Raft Distributed Consensus Algorithm over WebRTC (Web Real-Time Communication)

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Abstract

We describe the creation of a browser-based distributed peer-to-peer application using WebRTC, PeerJS and Raft.js. WebRTC is a protocol that modern browsers have begun to integrate which enables browser-to-browser communication. PeerJS is a WebRTC client library and extensible signaling server. Raft.js is an implementation in JavaScript of the Raft distributed consensus protocol. These technologies are combined to create a chat application that guarantees a shared, consistent and linearized view of the chat history without relying on a single centralized server. We also create a series of automated tests using the SlimerJS browser automation tool and use them to show the correctness of the chat application and also to measure the performance and scalability of the application over various cluster sizes.

1 Introduction

In the past few years, browsers have quickly evolved from document rendering engines into powerful platforms for building sophisticated applications. Modern browsers are integrating new protocols for communication including WebSockets for full-duplex browser-to-server communication and WebRTC for direct browser-to-browser communication. These new capabilities enable software developers to create distributed network applications with browser applications as first class citizens.

These new browser capabilities also bring with them the traditional challenges of building reliable distributed applications such as: concurrency, security, scalability, naming, discovery, replication, relocation, etc. The solutions to many of these challenges involve the proper management of distributed state. Distributed consensus is one category of distributed state management in which multiple nodes must come to agreement on the state of single value or set of values. The grandparent of distributed consensus algorithms is the Paxos algorithm described by Leslie Lamport. However, Paxos has been notoriously difficult to understand and to implement correctly. Raft is a more recent distributed consensus algorithm that aims for understandability and practicality of implementation.

In this study we will explore an implementation of the Raft protocol in a browser context using WebRTC as the communication channel. We will describe the results of the implementation and look at some of unique challenges (and opportunities) we encountered with the implementation. We will also explore some ideas for further study.

2 Motivation/Use Cases

The traditional use of distributed consensus is in the data-center (private or cloud based) especially as the core coordination component of a replicated data-storage system. In these examples, the set of nodes that compose the cluster tend to be fairly static and changes to cluster membership are usually for the purpose of replacing nodes. Changes to the long term size of the cluster are uncommon.

Most existing distributed browser applications are extensions of this model in which browsers connect to load-balanced web servers which are in close network proximity to the distributed consensus cluster. The backend web system delivers the web application to the browser and also serves as the conduit for all browser-to-browser communication and state management.

However, it is not always desirable for application messages and state to traverse the server back-end for various reasons including the following:

1. *security (privacy)*: application state and user messages may need to traverse directly from browser to browser rather than traversing through a server.

2. *locality*: the nature of the application may allow clusters of applications that are nearby on a network (e.g. intranet) to communicate and coordinate directly rather than passing all messages and state through a server back-end which may be considerable more remote.

3. *bandwidth and scale*: a distributed browser based application may be able to avoid expensive bandwidth into and out of the backend data-center by passing messages and state directly between browsers. Depending on the design of the application this may also enable better scalability then a more centralized model.

Here are some specific use cases in which Raft over WebRTC may be applicable:

1. *Consistent order secure chat*: a chat room system where all users see the same message order but messages never traverse through a central server even though the application code itself is delivered from a central web service. This application was implemented for this study.

2. *Private multiuser document editing/whiteboard system*: Google Docs among other systems has demonstrated a powerful model for collaborative document creation.

However, these systems all user central web services to coordinate shared state and handle browser-to-browser communication. Using a Raft over WebRTC could enable tools with similar functionality but without the central coordination and communication.

3. *Browser based network games*: most network games have certain states that must be agreed upon by all nodes (e.g. avatar dead or alive, etc). This can be achieved with Raft over WebRTC without requiring any data to traverse a central server. In addition to reducing browser to server bandwidth requirements, this may also allow lower latency gaming if games instances are partitioned by latency locality.

3 Background

3.1 Raft

Raft is a new distributed consensus algorithm designed to be understandable and practical without sacrificing safety and correctness.

The Raft algorithm breaks the problem of consensus into three core sub-problems: leader election, log replication, and safety. A full Raft solution must also address: cluster membership changes, log compaction, and client interaction. The Raft algorithm implemented and discussed in this paper is based on "Consensus: Bridging Theory and Practice" by Diego Ongaro (PhD Dissertation).

The Raft protocol is based on the concept of a distributed transaction log. Entries in the log represent a sequence of changes to an internal state machine. If all members of the Raft cluster have the same log entries, then their state machine on each node will have exactly the same state. Each node of a Raft cluster can be in one of three states: follower, candidate, or leader. The responsibility of the leader is to accept new transaction log entries and then replicated these entries in the same order to all other members of the cluster. The leader is the only member of the cluster that may make changes to the transaction log. In order to maintain leadership, the leader sends heartbeat RPCs to all the followers in the cluster. Followers that do not receive a heartbeat RPC within a certain time period become candidates and attempt to be elected by the other nodes as the new leader of the cluster.

