Utilising peer-to-peer networking for data transfer over the browser

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Chapter 1

Interim Planning & Investigation Report

1.1 Project Scope

Aims and Objectives

The ultimate objective of my project is to investigate the possibility of data transfer over web browsers (in particular audio and video streaming) without the need for a centralised client-server architecture, instead opting for a peer-to-peer network architecture. In order to do so, I plan to:

- Research peer-to-peer networking architecture
- Research WebRTC
- Research signalling protocols
- Research media streaming compression & protocols
- Research client-side JavaScript frameworks
- Develop a session signalling server in Java with the WebSockets protocol
- Develop a WebRTC Application to transfer and stream uploaded files.
- Implement a peer to peer network between peers transferring/streaming a file using the web application.

Stakeholders

The stakeholders involved in my project will be myself, my supervisor, Stelios Kapetanakis and the user.

Methods of Communication

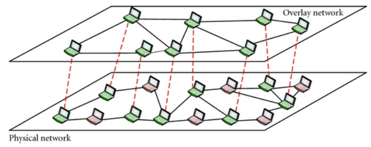
Stelios and I have set up a regular meeting once a week on Friday at 4pm to review progress and answer any questions. On top of this, we communicate regularly via email and Stelios has access to a project git repository on GitHub and a workflow board set up on my web server to monitor the progress of my project.

1.2 Research

Comparing Network Architectures

Peer-to-peer networking is the distribution of resources and processing between the nodes of a network. These networks tend to take the form of an overlay network. An overlay network is a topology describing a virtual network that sits on top of a physical network (e.g the internet) consisting of a subset of the nodes connected to the physical network.

Figure 1.1: An Unstructured Peer-To-Peer Overlay Network [6]



Although an overlay network does not need to manage the burden the physical network is responsible for, in order to discover new nodes and route them together, it needs to have some way of managing this subset of nodes. This is normally achieved in 3 different ways:

- Unstructured peer-to-peer: Peer connections in the overlay are established randomly. New peers copy the connections another peer has formed and develops it's own over time. [7]
- Structured peer-to-peer: Peers in a network are organised using a distributed hash table (DHT). The DHT consists of many peers maintaining a partial hash table containing the unique keys of it's neighbouring peers, allowing it to route to them. This way, a node can communicate with another by hopping a message through other nodes until it is reached. [7]
- Hybrid peer-to-peer: A combination of peer to peer and client-server models, peers are discovered and routed by a centralized signalling server. The responsibilities of this server can vary in degree as they can leave features that function better in decentralized networks to the nodes in order to improve performance whilst maintaining responsbility for the features that function better using a centralized server.[8]

My application will most closely resemble a hybrid peer-to-peer network as even though the actual channel for transferring data between peers will not go through a server, it will use a signalling server to initially establish a session between them.

Currently, the most common network architecture for file storage/transfer and video streaming services is the client-server model. In a basic sense, this is when the machines processing data (servers) are distinct from the computers requesting this processing (client), although in actuality, the boundaries between the two split dependent on how it is implemented. Whilst the shift of processing on to the service supplier means that the user of the application does not need a powerful machine to run it, it also means that the service supplier has the responsibility of maintaining and paying for these servers. Furthermore, although modern server architectures are designed in a distributed structure granting them to be more robust (as distribution allows for redundancy and load balancing), if these servers undergo too much load or in extreme cases, if the server farm completely crashes, the user would not be able to use the application as intended.

Using peer to peer architecture avoids these issues with the client-server model as processing is distributed between the nodes making up the network, resulting in it being cheaper to maintain and scale the service as load increases whilst being inherently robust. However, there are a number of considerations to be aware of in peer to peer networking. Security is a concern within peer-to-peer networking as there is no centralized authority to manage access control to resources. Thus, a malicious client could potentially launch a number of attacks versus other nodes in the network such as Denial Of Service (DoS) where a malicious group of nodes floods a network with a massive amount of fake data, potentially crashing nodes within it or Man in the Middle (MitM) attacks where a node is able to intercept data flowing between two other nodes and spy on or poison this data.[9]

Whilst it would be hard to completely avoid risks like this as it is an inherent flaw in peer-to-peer architecture, introducing the basic concept of trust into a peer-to-peer network can stop malicious peers from entering it in the first place. Trust will be implemented in my application by allowing users to enable authentication (via password) for the uploaded file giving them a level of access control that would prevent malicious attackers from being able to join the network of peers involved in the download or stream of the file.

