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Enriching Web Applications with Browser-to-Browser Communication

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To my family and friends.

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Abstract

An increasing number of web applications run totally, or partially, in client machines, examples range from collaborative editing tools and social networks to multi-user games. Although in many of these applications users interact among themselves, these applications continue to resort to an interaction model mediated through a centralized component. This central component, besides being a contention point on which all interaction between clients depends, can also introduce significative latency penalties specially for clients that are geographically close (or even on the same local network).

We propose to enrich the current architecture used by most Web applications with support for direct communication between browsers, improving latency among clients, scalability of the centralized component, and offering the potential to support disconnected operation (from the centralized component). In this dissertation, we present the design and implementation of a framework and supporting algorithms to allow clients to efficiently communicate with each other in a peer-to-peer fashion, allowing applications that execute in browsers to share replicas of objects across multiple instances of these applications (on different browsers on different machines), which they can concurrently modify and exchange among them.

Browsers propagate modifications directly using WebRTC, resorting to a server as intermediate when WebRTC is unavailable. Replica state convergence is achieve using a solution based on CRDTs.

We evaluate our implementation using both micro and macro benchmarks. The micro-benchmarks evaluate specific parts of the system, namely a comparison between implemented overlay networks and CRDT versions. The macro-benchmark compares the performance of our prototype to an existing industrial product, in particular the Drive Realtime API offered by Google. The results show that our solution reduces latency between geographically close clients and server load.

Keywords: client-based web applications; browser-to-browser; replication; CRDTs.

Resumo

Um número crescente de aplicações web executam total, ou parcialmente, nas máquinas do cliente, como por exemplo ferramentas de edição colaborativa, redes sociais e até jogos multi-utilizador. Embora em muitas destas aplicações os utilizadores interajam entre si directa e continuamente, estas aplicações continuam a recorrer a um modelo de interacção mediada por uma componente centralizada. Esta componente central, além de ser um ponto de contenção para todas as interações entre os vários clientes, pode também aumentar a latência de comunicação de forma significativa, especialmente entre clientes que estão geograficamente próximos (ou até na mesma rede local).

Para contornar esta tendência, propomos enriquecer a arquitetura atual usada pela maioria das aplicações Web, através da adição de suporte para comunicação direta entre os *browsers*, melhorando a latência entre os clientes, a escalabilidade da componente centralizada, e oferecendo suporte para a operação desconectada (da componente centralizada). Nesta dissertação, apresentamos o desenho e a implementação de uma framework e algoritmos de apoio para permitir aos clientes a possibilidade de comunicarem de forma directa e eficiente (comunicação entre pares), permitindo que as aplicações que são executadas em vários *browsers* partilhem réplicas de objetos, que podem ser simultaneamente modificadas e actualizadas através de comunicações par-a-par.

Os *browsers* propagam modificações diretamente usando WebRTC, recorrendo a um servidor como intermediário quando o WebRTC não está disponível. A convergência do estado das réplicas é alcançada usando uma solução baseada em CRDTs.

Foi feita uma avaliação da implementação usando micro e macro benchmarks. Os micro-benchmarks avaliam partes específicas do sistema, nomeadamente, uma comparação entre as várias redes sobrepostas implementadas e as diferentes versões de CRDTs. A macro-benchmark compara o desempenho do nosso protótipo com uma solução industrial existente, em particular, a Drive Realtime API oferecida pela Google. Os resultados mostram que a nossa solução diminui a latência para cliente próximos geograficamente e diminui a carga nos servidores.

Palavras-chave: aplicações-cliente baseadas em web; *browser-to-browser*; replicação; *CRDTs*.

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CHAPTER 1

INTRODUCTION

In recent years we have observed the proliferation and increase in popularity of web applications. A lot of these applications are centered on users, and many have aspects of collaborative editing or support for direct interaction between their users. Relevant examples of these applications are social networks, chat systems, games, and collaborative editing systems like Google Docs or Microsoft Office 365.

This increase in popularity has also led to the increase of the complexity of web based applications. Recent changes in web browsers and client devices have enabled programmers to build more powerful and interesting web applications. From collaborative editing tools to multi-user games, an increasing number of web applications run partially, or completely, on client machines. Recent frameworks and developer APIs for browser applications allow for easy development and deployment of such applications. The rise of HTML5[41] has paved the way for even more complex and interesting applications by providing additional control over the browser to developers, including aspects such as multi-threading and local storage, which previously were available only to desktop applications. This creates the opportunity for richer client browser applications to be developed.

Currently, even when web applications run in the client, they typically access information stored in servers (potentially in datacenters) and all communication between clients goes through this centralized component - e.g. all interaction in the Google Docs application is mediated by a server even when clients are nearby. The growth in browser support for WebRTC[42] (which allows for peer-to-peer like communication between browsers) makes a good promise for enabling client browser fully distributed architectures, moving from the client-server model to a server-to-server (or in this case, browser-to-browser) model.

Peer-to-peer systems have gained a lot of attention in research and have been widely adopted by industry. Peer-to-peer has been adapted for supporting file-sharing, streaming media, telephony, and even volunteer computing platforms and applications[27]. Peer-to-peer technologies have also been incorporated into other systems, for example Amazon's Dynamo[9], a NoSQL database that uses a DHT (discussed in Section 2.1.1) to distribute state over nodes as if it were a distributed hash table.

There are still few examples of frameworks that employ peer-to-peer techniques to improve web applications running in the clients. By moving significant parts of an application's logic to the client side in browsers and leveraging peer-to-peer (using recent browser based technologies) architectures (which are widely explored by academia and industry), powerful applications can be conceived.

In this work, we propose a framework to simplify the creation of such applications that follow this new paradigm. We further implemented and experimented with a prototype of this framework which acts as a proof of concept for our proposal.

1.1 Motivation

Real-time collaborative editing tools, where users can edit the same file simultaneously, have been deployed with significant client adoption. Current collaborative editing applications and developer frameworks are typically based the client-server model, with only a limited number of applications and frameworks resorting to peer-to-peer support at the browser level. User count can drastically increase in a short span of time and the user's need for fast propagation of updates (i.e., low latency between clients) has greatly increased over the last years as higher expectations are put on web applications. Collaborative editing is thus a good example of an application that can be greatly improved by exploring this design space.

In the current interaction model servers act as the data storage system, maintaining the data being edited, while also being used for access control and to expose, or resolve, divergence on concurrently edited objects.

The limitations to the client-server model are well-known and include: *i*) a significant increase in client-to-client latency, especially between those clients that are geographically close (or even those on a single local network); *ii*) the centralized component is a potential contention point that can limit the scalability due to limits on available resources (which cloud computing mitigates as we will address next); *iii*) the server is a single point of failure that prevents any interaction among users when the server becomes unavailable (i.e., if the server becomes unavailable the whole system becomes unavailable).

The rise of cloud computing mitigated significantly some of these limitations, especially scalability and fault tolerance, as techniques like replication can be used (i.e., the ability to increase the amount of servers that support an application according to load, typically using partitioning techniques). On the other hand, using cloud infrastructures adds monetary costs to application providers which is obviously undesirable or even poses a practical impediment for the operation in the case of small companies that try to launch an application in an already very competitive market.

Current frameworks for collaborative editing on the web[11, 15, 17] do not allow direct client-to-client (i.e, peer-to-peer) communication, and all user interactions are mediated by the central component. The main reasons for this are the following. First, relying on a central component simplifies the management of data and notifications to clients. Second, limitations to peer-to-peer networking in browsers made it hard to explore the peer-to-peer paradigm (as discussed previously) in the past. However, a number of recent developments are challenging these problems.

CRDTs[36, 37], Convergent (or Commutative) Replicated Data Types, allow for automatic reconciliation of state even in decentralized systems. Operations on CRDTs can immediately execute locally and, unaffected by network latency, faults, or disconnections, data replicas will eventually converge towards a common, single, state where these replicas are able to synchronize and updates are propagated across all replicas. This makes managing replicated data much simpler, as convergence of state no longer has to be ensured by application logic.

Regarding support for peer-to-peer networking, a number of new technologies have been introduced in browsers recently. These technologies incude WebRTC[42] (Web Real Time Communication), and the proposal of protocols (and publicly available services) to surpass firewall and NATs like STUN[18] and TURN[19]. These new technologies combined with advances of HTML5, pave the way to the development of frameworks that allow for enriching web applications with direct communication between user devices.

Using peer-to-peer for collaborative editing or other multi-user applications (i.e., social networks, games, and others) that have high user interaction in real-time can greatly reduce client-to-client latency and the load on the server as fewer server accesses might be needed. Ideally, scalability can be greatly increased and user experience improved.

Other recent improvements also contribute to the possibility of a system such as we propose (though not being fundamentally required). Improvements on local storage and threading support make it possible for objects to be cached or even permanently stored on the client, possibly removing the need for continuous access to the server, as to reduce access latency and bandwidth, server load or even reducing trust on the service provider[43].

1.2 Proposed Approach

To overcome the current limitations of centralized architectures, in this thesis we propose to enrich current web application architectures with support for direct browser-to-browser communication.

We diverge from the common model for designing web applications, and propose a technique that enriches cloud infrastructures with peer-to-peer techniques adapted for web-applications enviroments, to surpass the client-server model's limitations. The use of this enriched model has the potential to impact scalability and availability of web applications, latency experienced by geographically close clients, and operational cost for the operation of web application's centralized components.

To achieve this, we present the design of a framework that allows for applications executing on multiple browsers to keep up-to-date replicas of a set of objects which can be modified concurrently. This framework, which we have implemented as a prototype, is composed by the following main components: *i*) mechanisms for managing browser-to-browser connections that allow instances of a web application, according to some policy defined by that application, to establish direct connections between users that interact frequently and/or are geographically close; *ii*) a set of communication primitives that allow applications to share data by using browser-to-browser connections, and to coordinate between each other as to propagate information to the centralized component for both durability and also to enable the exchange of information with users that must access the application exclusively via the centralized component, due to the lack of WebRTC support of their browser, or due to overly restrictive networking policies; *iii*) mechanisms that enable the replication of shared objects, with updates being transparently propagated across all clients, guaranteeing that application state eventually converges; *iv*) a library of CRDTs with support for these mechanisms, written in JavaScript, based on the specifications in [36]. This simplifies the design and implementation of decentralized web applications that execute on the browser.

1.2.1 Main Contributions

The main contributions of this thesis are the following:

- A framework supporting direct browser-to-browser communication, without the need to install any kind of software or browser plugins.
- The design and implementation of a mechanism to replicate a set of objects in web applications, combining peer-to-peer interactions and a centralized component.
- A CRDT library that can be used with the replication mechanism for maintaining the state of different applications.
- An evaluation of the proposed system and comparison with an existing industry solution.

The design of our system's programmer interface was performed with close attention to existing frameworks' APIs. Besides the fact that such an approach enables existing applications to be easily converted to our solution, it enables us to compare our proposed solution to one that only uses the central component for communication, i.e., a client-server model.

The system we present in this thesis exposes a programmer interface for creating applications with similarities to the PeerJS API and other messaging systems (like *publish/subscribe*) for interaction based on sending messages. We combine this interface with an interface similar to the Google Drive Realtime API[17] for object replication and synchronization.

To evaluate the performance of our implementation we developed micro and macro benchmarks. The micro-benchmarks evaluate specific parts of the system, namely a comparison between implemented overlay networks and CRDTs. The macro-benchmark was created to compare the performance of our prototype to an existing industry solution, namely the Drive Realtime API offered by google. We show that we are able to improve on the total server load and on latency felt by clients.

Publications

Part of the results in this thesis were published in the following publication:

Enriquecimento de plataformas web colaborativas com comunicação browser-a-browser Albert Linde, João Leitão e Nuno Preguiça. Actas do sétimo Simpósio de Informática, Covilhã, Portugal, September, 2015

1.3 Document Organization

The remainder of the thesis is organized as follows:

Chapter 2 introduces fundamental concepts which are relevant for the context of the contributions presented in the thesis;

Chapter 3 presents the design of such a system as we propose in this thesis;

Chapter 4 describes how the prototype built can be used and also discusses relevant implementation details;

Chapter 5 presents the evaluation of our system and the results of comparison established industry solution;

Chapter 6 concludes this thesis by summarizing the thesis and discussing pointers for future work.

Chapter 2

Related Work

This thesis addresses the challenges of real-time collaboration leveraging on peer-to-peer communication directly on the browser. Therefore, various aspects have to be considered. The following sections cover the main aspects of these fields, in particular:
In **Section** 2.1 existing peer-to-peer technologies are studied and compared, with special interest on overlay networks and communication models.
In **Section** 2.2 web based service providers are discussed, in particular addressing the challenge of storing and accessing data.
In **Section** 2.3 the field of collaborative editing is explored, with emphasis on real-time collaborative editing.
In **Section** 2.4 we give an overview of the existing WebRTC standard, which serves as a basis for our work.

2.1 Peer-to-peer systems

Peer-to-peer systems typically have a high degree of decentralization, leveraging the use of resources from the server to the client. In other words, each peer implements both server and client functionality to distribute bandwidth, computation, and storage across all participants of a distributed system[27]. This is achieved by allocating state and tasks among peers with few, if any, dedicated peers.

Nodes are initially introduced to the system and typically little or no manual configuration is needed to maintain connectivity[1]. Participating nodes generally belong to independent individuals who voluntarily join the system and are not controlled by a single organization.

[1] To consider a network as connected, there should be at least one path from each node to all other nodes.

Peer-to-peer systems are interesting due to their low barrier to deployment, its organic growth (as more nodes join, more resources are available), resilience to faults/malicious attacks, and the abundance/diversity of systems.

Popular peer-to-peer applications include sharing and distribution of files, streaming media, telephony, and volunteer computing. Peer-to-peer technologies were also used to create a diversity of other applications, for example Amazon Dynamo[9] which is a storage system, which internally heavily relies on DHTs, demonstrating the benefits of leveraging peer-to-peer architectures. It is important to note that the network topology of the underlying network has a high impact on the performance of peer-to-peer services and applications. For client nodes to be able to cooperate they need to be aware of the underlying network. The typical approach is to create a logical network of nodes on top of the underlying network, called an overlay network.

Thus, in order to develop a peer-to-peer distributed service, it is of paramount relevance to study the mechanisms for creating and managing overlay networks that match the application requirements.

2.1.1 Overlay Networks and Communication models

An overlay network is a logical network of nodes, built on top of another network. Links between nodes in the overlay network are virtual links, being composed of various links of the underlying network. In peer-to-peer systems, overlay networks are constructed on top of the internet, each link being a connection between two peers.

To achieve an efficient and robust method of delivering data through a peer-to-peer technique an adequate overlay network is needed. When building an application, the programmer must first decide on the overlay to deploy and use, choosing between degrees of centralization as well as on structured vs unstructured designs.

Degree of centralization: Peer-to-peer networks can be classified by their use of centralised components.

Partly centralized networks leverage system components to dedicated nodes or a central server to control and index available resources. These centralised components are used to coordinate system connections, facilitate the establishment of communication patters, and coordinate node co-operation.

As an example, when client nodes want to execute a specific query, only the central component is contacted, which in turn can return the set of nodes that match the query. These systems are relatively simple to build but come with the drawback of a potential single point of failure and bottleneck. Therefore, this design can not be as resilient and scalable as a fully decentralized system. Examples include Napster[28], Bittorrent using trackers[5], BOINC[1], and Skype[2].

Decentralized networks avoid the use of dedicated nodes. All network and communication management is done locally by each participating node, using light coordination mechanisms. This way a singe bottleneck and point of failure is avoided, increasing potential for scalability and resilience.

In this type of architecture, nevertheless, a few selected nodes may act as supernodes, as to leverage potential higher CPU or bandwidth available, gaining additional responsibilities such as storing state or even becoming the entry point for new nodes. As queries cannot be sent to and executed by a central component, typically when using unstructured overlays they have to be flooded through the network or routed through the network when using a structured overlay. Example protocols include Gnutella[38] and Gossip[3].

Structured vs Unstructured: Choosing between structured and unstructured overlays depends mostly on the usefulness of key-based routing algorithms[2] and the amount of churn[3] that the system is expected to be exposed to.

Structured overlays: Each node gets an identifier which determines its position in the overlay structure. Identifiers are chosen in a way that peers are usually uniformly distributed at random over the key space. This allows to create a structure similar to a hash table, named DHT, distributed hast table. This type of overlay graph is typically chosen when efficient (logarithmic complexity) key-based routing is required. Structured overlays typically use more resources to maintain the overlay, but in return get efficient queries at the cost of poor performance when churn is high, in fact, churn is not handled well at all.

Unstructured overlays: There is no particular structure in the network links and queries are usually done by flooding the network. Each peer keeps a local index of its own resources and, in some cases, the resources of its neighbours. The connected peers are designated as a partial view. This is a (small) fraction of all peers in the system with whom that participant can interact directly. Ideally, the size of such partial views should grow logarithmically with the number of participants in the system. Queries are typically disseminated among the connected peers. To ensure that a query returns all possible results, the query must be disseminated to all participants.

Maintaining data: In partially centralized systems data is typically stored at the inserting and downloading nodes. The central component maintains metadata, i.e., an index, on the stored data including where it is located.

[2] Key-based routing is a lookup method, used in conjunction with distributed hash tables, that enables to find the node that has the closest identifier to the key being searched.

[3] Churn is the participant turnover in the network (the amount of nodes joining and leaving the system per unit of time).

In unstructured systems data is also stored on submitting and downloading nodes but to locate data typically the queries are flooded. For faster searches, nodes can distribute metadata among neighbours.

In structured overlays distributed state is maintained using distributed hash tables. Primitives are similar to any hash table, and easily implemented when a key-based routing function is available. On high churn it becomes very inefficient to store large amounts of data at peers responsible for the keys, therefore indirection pointers are commonly used, pointing to the node (or nodes) that effectively holds the data.

Coordination: In partially centralized systems the centralized component can trivially achieve coordination.

In unstructured overlays, epidemic techniques are typically used because of their simplicity and robustness to churn. Information tends to propagate slowly though and scaling to large overlays is costly. Spanning trees[4] can increase efficiency but maintaining the tree structure adds maintenance costs.

In structured overlays, key-based routing trees are the basis for potentially large sub-groups within the overlay, enabling fast coordination and good efficiency.