3.1.1 Leader Election

Each node of the Raft cluster maintains an ordinally increasing current term value. When a Raft follower node does not receive a heartbeat from a leader within a random election timeout period, it transitions to candidate state, votes for itself, increases its current term value by one and sends a requestVote RPC (with the new term) to all the other nodes in the cluster. When a node receives a requestVote RPC with a term that is greater than its own it will become a follower (if not already), update its term to the new term, and send a RPC response back that indicates a vote for that candidate. However, a node may only vote for one candidate in a given term. If a candidate receives votes from more than half of the cluster, then it immediately becomes a leader and sends out a appendEntries RPC to confirm leadership and reset the election timers on all the other nodes.

3.1.2 Log Replication

Each Raft node maintains a transaction log. The entries in the transaction log each contain a term value in addition to the action to apply to the state machine. Each node also keep track of the most recent log entry which is known to be committed (exists on more than half the nodes in the cluster).

When a new leader is elected, it begins sending appendEntries RPCs to other nodes in the Raft cluster. These RPC messages contain information about the state of its transaction log: index and term of the latest entry, and the most recently committed entry. If the node does not have an entry matching the most recent entry on the leader, then it replies with false to indicate that its log and not up to date. The next appendEntries RPC that the leader sends to that node will contain the next oldest index and term. This will continue until the leader discovers the latest log entry which is agreement (the node may not have any entries if it is new). Then the leader will begin to propagate entries to the node in subsequent appendEntries RPCs until the node is caught up. The leader continues sending empty appendEntries as heartbeat RPCs to all the nodes until it receives a new entry in its transaction log from a client.

3.1.3 Safety

In order to ensure that each state machine on every node executes exactly the same commands in exactly the same order, the Raft system provides safety guarantees. In particular, there are some restrictions on which Raft nodes are actually eligible to become a leader based on the state of their transaction log compared to other node transaction logs. A Raft node will only vote for a candidate if the candidate has a log that is more up-to-date than the voter. A log entry with a later term is always more up-to-date. If the log entries have the same term, then the entry with a higher index is more up-to-date. In addition, Raft leaders may not consider entries from a previous term to be committed until it has committed at least one an entry from its own term.

Raft nodes must also persist certain properties to durable storage before sending or receiving any RPCs. The properties that must be persisted are current term, current vote for this term (if any), and either the full transaction log or the state machine plus any unapplied transaction log entries.

3.1.4 Membership Changes

Raft leverages the replicated transaction log to accomplish live cluster membership changes. Membership changes are accomplished by adding or removing one Raft node at a time using special add/remove log entries. As soon as the new entry is added to the log it becomes effective without waiting for the entry to be fully committed across the cluster. However, a new cluster change entry may not be added to the log until the previous one is committed. Once the change entry is committed, the initiator of the change is notified, removed servers can be shut down and another change entry may be added to the log.

3.1.5 Log Compaction

In order to keep the replicated transaction log from growing indefinitely, Raft nodes should periodically compact their logs. There are many different ways to accomplish log compaction and the best way will depend on application requirements and on the specific nature of the distributed state machine. For example, if the state machine is simply a single shared counter, then any log entries before the most recently committed entry may be safely discarded. The simplest and most generic solution is often to provide some way of snapshotting the entire state machine (including the term and log index it represents) to disk at which point the all previously applied log entries can be discarded. Snapshotting the state machine also means that the implementation must provide a way for the leader to serialize and send the current state machine to other nodes if their transaction log is too old (e.g. when a new node is added).

3.1.6 Client Interaction

Clients of the Raft cluster are able to interact with the distributed state machine by sending RPCs to the leader in order to add command entries to the transaction log. Clients first find the address of any node in the cluster either via broadcast or via an external directory service. Once the client discovers a node of the cluster, that node can either forward messages directly to the leader node (if it is not the leader) or it can reject client requests with a response indicating the address of the leader for the client to redirect to.

The Raft protocol should also provide strict linearizable semantics for client commands/requests (reads and writes). In order to accomplish this, each client is given a unique ID, and the client assigns a unique serial number to each command. This prevents a single client command from being applied twice to the state machine (in case of a leader crash, or network duplication).

Linearizability is important not just for writes/updates to the state machine, but also for reads, which means that read requests must also be entered as transactions in the log and committed to more than half the Raft nodes before the leader can respond to the client. If reads bypass the transaction log then those reads are serializable but not necessarily linearizable. For example, a leader may have been deposed and not realize it if the cluster is partitioned which could result in reads of stale data (the leader of the larger partition may have already committed new entries).