Current Solutions to Data Transfer

Currently, solutions such as DropBox and Google Drive focus on cloud storage as a service whilst providing the ability to transfer your data as a secondary function of this service. Although this is an extremely valuable and convenient idea enabling consumers to back-up their data, not all want a service that stores the data they want to transfer as the practice of storing data in the cloud does raise ethical issues, namely with it's security, privacy and ownership once it has been uploaded. However, this can be hard to avoid when the online storage and transfer of data are so closely intertwined.

Once you have uploaded files to one of these services, there is a trust placed on the company to store the data securely. However, The "Cloud Security Alliance" released a regularly updated list of the top threats faced by cloud computing services in 2010 submitting there are numerous security issues faced within the cloud computing industry with many of the issues on this list occurring due to malicious intentions, for example the 6th item on the list, "malicious insiders" describes a situation in which current or former employees abuse their authorizations within system to gain access to sensitive information [1]. This suggests that even though companies can protect their client's data to an extent, there are still issues the industry face that consumers may want to avoid.

Whilst the uploaded data can be encrypted in a way that means only clients of the service can decrypt it (through end-to-end encryption), many people feel insecure leaving traces of their private information on a remote server belonging to a company that is not necessarily always acting in their interest. In 2013, this concern was legitimized by Edward Snowden when he disclosed information about government surveillance programs such as PRISM which coerced firms such as Google, Microsoft and Yahoo to provide private consumer data from their services to the government.[2] More recently, this idea has been reinforced by legislation such as the "Investigatory Powers Bill" being introduced in the UK that could prohibit companies from using end-to-end encryption techniques allowing the government to request decrypted client data [3]. Whilst we can assume the UK government have done this with non-malicious intentions, by weakening the encryption practices of these companies, they will potentially weaken the defence against malicious attackers.

As an alternative to using cloud storage solutions for sharing files, there is also specific file sharing sites such as WeTransfer that allows you to send files up to 2gb via their site. This works by sending a link through email to the person you want to send the file to. Whilst it does store the uploaded files, they are only kept for 7 days in order to prevent unnecessary storage usage [4]. Although this is better for data privacy, WeTransfer use third party providers such as Amazon Web Services (AWS) to store this data using AWS S3 Storage [5] meaning there is still the potential concern of data privacy and security highlighted previously.

Peer-to-peer networks have been also been popular for file sharing since the late 1990s due to the rise of applications such as Napster which allowed users to share music over one. Napster worked by using a central server to index files that each peer's made available on their machine. Each peer would then be able to search the index for copies of songs. However, this service was eventually shut down as the company ran into legal problems with copyright infringement. Since then, protocols such as BitTorrent have improved on the idea of peer to peer sharing. The BitTorrent protocol works by peers creating ".torrent" descriptor files that describe the file's metadata, this is then shared however it is seen fit. Each peer wanting to download this file then picks up this ".torrent" file, opens it with a BitTorrent client and becomes one of the many "leechers" trying to download this file. At the same time, this peer becomes one of the many "seeders" uploading this file as they download it so that other leechers can download it from them. The initial peer discovery is based on "trackers" which are servers hosting lists of seeders and leechers for the file. One of the issues with BitTorrent is that in order to share a file, you must go through the process of download a client, creating a ".torrent" file, add it to a trackers list and share the ".torrent" with the peers you wish to share it with and this can be seen as a arduous task for a basic consumer looking to share their files.

As an alternative to these current solutions, the application I plan to develop has the aim of separating the online transfer of data from it's storage by connecting the peers involved in the transfer directly with each other. This circumvents the problems with cloud storage as the data being transferred does not pass through any servers as well as the problems with current peer to peer file sharing as it does not require a desktop client or torrent files. To allow my application to share files online, WebRTC will be used.