2.1.2 Examples of peer-to-peer overlay networks

Chord[39] is a distributed lookup protocol that enables peer-to-peer systems to efficiently locate nodes that store a particular data item. It only offers one primitive: given a key, return the nodes responsible for the key. Keys are distributed over the nodes using consistent hashing and replicated over succeeding nodes. Nodes typically store their successor nodes, forming an ordered ring (considering node's identifiers), making it easy to reason about the overlay structure. For fault-tolerance a list of successor nodes is kept and for efficient lookups a finger table, shortcuts to nodes over the graph, is used to jump over nodes in the graph.

Gnutella[38] is a decentralized peer-to-peer file sharing protocol. When a node is bootstrapping to the network, it tries to connect to the nodes it was shipped with, as well as nodes it receives from other clients. It connects to only a small amount of nodes, locally caching the addresses it has not yet tried, discarding the addresses that are invalid.

Queries are issued and flooded from the client to all actively connected nodes, the query is then forwarded to any nodes they know about. Forwarding ends if the request can be answered or the Time-To-Live field ends. The protocol in theory doesn't scale well, as each query increases network traffic exponentially each hop, while being very unreliable as each node is a regular computer user, constantly connecting and disconnecting.

[4] A spanning tree of a graph is a tree connecting all nodes in the graph.

A revised version of the gnutella protocol is a network made of leaf nodes and ultra peers. Each leaf node is connected to a small number of ultra peers, while ultra peers connect to many leaf nodes and ultra peers. Leaf nodes send a table containing hashed keywords to their ultra peers, which merge all received tables. These tables are distributed among ultra peer neighbours and used for query routing, by hashing the query keywords and trying to match the tables.

Cyclon[40] is a membership management framework for large peer-to-peer overlays. The used membership protocol maintains a fixed length partial view managed through a cyclic strategy. This partial view is updated every T time units by each node though an operation called shuffle. In a shuffle, a node selects the oldest node in its partial view and exchanges some elements of its local partial view with it. When nodes initially join the overlay a random walk is used, ensuring that the in-degree of all nodes remains balanced. This work achieves an overlay topology with low diameter and low clustering coefficient with highly symmetric node degrees and high resilience to node failures.

Scamp[14] is a membership management framework for large peer-to-peer overlays. The Scamp protocol maintains two views, a PartialView to send gossip messages and an InView from which they receive messages. The PartialView is not of fixed length, it grows to a size logarithmic in scale to the number of nodes in the network without any node being aware of this number. The protocol uses a reactive strategy, in the sense that the partial views are updated when nodes join or leave the system. Periodically nodes send heartbeat messages as to detect and recover from isolation due to failures. Not receiving any heartbeats allows the node to assume that it is isolated, triggering the join mechanism to effectively rejoin the overlay.

HyParView[25] , Hybrid Partial View, is a reliable gossip-based broadcast protocol that maintains a small symmetric Active View (managed through a reactive strategy) for broadcasts and a larger Passive View (managed through a cyclic strategy) to recover timely from faults. Both strategies are very similar to Scamp and Cyclon. TCP is used as a reliable transport and to detect failures, being feasible as the Active View is small. Even with a small ActiveView, improving protocol efficiency as less network traffic is required for flooding messages, very good results in reliability are obtained. This work shows the importance of each reactive and cyclic strategies to maintain views of the network, and that the use of a reliable transport mechanism, like TCP, to timely encounter failures, can greatly improve results.

Using a structured network overlay as Chord, a really high network efficiency can be achieved as all request are routed directly to the right nodes. Unstructured network overlays typically have to flood the network, reducing efficiency, but create tolerance to network churn. Declaring some nodes as dedicated to the network, as done by Gnutella, can greatly reduce network traffic. Cyclon, Scamp, and HyParView each show the importance of reactive and cyclic strategies for maintaining partial views as well as the combination of both.

2.2 Data Storage

Web based services store client data on geographically distributed data-centers, trying to provide good latency, bandwidth, and availability for interactions with the data. Typically replication and distribution of state across geographically separated data centres is required to ensure low latency and fault tolerance. A problem arises, formally captured by the CAP theorem[16], which states that it is impossible for a distributed computer system to simultaneously provide all three of the following: **C**onsistency (all nodes see the same data at the same time), **A**vailability (every request receives a response about whether it succeeded or not) and **P**artition tolerance (the system continues to operate despite arbitrary message loss or partial failure of the system or unavailability due to network partitions). Unfortunately, due to how the internet works, partitions due to network or node failures are part of our lives so the question is to ask which to sacrifice, **C**onsistency or **A**vailability?

Strong Consistency. A system is said to provide strong consistency if all accesses to data are seen by all clients in the same order (sequentially). A distributed system providing Strong Consistency will come to a halt if nodes become network-partitioned. It is easy to understand that two nodes cannot decide on a value if they cannot reach one another. **C**onsistency can thus be maintained but the system will sacrifice **A**vailability.

Eventual Consistency is a consistency model used in distributed computing systems to achieve high availability which informally guarantees that, if no new updates are made to a given data item, eventually all accesses to that item will return the value of the last update. This allows these systems to, even during network partitions, always serve read and write operations over data. Eventual consistency may not be enough, one example: user A writes to a page and user B answers, due to network latencies user C sees B's answer before A's initial post. This shows that while this consistency model is correct, it can lead to confusion.

Causal Consistency. A system provides causal consistency if potentially causally related operations are seen by every node of the system in an order that respects these causal dependencies. Concurrent writes (i.e., write operations that are not causally related) may be seen in different orders by different nodes. When a node performs a read followed later by a write, even on different objects, the first operation is said to be causally ordered before the second, because the value stored by the write may have been dependent upon the result of the read. Also, even two write operations performed by the same node are defined to be causally related, in the order they were performed. Intuitively, returning on our previous example, such a system would never show B's updates before A's as they are causally related.

Using eventual and causal consistency (i.e., not strong consistency) usually comes with a cost: state divergence. To address state divergence, conflict resolution techniques such as the ones discussed in section 2.2.1 must be used.

One way to avoid state divergence, as achieved in Yahoo!'s PNUTS[6], is to funnel all state changing operations through a per record chosen primary site and lazily propagating to replicating nodes. This increases latency and reads can return stale data, but data exposed to users is consistent. The problem of this approach is availability as the primary site is a potential single point of failure.

When multiple nodes can write to the same data, object versioning control has to be done using for example logical clocks or version vectors[22, 31] and conflict resolution techniques have to be applied.

2.2.1 Conflict resolution techniques

Relaxing from a strong consistency model to a weaker model such as causal consistency, minimizes the amount of required synchronization among replicas at the expense of having to deal with state divergence. To do so, one resorts to conflict resolution techniques. Common conflict resolution techniques include:

Last Writer Wins: the idea is that the last write based on a node's system clock will overwrite an older one. Using a single server this is trivial to implement but when clocks are out of synch when writing on multiple nodes, choosing a write between concurrent writes is not trivial at all and can lead to lost updates.

Programatic Merge: letting the programmer decide what to do when conflicts arise. As an example, an application maintaining shopping carts can choose to merge the conflicting versions by returning a single unified cart. This conflict resolution technique requires replicas to be instrumented with a merge procedure, or alternatively, requires replicas to expose diverging states to the client application which then reconciles and writes a new value.

Commutative Operations: If all operations are commutative, conflicts can easily be solved. Independently of the order, when all operations have been received (and applied), the final outcome will be the same. An always incrementing counter, where each operation is uniquely marked by the writing node, is an easy example: independently of the order of operations, the final result will eventually be the same. Commonly used commutative operation techniques are:

OT, Operational Transformation. The idea of OT is to transform the parameters of an operation to the effects of previously executed concurrent operations, so that the outcome is always consistent. As an example, a text document contains 'abc' and there are two concurrently editing users. One user inserts 'x' at position 0 and the other deletes 'c' from position 2. If both execute their operation and later receive the operation of the other (due to network latency), the final states diverge to 'xac' and 'xab'. Transforming the operations solves this problem, the delete is transformed to increment one position and the insert can remain the same. Both outcomes become 'xab', independently of the order in which operations are applied.

Operational Transformation has been extensively studied, especially by the concurrent editing community, and many OT algorithms have been proposed. However, it was demonstrated that most OT algorithms proposed for a decentralized architecture are incorrect[30]. It is believed that designing data types for commutativity is both cleaner and simpler[36].

CRDT, Convergent or Commutative Replicated Data Types. CRDTs are replicated data types that guarantee eventual consistency while being very scalable and fault-tolerant. An example is a replicated counter, which converges because increment and decrement operations commute. No synchronization is required to ensure convergence, so updates always execute locally and immediately, unaffected by network latency, faults, or disconnections. CRDTs can typically be divided in two classes:

CvRDT, state-based Convergent Replicated Data Types. In the state-based class, the successive states of an object should form a monotonic semilattice[5] and replica merge computes a least upper bound. In other words, when merging diverging states the end result must be equal at each replica. State-based CRDTs require only eventual communication between pairs of replicas.

CmRDT, operation-based Commutative Replicated Data Types. In the operation-based class, concurrent operations commute. Operation-based replication requires reliable broadcast communication with delivery in a well-defined order, such as a causal order between operations.

Both classes of CRDTs are guaranteed to eventually converge towards a common, single, state (i.e., when all updates are received by all participating nodes). Practical use of CRDTs shows that they tend to become inefficient over time, as tombstones accumulate and internal data structures become unbalanced[36]. Garbage collection can be performed using a weak form of synchronization, outside of the critical path of client-level operations.

[5] An idempotent and commutative system that grows only in one direction.

2.2.2 Examples of data storage systems

Spanner[7] is a system providing strong consistency which uses the Paxos algorithm as part of its operation to replicate data across hundreds of data-centers. It also makes heavy use of hardware-assisted time synchronization using GPS clocks and atomic clocks to ensure global consistency.

One server replica is elected as the Paxos leader for a replica group, which will become the entry point for all transactional activity for that group. Groups may include read-only replicas, which do not vote in the Paxos algorithm and cannot become group leaders.

Furthermore all transactions in Spanner are globally ordered as they are assigned a hardware assisted commit timestamp. These timestamps are used to provide multi-versioned consistent reads without the need for taking locks. A global safe timestamp is used to ensure that reads at the timestamp can run at any replica and never block behind running transactions.

Spanner thus has very strong consistency and timestamp semantics, providing scalable data storage and synchronous replication.

Dynamo[9] is a highly-available key-value storage system. To achieve high availability, consistency is sacrificed using object versioning and application-assisted conflict resolution, exposing data consistency issues and reconciliation logic to the developers.

Data is partitioned and replicated using consistent hashing and vector clocks are used for object versioning. Dynamo uses a gossip based failure detection and membership protocol. This removes the need for manual configuration creating a completely decentralized system, ensuring that adding and removing nodes can be done without any manual effort. Each node is aware of the data being hosted at its peers. In contrast to other DHT systems, each node actively gossips the full routing table with other nodes in the system. This model works well in their expected scenario of a couple of hundred of nodes, scaling this design to a higher number of nodes can be troublesome as the routing table increases with the number of nodes in the system.

Gemini and its RedBlue consistency[26] build on the premise that while a system can be leveraged to use eventual consistency for higher performance, strong consistency may be necessary to ensure correctness of the applications.

RedBlue consistency labels operations as red or blue. Blue operations are to be fast (eventually consistent) while red operations are slow (strongly consistent). Blue is used when possible and red when needed. Gemini is a coordination infrastructure implementing RedBlue consistency. Experimental results show that RedBlue consistency provides substantial performance while being able to maintain application invariants, the downside is that transactions have to be individually modified and correctly labelled.

Riak[21] is a distributed NoSQL key-value data store that supports high availability by giving the possibility between strong and eventual consistency, using quorum read and write requests and multi-version concurrency control with vector clocks. Eventual consistency in Riak uses CRDTs at its core, including counters, sets, flags, registers, and maps. Partitioning and replication is done via consistent hashing using a masterless approach, thus providing fault-tolerance and scalability. The built-in functions determine how replicas distribute the data evenly, making it easy for the developer to scale out to more nodes.

SwiftCloud[34] is an eventual consistency data storage system with low latency that relies on CRDTs to maintain client caches. The main focus of this work is to integrate client and server-side storage. Responsiveness is improved when accessed objects are locally available at the cache, which allows for disconnected operation.

In the presence of infrastructure faults, a client-assisted failover solution allows client execution to resume immediately and seamlessly access consistent snapshots without blocking. Additionally, the system supports merge-able and strongly consistent transactions that target either client or server replicas and provide access to causally-consistent snapshots efficiently.

Systems like Spanner have been designed to provide strong consistency on geographically distributed data-centers. These systems use very complicated algorithms or specialised underlying hardware and are not trivial to deploy. Systems supporting weaker consistency models have been developed, like Dynamo, that support writes on different clients increasing scalability and fault tolerance, but need a way to address state divergence. A system like Gemini that supports both eventual and strong consistency can be used to have the best of both. It can be very difficult to reason in detail on such a system, especially on what has to be strong or what can be eventually consistent. The use of CRDTs, like in Riak and SwiftCloud, can greatly improve latency as all updates can always execute and merging diverging state isn't an issue, as data converges to a single final consistent state.

2.3 Collaborative Editing

A collaborative editor is a piece of software that allows several people to edit files using different client-devices, working together through individual contributions. Collaborative editing can be divided in two types: real-time and non-real-time. In real-time collaborative editing systems users can edit the same file simultaneously while in non-real-time collaborative editing systems editing the same file at the same time is not allowed.

In collaborative editing the main challenge is to figure out how to apply edits from remote users, who produced these edits on versions of the document that possibly never existed locally, and that can potentially conflict with the user's own edits. Users may write on previously decided sub-parts of the document, facilitating merges, or, on the other end of the spectrum, work together on the same task.

2.3.1 Handling concurrent updates

There are several approaches in creating a collaborative editor. The basic needs for such a system are the possibility for concurrent (possibly in real time) editing of objects while preserving user intent. Some approaches include:

Turn taking where one participant at the time 'has the floor'. This approach lacks in concurrency but is easy to comprehend, preserving user intent.

Locking based techniques, where concurrent editing is trivially possible as users work on different objects. Pessimistic locking introduces delays and optimistic locking introduces problems when the lock is denied or when user edits have to be rolled back to a previous state.

Serialization can be used to specify a total order on all operations. Non-optimistic serialization delays operations until all preceding operations have been processed while in optimistic serialization, executing operations on arrival is allowed, but there might be the need to undo/redo operations to repair out-of-order executions (as in version control systems).

Commutative operations can be leveraged to address the challenge of collaborative editing systems. By using OT or CRDTs (as described in section 2.2.1) a high degree of concurrency can be achieved while capturing and preserving user intent.

A collaborative editor can be designed using a client-server model. The server ensures synchronization between clients, determining how user operations should affect the server's copy and how to propagate these operations to other clients. Though easy to implement, this approach possibly lacks scalability and can detoriate user experience by increasing latency. A more sophisticated solution is one that does not require a server, avoids to resort to locking, and supports any number of users.

Though good enough for non-real-time collaborative editing, to provide the basic needs for a real-time collaborative editor it is easy to see that approaches as turn-taking, locking, and serialization are insufficient. Besides not allowing real-time concurrent editing of the same data, the coordination algorithms of underlying systems can be unnecessarily complicated while scalability and fault tolerance are non trivial to reason about. Using commutative operations is thus widely accepted as the *de facto* solution.

2.3.2 Examples of collaborative editing systems

Etherpad[11] (or Etherpad Lite), is a web-based collaborative real-time editor, allowing authors to simultaneously edit a text document, and see all of the participants' editions in real-time, with the ability to display each author's text in their own color. There is also a chat box in the sidebar to allow direct communication among users. Anyone can create a new collaborative document, known as a *pad*. Each pad has its own URL, and anyone who knows this URL can edit the pad and participate in the associated chats.

The software auto-saves the document at regular, short intervals, but participants can permanently save specific versions (checkpoints) at any time. A *time slider* feature allows anyone to explore the history of the pad. Applying concurrent operations is handled by Operational Transformation using a Client-Server model.

Dropbox Datastore[10] is an API that allows developers to synchronise structured data easily supporting multiple platforms, offline access, and automatic conflict resolution.

The server doesn't attempt to resolve conflicts. OT-style conflict resolution is done on the client, the server simply serializes all operations. Conflict resolution is allowed to be defined by the client, the application created by a developer, by choosing from the following conflict resolution rules: choose remote value, choose local value, choose the maximum value, choose the minimum value, and sum the values.

A datastore is cached locally once it is opened, allowing for fast access and offline operation. Changes to one datastore are committed independently from another datastore. When data in a datastore is modified, it will automatically synchronize those changes back to Dropbox (i.e., upload local changes and download and apply remote modifications).

Google Drive Realtime API[17] is a client-only library that can merely be used in combination with Google servers. The API can be used by developers to implement a real-time application by using its collaborative objects, events and methods. It uses Operational Transformation to resolve concurrency issues and thus local changes are reflected immediately, while the server transforms changes to the data model so that every collaborator sees the same (final) state. In contrast to Google's own collaborative web applications, such as Google Docs, anonymous users are not permitted and as such using the API requires the end users to have an active Google Drive account.

This API itself is limited to document-based synchronization, such as lists, strings, and key-value maps. It does not specifically support model-based synchronization and complex object graphs. The developer would need to deal with the intricate details of merging model instances while also handling complex situations such as relations to abstract classes or cycles in the object graph.

ShareJS[15] is a server and client library to allow concurrent editing of content via Operational Transformation. The server runs on NodeJS and the client works in NodeJS or a web browser. In the local browser, edits are visible immediately. Edits from other clients get transformed. ShareJS currently supports operational transform on plain-text and arbitrary JSON data.

If multiple users submit an operation at the same version on the server, one of the edits is applied directly and the other user's edit is automatically transformed by the server and then applied. In contrast to previous systems ShareJS gives complete control to the developer over both the client and server logic and over the Operational Transformation protocol.

Jupiter[29] is a tool that supports multiple users for remote collaboration using Operational Transformation. The synchronization protocol is not applied directly between the clients as each client synchronises only with the server. The server is thus used to serialise all operations and disseminate those operations to other clients.

Operations are directly executed at the local client site when generated. They are then propagated to the central server which serialises and transforms operations before executing on the server's copy. Finally the transformed operations are broadcast to all other client sites. When receiving an operation from the server a client may transform this operation if needed, and then execute on the local copy.

SPORC[12] is a cloud-based framework for managing group collaboration services. It uses Operational Transformation to merge diverging state and, as an example, showcases a collaborative editor that allows multiple users to modify a text document simultaneously via their web browsers and see changes made by others in real-time providing an application similar to Google Docs and EtherPad. However, unlike those services, it does not require users to trust on the centralised server. The purpose of the server is to order and store client generated operations. As the server only sees an encrypted history of the operations all application logic is leveraged to the client.