3.2 Raft.js

Raft.js is an implementation of the Raft algorithm in JavaScript that was creator by the author of this paper. Raft.js is designed to run in either a browser environment or within node.js (server-side JavaScript). Raft.js implements the Raft algorithm as described in "Consensus", Ongaro (dissertation).

3.2.1 Modular Design

The implementation of the Raft algorithm is implemented in the RaftServerBase class (in base.js). The base class does not directly implement stateful functions such as RPC communication, durable storage, scheduling/timeouts, or state machine command/management. In order to create a working implementation these functions must be provided either as configuration parameters when the class is instantiated or by sub-classing the class.

The modular design of the Raft.js implementation allows it to be easily used in several different contexts. For example, the RaftServerLocal class (local.js) implements RPC calls using plain JavaScript function calls between different instances of the class (nodes) in the same execution context. The online Raft visualization at http://kanaka.github.io/raft.js/ uses the RaftServerLocal class and instantiates it using a scheduling functions that fire when the user clicks on the "Step" button rather than due to the passage of time.

The RaftServerHttp class ('http.js') embeds a simple web server into each Raft node and then uses normal HTTP requests to send RPC messages between Raft nodes.

For this paper, the RaftServerLocal class was extended with the capability of sending RPCs over the WebRTC Data Channel.

3.2.2 Differences between Raft and Raft.js

The current Raft.js implementation is based on "Consensus", Ongaro and implements most of the Raft algorithm including: leader election, log replication, safety, membership changes, basic client interaction, and read-only operation optimizations. Log compaction and full client linearizability are not currently implemented (clients and client transactions are not assigned unique IDs). Log compaction is important for long-running production systems in order to limit memory growth and allow new nodes to catch up their logs more efficiently. However log compaction is not required for verifying overall system correctness and does not significantly impact short-running test configurations. Full client linearizability is important feature for simplifying certain types of distributed applications, however the implementation requires some significant additions to the Raft implementation (e.g. assigning client IDs and sequence IDs and tracking timed-out clients). For the chat application, simply adding one atomic check-and-set state machine command is sufficient to achieve a linearized chat history. This does impose a slight additional cost on the client because it must retry the atomic operations until they succeed. The author intends to implement both log compaction and full client linearizability in the future but, however these do not have a significant impact on the ability to test Raft combined with WebRTC.

The Raft protocol description in "Consensus", Ongaro, has undergone some revisions since the original draft paper (including incorporation of suggestions by the author of this paper). These changes include cluster membership simplifications, separating the concepts of committing entries to the log and applying entries to the state machine, clearer client interaction, etc. As part of the work for implementing Raft over WebRTC, the Raft.js implementation was updated to reflect these changes to the Raft protocol definition.

The are also some other differences between Raft.js and the Ongaro paper:

1. The paper describes the Raft RPCs in terms of the traditional request/response model where requests and responses are automatically correlated together by a

lower level of the software stack. For example with HTTP 1.0 a separate network session is established for each request/response pair (HTTP 1.0). Other examples including using a unique per session ID to correlate request and response. WebRTC DataChannel connections are message based rather than request/response based. The original Raft definition of RPC and RPC response messages almost contain enough information to function correctly in an environment without builtin request/response correlation. Raft.js includes a small modification to the response RPCs for requestVote and appendEntries to add a 'sourceId' field.

2. The Raft paper is clear that each membership change must be delayed until all the previous membership change is committed (sequential). It is implied that the leader node should track all pending membership change requests and apply them one at a time. However, in Raft.js, if a client requests a membership change while another change request is pending (uncommitted), the leader will reject the change request with a status of 'PENDING_CONFIG_CHANGE'. This keeps the core Raft implementation simpler with the trade-off being that the client implementation of membership change requests is slightly more complex.

3.3 WebRTC

WebRTC is name of a collection of browser APIs and network protocols that are currently being standardized the W3C and IETF respectively. Collectively, these APIs and protocols work together to enable real-time peer-to-peer video, audio and data communication between browsers.

3.3.1 WebRTC Discovery and Signaling

One of the challenges with browser-to-browser communication (and direct peer-to-peer communication in general) is that the web browser environment or user agent (UA) is usually running behind a firewall in a private internet subnet. This complicates direct communication in several ways: the public Internet address of the user agent is unknown and dynamic, inbound connections are denied by default, and outbound connections are network address translated (NATed).