WebRTC

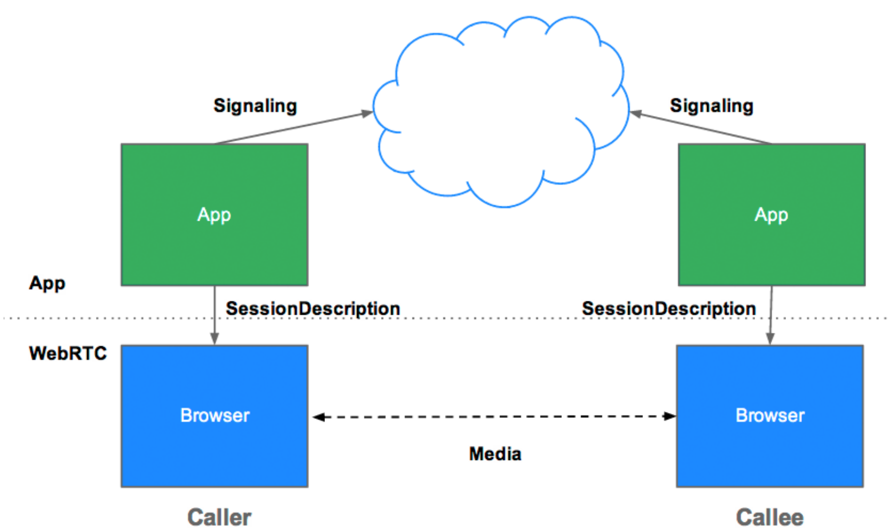
WebRTC (Real Time Communication) is an emerging web technology that enables browsers to communicate real time via a peer-to-peer connection, avoiding the need for a centralized server to transfer data between clients. This was first released by Google as an open source project in May 2011 [11] and was later drafted as an API definition by W3C which is still a work in progress. [12]. WebRTC has yet to be fully implemented in every web browser but Chrome, Firefox and a WebRTC specific browser called Bowser are leading the way in implementing the API definition. Firefox (Nightly) is leading the way in this so the application will specifically be developed towards this browser[13]. The 3 main WebRTC APIs supported at this time are:

- RTCDataChannel: "The RTCDataChannel interface represents a bi-directional data channel between two peers of a connection." [14]
- RTCPeerConnection: "The RTCPeerConnection interface represents a WebRTC connection between the local computer and a remote peer. It is used to handle efficient streaming of data between the two peers." [14]
- getUserMedia: "Prompts the user for permission to use one video and/or one audio input device such as a camera or screensharing and/or a microphone. If the user provides permission, then the successCallback is invoked with the resulting MediaStream object as its argument. If the user denies permission or media is not available, then the errorCallback is called with PermissionDeniedError or NotFoundError respectively. Note that it is possible for neither completion callback to be called, as the user is not required to make a choice." [14]

In order to achieve it's aim, the application will utilize the RTCPeerConnection and RTCDataChannel APIs. The former to establish a peer connection between two clients and the latter to create a data channel over this peer connection to transfer data.

As mentioned in the Javascript Session Establishment Protocol (JSEP) standard[15], although WebRTC and the browser is used to transfer the data between two peers, it purposely does not handle signalling. Signalling is a concept that came from telecommunications and VoIP and is the process of organising the communication between two clients, handling the exchange of metadata that creates and manages a session. The rationale behind this technology being signalling protocol-agnostic is that different applications will require particular protocols in order to, for example, fit into previously existing architecture. In the case of my application, to handle signalling, Session Initiation Protocol (SIP) over WebSockets will be used.

Figure 1.2: WebRTC Architecture as defined by JSEP [15]



WebSockets is a protocol implemented by browsers and web servers running over TCP to provide bi-directional communication between them. Once it establishes a session with a client connecting to the server, this session is left open for the server to send messages to the client and vice versa, making it a good candidate for signalling with a WebRTC application which needs peers to be able to reliably signal messages back and forth through the server. SIP is a communication protocol that does not specify how signals are transported (e.g WebSockets) but how these signals are defined. The transaction model defined by SIP is similar to HTTP with each SIP request being matched by a SIP response.

Figure 1.3: SIP over websockets transaction example[16]

Initial handshake with WebSocket signalling server over HTTP
Alice -> proxy.example.com (TLS)

```
GET / HTTP/1.1
Host: proxy.example.com
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Key: dGhlIHNhbXBsZSBub25jZQ==
Origin: https://www.example.com
Sec-WebSocket-Protocol: sip
Sec-WebSocket-Version: 13
```

Switching to WebSocket protocol
proxy.example.com -> Alice (TLS)

```
HTTP/1.1 101 Switching Protocols
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Accept: s3pPLMBiTxaQ9kYGzzhZRbK+xOo=
Sec-WebSocket-Protocol: sip
```

SIP REGISTER request letting the server know of Alice's "location"
Alice -> proxy.example.com (TLS)

```
REGISTER sip:proxy.example.com SIP/2.0
Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bKasudf
From: sip:alice@example.com;tag=65bnmj.34asd
To: sip:alice@example.com
Call-ID: aiuy7k9njasd
CSeq: 1 REGISTER
Max-Forwards: 70
Supported: path, outbound, gruu
Contact: <sip:alice@df7jal23ls0d.invalid;transport=ws>
;reg-id=1
;+sip.instance="<urn:uuid:f81-7dec-14a06cf1>"
```