EtherPad is a completely open-source real-time collaborative word processing tool that can freely be hosted on any server while Google's Realtime API and Dropbox's Datastore require the use of the provider's servers. These systems provide the user with a history view for each document. ShareJS shows that Operational Transformation is possible over JSON data. Jupiter and SPORC emphasise the importance of a central server to serialise editions, though in SPORC the server doesn't ever see the content of the documents while still being able to provide all functionality.

2.4 WebRTC

WebRTC[42] was created for supporting real-time, plugin-free video, audio, and data communication directly on browsers. Real Time Communication is used by many web services (Skype, Google Hangouts, etc..), but requires large downloads, the use of native apps, or plugins. Downloading, installing, and updating plugins can be complex for both the developer and end user. Overall, its often difficult to persuade people to install plugins, which impacts the adoption of applications with this requirement. WebRTC was designed to address these challenges.

To acquire and communicate streaming data, WebRTC offers the following APIs: MediaStream, to get access to multimedia data streams, such as from the user's camera and microphone; RTCPeerConnection, for audio or video calling, with facilities for encryption and bandwidth management; RTCDataChannel, for peer-to-peer communication of generic data.

WebRTC audio and video engines dynamically adjust bitrate of each stream to match network conditions between peers. When using a DataChannel this is not true, as it is designed to transport arbitrary application data. Similar to WebSockets, the DataChannel API accepts binary and UTF-8 encoded application data, giving the developer choices on message delivery order and reliability.

Though designed for peer-to-peer like applications, in the real world WebRTC resorts to servers, so each of the following interactions, mediated by a centralized server, can happen:

- Before any connection can be made, WebRTC clients (peers) need to exchange network information (signalling protocol).

- For streaming media connections, peers must also exchange data about media such as video format and resolution.

- Additionally, as clients often reside behind NAT gateways and firewalls, these may have to be traversed using STUN (Session Traversal Utilities for NAT) or TURN (Traversal Using Relays around NAT) servers.

2.4.1 Signalling

Signalling is the process of coordinating communication in WebRTC. In order for a WebRTC application to set up a 'call' (i.e., a connection), clients need to exchange information: session control messages used to open or close communication channels; error messages; media metadata such as codecs and codec settings, bandwidth and media types; key data, used to establish secure connections; network data, such as a host's IP address and port as seen by the outside world.

Figure 2.1[6] depicts the operation of the signalling protocol to establish a WebRTC connection. A signaling channel can be any medium that allows messages to go back and forth between clients. This mechanism is not implemented by the WebRTC APIs: it has to be implemented by the developer. It can be as rudimentary as using e-mail or an instant messaging application (i.e., an out of band mechanism), it can be done via a centralised server, and, theoretically, WebRTC's DataChannels can be used. As depicted in the figure, when peers reside behind firewalls or NATs they have to make use of STUN or TURN to establish connections. STUN is used only to obtain the public address for a peer to pass along via the signalling mechanism. If no connection can be made between two peers WebRTC can resort to the use of TURN. TURN servers are used to relay the data between peers.

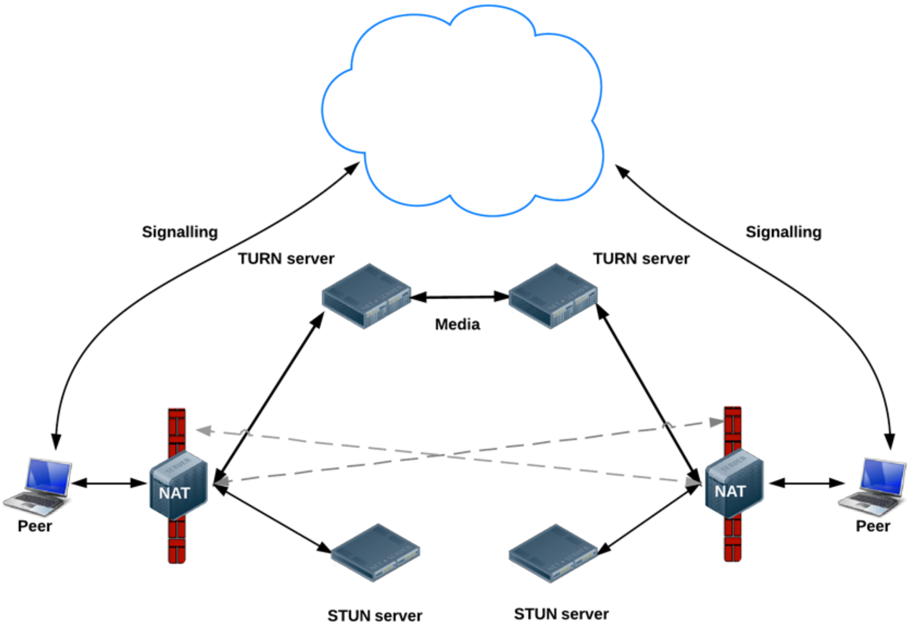


Figure 2.1: WebRTC Signalling

Table 2.1 shows the current support from different browsers for WebRTC and various HTML5 APIs and their current market share[7]. Though support for communication between different browsers is currently in development, support by Chrome alone already constitutes a majority of the users and is therefore the best candidate for the development of the work pressented in this thesis.

2.4.2 Examples of WebRTC enabled systems

PeerJS[32] is a WebRTC enabled framework for creating applications which enables clients to connect to each other, creating a media stream or data connection to a remote peer.

[6] Taken from http://www.html5rocks.com/en/tutorials/webrtc/infrastructure/

[7] Last checked on 20th September 2015.

Table 2.1: Browser Support for HTML APIs

Browser	WebRTC	LocalStorage	WebWorkers	Market Share
Chrome	Yes	Yes	Yes	60.1%
Firefox	Yes	Yes	Yes	23.4%
Opera	Yes	Yes	Yes	1.6%
Microsoft Edge	Partial	Yes	Yes	9.8%
Safari	No	Yes	Yes	3.7%

WebRTC Experiment[20] is a collection of frameworks and examples of the use of WebRTC, exposing the complete stack to the programmer.

Both these systems allow for the usage of the existing signalling server and the use of private servers. The implemented examples only illustrate the connection between two peers and have little to no support for a large number of users (i.e., no peer-to-peer overlay network creation or support for scalable gossip communication primitives between nodes).

2.5 Summary

This chapter discussed previous work in the areas related to the development of the work presented in this thesis.

In the peer-to-peer context the need for an overlay network has been described, explaining that different application requirements can require different types of overlays. Overlays can generally be described by degree of centralization and structured versus unstructured topology.

In the data-storage context we discussed the inherent choice between strong consistency and high availability in distributed systems (that by definition should deal with network partitions). Different consistency models have been explored and, in the case of eventual consistency, several techniques for conflict resolution have been described.

In the collaborative-editing context, various commonly used approaches have been presented and disscussed, describing how concurrency is handled in real-time editing in each of them.

WebRTC has been presented, describing how peer-to-peer connections can be made between browsers and what mechanisms exist to facilitate these connections.

Chapter 3

A Browser-to-browser Framework

In this chapter, we show our proposal of an architecture to enrich the operation of web based applications with direct browser-to-browser communication. The focus of this design will be applications that have a high degree of direct user-to-user interaction, including but not limited to, social networks, chat systems, games, and collaborative editing systems.

Typically web based applications use a client-server communication model. In this work we propose to enrich applications as to allow the use of peer-to-peer, or more precisely in this case: browser-to-browser, communication. Figure 3.1 depicts a parallel between the commonly used client-server model of web applications (left) and the enriched browser-to-browser model we propose in this work (right).

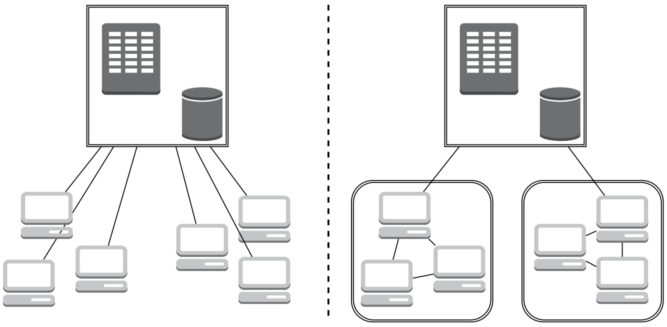


Figure 3.1: Communication model (client-server) | (browser-to-browser)

Note however that in the proposed architecture the centralized component is still a relevant component which is used for several purposes, namely for data persistence and to support clients that operate on non WebRTC compliant browsers (i.e., clients whose browsers do not support direct communication with other browsers). Clients that due to overly restrictive firewalls policies or due to Network Address Translations (NAT) are unable to estxablish connections to other browsers will also resort to the centralized component to propagate operations and interact with the system. Additionally, as it will become clear later in this chapter, the centralized component itself is also used to allow clients to establish the initial connections between each other when a new client joins the system.

3.1 Requirements

As previously stated, the work presented in the thesis focuses on Internet applications and services. The fundamental requirements that a system such as we propose in this thesis must answer to are: first, improve latency between clients as they are able to directly communicate between them; second, reduce the dependency on the server as a mediator for interaction, thus reducing the network load on the centralized component and the need for constant connection.

Considering these high level goals, we must also provide a good basis for creating web applications. To simplify the adoption of the hybrid communication model discussed before, we propose the design of a framework that supports the creation of web applications without requiring end users to install any kind of software or browser plugins, which makes the use of the proposed architecture completely transparent and non-intrusive to clients of web applications built on top of our framework. We must also aim to provide an extendable API, with a basis similar to existing frameworks.

While we could propose a new API for our framework, we believe that such an approach would make the adoption (and testing) of our proposal harder. Therefore, we planned to provide close integration with existing frameworks, mapping existing APIs to our framework, thus allowing application developers to continue using familiar APIs while taking advantage of additional support on the clients with data replication and browser-to-browser communication and synchronization mechanisms (we achieve this by providing shared data types that are causally consistent and have durable storage). However, to allow this we had to study the APIs offered by commonly used frameworks, such that we could steer the design of our framework towards a direction that allows for supporting such an API, and also to leverage our proposed design to improve those frameworks.

In this context, we have studied the following frameworks and respective APIs:

- Google Drive Realtime API[17], Dropbox Datastore[10], and ShareJS[15]. All of these frameworks are used by developers to build applications requiring real-time concurrent editing of data by multiple users. Each of these can profit from using the described framework to reduce client-to-client latency and improving scalability of the centralized component.

- Redis.io[35] is a key-value cache and store. Keys can contain strings, hashes, lists, sets, among others and is therefore often referred to as a data structure service. Our framework could expand Redis.io to provide clients with the ability to locally apply operations over these data structures, while reducing client-to-client latency and server load by sharing them (and respective updates) directly among web clients.

- PeerJS[4] is a peer-to-peer framework supporting the establishment of connections between a pair of browsers providing a generic data-channel between them. Not supporting any kind of data-type abstraction, we could easily enrich this framework with CRDTs and even with group communication primitives that could simplify the design of applications on top of this framework.

- Priv.io[43] and Sporc[12] are systems that resort to a centralized server for storing confidential data. Leveraging our framework, it would be possible to leverage peer-to-peer communication patterns to improve performance and lower trust in centralized components.

We opted to initially create a browser-to-browser communication layer, offering the creation of overlay networks with a messaging API, similar to PeerJS and other messaging systems. On top of the connection and messaging systems, we have decided to add an interface similar to the one offered by the Google Drive Realtime API, using CRDTs internally, to simplify both object replication and synchronization.

3.2 Interaction model

As referred previously, in the classic client-server model all interactions between clients are mediated trough a centralized component. This component is responsible for serving all data objects that the application requires, while also processing all operations that update and modify application state executed by any client.

In our proposal we extend this traditional model as we also offer to the clients the capability to serve replicas of objects directly between them (especially between those that are close, for example, on the same local network), while being able to operate over those objects locally and propagate updates in a decentralized fashion. This allows us to substantially reduce the dependency on the centralized component, which brings several advantages such as the possibility to lower server load and improve latency between clients. In contrast to the client-server communication model this also allows for interaction between clients even when the centralized component becomes temporarily unavailable.

These additional mechanisms imply that the interaction model of client applications with the centralized component that supports a particular web application or service will change substantially from the use commonly found. In more detail, the interaction model we consider in this work is as follows:

First, a user uses his browser to access a web application, residing on a web server. By obtaining the web page, the browser also obtains, transparently to the user, our framework, which is distributed as JavaScript code, and thus included in the web application. After loading these resources, the client's browser initializes the application which will then be responsible for initializing our framework (through a well established API).

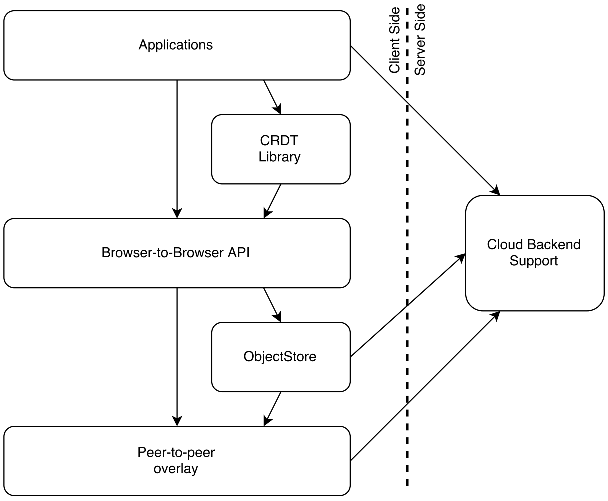


Figure 3.2: Interaction overview of the proposed system

As detailed in Figure 3.2, an application relies on our API to initialize the framework[1]. The framework will join the new client to an overlay network by first establishing a connection (through WebSockets) to a server process.

As soon as a client makes an initial connection to the server, the application can request the framework to obtain copies of resources associated to it – as an example, in case of a chat application, a resource can be a list of messages exchanged previously in a chat room that the user wants to join; alternatively in the context of a collaborative editing application, the resource can be the actual content of a shared document.

This server is responsible to help various clients to establish connections among each other. The logic that defines if two clients should or not establish a connection is application dependent. However, and as we will discuss later, we designed some connection management policies that might benefit various application types.

When the server receives a new client connection, it propagates its connection request to clients that were previously connected, mediating this way the initial connection between multiple browsers. The clients that receive this request can chose to send an answer or to propagate this request to other clients. This answer will then be sent back to the new client. When the new client obtains answers from other clients it initiates a direct

[1] This image shows an overview of interaction with the framework, to help understand developers how accesses are executed. Although it provides some hints to, this figure does not expose the details concerning the internals of our proposed architecture.

browser-to-browser connection. This new client is thus responsible, using the server as mediator, to connect, in a peer-to-peer fashion, to other clients.

To this end we resort to the mechanisms offered by Web Real Time Communication (WebRTC[42]), that, on its own, can resort to STUN and TURN servers[2] for support to surpass firewalls and NATs that restrain browser-to-browser connections.

When connections are established to other clients the communication that is done initially via the server is, transparently to the user, re-directed to use the browser-to-browser communication model.

The resources associated to the application that a client has requested can thus now be requested to other clients. Objects are replicated between clients using the appropriated CRDTs for the kind of data the application needs. CRDTs allow to minimize coordenation between clients when executing operations on their local replicas. Also, these data types allow for replicated objects to converge to a common single shared state. They also allow clients to aggregate various operations of various neighbours to a single message to be propagated to the central component, creating the opportunity to minimize load on this component. This way we are able to reduce client-to-client latency especially on nearby clients, and reduce server load when groups of clients collaborate among each other to aggregate updates. It also becomes possible to avoid the central point of failure. In current architectures, the server is the only communication medium and when it becomes even temporarily unavailable all user interaction is halted. When we allow for browser-to-browser communication all clients can interact between each other, interaction can continue even with temporary server failures.

Considering this interaction model, the framework is composed of the following main logical components:

- A logical peer-to-peer connectivity layer between end users' browsers, supporting the choice between various overlay network configurations. As shown in Section 2.1.1, the applicational needs weight heavily in the choice of the overlay network. Therefore the chosen overlay must be the one that best fits the requirements of possible applications. WebRTC is used as a generic browser-to-browser data transport layer and WebSockets[3] are used for communication with the centralized component of the web application or service. This allows for a generic messaging system between browsers to operate on top of the overlay network.

- Object replication mechanisms which provide the ability to implement CRDTs enabling real-time concurrent manipulation and synchronization of application data objects in a fully decentralized and distributed fashion. The previously described

[2] While these can be privately deployed, we resort to those provided publicly by Google.

[3] WebSockets are implemented by all major browsers and allow for a TCP connections to be established between a browser and a web server, enabling asynchronous client-to-server and even server-to-client communication.

peer-to-peer network module is used to propagate updates among multiple clients that own replicas of each applicational object.

- An extensible CRDT library is provided by our framework as programmers should be able to select or create new CRDTs that better match their application needs. Therefore we provide commonly used data objects in concurrent editing systems, namely List, String, and Map, among others originally specified in [36].
- Logical centralized component to act as signalling service, enabling for easy deployment of a peer-to-peer network. These, as explained further on in this chapter, are also used as in the classical communication model for persistence, and to act as a communication medium between browsers that cannot directly communicate.

In the following section we will explain in greater detail the actual architecture and responsibility of each of its components.

3.3 Architecture Overview

Our proposed solution can be divided into two main components: the connectivity layer and CRDT replication mechanism that run in the user's browser and the other that runs in a centralized, possibly cloud, service. In this section we present a detailed architecture of both, which we will call the *B2BClient* and *B2BServer* components.

With respect to the general interaction model in the previous section, in Figure 3.3 we show the architecture of the system that we propose in this work.

3.3.1 *B2BClient*

The client component of our framework can be divided into two main modules: the object replication module and the connectivity and communication module.

The object replication module, or object store, is responsible for managing objects that the application accesses, using the connectivity and communication module to propagate and receive updates. The communication sub-module is responsible to carry out all communication with other browsers and the centralized component of our system (i.e., the B2BServer component). The connectivity sub-module is responsible to maintain the connections to other clients and to the server. Though these sub-modules are closely related, the logic, or rule-set, that defines how messages should be propagated or connections be made can be very different, hence the separation.

An application that executes in the browser, typically written in JavaScript, is responsible for interacting with the presentation layer (i.e., the web page) and to access the API offered by our framework.

