennomføring av baseline	Srudere resultatene og skrive referat	

Uke 10	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	3 timer	3 timer	4 timer	3 timer	4 timer	3 timer	27
Aktivitet	Gjennomføre testing me latency	Planlegge møte med Telenor og delta på møtet	Planlegging av test	testing av hønsenetting og i kjeller	Studere resultatene og skrive referat	testing av hønsenetting og i kjeller i helga	Srudere resultatene og skrive referat	
Uke 11	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	6 timer	4 timer	5 timer	4 timer	6 timer	2 timer	2 timer	29
Aktivitet	Testing	Srudere resultatene og skrive referat	Hovedrapporten	Hovedrapporten	Møte med veileder og jobbe med Hovedrapporten	Forberedelser til testing	testing	
Uke 12	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	4 timer	5 timer	6 timer	5 timer	5 timer	3 timer	4 timer	32
Aktivitet	Studere resultatene og skrive referat	Diskusjon + skriving	Møte med Telenor og jobbe med Hovedrapportenen	Hovedrapportenen	Møte med veileder og jobbe med Hovedrapporten	Hovedrapporten	testing	
Uke 13	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	7 timer	5 timer	4 timer	3 timer	7 timer	4 timer	35
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	
Uke 14	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	6 timer	5 timer	7 timer	6 timer		5 timer	34
Aktivitet	Hovedrapporten	Hovedrapporten	Jobbe med utkast til veileder	Jobbe med utkast til veileder	Møte med veileder og obbe med utkast til veileder	Reise	Ferdigstille utkast til veileder	
Uke 15	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer						4 timer	5 timer	9
Aktivitet	Påskeferie	Påskeferie	Påskeferie	Påskeferie	Påskeferie	Diskusjon og planlegging videre	Hovedrapporten	
Uke 16	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	2 timer	7 timer		5 timer	5 timer	7 timer	8 timer	34
Aktivitet	Hovedrapporten	Hovedrapporten	Reise	testing	Møte med veileder og testing	Studere tester og ta notater	Hovedrapporten	
Uke 17	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	5 timer	6 timer	6 timer	5 timer	8 timer	5 timer	40
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Planlegge nye tester og jobbe med Hovedrapporten	Møte med veileder og testing av packetloss	Studere tester og ta notater	Hovedrapporten	

Uke 18	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	6 timer	7 timer	6 timer	7 timer	4 timer	5 timer	42
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Møte med veileder og jobbe med Hovedrapporten	Hovedrapporten	Hovedrapporten	
Uke 19	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	8 timer	9 timer	5 timer	8 timer	9 timer	9 timer	48
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Møte med veileder og jobbe med Hovedrapporten	Ferdigstille rapporten	Ferdigstille rapporten	
Uke 20	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	9 timer		8 timer	4 timer				21
Aktivitet	Ferdigstille rapporten	17.mai	Ferdigstille rapporten	Ferdigstille og levere rapporten				
Antall timer totalt								535