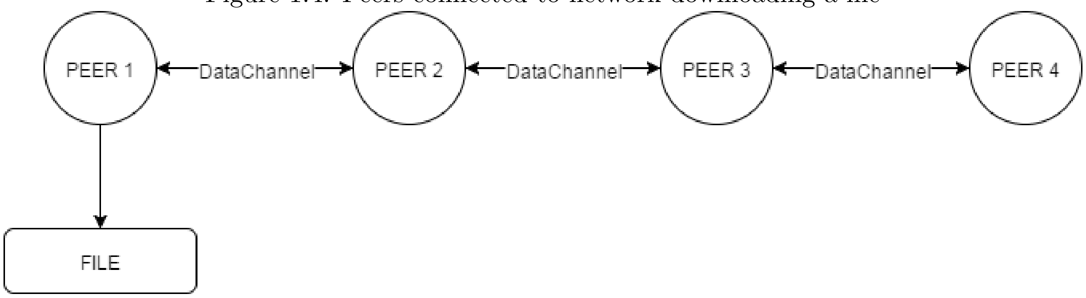
OK response
proxy.example.com -> Alice (TLS)

```
SIP/2.0 200 OK
Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bKasudf
From: sip:alice@example.com;tag=65bnmj.34asd
To: sip:alice@example.com;tag=12isjljn8
Call-ID: aiuy7k9njasd
CSeq: 1 REGISTER
Supported: outbound, gruu
Contact: <sip:alice@df7jal23ls0d.invalid;transport=ws>
;reg-id=1
;+sip.instance="<urn:uuid:f81-7dec-14a06cf1>"
;pub-gruu="sip:alice@example.com;gr=urn:uuid:f81-7dec-14a06cf1"
;temp-gruu="sip:87ash54=3dd.98a@example.com;gr"
;expires=3600
```

Once two peers have registered with the server, a peer can send an invite to another through it in the form of an INVITE request to form a signalling channel between them which WebRTC can use to establish a peer connection and data channel. To create this peer connection, it uses interactive connectivity establishment (ICE) to find a list of the possible IP's and ports of each peer which are then sent through the server. Once this is done, the data channel can be formed and packets can be sent peer-to-peer. These packets are managed and secured by Secure Real Time Transport Protocol (SCTP) and encrypted by Datagram Transport Layer Security (DTLS).

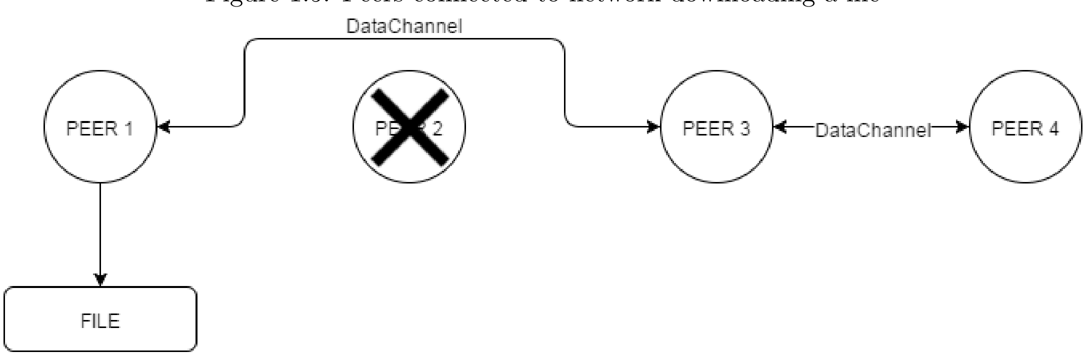
Whilst this one-to-one peer connection is formed and maintained by WebRTC, it does no more than this. To form a scalable network of peers, my signalling server and client side application will have to manage how the peers connect to each other. The idea behind this is that when a new peer connects, they will connect to the second newest peer and so forth.

Figure 1.4: Peers connected to network downloading a file



If a peer disconnects, then the network should reroute the neighbouring peer to the peer before that.