This API allows direct communication with other clients and to access transparently syncronized local replicas of a set of shared data objects. Next we will describe each module of the B2BClient component in more detail:

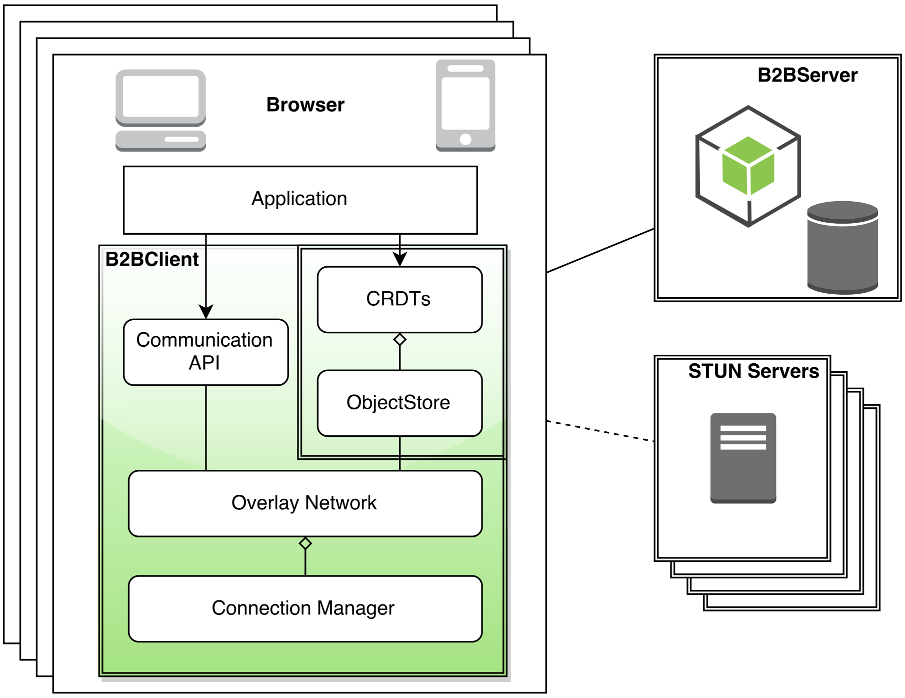


Figure 3.3: Architecture

Communication and connectivity module: The communication and connectivity module is composed by the following sub-modules that execute on the client's browser:

Communication API: Offers a set of communication primitives used to exchange information with other clients (using the information maintained by the *Overlay Network* module). This module provides the application with both unicast and multicast messaging primitives in the context of a logical *Group* that the client has previously joined (we explain further ahead how we leverage this group abstraction). All communication is performed through messages encoded in the JSON format. The mechanisms that are responsible for supporting the communication API use well known epidemic broadcast techniques to propagate information between all elements of a group[3].

Overlay Network (per *Group*): This component manages all information concerning overlay networks to which the client belongs and that are used to propagate information between clients. To achieve this we use a *Group* abstraction, where each client can join multiple groups, and each group can have a set of objects associated with it. To this end, our API provides a Join operation that receives the identifier of a group. As previously discussed, a connection with the server is initially established to send a request that other clients can answer to. These messages are thus sent, by the server, to clients that belong to the same group. This server connection is also used to execute the *Signalling* protocol required by WebRTC (as explained in Section 2.4.1), thus creating new connections to neighbours. The final result of this process is that the clients, for each group, become connected to an overlay network whose topology is similar to a random graph with bi-directional connections, with properties that closely map on the overlay networks maintained by algorithms used in peer-to-peer systems like Cyclon[40], Scamp[14], or HyParView[25]. This component also keeps information on the logical neighbours (i.e., other clients) of the client in each logical overlay network (i.e., one per group) to which the client has joined, while it also implements algorithms for efficient propagation of messages within a group.

Though the server is needed to establish initial connections to other clients, the connections to other clients themselves can be used to find new neighbours and even execute the *Signalling* protocol. This way, within a group, more connections can be created. As neighbours can possibly be members of various groups the client has not yet joined, the client's neighbours themselves can be used directly to join new groups.

To avoid overloading the central component with excessive updates relative to operations by clients over data objects, only a sub-set of the clients maintain a connection to the server. In order to decide which client is responsible for this task we resort to a *Bullying*[8] algorithm that is used to perform (multiple) leader elections in the context of each logical group[4]. In this algorithm each client starts in an *active* state and, periodically, emits to each neighbour a message containing its identifier. Each time a client receives an identifier that is lower than its own, the client changes its state to *inert*, stops broadcasting the periodic message and disconnects from the centralized component. An *inert* node that doesn't receive these messages for a long period of time changes its state back to *active* and re-connects to the central component (restarting to emit the periodic messages). Only nodes in *active* state are thus responsible for interacting with the centralized component, which includes sending messages and aggregating and propagating operations performed over replicated objects for persistence at the central component.

Connection Manager: This component is responsible for the creation and management of connections between clients and also to the centralized component (when required). The use of this component enables us to avoid redundant connections between clients that join multiple groups and with the centralized component. The ConnectionManager resorts to the WebRTC API to create connections between clients and to WebSockets to create connections from clients to the server. WebRTC has native support in the most used web browsers, namely Chrome and Firefox, being possible to resort to instalable plugins to support WebRTC in other browsers. One of the main reasons that motivates the use of WebRTC is the native support in these browsers, as this allows for the use of our framework without any effort (e.g,

[4] We say multiple leader election because, as the network resembles a random graph and not all clients are connected to all clients, within a single group multiple leaders can be elected.

installing software or configuring firewalls and NATs) by the end user. WebRTC always uses secure connections, where all data is encrypted which minimizes the concerns concerning privacy breaches due to eavesdroppers[5].

The existence of Firewalls and NATs have always been a challenge in the deployment and adoption of peer-to-peer systems in the past, as they difficult (or impede) the establishment of connections and direct communication between clients that are behind one of these components. To bypass this practical challenge, WebRTC resorts to services specifically designed to facilitate establishment of connections in these scenarios. There are two kinds of services that can be used. STUN which allows nodes to determine their public IP addresses (in the case of NAT) and install state in firewalls or NATs to allow for direct communication. A second alternative is using TURN services, where external TURN servers are used to redirect traffic between each pair of communicating clients. In the context of our work the use of TURN has been disabled by default (though enabling it is possible). The usage scenarios that we focus on it is more efficient for clients to communicate indirectly through the centralized component of the system than to have all their direct communication redirected by a server the application has no control over. In these cases, just as when a browser doesn't support WebRTC, the framework resorts to the classic client-server interaction model of Web applications, where all user actions are mediated by a centralized component.

In order to establish a direct browser-to-browser connection, WebRTC uses a *Signalling* protocol (as discussed in Section 2.4.1) that requires both clients to exchange information between them. Any mechanism that allows for an exchange of text can be used to materialize this signalling protocol. In our framework, we opt to use the centralized component, by leveraging the existing WebSocket connections, to carry out the necessary information exchange to establish these connections. This decision is also motivated by the fact that most Web services require an authentication process by the user, which necessarily requires sending information to the centralized component when the client starts using the service.

Object store: This module allows, to an application that executes in a browser, to share objects with other client applications that execute remotely in other browsers – typically, instances of the same application. These shared objects contain data manipulated by the application allowing users to co-operate and exchange information by concurrently modifying these same objects. The object store, present in each browser, is composed of the following:

Object replication : This sub-module is responsible for using the group communication primitives to obtain the initial state of objects and to keep these objects updated.

[5] As the reader might expect, the use of this framework rises preoccupations concerning base privacy and security. We intend to address these issues in future work.

To maintain objects up to date, the system leverages the Overlay Network's communication primitives to propagate updates made by individual clients among them. All updates are shared with neighbours and also with the centralized component for durability. The system allows for updates to be queued before propagation, in order to allow for multiple updates to be aggregated and sent in a single message.

This module allows to obtain a string representation of the object's state and to create an object from such a string representation. This allows for the usage of any type of client-side persistence to keep local persistent copies of CRDTs. This enables the possibility for clients that are initializing to load a previously known state from the client device, this immediately enables working with the object while updates to it will eventually, as connections are established and used, be added to the local copy.

CRDT Library : This sub-module provides a set of data types based on CRDT principles coded in JavaScript. This library is used by the application to access objects relevant to the application. We provide a set of commonly used data types (List, String, Map, and Set) which can be used as building blocks for creating applications. This library can be extended by the programmer by defining new types of CRDTs that are better adjusted to the application requirements.

3.3.2 *B2BServer*

To support the services offered at the client side and to allow browser-to-browser communication, our system also includes the following modules that are executed in a centralized component, potentially in a cloud computing enviroment:

B2BServer : The centralized component is materialized trough a server process written in NodeJS. This server receives connections from the client's ConnectionManager (through WebSockets) and acts as an entry point of the clients to our framework.

This component acts as an intermediary during the establishment of direct connections between clients. Also, this component is responsible for mediating the access between clients and the persistence layer of the system, including the access to objects that are not yet available in the clients, and to receive updates on those objects to guarantee the correct storage in the (centralized) persistence layer.

This component is also responsible to serve the clients that are unable to establish browser-to-browser connections with other clients.

STUN Servers : These servers are out of the scope of this work and thus out of the control of our solution. Though possible to be deployed by the application operator, in the context of this work we use those publicly available from Google. STUN servers are used to allow clients that are behind firewalls and NATs to establish browser-to-browser connections. It is also possible to use TURN servers although, as previously discussed, the usage of these is disabled by default (but can be explicitly enabled).

Chapter 4

Browser-to-browser interaction framework

This chapter gives an overview of the API provided by our framework. We detail the available API calls and, subsequently, the designed algorithms that support the functionalities provided by the API.

We start by giving a general overview of how the interface can be used. As depicted in Figure 4.1 we show that in the general use case an application has the following interactions with our interface:

To initialize our framework making it available to the application a *ConnectionManager* object has to be created (*a*). This object receives as a parameter the server address that it connects to (when needed). This object allows the application to *Join* and *Leave* groups. When a Group is joined we enable sending and receiving multicast and point-to-point messages (*b*) in the context of this Group (Section 4.1). A Group offers an object-store to create and interact with objects. Though objects and the object-store can be interacted with directly, normally CRDTs (*c*) will be used as an encapsulation and thus applications interact with them instead (Section 4.2).

Furthermore, in Section 4.3 we discuss relevant parameters and their effects on the framework and system behaviour, in Section 4.4 we explain how to perform server setup, and section 4.5 provides a description of our prototype implementation discussing the most relevant implementation decisions.

4.1 Browser-to-Browser Communication

As explained previously, in our approach communication is performed within the context of groups (detailed in section 4.3). When a *Group* object is obtained, the application is supplied with an interface to interact with the elements of that group (i.e., other instances of the application that also joined the group). The available calls are presented in Listing

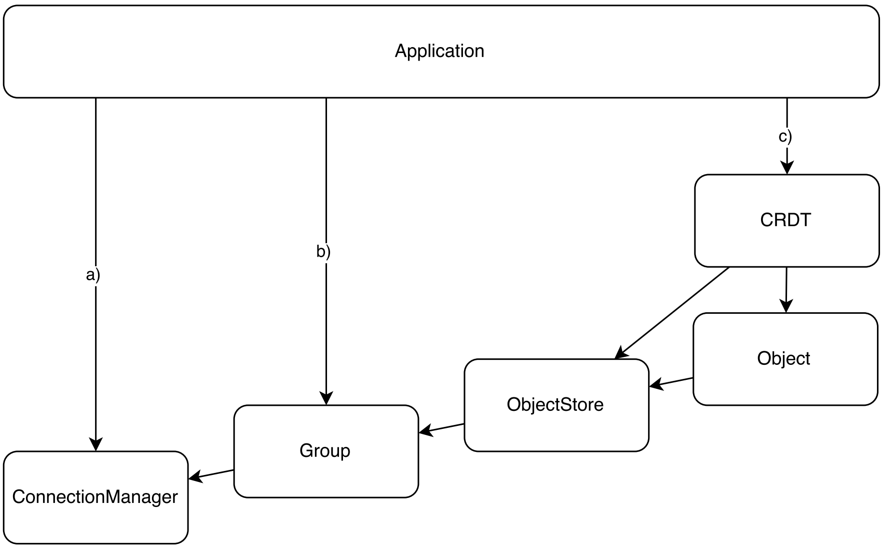


Figure 4.1: API overview

4.1.

```
Group.multicast = function (message);
Group.send = function (peer_id, message);

Group.onReceiveMulticast = function (fn);
Group.offReceiveMulticast = function (id);

Group.onReceive = function (fn);
Group.offReceive = function (id);

Group.sendReply = function (peer_id, message, callback, timeout);
Group.onSendReply = function (fn);

Group.sendAck = function (peer_id, message, callback, timeout);

Group.emit = function (tag, message);
Group.on = function (tag, callback);
```

Listing 4.1: Communication interface

The *Multicast* method is used for messages that are to be delivered to every client in the group. These messages can be in any textual format, from plaintext to JSON objects. *Unicast* messages are only delivered to the client's application whose identifier matches that of the one given as argument (client identifiers are generated by the framework if not given as argument at initialization). *OnReceive* and *onReceiveMulticast* are used to add callbacks that are executed when messages sent from other clients are received locally. These return an integer (i.e., an identifier) that can be used to disable these callbacks.

The *SendReply* operation is used to send messages to a particular destination (i.e., client) and accepts a callback for processing the corresponding answer. The callback is executed with the answer or *null* if the timeout (configured with the timeout value) expires without the reception of the expected answer from the other endpoint. To be able to send replies at the receiver side, the *onSendReply* method has to be previously executed (typically at initialization time) to explicitly configure a handler which accepts the message (type) and returns the intended answer. The *SendAck* operation is similar to *SendReply* as it waits for an acknowledge from the client where the message is delivered at, but no handler has to be specified (and thus only a boolean value is returned meaning the message has been received at the destination or reporting that no delivery has been guaranteed within the specified timeout).

To provide a similar interaction to that of *publish/subscribe* systems, we provide the *emit* function to emit messages that are labelled with a particular (or multiple) tag, while the *on* method can be used to add callbacks for handling messages disseminated through the emit operation, based on their tags.

With respect to this API it is clear that there are no guarantees that all clients will always receive every message (as in, *at least once delivery* guarantees), and that the order of delivery of messages will be the same at all clients. We use optimistic propagation mechanisms (as we detail in Section 4.3) and ensure that each message explicitly sent is delivered at most once (*at most once delivery*).

Algorithm 1 shows the algorithms used by our framework to support these calls. *Upon* denotes a call that was performed by the programmer explicitly and *OnReceive* denotes the reception of a message from the network. All propagation of messages[1] is done through the use of flooding techniques over the available overlay (i.e., logical) network, with respect to the algorithms that can be selected when initializing the framework. The *Propagate* call is thus detailed in Section 4.3. For the sake of simplicity, message identifiers and duplication detection are omitted from the algorithms.

4.1.1 Example

To show how the programmer interface can be used, a small example is provided in Listing 4.2. The details regarding the user interface have been omitted (these details include aspects as the HTML code and the HTML manipulating JavaScript code). The example shows the relevant code to create a simple chat application where users can join rooms.

As previously stated, we first have to create the ConnectionManager object (Line 1). This object can then be used to Join groups. Each room can be joined (or created) by providing a name, which serves as the group identifier (Line 5). These groups can be stored for future use and to add event handlers, where in this case we listen for multicast messages (Line 7). When we want to send messages ourselves, the right group's method for disseminating messages has to be used, in this case, multicast (Line 15).

[1] Except those that follow a point-to-point communication pattern.

Algorithm 1 Communication

```
Upon multicast(message):
    m ← newMessage(message)
    Propagate(m)

Upon send(id, message):
    m ← newMessage(message)
    m.receiverID ← id
    Propagate(m)

Upon sendReply(id, message, fn, t):
    m ← newMessage(message)
    m.receiverID ← id
    m.needsReply ← true
    Propagate(m)
    Upon answer(answer):
        fn(answer)

    Upon timeout(t):
        fn(null)


Upon sendAck(id, message, fn, t):
    m ← newMessage(message)
    m.receiverID ← id
    m.needsAck ← true
    Propagate(m)
    Upon ack():
        fn(true)

    Upon timeout(t):
        fn(false)



Upon emit(tag, message):
    m ← newMessage(message)
    m.tag ← tag
    Propagate(m)

OnReceive Message(m):
    if m.receiver_id then
        if m.receiver_id equals my_id then
            if m.needsReply then
                answer ← OnReplyCallBack(m)
                r ← newReply(answer)
                r.destination ← m.sender
                Propagate(r)
            else if m.needsAck then
                a ← newAck(m.sender)
                Propagate(a)
                DeliverUnicast(m)
            else
                DeliverUnicast(m)
            return
    else if m.tag then
        fns ← CallbacksFor(m.tag)
        for each fn in fns do
            fn(m)
    else
        DeliverMulticast(m)
    Propagate(m)

```

```
var cm = new ConnectionManager({IP: "server_ip", PORT: "server_port"});
var groups = [];

function create_or_join_room(room) {
  cm.joinGroup({id: room}, function (group) {
    groups[room] = group;
    group.onReceiveMulticast(function (message) {
      showMessage(room, message.username, message.time, message.text);
        });
    });
}

function send_message(room) {
  var m = {username: "name", time: new Date(), text: "some input"};
  groups[room].multicast(m);
}
```

Listing 4.2: Example: Chat Room

4.2 Data Model

Our interface for object interaction was created with special care to promote and support the use of CRDTs. We start by explaining how the ObjectStore is used.

The ObjectStore API is provided in Listing 4.3. An ObjectStore exists in the context of a Group, therefore an object-store object can only be obtained from a Group object. The object-store can then be used to get, create or delete objects. The *Get* method calls the given callback if the object is found to exist with as argument the object or with null if no such object could be found (the type of the object that is given as argument to the callback will be detailed further on in this chapter). The *NewObject* attempts to create a new object. The type argument is required so that neighbouring clients and the centralized component know which CRDT implementation (and synchronization mechanisms) to use with new object. The callback is called with the newly created object, an existing object if the object with the given identifier already existed, or null if no answer was received. The *DeleteObject* method attempts to delete the object. This results in propagating the deletion to the network, executing the callback (with true or false) if the deletion has been registered at the persistence layer.

```
Group.joinStore = function ();
Group.leaveStore = function ();

ObjectStore.get = function (object_id, callback);
ObjectStore.newObject = function (object_id, type, callback);
ObjectStore.deleteObject = function (object_id, callback);
```

Listing 4.3: Object Store Interface

Algorithm 2 shows the object-store's internal algorithms that support its programmer interface. As previously stated, the *Propagate* call is detailed in Section 4.3.