A.2.2 Knut-Magnus

Timelogg for Knut-Magnus Karlsen								
Uke 2	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer		2 timer	6 timer	2.5 timer	1.5 timer		2 timer	14
Aktivitet		Lynkurs	Telenor kickoff	Disposisjon	PDF & disposisjon		Møte, forprosjekt	
Uke 3	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	3 timer	2.5 timer	4 timer		2 timer			11,5
Aktivitet	Møte, forprosjekt	Forprosjekt	Forprosjekt		Teknologi møte			
Uke 4	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	3.5 timer	6 timer	4 timer	2 timer	5 timer	6 timer	31,5
Aktivitet	Forprosjekt	Møte og kontrakt	Forprosjekt	Forprosjekt	Møte, planlegging	Forprosjekt	Ferdigstille forprosjekt	
Uke 5	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	3 timer		2 timer	4 timer	2 timer	3 timer	5 timer	19
Aktivitet	Levere forprosjektet og gjøre research		Hovedrapporten Teoridel	Planlegging og valg av temaer	Møte & diskusjon	Reasearch av mulige temaer	Hovedrapportenen	
Uke 6	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	3 time	4 timer		4 timer	3 timer	4 timer	23
Aktivitet	Hovedrapporten Teoridel	Motta utstyr fra Telenor og undersøke dette	testing & innkjøp		Møte med veileder og internt med gruppa	Hovedrapportenen	Hovedrapportenen	
Uke 7	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	7 timer	1 time	2 timer	4 timer	5 timer	5 timer	29
Aktivitet	Finpusse forprosjekt etter tilbakemelding.	Møte og jobbe med Hovedrapporten	Hovedrapportenen	Hovedrapportenen	Hovedrapportenen	Hovedrapportenen	Finpuss av første utkast	
Uke 8	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	4 timer	4 timer	4 timer	3 timer	4 timer	5 timer	2 timer	26
Aktivitet	Testing	Samle data fra testene og lage presentasjon	Møte med Telenor og jobbe med tilbakemeldinger	Hovedrapportenen	Møte med Ernst og møte med gruppa	Hovedrapportenen	Hovedrapportenen	
Uke 9	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	4 time	4 timer	2 timer	5 timer	5 timer	4 timer	31
Aktivitet	Studere resultater og jobbe med Hovedrapportenen	Researche temaer	Hovedrapportenen	Hovedrapportenen	Møte med veileder diskusjon rundt videre arbeid og planlegge testing	Gjennomføring av baceline	Srudere resultatene og skrive referat	

Uke 10	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	3 timer	3 timer	5 timer	3 timer	4 timer	3 timer	28
Aktivitet	Gjennomføre testing med latency	Planlegge møte med Telenor og delta på møtet	Planlegging av test	testing av hønsenetting og i kjeller	Studere resultatene og skrive referat	testing av hønsenetting og i kjeller i helga	Srudere resultatene og skrive referat	
Uke 11	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	6 timer	4 timer	5 timer	4 timer	6 timer	2 timer	3 timer	30
Aktivitet	Testing	Srudere resultatene og skrive referat	Hovedrapporten	Hovedrapporten	Møte med veileder og jobbe med Hovedrapporten	Forberedelser til testing	testing	
Uke 12	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	4 timer	5 timer	4 timer	5 timer	5 timer		4 timer	27
Aktivitet	Studere resultatene og skrive referat	Diskusjon + skriving	Møte med Telenor og jobbe med Hovedrapportenen	Hovedrapportenen	Møte med veileder og jobbe med Hovedrapporten		testing	
Uke 13	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	7 timer	6 timer	6 timer	4 timer	8 timer	4 timer	40
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	
Uke 14	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	6 timer	5 timer	5 timer	6 timer	7 timer	5 timer	39
Aktivitet	Hovedrapporten	Hovedrapporten	Jobbe med utkast til veileder	Jobbe med utkast til veileder	Møte med veileder og obbe med utkast til veileder	Jobbe med utkast til veileder	Ferdigstille utkast til veileder	
Uke 15	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer						4 timer	5 timer	9
Aktivitet	Påskeferie	Påskeferie	Påskeferie	Påskeferie	Påskeferie	Diskusjon og planlegging videre	Hovedrapporten	
Uke 16	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	7 timer	8 timer		5 timer	7 timer	8 timer	42
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten og forberedelser til testing		Møte med veileder og testing	Studere tester og ta notater	Hovedrapporten	
Uke 17	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	5 timer	5 timer	5 timer	7 timer	8 timer	5 timer	42
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Møte med veileder og testing av packetloss	Studere tester og ta notater	Hovedrapporten	

Uke 18	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	8 timer	6 timer	7 timer	6 timer	7 timer	5 timer	6 timer	45
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	
Uke 19	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer			5 timer	8 timer	7 timer	8 timer	33
Aktivitet	Wireshark			Wireshark	Møte med veileder og jobbe med Hovedrapporten	Wireshark og rapporten	Ferdigstille rapporten	
		Reising						
Uke 20	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	9 timer		8 timer	4 timer				21
Aktivitet	Ferdigstille rapporten	17.mai	Ferdigstille rapporten	Ferdigstille og levere rapporten				
Antall timer totalt								541