Figure 1.5: Peers connected to network downloading a file



Network topology

Media streaming

Media Streaming

1.3 Specification

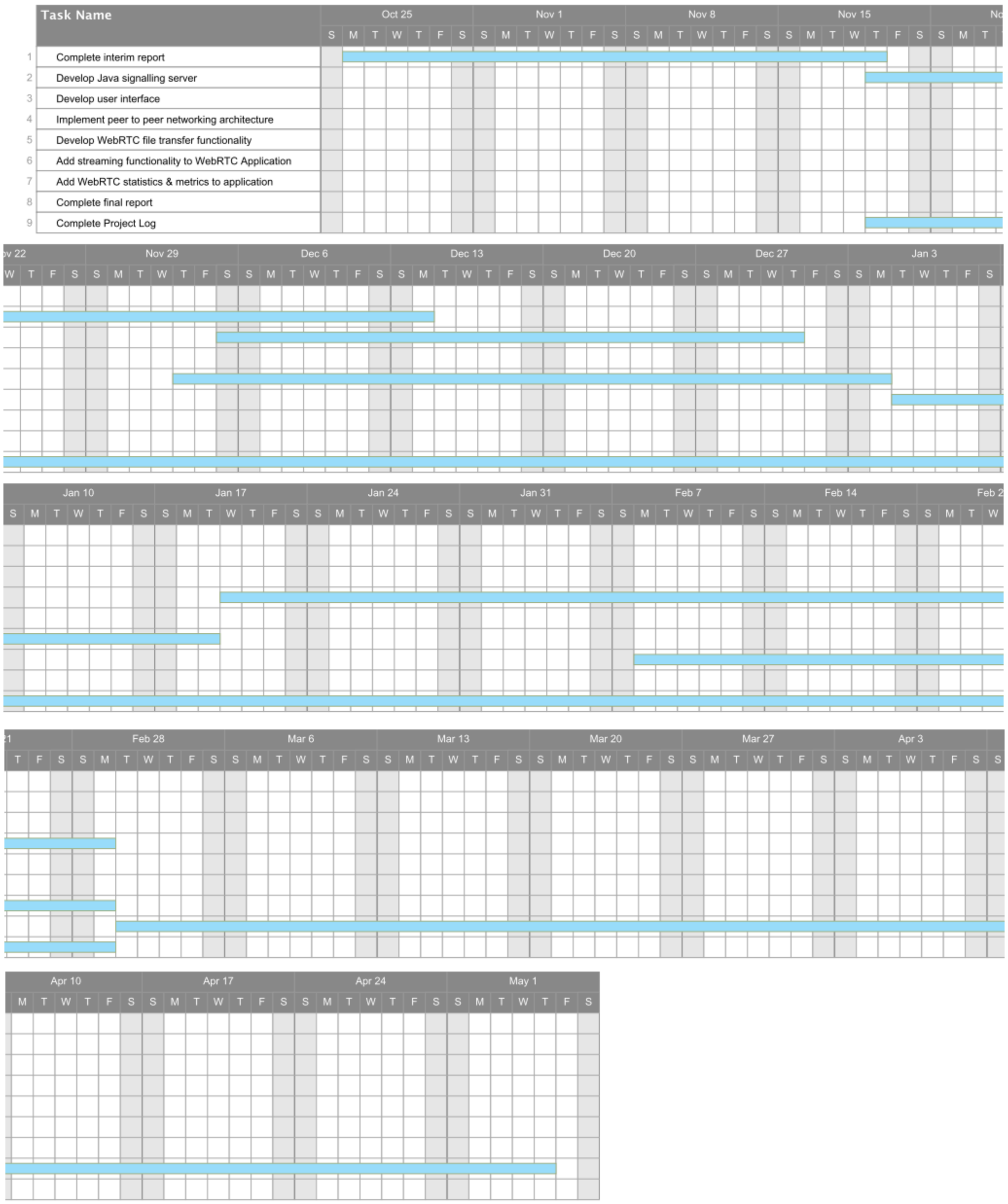
Deliverables

The first intermediate product will be the signalling server written in Java. This will handle the exchange of client meta data in order to establish the connection between two peers using the web application. I plan to overlap the development of this with the development of the data transfer functionality and user interface of my second deliverable, the web application as manually testing the signalling server will be a lot easier with a partially developed application to test it with.

The first end deliverable will be the client side web application the user interacts with in order to select a file as well as handling sending the meta data to the signalling server and managing the peer-to-peer data transfer and media streaming. This is broken down into several intermediate products, the data transfer functionality, the media streaming functionality, the peer-to-peer network and the user interface. I plan to produce the data transfer functionality first along side the user interface to allow for manual testing. After I have implemented data transfer, I will work on media streaming and forming the peer-to-peer network topology.