Algorithm 2 ObjectStore

```
Upon ObjectStore.Get(oid, callback):
    request ← GetRequest(oid)
    Propagate(request)
    Upon Answer(ret):
        if ret not null then
            o ← new Object(oid, ret)
            callback(o)
        else
            callback(null)


Upon ObjectStore.new(oid, type, callback):
    request ← NewRequest(oid, type)
    Propagate(request)
    Upon Answer(ret):
        if ret not null then
            o ← new Object(oid, ret)
            callback(o)
        else
            o ← new Object(oid, null)
            callback(o)


Upon ObjectStore.delete(oid, callback):
    request ← DeleteRequest(oid)
    Propagate(request)
    Upon Answer(ret):
        callback(ret)
```

When the object-store is not used (i.e., has not been initialized) a client simply propagates to its neighbours all object related messages it receives (i.e., the client does not respond or act as a result to receiving messages that relate to objects, it simply propagates these messages to neighbour nodes). When *JoinStore* is called an *ObjectStore* object is created and all object related messages will instead be sent to and handled by this ObjectStore (instead of being propagated across clients). When *LeaveStore* is executed by the client the previous interaction is resumed. The object-store, when created, is thus responsible to answer to and/or propagate all object related messages. In other words, it retains the responsibility to handle all object related events, being them locally created or received over the network.

All events an object-store receives with respect to an object it does not contain (i.e., the application has not yet tried to access), will simply be propagated. The object returned after calling *Get* or *newObject* is actually responsible to handle the messages received over the network in respect to the object, resorting to the ObjectStore to request propagation of state or operations, which will in turn use the communication primitives available in a Group context.

The objects in our current system, returned after calling *Get* or *newObject*, are CRDTs, which can then be of two kinds: CvRDT (state based) and CmRDT (operation based).

The CRDT data model has been created with respect to the specifications in [36]. The specified CRDTs can resolve diverging state by merging two different states (CvRDT) or by propagating commutative updates in a well-defined order (CmRDT). This requires the system to propagate state (in the case of CvRDT) or operations (in the case of CmRDT). To achieve the propagation of state and operations as response to user interactions the object methods *setValue* and *doOp* have to be called for each CvRDT and CmRDT. The differences between these two types of CRDTs, and their object methods, is covered further ahead in this chapter.

We provide a library of CRDT implementations of some commonly used data types such as Sets, Lists, Maps, among others. Listing 4.4 shows an example of how the existing implementations of CRDTs can be used. As in the previous example, we first have to create the ConnectionManager object (Line 1). When we obtain a Group from this object (Line 3) we can request it to instantiate an ObjectStore (Line 4). In this particular case we then create an *ORSet* (Line 5), which is an Observe-Remove Set as specified in [36]. Our CRDT implementations receive, as argument, an identifier and an ObjectStore which is responsible to handle changes and network events related with this object. Note that this ensures that the object will exist in the context of a Group, and only clients within this group can see or operate on this object. When a CRDT is created we can add event handlers to show the state when the object was first returned (Line 6) and for each following change to the object's state (Line 9) . In this case, as we use a Set, methods as *remove*, *add*, and *lookup* are available (Lines 14,15, and 16), which accept as value any textual representation or JSON objects.

All other CRDTs provided in the CRDT library can be used in a similar way.

```
 1  var cm = new ConnectionManager({IP: 'server_ip', PORT: 'server_port'});
 2
 3  cm.joinGroup( {id: 'group_name'}, function (group) {
 4    objectStore = group.joinStore();
 5    set = new ORSet('object_id', objectStore);
 6    set.setOnInit(function (state) {
 7          //use initial state
 8      });
 9    set.setOnStateChange(function (state) {
10          //use new state
11      });
12  }
13
14  function remove(value) { set.remove(value); }
15  function add(value) { set.add(value); }
16  function find(value) { return set.lookup(value); }
```

Listing 4.4: Example: Object Usage

4.2.1 CvRDT

A CvRDT (Convergent Replicated Data Type) resolves diverging state of two replicas by merging two different states. This requires both replicas to propagate the whole state among each other. As propagation of single operations never occurs, no problems arise in maintaining causal consistency due to the order of how operations are delivered. This ensures that causal consistency is always guaranteed, requiring only eventual communication between pairs of replicas.

Convergent Replicated Data Types, with respect to the specification in [36], require:

Payload, a value for object instantiation.

Queries, for local evaluation of the state without any side affects.

Operations, state affecting operations that must be commutative.

Compare, for deciding if a merge operation should be applied.

Merge, to merge two different replicas.

Listing 4.5 depicts our generic interface of CvRDTs which follows from the specification presented above. *GetValue* is used to obtain the state to execute local queries. *SetOnStateChange* is used to register a callback that is executed whenever the internal state changes due to local operations or remote operations received over the network. *SetValue* has to be executed when state has locally been changed, ie, after each change on *object.state*. This will result in the activation of the inner mechanisms that propagate the new state. *SetCompare* is used to set the Compare function. The Compare function receives as argument two states and must return -1, 0, or 1 (if the first argument is, respectively, older, equal, or newer than the second argument) or *nill* if no conclusion can be made (i.e., each has updates the other hasn't seen yet and thus Merge must be applied). *SetMerge* is used to set the Merge function. This function requires two arguments (the state of each replica) and must return a single state. The *SetToJSONString* and *SetFromJSONString* are used, respectively, to obtain string representations of the object and to convert a string representation back to object representation. These are used to serialize the object for propagation and persistence purposes.

```
BasicObject.getValue = function (); // or BasicObject.state

BasicObject.setOnStateChange = function (callback);
BasicObject.setValue = function ();

BasicObject.setCompare = function (fn);
BasicObject.setMerge = function (fn);

BasicObject.setToJSONString = function (fn);
BasicObject.setFromJSONString = function (fn);
```

Listing 4.5: CvRDT Interface

Algorithm 3 presents details on the operation of a CvRDT object. Note that simple optimizations, such as not sending an updated object back to where the update came from, have been omitted from the algorithm for brevity and clarity.

Algorithm 3 CvRDT

```
Upon setValue():
    if onStateChange then
        onStateChange(self.state)
    self.store.propagate(self.id)

OnReceive FromNetwork(state):
    if fromJSON then
        data ← fromJSON(state)
    else
        data ← state;
    c ← compare(data, self.state)
    if c == 0 then
        //equals
    else if c == -1 then
        //data is newer
        self.state ← data
        if onStateChange then
            onStateChange(state)
        self.store.propagate(self.id)
    else if c == 1 then
        //self.state is newer
        self.store.propagate(self.id)
    else
        //must merge
        self.state ← merge(self.state, data)
        if onStateChange then
            onStateChange(state)
        self.store.propagate(self.id)
```

4.2.1.1 Example

To show how the CvRDT interface can be used, we provide an example of a CvRDT implementation. Listing 4.6 has the required functions for a last writer wins register. Methods starting with *CRDT_LWWRegister* are shared with the server and methods starting with *CLIENT_LWWRegister* are only available to the client. The Compare, merge, fromJSONString, and toJSONString are functions that must be in a shared library with the server. As the server acts as a regular node in the system when it comes to object replication (which also acts as a medium to the persistence layer), it also needs to be able to operate over the objects. Set and Get are only used on the client side, as they are executed by the application based on user interaction. As a more complex example, an Observe-Remove Set, as specified in [36], can be found in Annex A.

```
CRDT_LWWRegister.compare = function (v1, v2) {
    var first = (v1.t > v2.t);
    var second = (v2.t > v1.t);
    if (first && !second) return -1;
    if (!first && second) return 1;
    if (!first && !second) return 0;
    return null;
};

CRDT_LWWRegister.merge = function (v1, v2) {
    if (v1.t > v2.t) return v1;
    else return v2;
};

CRDT_LWWRegister.fromJSONString = function (string) {
    var state = [];
    state.x = string.split(" ")[0];
    state.t = parseInt(string.split(" ")[1]);
    return state;
};

CRDT_LWWRegister.toJSONString = function () {
    return this.state.x + " " + this.state.t;
};

CLIENT_LWWRegister.set = function (value) {
    this.object.state.x = value;
    var newT = new Date().getTime();
    this.object.value.t = newT;
    this.object.setValue();
};

CLIENT_LWWRegister.get = function () {
    return this.object.value.x;
};
```

Listing 4.6: Example: Last Writer Wins Register

4.2.2 CmRDT

A CmRDT (Commutative Replicated Data Type) resolves diverging state of two replicas by propagating operations that modify the state of the object between replicas. This requires both replicas to propagate updates in a well-defined order. Because the whole state is no longer sent over the network, there are no by-default guarantees on causal consistency using these data-types given our communication patterns (we will address this problem later in this section).

Commutative Replicated Data Types, with respect to the specification in [36], require:

Payload, an initial value for instantiation.

Queries, local evaluation of the state without side affects.

Operations, state affecting operations that must be commutative.

When operations are executed, the whole CRDT state doesn't have to be transmitted. Instead, only the operation and respective arguments have to be propagated. Along with each operation we also propagate a version vector to track and ensure causality (as in [22, 31]).

Our implementation of CmRDTs actually diverges from the original specification as we construct CmRDTs on top of CvRDTs. This results in a CmRDT becoming available to be used as a CvRDT, in fact, when an object is initially requested, the interaction with the framework is similar as to in the case of CvRDTs. A client that requests an object initially acts as if it required a CvRDT and, consequently, obtains the whole state. From this point onward, it propagates and receives only operations, thus avoiding the overhead of sending the whole state. Due to this, our implementation of a CmRDT must answer to all CvRDT requirements.

Listing 4.7 shows the extension to the interface made available to programmers to create CmRDTs. *SetOp* adds an operation to the CmRDT. This function requires three arguments: *op*,*fnLocal*, and *fnRemote*. *op* is the identifier, or name, of these operations. Local operations are then executed by *fnLocal* and remote operations by *fnRemote*. *fnLocal* is expected to change the local state and its return value is propagated to all other replicas. These replicas execute *fnRemote* with this value as the argument. The reasoning behind this separation is as follows: take as an example an OR-Set. Locally we aim at executing *ADD*(*element*) but this cannot be propagated to other clients as it would break the observe-remove specification[36]. The local *ADD* would return, as an example, a vector with *element*, *unique*, and the remote operation uses the unique value to ensure the correct operation on each replica. To ensure this proposed execution the *doOp* operation has to be called to execute operations that modify the local state. This method will call the previously given *fnLocal* and propagate the return value to be used on other replicas.

```
//OPBasedObject inherits all methods from BasicObject;
OPBasedObject.setOp = function (op, fnLocal, fnRemote);
OPBasedObject.doOp = function (op, args);
```

Listing 4.7: CmRDT Interface

Algorithm 4 presents details concerning the internal operation of a CmRDT object. To execute an operation we call the local function and store this call in a local operation history (which's usage will be explained further ahead in this chapter). The operation is then propagated, along with the return value from the local operation handling function, and the current version vector. When a replica receives operations it first checks if all dependencies are met and, if this is not the case, sends back its own version vector as to request missing updates (Section 4.3.2). If all dependencies are met the operations are executed, delivered to the interface (the application), and propagated.

Algorithm 4 CmRDT

```
Upon DoOp(op, args):
    newArgs ← self.ops[op](args)
    vv ← self.getVV( )
    opNum ← self.addToHistory(op, newArgs, vv)
    PropagateOp(self.id, vv, opNUm, op, newArgs)
    DeliverToInterface(op)

OnReceive FromNetwork(opGroup):
    if not depsMet(opGroup.vv then
        self.sendBackVV
        return
    else
        for op in opGroup do
            self.ops[op.op](op.args)
            history.add(op)
            DeliverToInterface(op)
        ForwardOps(opGroup)
```

4.2.2.1 Example

For completeness, we provide an example of a CmRDT implementation. Listing 4.8 has the methods required to extend a CvRDT implementation of an observe-remove set (which is presented in Annex A). Methods starting with *OP_ORSet* (lines 2 and 6) and *obj.setOp* (lines 11 and 30) are to be executed at the client side, while methods starting with *CRDT_ORSet* are within a shared library between clients and server.

We start by creating a programmer interface which calls the inner *DoOp* method. To call the *doOp*, *setOp* has to be previously executed with the correct argument, typically when the object is initialized. The functions added to *setOp* have to provide as return value the arguments that are to be sent over the network to other nodes. The remaining functions are those that are called when an update is received over the network. Note that, in the case of a Set, we could not simply send a *remove* as this would affect causality. We have to make sure that the added and removed unique identifiers associated to elements are exactly the same, i.e., we have to ensure that all operations are idempotent on all nodes that replicate the object.

```
//interface methods:
OP_ORSet.remove = function (elem) {
    self.doOp("rm", [elem]);
};

OP_ORSet.add = function (elem) {
    self.doOp("add", [elem]);
};

//local remove:
obj.setOp("rm", function (arg) {
    var v = self.state; var timeStamps = v.elements[arg];
    v.tombstones = v.tombstones.concat(timeStamps);
    delete v.elements[arg];
    return [arg, timeStamps];
}, CRDT_ORSet.rm);
//from network remove:
CRDT_ORSet.rm = function (element, ids) {
    var v = self.state;
    v.tombstones = v.tombstones.add(ids);
    if (v.elements[element]) {
        v.elements[element] = v.elements[element].except(ids);
        if (v.elements[element].length == 0) {
            delete v.elements[element];
        }
    }
};

//local add:
obj.setOp("add", function (arg) {
    var v = self.state, timeStamps;
    if (v.elements[arg]) {
        timeStamps = v.elements[arg];
        v.tombstones = v.tombstones.concat(timeStamps);
        delete v.elements[arg];
    }
    v.elements[arg] = [];
    var newTimestamp = self.generateTimeStamp();
    v.elements[arg].push(newTimestamp);
    return [arg, newTimestamp, timeStamps];
}, CRDT_ORSet.add);
//from network add:
CRDT_ORSet.add = function (rand, element, removed_timeStamps) {
    var v = self.state;
    if (removed_timeStamps) {
        v.tombstones = v.tombstones.add(removed_timeStamps);
        v.elements[element] = v.elements[element].except(removed_timeStamps);
    v.elements[element].push(rand);
};
```

Listing 4.8: Example: Operation Based OR-Set

4.3 Initialization and Parameters

In this section we detail how to initialize our system and also detail different implementation options to some of the modules of the framework and discuss the implications on their usage.

There are in fact three fundamental components of the framework that can be differently initialized in this step:

Communication and Propagation protocols (Section 4.3.1), which is responsible to decide which neighbours to communicate or propagate messages to. In other words, this sub-system defines and controls communication patters.

Object protocols (Section 4.3.2), which are responsible to handle all object related events, including applying received updates to objects and use the propagation mechanisms to propagate these updates to other nodes in the system.

Membership protocols (Section 4.3.3), which is responsible for handling affiliation (in our case, connection to neighbours creating overlay networks).

In reality, the initialization of our system with the full instancing of all chosen protocols (each detailed later in this section) is shown in Listing 4.9.

```
var server = {IP: "ip", PORT: "port"};
var options = {
  messageProtocol: {"Bully | Flood"}, // Discussed in Section 4.3.1
  objectProtocol: {"ByTime | Immediate"}, // Discussed in Section 4.3.2
  membershipProtocol: {"Clique | RandomGraph"} // Discussed in Section 4.3.3
};
function init() {
  manager = new ConnectionManager(server, options);
}
function join_group(group_identifier) {
  manager.joinGroup(group_identifier, function (group) {
    if (!group) {
      //no available connection to this group
    } else {
      //use group
    }
  }
}
```

Listing 4.9: Initialization and Parameters

4.3.1 Communication and Propagation Protocols

On the server side, all messages are simply propagated (respecting the context defined by groups) to all connected clients.

On the client side, in our current prototype, we provide the programmer implementations of two different strategies for defining and controlling communication patterns: *Flood* or *Bully*.

When opting for *Flood*, all messages are sent from all nodes to all available connections to other clients (browser-to-browser connections) and the server.

When choosing *Bully*, internally a protocol for (multiple) leader election[8] is used. A bullying mechanism consists in each node periodically sending a message to other nodes, whose contents have, for instance, the node's identifier. When a node receives such a message with an identifier lower than its own, it decides that the sender of the message is its bully node. When a node decides on a bully node, it stops sending the periodical message to neighbours and disconnects from the centralized component. When a node that was bullied stops receiving the periodical messages for a long enough interval of time, it restarts to emit the messages periodically and re-connects to the centralized component. There are thus some chosen clients (bullies) that will be used for ensuring propagation of operations to the central component and all remaining clients (non-bullies) will disconnect from the centralized component and only interact with client neighbours. Due to how connections are established among clients (as detailed in section 4.3.3), typically multiple bullies will exists in each overlay network as to not create a bottleneck on these bully nodes and to avoid a single point of failure. Algorithm 5 provides additional details on this mechanism.

Algorithm 5 Bullying Protocols

```
Upon Init():
    isBully ← $true$
    myBully ← $null$
    lastTimeBullied ← $null$
    doBully( )

Every $\Delta Tsend$ do:
    if isBully then
        doBully( )

OnReceive Bully($id$, $time$):
    if $id$ less than self.id then
        if $id$ less than or equals myBully then
            myBully ← $id$
            lastTimeBullied ← $time$

Every $\Delta Tcheck$ do:
    $\Delta t$ ← currTime - lastTimeBullied
    if $\Delta t$ more than $\Delta Tinterval$ then
        Init( )

Upon doBully():
    $t$ ← new Time( )
    $b$ ← newBully(self.id, $t$)
    Propagate($b$)
```

When the Bully algorithm is applied the two types of resulting clients act as following: the bully propagates to the server and to all client connections, a non-bully propagates to its bully and to a sub-set of its current non-bully connections. This ensures the emergence of a gossip like mechanism, where the size of the sub-set to which non-bullies send messages is a system parameter, operating in a way similar to the fanout parameter of gossip[25]. Bully nodes always propagate to the server and to all clients they are connected to. The remaining clients propagate messages using a gossip like mechanism, sending to a sub-set of their connections and always to their bully. This means a fanout value has to be set. The fanout value is the amount of clients from the current connections that messages are propagated to.