A.2.3 Ole Morten

Timelogg for Ole Morten Ystad Karlsen								
Uke 2	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer		2 timer	6 timer	2 timer			2 timer	12
Aktivitet		Lynkurs	Telenor kickoff	Disposisjon			Planlegging av møte med veileder	
Uke 3	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	3 timer	2,5 timer	4 timer					9,5
Aktivitet	Møte med veileder + jobbing med forprosjekt	Forprosjekt	Forprosjekt	Hemsedal	Hemsedal	Hemsedal	Hemsedal	
Uke 4	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	4 timer	4 timer	6 timer	4 timer	1 time	5 timer	6 timer	30
Aktivitet	Forprosjekt	Møte med Telenor + opprette kontrakter	Forprosjekt	Forprosjekt	Møte	Forprosjekt	Finpuss på forprosjekt	
Uke 5	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	3 timer	3 timer	3 time	4 timer	2 timer	3 timer	5 timer	23
Aktivitet	Levere forprosjektet og gjøre research	Hovedrapporten	Kildeføring og referering	Planlegging og valg av temaer	Møte og diskusjon	Reasearch av mulige temaer	Hovedrapportenen	
Uke 6	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	3 time	4 timer	1 time	4 timer	3 timer	4 timer	24
Aktivitet	Reasearch bandwidth og throghput	Motta utstyr fra Telenor og undersøke dette	teste utstyr og kjøpe utstyr	Hovedrapporten	Møte med veileder og internt med gruppa	Hovedrapportenen	Hovedrapportenen	
Uke 7	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	7 timer	5 timer	2 timer	4 timer	5 timer	5 timer	33
Aktivitet	Finpusse forprosjektet og endringer generelt etter tilbakemeldinger	Møte med Telenor og jobbe med Hovedrapportenen	Hovedrapportenen	Hovedrapportenen	Hovedrapportenen	Hovedrapportenen	Finpuss av første utkast	
Uke 8	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	4 timer	4 timer	4 timer	3 timer	4 timer	5 timer	2 timer	26
Aktivitet	Testing	Samle data fra testene og lage presentasjon	Møte med Telenor og jobbe med tilbakemeldinger	Hovedrapportenen	Møte med veileder og møte med gruppa	Hovedrapportenen	Hovedrapportenen	
Uke 9	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	4 time	4 timer	2 timer	5 timer	5 timer	4 timer	31
Aktivitet	Studere resultater og jobbe med Hovedrapportenen	Researche temaer	Hovedrapportenen	Hovedrapportenen	Møte med veileder diskusjon rundt videre arbeid og planlegge testing	Gjennomføring av baceline	Srudere resultatene og skrive referat	

Uke 10	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	3 timer	3 timer	5 timer	3 timer	4 timer	3 timer	28
Aktivitet	Gjennomføre testing med latency	Planlegge møte med Telenor og delta på møtet	Planlegging av test	testing av hønsenetting og i kjeller	Studere resultatene og skrive referat	testing av hønsenetting og i kjeller i helga	Srudere resultatene og skrive referat	
Uke 11	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	6 timer	4 timer	5 timer	4 timer	6 timer	2 timer	3 timer	30
Aktivitet	Testing	Srudere resultatene og skrive referat	Hovedrapporten	Hovedrapporten	Møte med veileder og jobbe med Hovedrapporten	Forberedelser til testing	testing	
Uke 12	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	4 timer	5 timer	6 timer	5 timer	5 timer	3 timer	4 timer	32
Aktivitet	Srudere resultatene og skrive referat	Diskusjon + skriving	Møte med Telenor og jobbe med Hovedrapportenen	Hovedrapportenen	Møte med veileder og jobbe med Hovedrapporten	Hovedrapporten	testing	
Uke 13	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	7 timer	6 timer	6 timer		8 timer	4 timer	36
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Reising	Hovedrapporten	Hovedrapporten	
Uke 14	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	6 timer	5 timer	5 timer	6 timer	7 timer	5 timer	39
Aktivitet	Hovedrapporten	Hovedrapporten	Jobbe med utkast til veileder	Jobbe med utkast til veileder	Møte med veileder og obbe med utkast til veileder	Jobbe med utkast til veileder	Ferdigstille utkast til veileder	
Uke 15	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer						4 timer	5 timer	9
Aktivitet	Påskeferie	Påskeferie	Påskeferie	Påskeferie	Påskeferie	Diskusjon og planlegging videre	Hovedrapporten	
Uke 16	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	7 timer	8 timer	5 timer	5 timer	7 timer	8 timer	47
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten og forberedelser til testing	testing	Møte med veileder og testing	Studere tester og ta notater	Hovedrapporten	
Uke 17	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	7 timer	5 timer	6 timer	6 timer	5 timer	8 timer	5 timer	42
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Planlegge nye tester og jobbe med Hovedrapporten	Møte med veileder og testing av packetloss	Studere tester og ta notater	Hovedrapporten	