The second end deliverable will be the final report containing documentation and analysis using the metrics from my application, comparing how it and technologies behind it perform in comparison to others, focusing in particular on how peer-to-peer over the browser (WebRTC) compares to other methods of data transfer and media streaming.

Schedule of Activities



Risk Analysis

Risk	Probability (1-5)	Impact (1-5)	Mitigation	Contingency
Illness/Injury	4	3	Reserve time for illness Be hygienic Eat healthy Exercise	Allow time for recovery Take medicine to aid recovery
Inaccurate estimations	3	3	Be liberal with estimations Reserve time for deliverables behind schedule	Adjust scope of project
Data loss	1	5	Use a version control system Keep local backups	Recover data from Git
Uncommunicative stakeholder	1	3	Ensure regular meetings with stakeholder	
Stakeholder turnover	1	4	N/A (out of my control)	Get new stakeholders
Project scope too large	3	4	Research enough to be certain in project scope Be liberal with estimations	Adjust scope of project
Technologies too immature/insufficient for project	2	4	Research technologies beforehand	Find alternative technologies Adjust scope of project

write more

Quality Analysis

The main measure of success will be the web application's performance in it's ability to transfer & stream data between two peers. To track this, I will implement metrics using the WebRTC "getStats" statistics API, which allows for monitoring of peer connections. Further to this, once I have implemented a many-to-many peer-to-peer network, I will test the application with multiple peers in order to replicate load. The application will also be user tested to ensure it is user-friendly. The signalling server will be load tested in isolation to make certain it can handle multiple requests.

1.4 Methodology

I chose to use an alternative methodology to the waterfall model because it lacks the ability to adapt to changes in a project deadline. Due to the way waterfall is structured into different phases that must be completed sequentially, often when changes such as new requirements occur, all these phases must be repeated in order to account for this. Iterative methodologies take an approach that can adapt to these changing requirements because they utilise short development cycles and focus on developing small modules of a product at one time, making it easier to revise a product if necessary. This is particular useful in my project as it is relatively experimental and the requirements of it may change regularly.

Thus, as a way of tracking the progression of my project, I plan to use an iterative and evolutionary methodology. This is used within software development as a way of incrementally developing applications through small cycles. Whilst it will be similar to scrum, it will not have the stricter framework surrounding it that requires a product owner and scrum master. This methodology will be relatively simple and based around a backlog of tasks from which a developer pulls from in a limited amount, normally 1 or 2 tasks at a time which will are then pushed through the development work flow.

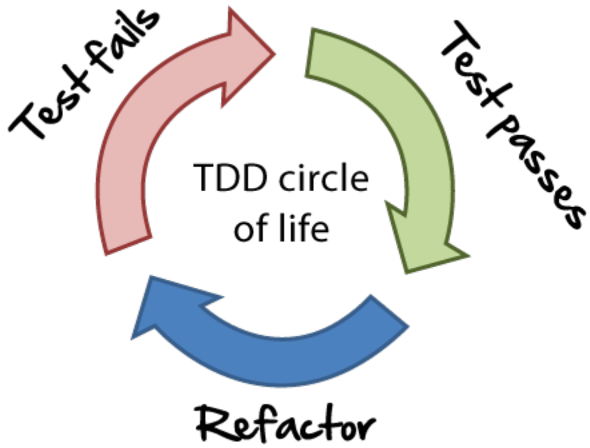
In my case, the work flow will be relatively simple:

1. To Do
2. In Progress
3. Code Review
4. Manual Testing
5. Done

During the "In Progress" step of the work flow, I will use a test driven development (TDD) process in which tests are written first and then code is wrote to make the test work. However, I will be fairly lenient with this, only using this process on parts of the code that require stringent testing as writing unnecessary tests will take up development time.

During the "Code Review" step, I plan to self-evaluate the task I have completed. On top of this, I will run static code analysis tools (such as FindBugs and PMD for Java) if the task is a coding task and use the

Figure 1.6: Test-Driven Development (TDD) Cycle [17]



code review section StackExchange to get second opinions. If it passes the code review step and it is possible to do so, I will black-box test the task from a user's perspective to see that it actually works as intended. Once every task forming a feature is completed, I will manually test the feature as a whole to see that each task has integrated together as planned.

To visualise this workflow, I will use the open source web application "Kanboard" which I have hosted on an AWS EC2 Instance. This will allow me to keep track of progress throughout the duration of the project. To manage the actual changes made to the project files, I have utilised the version control system Git with a private GitHub repository containing my project.

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