In Listing 4.10 we show how this is achieved. The fanout value is verified dynamically, this means that, as fanout we can give a function which is evaluated each time the fanout is used. This way fanout can depend for example on the amount of connected nodes or browser and device type, among other aspects.

```
var options = {
  messageProtocol: {name: "Bully", fanout: 2}
  //messageProtocol: {name: "Flood"}
};
```

Listing 4.10: Parameters on Messaging Mechanisms

4.3.2 Object Management Mechanisms

We give the option to aggregate updates during a time interval as to lower the total bandwidth usage from clients to the central component or even among clients. Delaying updates naturally results in an artificial delay in which clients observe updates (we will detail the implications of this further on). The mechanisms that manage object related events are thus queues of updates where information (the object for state based propagation or the operations for operation based propagation) is queued until the next propagation phase.

The object store can thus be seen as to have two main phases: the aggregation phase and the propagation phase.

There are two parameters that control the duration of the aggregation phases, *peerTime* and *serverTime*. These define the time interval of the aggregation phase and each time this interval ends, propagation of the updates is done. The division of these times comes with the fact that if we want to keep low latencies between clients we might want to have a small aggregation phase before propagating updates to neighbouring clients while we could aggregate for a longer time when we sent data to the server. In other words, the delay in propagating updates between clients and between clients and the server might have application specific requirements.

When browsers are able to connect we expect that sending data is inexpensive and can be achieved with low latencies. We can thus use a lower interval (or zero) for delaying propagation across browser-to-browser connections. On the other hand, sending data to the server can be expensive for the application provider and the impact on delays on this propagation phase might be lower to clients.

Consider, as an example, that we have a company that uses a collaborative editing application where multiple users can edit a text document, and that this company has two datacenters, where each is used to support approximately half of their users (effectively dividing users into two groups). We could set the parameters as to have a very low aggregating time between client nodes (0 - 200 ms) and a high value for aggregating before sending to a server (2-5 seconds). These parameters do not give a degradation in what the user expects of the application, but does in fact potentially improve the load and hence, scalability of the centralized component.

Algorithm 6 depicts the algorithms used for object propagation. *PropagateObject* is used by CvRDTs and *PropagateOP* by CmRDT to, respectively, propagate state or operations. The objects' identifier or operation is stored in a queue for the server and an additional queue for the node peers. These queues are flushed every time the corresponding propagation phase is executed. Clearing the state queues results in the propagation of the queued objects. Clearing the operation queue results in the propagation of the queued operations. When a queue has multiple elements only a single message has to be propagated, encoding the collection of elements to be sent.

Note the existence of a a Shutdown call. This method is called when *LeaveStore* is executed by the application on a Group. Typically this method is explicitly called when a page is closed to ensure that operations are propagated to the network (either to other clients or the centralized component).

The methods that propagate state or operations (*SendObjects* and *SendOperations*) respect the messaging protocols described in Section 4.3.1. The first method sends, for each object id, the current state across available connections. The second method sends a single message with all operations.

Algorithm 6 Object Protocols

```
Upon PropagateObject(oid):
    QpeerState.APPEND(oid)
    QserverState.APPEND(oid)

Upon PropagateOp(oid, vv, op, args):
    operation ← QueueOp(oid)
    operation.op ← op, args, vv
    QpeerOps.APPEND(operation)
    QserverOps.APPEND(operation)

Every $\Delta T peerInterval$ do:
    clearQueues
        (QpeerState, QpeerOps)

Every $\Delta T serverInterval$ do:
    clearQueues
        (QserverState , QserverOps)


Upon clearQueues(stateQ, operationQ):
    if peerQueues then
        cs ← peerConnections
    else
        cs ← serverConnections
    SendObjects(cs, stateQueue)
    SendOps(cs, operationQueue)

Upon Shutdown():
    clearQueues
        (QpeerState, QpeerOps)
    clearQueues
        (QserverState , QserverOps)
```

Listing 4.11 shows a typical configuration of the propagation mechanism. When *ByTime* is selected an aggregation phase is used for both client and server connections based on the intervals given. When the interval is zero then all updates are immediately sent (i.e., no queue is used). When initiated with *Immediate* internally it actually executes *ByTime* with the remaining arguments taking a value of zero.

```
var options = {
  objectProtocol: {name: "ByTime", peerTime: 200, serverTime: 3000}
  //objectProtocol: {name: "Immediate"}
};
```

Listing 4.11: Parameters on Object Propagation Mechanisms

4.3.3 Membership and Overlay Management Protocols

As previously mentioned, membership management relies on the notion of groups of clients. Each group is materialized by an overlay network which is leveraged to support communication mechanisms across that group.

There are two types of overlay networks that can be selected by the programmer when instantiating our framework in the current version of the prototype:

A **Clique**, where all clients attempt to connect to all clients in the system.

A **Random Graph**, where clients connect only to a random sub-set of other clients.

To join a group a client has to send a *JoinRequest* message to other nodes, which can propagate this message to other clients and also reply with a *JoinAnswer* message. When a node receives a *JoinAnswer* it can initiate a WebRTC connection and propagate the required textual data over the existing connections.

When a client initializes the framework and requests to join a group a connection to the server first has to be created. The server, when receiving a *JoinRequest*, propagates this message to clients it is connected to, with respect to the context of the group which the client wants to join.

When a clients attempts to join a group which forms a clique all clients which receive the request will propagate this request and also send an answer, this effectively builds a partial clique where the connections that lack are those impossible to be established due to limitations of the browser of one client or due to the existence of firewalls and NATs.

To build a logical connection with properties similar to those of a random graph we implemented a mechanism where in the end, for each group, an overlay network is established by leveraging design principles of overlay networks maintained by algorithms used in peer-to-peer systems like Cyclon[X], Scamp[Y], or HyParView[Z].

The main difference between a clique and a random graph is that a clique attempts to connect to all nodes and propagates all membership related messages to all neighbours while with random graphs only a sub-set of the available clients is connected to, using random walks over the existing logical network links to build a random graph. A random walk in this context is the propagation of a message over the network (in our case a *JoinRequest*) in a fixed maximum number of steps. At each propagation point selecting the next step for the message is made at random from the set of neighbours of the node forwarding the message.

The algorithm, detailed in Algorithm 7, requires the following parameters to be set: *min*, which controls the minimum browser-to-browser connections a client strives to maintain; *max*, which controls the maximum acceptable number of connections; *initial_n*, the amount of connections a client requests when joining a group; *initial_ttl*, the time-to-live value of random walks; *meta*, time interval to send meta-data to peers (which's usage will be clear next); *timeout*, time interval to check for maximum and minimum connection count; and *rand*, the probability factor for connections (as explained next).

The way this mechanism works is as follows: first the client establishes a connection to the server. The client then sends a *JoinRequest* over this connection, adding to the message a time-to-live (TTL) for random walks, and a number, N, which represents the number of connections the node wishes to establish. The server will propagate this message to a maximum of N nodes, modifying the N value of the message as to divide N between all clients, and decrements the TTL value (as an example, the propagation of a request with N equal to eight being sent to four nodes, each receives a request with a N value of two). When a client node receives a *JoinRequest* it acts, in order of preference, as follows: *i*) if the current amount of connections is below the minimum specified then send a *JoinAnswer*; *ii*) if the request's TTL is zero then send a *JoinAnswer*; *iii*) if current connection count is below the maximum then the client sends a *JoinAnswer* with a given probability which is a configuration parameter (*rand*). If the value returned by this function falls below a certain threshold (which can be changed by parameter) the TTL of the message is decremented and, if a *JoinAnswer* was send (i.e., the new client was included as overlay neighbour of the current node), N is also decremented. Finally, the request is propagated to at most N nodes, dividing N by the number of nodes to which the message is propagated to.

To ensure that the minimum amount of connections is maintained, an interval is specified to check the current connection count. Every interval of *timeout* duration, each node verifies the number of overlay neighbours it owns currently. If this count falls below the minimum then a new *JoinRequest* is sent, as previously described, but with N equal to *max* minus the current amount of neighbours. If the count is above the maximum, then a peer is chosen to be disconnected and removed as neighbour. To ensure we do not remove the peers that would immediately have to re-initiate new connections to clients we opt to remove the best connected peer from the connected peers.

To be able to chose a best connected peer, each peer sends, each *meta* milliseconds, a message with the count of number of peer-to-peer connections and server connections it has. The peer to be removed is the one with the most browser-to-browser connections, where the count of server connections is used as disambiguation. If no distinction can be made between peers then the peer to be removed is chosen at random among the existing candidates.

Algorithm 7 Membership Protocols

```
Set: $min, max, initial\_n, initial\_ttl$
    $meta, timeout, rand$

Every $\Delta T meta$ do:
    SendPeerMetadata( )

Upon ServerConnection():
    $request \leftarrow$ new JoinRequest( )
    $request$.N $\leftarrow initial\_n$
    $request$.TTL $\leftarrow initial\_ttl$
    Propagate($request$)

Every $\Delta T timeout$ do:
    $count \leftarrow$ getConnectionCount( )
    if count less than min then
        $request \leftarrow$ new JoinRequest( )
        $request$.N $\leftarrow max - count$
        $request$.TTL $\leftarrow initial\_ttl$
        Propagate($request$)
    else if count higher than max then
        RemoveBestPeer( )

OnReceive JoinRequest($req$):
    if alreadyConnected then
        $req.TTL--$
        propagateToN($req.N$, $req$)
        return
    $done \leftarrow false$
    $count \leftarrow$ getConnectionCount( )
    if count less than min then
        SendAnswer($req$)
        $done \leftarrow true$
    else if $req.TTL$ equals 0 then
        SendAnswer($req$)
        $done \leftarrow true$
    else if $count$ less than max then
        if Random( ) less than rand then
            SendAnswer($req$)
            $done \leftarrow true$
    $req.TTL--$
    if $done$ then
        $req.N--$
    if $req.N$, $req.TTL$ higher than 0 then
        propagateToN($req.N$, $req$)

```

Listing 4.12 shows how parameterization with respect to the previously mentioned mechanisms can be done. The programmer can chose between *Clique* or *RandomGraph*. When *RandomGraph* is chosen values for each of the algorithm's parameters can also be set. Each of these values can be a function that returns an integer, enabling the system to dynamically adjust its behaviour if there was some need to increase or decrease the number of connections owned by each client.

```
var options = {
  //membershipProtocol: {name: "Clique"}
  membershipProtocol: {
    name: "RandomGraph",
    min: 2,
    max: 5,
    initial_n: 3,
    initial_ttl: 3
    }
};
```

Listing 4.12: Parameters on Membership Mechanisms

Though all peer-to-peer (more precisely, browser-to-browser) connections are used *under-the-hood*, in Listing 4.13 we give options that control what kind of connections should be employed by a particular application (or, as these values can be set before the initialization of the framework, they can be set differently per client device). With *PC_host* set we enable WebRTC connections that do not require the use of STUN or TURN (i.e., in the common case, WebRTC connections become restricted to local network connections or to computers that directly expose their public ip addresses). *PC_reflexive* and *PC_relay* enable, respectively, the usage of STUN and TURN servers. By default we disable the usage of TURN servers as our centralized component is already used to provide the same functionality. The usage of TURN can be enabled to attempt to further reduce load on the centralized component but can incur to unexpected latencies as clients interact via a, possibly distant, server.

```
var PC_host = true;
var PC_reflexive = true;
var PC_relay = false;
```

Listing 4.13: Peer-to-peer connectivity defaults

So far we discussed the system as if no failures ever happen. In our model we expect clients to be removed from the network and servers to become temporarily unavailable.

The first case that we consider is the fact that a client can, at any given time, be removed from the network (for instance, because the user closes its browser). To ensure that new client connections (instantiations of WebRTC connections) are not waiting for an eventual answer from a non-existing client, we added a *PEER_INIT_TIMEOUT*. When this timeout is reached the client is assumed to be no longer available (due to departure or fault).

The second case that we consider is the fact that the connection to the server can fail (failure of direct connections are easy to discover, but when nodes are using other nodes to propagate messages, it has to be clear when a path to the server no longer exists). To this end we added a heartbeat mechanism to our overlay management algorithm, where heartbeats are propagated between clients in the context of a group. This ensures that, using the overlay network, clients are sure of an eventual propagation to the centralized component. When a client does not receive the heartbeat for a long enough amount of time (i.e., the *SERVER_HB_TIMEOUT* timeout expires), it attempts to re-connect to the server.

Note, however, that when the Bullying protocol is used only the bully nodes are responsible to re-connect to the server. Non-bully nodes do not need to check for server heartbeats as they have a node that takes over this responsibility for them.

It is important to understand that when a Bully node is removed from the network, possibly several clients have their path to the centralized component removed, and thus all would eventually fire the timeout. It is thus important to ensure that the time interval to detect a failure on the bully node is small and that re-establishing new bullies is done in a small interval of time.

Note that all the previously explained timeouts can be parameterized at the initialization of the framework.

4.4 Server Setup

The client application code, which is composed by web pages and JavaScript code of both framework and application, can be served by any Web server.

The server component which act as support for the framework is materialized trough a server process written in NodeJS in our prototype. As it is not only used as an entry point for new nodes, but also acts as an intermediary between the browser-to-browser infrastructure and the centralized application persistence layer, we have to provide extensibility of the server component.

The server contains two functions that can be overridden to interact with any kind of persistence layer or external server. The *Deliver* function is called with each *Message* that has been sent using the messaging API. The *DeliverObject* function is called with each single object related message, from creation and deletion to single updates of state. By overriding these two methods the programmer can interact with any abstract persistence layer, which can range from a text file to a database running on a different machine or set of machines.

To extend the CRDT library with new CRDT implementations, being them Cv or Cm RDTs, we can extend the CRDTObject in the server side code, just as the CRDTs already provided in the library do. Note that this code is a sub-set of the code required by the client, as typically the server process has no applicational execution (i.e., doesn't explicitly perform local mutations of the state due to human interaction).

The server packages are maintained with *npm*, Node Package Manager. This ensures that the minimal required server setup is as simple as downloading the code, executing *npm install* and running the server by executing *node manager.js*.

Authentication to the Web service is responsability of the developer. This comes from the fact that most developers or organizations already have their own methods for authenticating clients across their applications. We do however intend, in future work, to add support that simplifies the extension of our framework to add simple authentication mechanisms.

4.5 Implementation Details

In this section we present some details of our implementation of the framework prototype that may have a minimal, albeit visible, effect on measurements performed in our experimental work, reported in the next chapter. Our implementation, including code of the centralized component, consists of 5,526 lines of JavaScript. Using an automated JavaScript code minimizing tool[2], the client side code results in a 39 KB size file. This implies that an additional 39KB can be downloaded from the web server when loading the web application into the browser.

To connect with the *B2BServer* we use WebSockets, available in all major browsers.

To connect from browser-to-browser we use WebRTC, as previously explained.

All communication between clients and server nodes is performed over JSON (JavaScript Object Notation), as it is easy for humans to read and computers to generate, being widely used in JavaScript applications.

All data that is sent over the network is compressed, which is expected to greatly reduce network traffic. To compress messages we call *compress* and *decompress* before sending messages. Our current implementation uses a LZMA implementation[3] and can be exchanged with any other compression method by overriding the *compress* and *decompress* functions on both the client and server.

[2] Namely the tool available at https://github.com/mishoo/UglifyJS2.

[3] In particular, the one available at http://github.com/nmrugg/LZMA-JS.

Chapter 5

Evaluation

In this chapter we present the evaluation of our work. In Section 5.1 we show results that evaluate alternatives to the materialization of modules in our framework using Micro-Benchmarks. As our main intent is to show the feasibility of using browser-to-browser as a communication model for richer applications, in Section 5.2 we show results of the comparison of our solution to an extablished industry solution using the traditional communication model (client-server).

All experiments reported here were conducted in a setting that uses server and clients nodes running in a cloud platform (Amazon AWS).

5.1 Micro-Benchmarks

In this section we show results of micro-benchmarks over different implementation alternatives in modules of our system. We also present a series of micro-benchmarks to evaluate specific functionalities of the system. These benchmarks help us ensure that the most adequate implementation and parameters are used for a valid comparison with an industry solution.

We start off, in Section 5.1.1, by showing the impact of adding support for the propagation of operations of CRDTs instead of propagating the whole state (i.e., the use of CvRDT versus CmRDT). In Section 5.1.2 we compare the impact of using different overlay networks. In Section 5.1.3 we evaluate a specific overlay network's properties (an unstructured overlay network that establishes a random graph across nodes). In Section 5.1.4 we briefly evaluate the support for disconnection and other execution environments.

5.1.1 Comparison on the usage of CvRDT and CmRDT

To evaluate the effectiveness of propagating only the operations we directly compare the usage of CvRDTs (state-based propagation) and CmRDTs (operation-based propagation). The data type we use, for both CRDTs, is an observe remove set, as specified in [33, 36].

In this experiment we run two clients: a writer and an observer. Both of these clients join a single group (i.e., they connect to each other) and both obtain the same two objects, one materialized by a CvRDT and the other materialized by a CmRDT. The writer client executes write operations on the objects while the observer client receives updates and measures the size of messages it received by it.

The writer client adds a total of 2,000 random words (4 characters each), always executing the same operation on both objects. The observer client logs to the console the message size (in bytes). Figure 5.1 summarizes the size of messages received by the observer as the number of write operations increases.

As expected, state based propagation has a high increase in message size based due to the constant increase in object size produced by this workload, while operation based propagation continually transmits small messages. This happens because, as the size of the object increases linearly with the number of operations, and as such, so do the messages that have to be sent encoding this state. Operation based only propagates operations and associated meta-data (as explained in Section 4.2), keeping the message sizes small.