Uke 18	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	8 timer	6 timer	7 timer	6 timer	9 timer	5 timer	6 timer	47
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Møte med veileder og jobbe med Hovedrapporten	Hovedrapporten	Hovedrapporten	
Uke 19	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	5 timer	8 timer	9 timer	5 timer	8 timer	9 timer	8 timer	52
Aktivitet	Hovedrapporten	Hovedrapporten	Hovedrapporten	Hovedrapporten	Møte med veileder og jobbe med Hovedrapporten	Ferdigstille rapporten	Ferdigstille rapporten	
Uke 20	Mandag	Tirsdag	Onsdag	Torsdag	Fredag	Lørdag	Søndag	Timelogg for uken
Antall timer	9 timer		8 timer	4 timer				21
Aktivitet	Ferdigstille rapporten	17.mai	Ferdigstille rapporten	Ferdigstille og levere rapporten				
Antall timer totalt								571,5

A.3 Testing Scenarios Excess Data

Excess data regarding the different testing scenarios gathered from Wireshark.

Baseline

Metric	Value
Amount of P frame packets sent	57 993
P frames average Kbps sent	2 572 Kbps
Amount of P frame packets received	54 553
P frames average Kbps received	2 487 Kbps
Amount of I frame packets sent	2 510
I frames average Kbps sent	126 Kbps
Amount of I frame packets received	3 959
I frames average Kbps received	146 Kbps

Basement in Atrium

Metric	Value
Amount of P frame packets sent	3 461
P frames average Kbps sent	153 Kbps
Amount of P frame packets received	23 146
P frames average Kbps received	1049 Kbps
Amount of I frame packets sent	366
I frames average Kbps sent	12 Kbps
Amount of I frame packets received	989
I frames average Kbps received	30 Kbps

Campus Cafeteria

Metric	Value
Amount of P frame packets sent	12 939
P frames average Kbps sent	504 Kbps
Amount of P frame packets received	20 216
P frames average Kbps received	782 Kbps
Amount of I frame packets sent	221
I frames average Kbps sent	11 Kbps
Amount of I frame packets received	608
I frames average Kbps received	13 Kbps

Atriet with human blockage

Metric	Value
Amount of P frame packets sent	2 059
P frames average Kbps sent	93 Kbps
Amount of P frame packets received	4 501
P frames average Kbps received	218 Kbps
Amount of I frame packets sent	82
I frames average Kbps sent	17 Kbps
Amount of I frame packets received	101
I frames average Kbps received	5 Kbps

A.4 Minutes of Meeting With Supervisor

Møtereferat for møte 17.01.2022

Deltakere: Ernst Gunnar Gran (veileder), Dennis Jensen, Knut-Magnus Karlsen, Ole Morten Ystad Karlsen