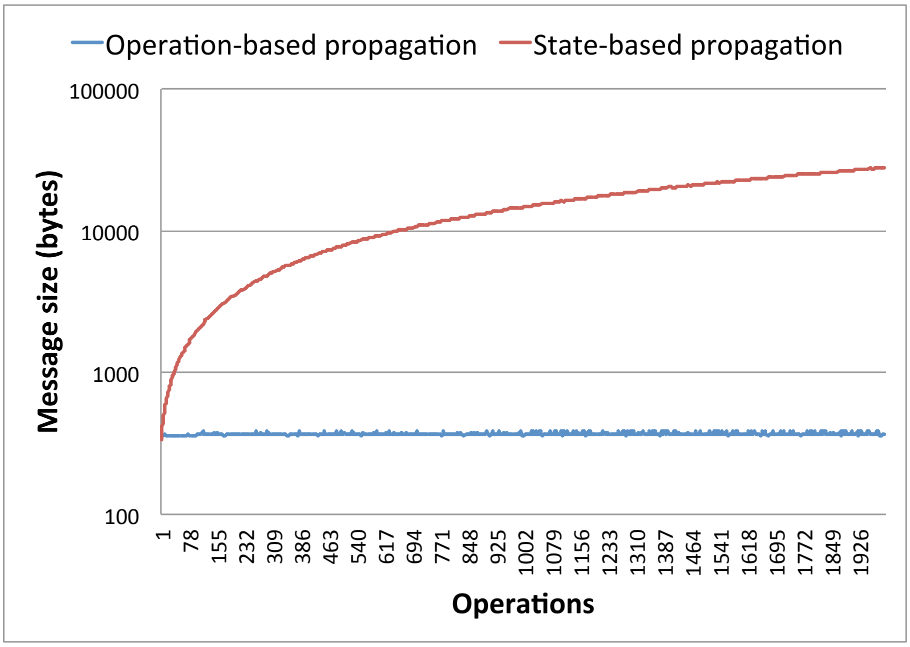


Figure 5.1: Message sizes of state and operation based propagation

5.1.1.1 Impact of compression

Figure 5.2 shows the effects of message compression on message size when using both types of CRDTs. In this experiment we run the previous example, but only up to 50 updates. The objects used are, as in the previous experiment, a state-based (CvRDT) and an operation based (CmRDT), Observe-Remove Sets. *C Cm* and *C Cv* are the compressed values for operation based and state based propagation, while *Cm* and *Cv* show the results with no compression.

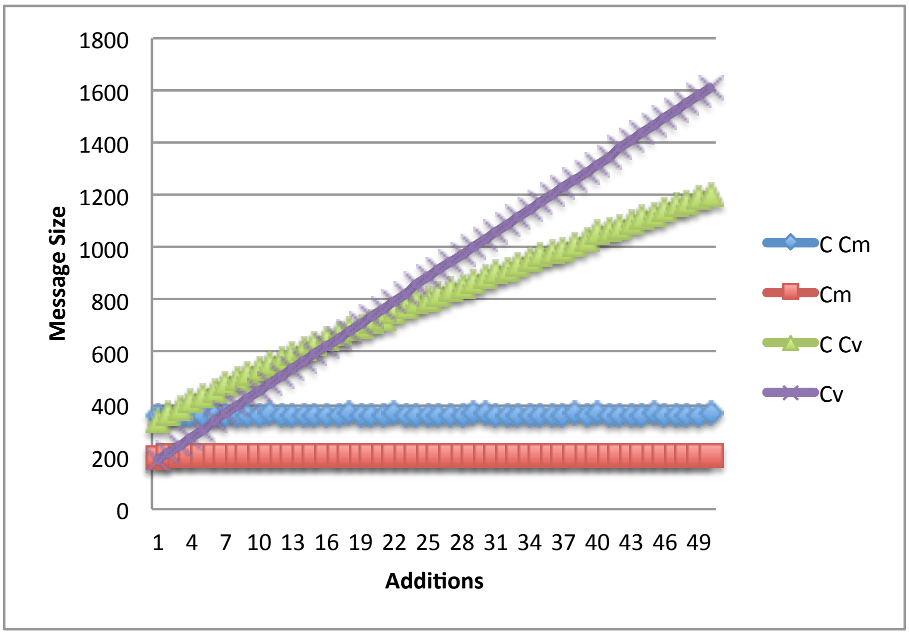


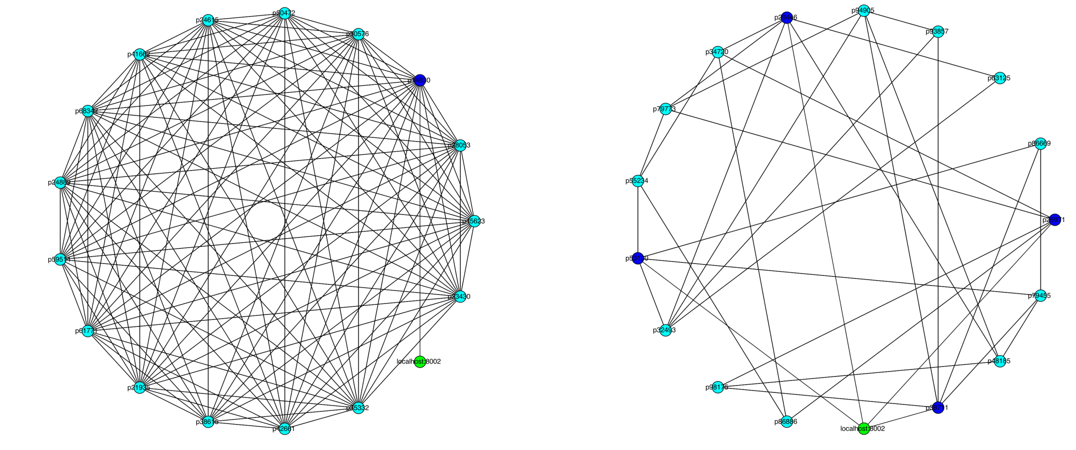
Figure 5.2: Impact of compression

As expected, compression has a more noticeable positive impact on messages with larger sizes. On the other hand, when using small messages the overhead introduced by compressing becomes non-negligible. Due to this result we decided to have all messages to be sent in their smaller format (i.e., large messages are typically compressed while small ones are sent as they are).

5.1.2 Impact of different overlay networks

The initial implementation of our prototype relied on an overlay network that attempts to establish connections between all nodes (i.e., forming a clique). This ensures that communication latency among peers is kept low by avoiding logical connection paths with multiple hops. Unfortunately, such a system does not scale to high numbers of participants. Furthermore, if we use a bullying protocol there would be only one bully in each group of clients, creating a contention point at this node when interacting with the central component. The amount of connections in small groups pose no problem, but when the amount of clients increases so does the connections kept active, and the overhead produced due to redundant messages. Thus we implemented a mechanism to build an overlay network that builds a random graph, as previously described in Section 4.3.3.

Figure 5.3 visually captures the parallel between the usage of each of our implemented overlay networks. A line between two nodes represents a data-channel that has successfully been established. Darker nodes are *bullies* and the green node is the server (labeled *localhost*). These figures were generated by executing an experiment, which consists in initializing the framework and joining a group only changing, between experiments, the overlay network parameter from *Clique* to *RandomGraph* (the remainder of the loaded page and code are exactly the same). We run a single server and 16 clients all on a local network, and each of these clients attempts to join the same group.



(a) Clique

(b) Random Graph

Figure 5.3: Overlay network comparison

Note that when all nodes connect to each other only one bully exists, which generates the previously discussed bottleneck when the number of clients grows. With a random graph the algorithm elects multiple bullies. This ensures that no bully will be overloaded while still reducing the amount of connections (and network traffic) to the server.

When running the experiment using the random graph overlay, we used the following parameters: *min* was set to 2; *max* had a value of 5; *initial_N* was set to 3; *initial_TTL* with a value of 3.

Figure 5.4 shows the same random graph in a different layout. This example shows that: bullies are never connected (as one would *bully* the other which the algorithm does not allow); only bullies are connected to the server; the *min* and *max* parameters are respected.

Multiple runs gave very similar results, and have shown that the implementation of the protocol leveraging this overlay was correct.

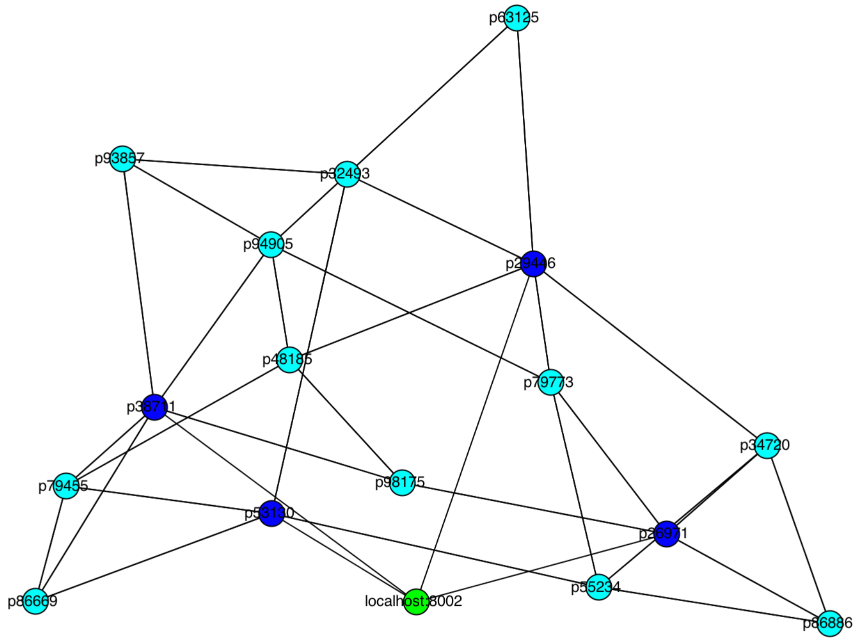


Figure 5.4: Random Graph Detail

5.1.3 Random Graph Properties

To effectively use the framework the correct parameters have to be set to ensure that the random graph built by our algorithm have adequate properties for the application (a discussion of these can be found in [23]). These parameters will heavily impact on the overlay network and thus affect the time of message propagation and the reliability of dissemination. To this end, we evaluate the following relevant properties:

- **Connectivity.** The overlay should be connected.

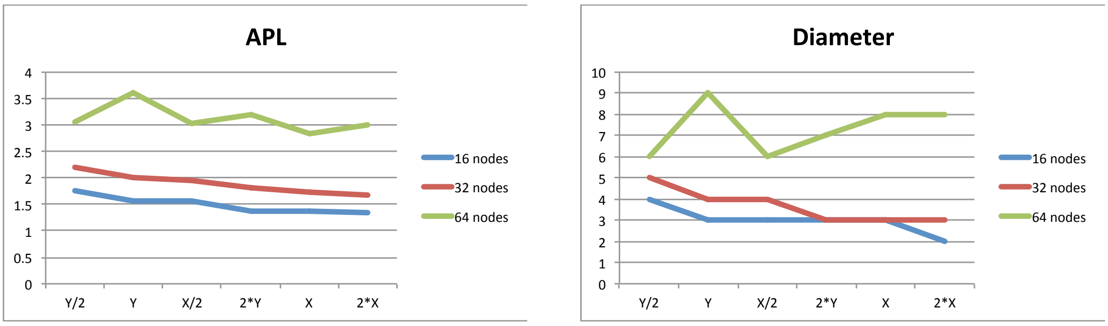
- **Degree Distribution.** The connections between nodes should be evenly distributed. Nodes shouldn't connect to all nodes (which can overload a node) and nodes should have a minimum number of connections (for fault tolerance).

- **Diameter and Average Path Length.** Diameter is the length of the minimum path between the two furthest nodes in the network. The average path length is the average of the length along the shortest paths between all pair of nodes in the overlay. Enforcing a small value for overlay diameter and keeping a low average path length ensures efficiency of information dissemination, as it influences the time a message takes to reach all nodes.

- **Clustering Coefficient.** The clustering coefficient of a node is the number of edges between that node's neighbours divided by the maximum possible number of edges across those neighbours. It has a value between 0 and 1 and it represents the density of neighbour relations across the neighbours of a node. Keeping a low average of clustering coefficients across all nodes ensures a small amount of redundant messages. This value also has impact on fault tolerance, as closely knit groups of nodes can easily become isolated from the rest of the graph.

To evaluate the properties of our random graph overlay we run the following experiment: we execute a varying number of client nodes that attempt to connect to a single group. The parameters vary as the amount of clients increases. If N is the amount of clients, the parameters are set as follows: min is set to $ln(N)+1$; $initial\_N$ is set to $(min+max)/2$ and $initial\_ttl$ is set to 3. The max value varies, equalling to the min value adding each of the following: $ln(N)$; $ln(N)/2$; $ln(N)*2$; $log_{10}(N)$; $log_{10}(N)/2$; $log_{10}(N)*2$.

Figure 5.5 depicts how the maximum amount of connections (per node) affects the overlay in terms of network diameter and average path length. Y denominates $ln(N)$ and X denominates $log_{10}(N)$ (note that these values have been ordered as to increase per step on the x-axis). As the max parameter grows we effectively add more possible paths between nodes. This results in a decrease on the Average Path Length and network Diameter by having each client maintain (on average) a larger number of browser-to-browser connections.



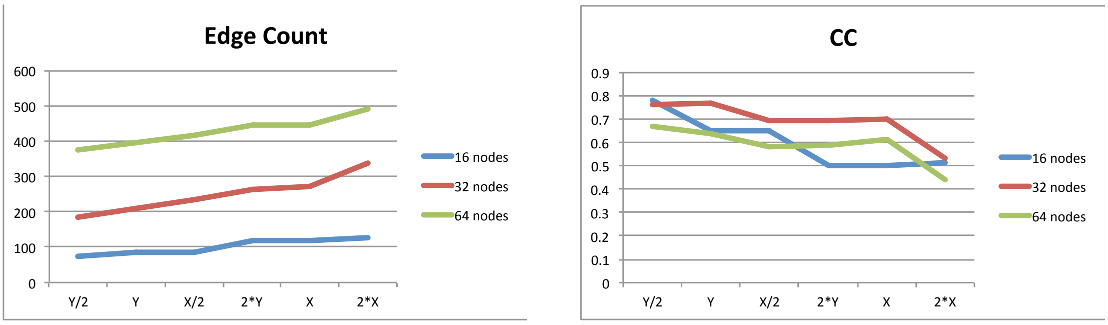
(a) Average Path Length

(b) Diameter

Figure 5.5: Average Path Length and Diameter

Figure 5.6 shows how the maximum amount of connections per node impacts the overlay in terms of total edge count and average clustering coefficient. The increase of the amount of edges compared to the decrease in clustering coefficient, makes us believe that the actual increase in connections reduces clustering (up to some extent: if we drastically increase the amount of connections to a fully connected graph we naturally increase clustering). In other words, new connections to other nodes are well distributed (non-biased) over the network.

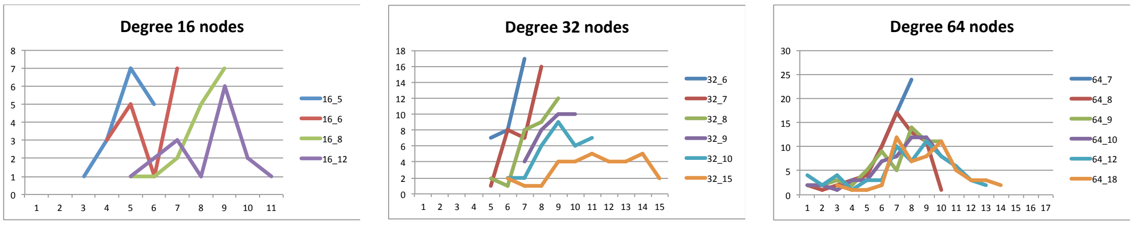
Figure 5.7 shows how the maximum amount of connections per node influences the overlay in terms of degree at each node (we remind the reader that connections among clients are symmetric). Each line N_M depicts the results of a graph with N nodes with the max parameter set to M. An increase in maximum degree distributes the degree of all participating nodes over the degree space. To ensure good load balancing properties of a system it is relevant that all nodes have a similar number of edges, otherwise nodes with



(a) Edge Count

(b) Clustering Coefficient

Figure 5.6: Edge Count and Clustering Coefficient



(a) Degree on 16 nodes

(b) Degree on 32 nodes

(c) Degree on 64 nodes

Figure 5.7: Connection degree

more connections have to, potentially, process and forward a larger number of messages. Results show that a smaller interval between the parameters keeps the degree distribution compact, improving load balancing.

5.1.4 Support for disconnection and other execution environments

Support for disconnection: We conducted experiments where we disabled Internet access (i.e., access to the centralized component) during the experiment. This experiment revealed that clients that have direct connections among them are able to continually co-operate. When a connection can later be established with the centralized component, all operations (those that affect objects) are propagated to the server by bully nodes. These results validate this design aspect of out framework.

Performance on computers and handheld devices: We verified the ability of using our framework on a diverse set of devices, including portable laptops, handheld phones, and tablets. We verified the creation of connections between these devices, on local networks and on separate networks behind NAT (requiring STUN), allowing communication between devices without the necessity to contact the centralized component. We also verified the ability to manually enforce high identifiers for clients running on mobile devices, while also disabling STUN on those clients only. The result is that we are able to ensure that client devices that are limited in resources (e.g. power) can always rely on other, more powerful nodes, to connect to the browser-to-browser infrastructure, by having those nodes act as their bullies.

5.2 Case Study: Google Drive Realtime API

In this section we provide a comparison on creating web-applications using the common client-server communication model and our proposed communication model (browser-to-browser). To this end we provide a comparative study between the use of the prototype of our framework and an existing industry solution for building a concrete application, that serves as a case study. We chose an existing collaborative editing framework as the baseline, the Drive Realtime API offered by Google.

In contrast to our proposed solution, the interaction between clients using the Google Drive Realtime API is always mediated by a central component, the server, which decides how operations are executed and propagated to other clients.

To evaluate our framework we compare the performance of an application designed to use our framework with the performance obtained of a similar application that relies on the Google Drive Realtime API. In this evaluation we want to be able to compare both approaches with respect to the following metrics: *i*) client-to-client latency observed on object updates; *ii*) the amount of data and messages exchanged over time between the server and the clients in both solutions (and between clients in our solution).

5.2.1 Design

The Google Realtime API provides a service for creating applications where users can collaborate on maintaining data objects. The Realtime API lets the programmer design applications that can use maps, lists, and string objects. The propagation of operations and resolution of conflicting updates is done automatically by the server. Changes are applied immediately to the, in-memory, local copy of the document. The API will handle the propagation of changes so that this change can be applied at the server and, eventually, at other collaborators. Conflicts are automatically resolved, so users never receive any errors on edit conflicts.

The Realtime data models are eventually consistent. The API guarantees that, if all collaborators stop editing, eventually all collaborators will see the same data model, without giving guarantees of a timely delivery, neither on the order of delivery of changes.

The Realtime objects internally use OT, Operation Transformation. Operation Transformation based systems typically adopt a replicated server architecture to ensure good responsiveness. The documents are replicated at each collaborating site, so that editions can be immediately performed locally and then be propagated to remote sites. Remote operations arriving at a site are then transformed before being executed. This transformation ensures that consistency is maintained across all replicas.

5.2.2 Implementation

To compare the systems we create a simplistic application using both frameworks, striving to maintain these implementatons as close as possible (to ensure that they are comparable). We say simplistic application as we minimize the HTML code and HTML affecting JavaScript code. In other words, we create methods to modify the objects and to listen for changes on the objects, but we never present to the interface the state or single updates received. This ensures that the evaluation of both systems will be independent of interface related effects.

For our evaluation we wrote code which can be divided into two parts: the listener and the writer methods. The listener (Listing 5.1) can be added to an initialized object to show the exact time a client receives an update. The writer (Listing 5.2) can be run on an initialized object to apply writes to an object. The writer can be initialized with three arguments: a wait time before the experiment begins, an interval between updates, and a total amount of updates. The *write_[gapi|b2b]* writes a single character at a random position on the selected system.

```
var got = 0;
function listener() {
  var date = new Date();
  console.log(++got + "-" + date.getTime());
}
```

Listing 5.1: "Listener"

```
var TIME_TILL_START, TIME_BETWEEN_SEND, AMOUNT;
setTimeout(write, TIME_TILL_START);

function write() {
  if (AMOUNT-- > 0) {
    write_[gapi|b2b](random(0, Selected.size());
    setTimeout(write, TIME_BETWEEN_SEND);
  }
}
```

Listing 5.2: "Writer"

5.2.3 Experimental Setup

All reported experiments using our case study were run on Amazon Web Services EC2, using *m3.xlarge* machine instances.

In all tests we setup our single server at us-west-1 (California) and we divide the clients equally over 16 machines, 8 at us-west-2 (Oregon) and 8 at us-east-1 (Virginia). This way we divide the clients in two equal sized groups (whose size varies in our experiments), and we allow only for direct communication between clients deployed on the same local network (i.e., same region). This restriction does not exist in our framework.

In all experiments we use the server in California to serve the static web pages and also to execute the centralized component of our framework. Clients are executed by loading a web page into Chromium, the Google Chrome open-source code. We execute the program by registering all activities to a virtual frame-buffer server, Xvfb, in memory. This allows us to run an as close as possible to a real world scenario, while keeping graphical related overheads low, and enabling us to control the experiments.

5.2.4 Latency

In this test we use a single client executing write operations (located in Virginia) and we measure the time till writes are propagated to other clients. The client that executes write operations, writes a single character on a *List* at each of its operations.

We present the latency results between clients on the same local network (i.e., those that use direct browser-to-browser communication) and clients on seperate networks (i.e., operations have to be propagated trough a centralized component). To facilitate the comprehension of these results we show the latency values between the machines used in this test, measured with the `ping` tool: i) between Oregon and the server at California: 20 ms; ii) between Virginia and the server at California: 80 ms; iii) between Oregon and the server of Google: 13 ms; iv) between Virginia and the server of Google: 12 ms; and v) between Oregon and Virginia: 70 ms.

Note that, to ensure that we measure correct latency results, we use the NTP tool that executes the network time protocol. As all results have been obtained (as an average) trough multiple runs on the same configuration, we expect that any discrepancies produced by NTP are minimized.

Figure 5.8 shows average latencies. The results show that using direct connections between clients results in a latency which is, as expected, substantially lower than when communication is mediated through a central component. The Realtime API shows no significant variation in clients being or not geographically close, which is expected as all client interactions are mediated by the central component. Our solution, due to the location of the server and lack of optimizations on the server component, presents higher latency values when operations are mediated through the central component.

Figures 5.9 (a) and 5.9 (b) present the latency values for, respectively, the 95th percentile and the maximum observed value. These results corroborate the previous observations and furthermore, show that in the case of communication mediated trough a central component, latency values increase substantially, especially in the worst case scenario. Also, these results show that latency observed by clients when they use our framework isn't significantly affected by the increase in the number of clients. The results show equally that the system based on the Google Realtime API is very sensitive to the number of clients, as the latency values rise substantially with the number of clients.

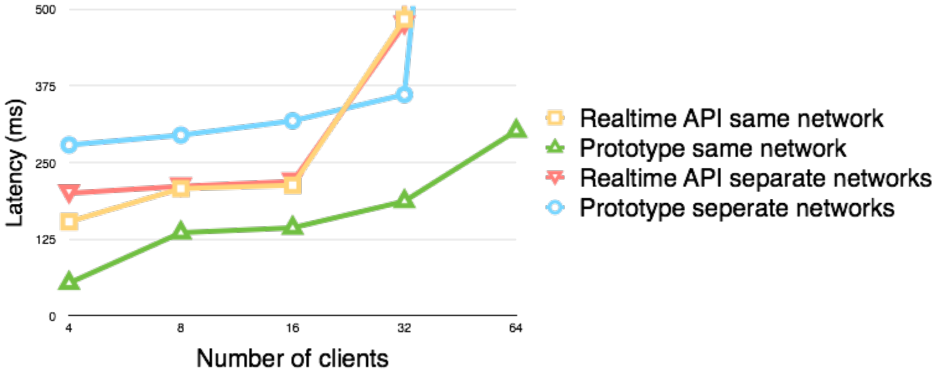
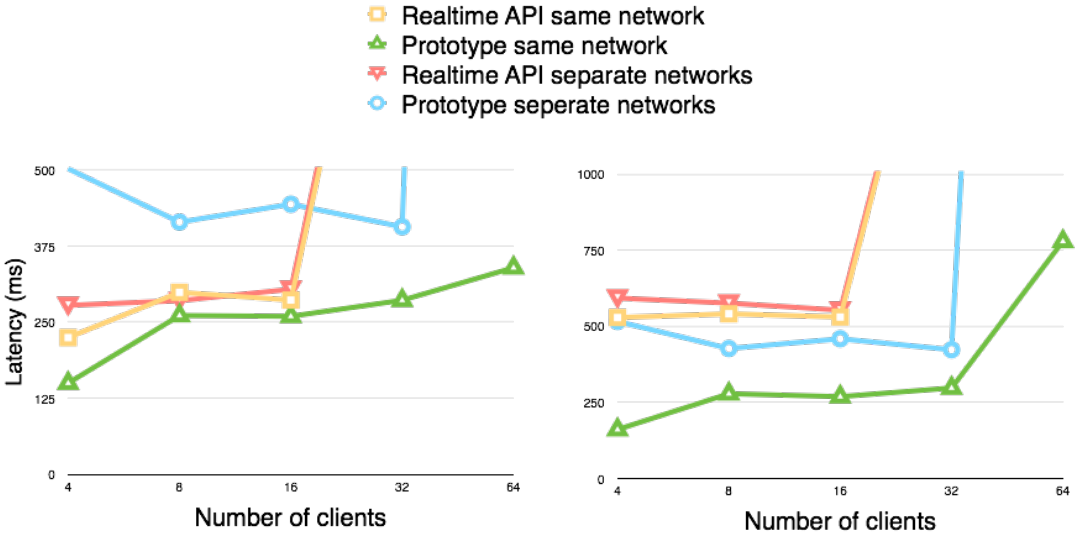


Figure 5.8: Average Latency



(a) 95th percentile

(b) Maximum

Figure 5.9: Latency: 95th percentile and maximum

5.2.5 Bandwith

In this section we present results concerning the amount of data exchanged over time (network bandwidth usage) between the server and the clients in both solutions and between clients in our solution.

In this experiment, two write operations are executed every second by each client node, during a short interval of time, again varying the number of clients. In each write operation, a single character is added to a *List*. We compare the usage of the Realtime API with our prototype, separating the usage of CRDTs based on the propagation of state and those that propagate operations. As parameters for the aggregation step of object updates (as detailed in Section 4.3.2), we vary between zero for both clients and server and 200 and 2,000 milliseconds, respectively for propagation to clients and to the server.

The total bandwidth measured on the server is the aggregation of all bandwidth observed at client nodes to the server, using the *iptraf* tool.

Figure 5.10 shows the average bandwidth usage per second on the server. The presented results are as expected. State based propagation incurs to a high load when the state of objects rises. Operation based propagation mitigates this problem as only the operations are sent over the network. The results from our prototype comparing to the Realtime API's results are very promising, and, in fact, we show we are able to obtain a smaller total load on the centralized component.

The reader should note that these results show the accumulated traffic over the total time of the experiment, which includes transmission of the necessary data to load HTML and JavaScript code from the server, as well as establishing connections between clients (Signalling). These values are though amortized over time and longer execution hides this initial overhead.

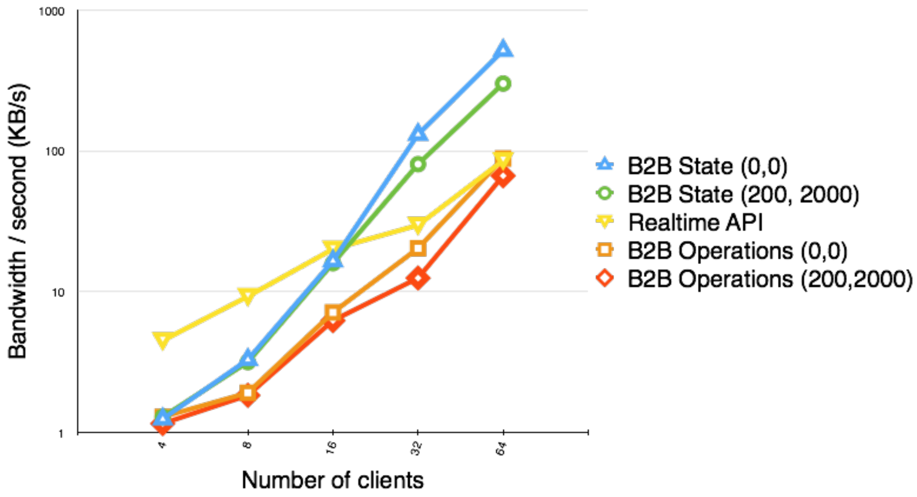


Figure 5.10: Server bandwidth usage

Figure 5.11 shows the average of traffic per second on each client. We do not show a comparison to the usage of the Realtime API as it does not support direct connections between clients. As expected, as the number of clients rises, so does the average bandwidth usage per client. This is due to both the facts that more messages are sent between clients to maintain the overlay network and to create new connections, and also to propagate operations. We believe we could further imrpove these results by leveraging more sophisticated group-based dissemination strategies, such as the ones found in *Plumtree[24]* and *Thicket[13]* protocols.

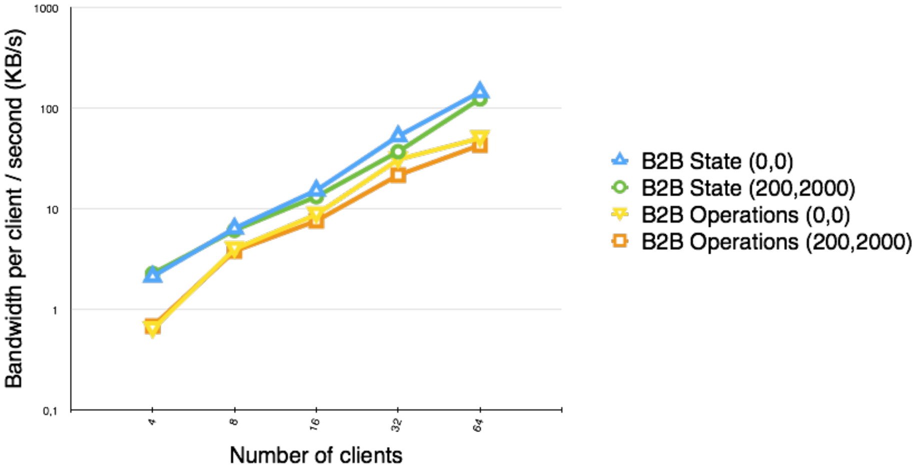


Figure 5.11: Client bandwidth usage

Chapter 6

Conclusion

Existing web applications, from collaborative editing, social networks, to multi-user games are fundamentally centered on users and the interaction between them. Even those applications that run partially or totally on the client machine resort to a model where interactions between clients are mediated by a centralized component. This central component, besides being a contention point on which all interaction between clients depends, may also introduce latency penalties on those interactions.

In this thesis we proposed an alternative architecture to support web applications with high user interaction by adding direct and transparent browser-to-browser communication. This approach has the benefits of improving client-to-client latency, lowering bandwidth usage on the server (as clients can coordinate to aggregate updates), and support disconnected operation from the centralized component as clients can directly exchange information and operate without resorting to the centralized component.

We designed and implemented a prototype of a framework to create web applications that follows the proposed model, using WebRTC to materialize direct communication between browsers, and resorting to CRDTs to materialize local object replicas on clients. Operations can be propagated directly between them, locally updating and reconciling with other clients the state of these shared and replicated objects. Persistence is achieved as propagation is eventually done to the centralized component, which is also leveraged to support clients that operate on non WebRTC compliant browsers. To demonstrate the benefits of the proposed solution, we implemented a system that exposes an API similar to the one offered by the Google Drive Realtime.

To evaluate the performance of our implementation we developed micro and macro benchmarks. The micro-benchmarks evaluate specific parts of the system, namely a comparison between implemented overlay networks and CRDTs, which validate our proposal. The macro-benchmark was created to compare the performance of our prototype to an existing industry solution, in particular the Drive Realtime API offered by Google. Our results show that we are able to improve on the total server load and on latency between clients when moving from the traditional client-server communication model to a browser-to-browser communication model. Our approach can also support client operation even in periods where the centralized infrastructure is not available.

In summary, the main contributions of the work presented in this thesis are as follows:

- A framework for creating web applications supporting direct browser-to-browser communication, without the need to install any kind of software or browser plugins.
- The design and implementation of a mechanism to replicate a set of objects in web applications, combining peer-to-peer interactions and a centralized component.
- A CRDT library that can be used with the replication mechanism for maintaining the state of different applications, with guarantees of covergence.
- An evaluation of the proposed system and comparison with an existing industry solution.

Publications

Part of the results in this thesis were published in the following publication:

Enriquecimento de plataformas web colaborativas com comunicação browser-a-browser Albert Linde, João Leitão e Nuno Preguiça. Actas do sétimo Simpósio de Informática, Covilhã, Portugal, September, 2015

6.1 Future Work

In the course of this work, a number of directions for future improvements have been identified.

The results of our overlay network implementations (presented in Section 5.1.3) meet our initial requirements but have some limitations. In future work we want to explore alternative overlay networks. One example is to modify the policy of connection between clients that are geographically close to each other by giving preference to close clients while far-away nodes would have fewer connections, if any.

The current library of CRDTs doesn't provide a realistic solution to all types of applications. We intend to improve the support for more complex data-types that better support the creation of different types of applications, as an example, CRDTs that can guarantee application invariants.

Having clients directly and transparently interacting and exchanging information among them creates significant challenges from the point of view of security, and in particular from the perspective of user privacy and data integrity. We aim to explore fundamental security mechanisms to mitigate attacks that can originate from potential misbehaving clients.

It would be interesting to integrate persistence with legacy systems, allowing users running new and old applications to interact. In other words, an existing application which uses the client-server communication model should be able to run alongside with clients that use applications that leverage our proposed framework. This would also have the benefit of allowing to see our systems as an extension to these legacy systems.

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Appendix A

Appendix 1

Listing A.1 has an observe remove set as specified in [36].

```
CRDT_ORSet.compare = function (v1, v2) {
    var first = false, second = false;
    //Compare tombstones:
    var v1_minus_v2Tombstones = v1.tombstones.filter(v2.tombstones);
    if (v1_minus_v2Tombstones.length > 0)
        first = true;

    var v2_minus_v1Tombstones = v2.tombstones.filter(v1.tombstones);
    if (v2_minus_v1Tombstones.length > 0)
        second = true;

    if (first && second) return null;
    //Compare elements:
    var keys1 = Object.keys(v1.elements);
    var keys2 = Object.keys(v2.elements);
    if (!second)if (keys1.length > keys2.length) {
        first = true;
    }
    if (!first)if (keys2.length > keys1.length) {
        second = true;
    }
    if (first && second) return null;
    if (!first) {
        for (var keyID1 = 0; keyID1 < keys1.length; keyID1++) {
            var key = keys1[keyID1];
            if (!v2.elements[key]) {
                var ts = v1.elements[key].filter(v2.tombstones);
                if (ts.length > 0) {
                    first = true;
                    break;
                }
            } else {
                var v1_times_minus_v2_times = v1.elements[key]
                      .filter(v2.elements[key]);
                if (v1_times_minus_v2_times.length > 0) {
                    first = true;
                    break;
                }
            }
        }
    }
    if (!second) {
        for (var keyID = 0; keyID < keys2.length; keyID++) {
            var key2 = keys2[keyID];
            if (!v1.elements[key2]) {
                var ts2 = v2.elements[key2].filter(v1.tombstones);
                if (ts2.length > 0) {
                    second = true;
                    break;
                }
            } else {

                var v2_times_minus_v1_times = v2.elements[key2]
                      .filter(v1.elements[key2]);
                if (v2_times_minus_v1_times.length > 0) {
                    second = true;
                    break;
                }
            }
        }
    }
    if (first && !second) return -1;
    if (!first && second) return 1;
    if (!first && !second) return 0;
    return null;
};

CRDT_ORSet.merge = function (v1, v2) {
    var newState = v1;
    v1.tombstones = v1.tombstones
        .filter(v2.tombstones)
          .concat(v2.tombstones);

    var keys2 = Object.keys(v2.elements);
    for (var i = 0; i < keys2.length; i++) {
        var key2 = keys2[i];
        if (!v1.elements[key2]) {
            v1.elements[key2] = [];
        }
        v1.elements[key2] = v1.elements[key2]
```

```
            .filter(v2.elements[key2])
              .concat(v2.elements[key2]);
    }

    var keys1 = Object.keys(v1.elements);
    for (var j = 0; j < keys1.length; j++) {
        var key = keys1[j];

        v1.elements[key] = v1.elements[key].filter(v1.tombstones);
        if (v1.elements[key].length == 0) {
            delete v1.elements[key];
        }
    }
    return newState;
};

CRDT_ORSet.fromJSONString = function (string) {
    var ret = [];
    ret.tombstones = string[0];
    ret.elements = [];
    for (var i = 1; i < string.length; i += 2) {
        ret.elements[string[i]] = string[i + 1];
    }
    return ret;
};

CRDT_ORSet.toJSONString = function () {
    var ret = [];
    ret.push(this.value.tombstones);
    var keys = Object.keys(this.value.elements);
    for (var i = 0; i < keys.length; i++) {
        var key = keys[i];
        ret.push(key);
        ret.push(this.value.elements[key]);
    }
    return ret;
};
```

Listing A.1: JavaScript OR-Set Implementation