Sekretær for møtet: Ole Morten Ystad Karlsen

Tema for møtet: Planlegging av møter og diskusjon rundt oppgaven

- Det ble avklart at møter med veileder skal holdes hver uke på fredager klokken 11:00. Endringer i møteplanen skal avklares på forhånd. Eks. møte 28.01 som ble flyttet fra klokken 11:00 til 12:00.
- Ernst kontaktet Erik, og fikk tak i en mal for eksamensskriving i Overleaf latex
- Videre ble det avklart litt rundt oppgaven, slik som at problemstillingen og prosjektplanen skal være presise og tilpasset oppgaven.
- Det ble anbefalt av Ernst å benytte google, spesifikt google scholar, til å finne relevant lesestoff til oppgaven.
- Gruppen hadde i utgangspunktet planlagt å utføre tre ulike senarioer, men etter litt diskusjon med veileder ble det anbefalt å snakke med arbeidsgiver om hva som skaper mest verdi for dem.
- For å utføre de ulike senarioene har gruppen behov for innkjøp av webkameraer. Det viste seg at NTNU ikke hadde noen ordning på finansiell støtte til utstyr på bacheloroppgaver. Det ble dermed bestemt å ta kontakt med Telenor om eventuelt sponsing av slikt utstyr.

Varighet for møtet: 40 min

Neste møte: fredag 28.01 klokken 12:00

Møtereferat for møte 04.02.2022

Deltakere: Ernst Gunnar Gran (veileder), Dennis Jensen, Knut-Magnus Karlsen, Ole Morten Ystad Karlsen
Sekretær for møtet: Ole Morten Ystad Karlsen
Tema for møtet: Tilbakemeldinger på forprosjekt og videre arbeid

- Rapporten ser bra ut ved første øyekast.
- Unngå bruk av «vi» og andre liknende ord som personaliserer teksten i selve oppgaven. Hvis man unngår dette, vil det gjøre oppgaven mer akademisk.
- Gantdiagrammet burde deles i flere deler. Del opp rapportskrivingen litt mer i perioder, slik at man kan konkretisere når forskjellige deler av rapporten skal være ferdig.
- Dersom det er noen spørsmål til Telenor eller veileder om teknologier, så bør det gjøres litt reasearch på forhånd. Dette er for å få bedre innsikt rundt teknologiene.
- Tilgang til eget rom på skolen for utstyret er aktuelt. Send en e-post til Ernst etter helga.
- Fysisk møte neste uke neste uke, 11.02, i det Topaz-bygget på campus. Møtes i 5.etasje.

Varighet for møtet: 18 min
Neste møte: fredag 11.02 klokken 11:00

Møtereferat for møte 22.04.2022

Deltakere: Ernst Gunnar Gran (veileder), Dennis Jensen, Knut-Magnus Karlsen, Ole Morten Ystad Karlsen

Sekretær for møtet: Ole Morten Ystad Karlsen

Tema for møtet: Første møte etter påskeferie, status og arbeid videre

- Kommentarer til kapitel 4 (generelle kommentarer etter å ha skumlest teksten)
    - God setningsoppbygging
    - Oppgaven mangler en tydelig problem statement som detaljerer hva oppgaven går ut på.
    - Oppgaven bør omstruktureres slik at det blir en tydelig «rød tråd» gjennom hele teksten. Dette handler blant annet om å ha teoretisk bakgrunn før teknologisk bakgrunn.
    - Hvordan skal resultatene presenteres og fremstilles slik at de skilles fra hverandre (god, medium, dårlig kvalitet)?
    - Forklar hvordan man vet hvilke faktorer som påvirker testene. Gå i detalj på hvorfor det blir dårlig.
- Sjøkablene som nevnes i oppgaven kan skrives bedre inn i oppgaven. Dette kan være et godt sted å introdusere internett og andre relevante begreper som bandwidth.
- Spør Telenor om hvorfor mobilapplikajsonen som de anbefaler, ikke samsvarer med finnsenderen når det kommer til geolokasjon.
- Lag mest mulig figurer og tabeller som samsvarer med hverandre. Disse bør lages selv slik at oppgaven blir mer personlig.

Varighet for møtet: 40 min

Neste møte: Fredag 29.04 klokken 11:00

Møtereferat for møte 29.04.2022

Deltakere: Ernst Gunnar Gran (veileder), Dennis Jensen, Knut-Magnus Karlsen, Ole Morten Ystad Karlsen

Sekretær for møtet: Ole Morten Ystad Karlsen

Tema for møtet: Progresjon og tilbakemelding av oppgave 1-3

- Oppsummering av arbeid så langt
- Gjennomgang av tilbakemelding
    - Hadde vært fint med en tydeligere definisjon av hypotese i kap. 1
    - Justere rekkefølgen til hovedkapitler og delkapitler slik at det blir bedre overgang og flyt mellom kapitlene. Dette gjelder kap. 2 og 3.
    - Savner en introduksjon til om viktige uttrykk som IP, UDP og TCP. Trenger ikke være mer enn ett par setninger.
    - Fjerne temaer som ikke er relevante til oppgaven. Dette gjelder blant annet teknologier som peering.

Varighet for møtet: 40 min

Neste møte: fredag 06.05 klokken 11:00

Møtereferat for møte 06.05.2022

Deltakere: Ernst Gunnar Gran (veileder), Dennis Jensen, Knut-Magnus Karlsen, Ole Morten Ystad Karlsen

Sekretær for møtet: Ole Morten Ystad Karlsen

Tema for møtet: Progresjon og videre arbeid

- Prøv å forklare begreper og definisjoner i kap 2-4. Nye kapitler bør ikke introduseres i kapitel 6. Kapitel 6 skal reflektere over det som allerede er blitt introdusert.
- Prøv å engasjere leseren, både gjennom tekst og figurer. Figurer gjør det lett å visualisere tematikken.
- Gjennomgang av kap 4
  - Kapitlet bør starte med å introdusere de forskjellige testene. Start med å presentere baseline. Bruk gjerne en figur for å visualisere og referere til hvordan ting henger sammen. Det er viktig å fremheve hvordan senarioene fungerer på en forståelig måte.
  - Kapitlet nevner at det ønskes «god kvalitet», uten å spesifisere hva som menes med dette.
- Ikke bruk ord som "hopefully" som skaper usikkerhet. Skriv heller at dette var noe som vi fant ut, slik at det høres troverdig ut.
- Spør Telenor om begrepene «kapasitetsbånd» og «dekningsbånd», for å finne ut hva de omhandler.
- Markere tydelig hvilke dokumenter som vi har fått privat i kildeliste. Muntlige kommentarer fra Telenor og andre pålitelige kilder kan også brukes.
- Visualiser hvor de ulike mobiltårnene står. Bruk gjerne mazemap til dette.
- Prøv å beskrive valg så mye som mulig, f.eks. hvorfor bruker vi discokuler?
- Prøv å konkretisere teksten hvis det er mulig. Gjørdet kort og konsist. Det er bedre å end opp på 50-70 sider enn å prøve på å skrive mer. Det bør være veldig bra dersom man skal noe særlig over 70s. Bør ikke forklare ting flere ganger.

Varighet for møtet: 60 min

Neste møte: fredag 13.05 klokken 11:00

Møtereferat for møte 13.05.2022

Deltakere: Ernst Gunnar Gran (veileder), Dennis Jensen, Knut-Magnus Karlsen, Ole Morten Ystad Karlsen

Sekretær for møtet: Ole Morten Ystad Karlsen

Tema for møtet: Siste møte før levering, progresjon og videre arbeid

- Kommentarer til kapitel 5 og testing
    - Hva betyr latency (sender, mottaker, ende-til-ende)? Hva gjør at latency har så stor forskjell? Hvorfor skjer dette?
    - Kan det være forsinkelse på lag 2 i OSI-modellen? Kan denne forsinkelsen gjøre at latency treigere? Er det noe levringsgaranti i lag 2 i OSI-modellen?
    - Stemmer konklusjonen vår om justering av nettkvalitet? Skjer det pga. "fare" overføring (sender like mye audio pakker som video pakker)? Kontakt Cisco om dette.
- Hypotesene bør forklares bedre og begreper som «dårlig forbindelse» må defineres.
- Noe av det som står i kapittel 6 omhandler metode og burde egentlig å ha stått i kapitel 4.
- Kilder i kapitel 5 og 6 kan brukes til å sammenlikne liknende resultater. Dette gjelder spesielt påstander.
- Skriv litt om i frames og p frames, og legg til andre relevante teknologiste beskrivelser i teoretical background.

Varighet for møtet: 90 min

Neste møte: Starten av juni, før presentasjonen.

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