To establish a direct communication channel between browsers, WebRTC uses mechanism that is similar to the Session Initiation Protocol (SIP) commonly used for Voice-over-IP (VoIP) communication. This involves a third party server known as a signaling server. Each user agent (browser) connects to this server at a well known address in order to establish a direct connection to other user agents. The signaling server is also uses to communicate session control messages (open, close, error) and to negotiate media capabilities. The signaling server may optionally provide other services such as higher level session management, HTTP serving of the web application, address book service, client presence/availability, etc.

3.3.2 WebRTC APIs

The WebRTC browser APIs enable web applications written in JavaScript to establish browser-to-browser network connectivity in order to stream real-time video, audio and data communication. These APIs are currently in the process of being standardized by the W3C organization (http://www.w3.org/TR/webrtc/)

There are three primary WebRTC API definitions:

1. RTCPeerConnection: The RTCPeerConnection API represents a direct connection between two browsers. Each RTCPeerConnection object has an associated Interactive Connectivity Establishment (ICE) agent that is used to establish network connectivity through firewalls and NAT services.

2. MediaStream (aka getUserMedia: The MediaStream API represents a stream of audio or video data. MediaStreams can easily be sent and received over RTCPeerConnections. The MediaStream API definition provides a mechanism that allows JavaScript to enumerate the native MediaStreams that are provided by the browser such as a local web camera or a live video stream of the desktop (for desktop sharing). Although support for MediaStreams was the original impetus for the creation of the WebRTC APIs and protocols, the Raft over WebRTC implementation does not make use of the MediaStream API.

3. RTCDataChannel: The RTCDataChannel API provides a mechanism for sending and receiving generic data. A single RTCPeerConnection can contain multiple RTCDataChannels. The RTCDataChannel API is modeled after the WebSocket API, however, unlike the WebSocket API which is reliable and guaranteed order transport (like TCP), each RTCDataChannel can be configured with different reliability and ordering settings. For example, by relaxing the reliability and ordering constraints on a RTCDataChannel, the average latency and jitter per message can be decreased even though some messages may be dropped or arrive out of order. The Raft protocol is specifically designed to support dropped or out-of-order messages so the transport options available with RTCDataChannel can be used to gain extra efficiency.

3.3.3 WebRTC Protocol

There is a large suite of protocols that are required for a working WebRTC system. The Internet Engineering Task Force (IETF) organization is responsible for specifying and/or standardizing (where necessary) the protocols that are part of WebRTC. In many cases an existing protocol is applicable and is referenced or extended as part of the WebRTC suite.

Describing all the protocols that make up the WebRTC suite is beyond the scope of the this document. However, here is a (incomplete) list of WebRTC protocols drafts and related protocols that are part of WebRTC:

1. General:
   (a) WebRTC: Overview
   (b) WebRTC: Security Architecture
   (c) WebRTC: Security Considerations
2. Data transport:
   (a) WebRTC: Data transport overview

   - (b) WebRTC: DSCP and other packet markings for RTCWeb QoS
   - (c) DTLS Encapsulation of SCTP Packets
   - (d) Stream Control Transmission Protocol (SCTP)-Based Media Transport in the Session Description Protocol (SDP)
   - (e) Stream Schedulers and a New Data Chunk for the Stream Control Transmission Protocol
   - (f) WebRTC: Media Transport and Use of RTP
   - (g) WebRTC: Data Channels
   - (h) WebRTC: Data Channel Establishment Protocol
   - (i) WebRTC: Application Layer Protocol Negotiation for Web Real-Time Communications (WebRTC)
   - (j) RFC5245: Interactive Connectivity Establishment (ICE)
   - (k) RFC5128: Peer-to-Peer (P2P) Communication across Network Address Translators (NATs)
   - (l) RFC5389: Session Traversal Utilities for NAT (STUN)
   - (m) RFC3489: STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
   - (n) RFC5766: Traversal Using Relays around NAT (TURN)
   - (o) RFC6156: Traversal Using Relays around NAT (TURN) Extension for IPv6
   - (p) RFC6544: TCP Candidates with Interactive Connectivity Establishment (ICE)
   - (q) RFC4571: Framing Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP) Packets over Connection-Oriented Transport
   - (r) RFC5764: Datagram Transport Layer Security (DTLS) Extension to Establish Keys for the Secure Real-time Transport Protocol (SRTP)

3. Data framing and security
   - (a) RFC3550: Real-time Transport Protocol (RTP)
   - (b) RFC3711: Secure Real-time Transport Protocol (SRTP)
   - (c) WebRTC: Media Transport and Use of RTP
   - (d) WebRTC: Data Channels
   - (e) WebRTC: Data Channel Establishment Protocol

4. Data formats (audio)
   - (a) WebRTC: WebRTC Codec and Media Processing Requirements
   - (b) WebRTC: Audio Codec and Processing Requirements
   - (c) RFC6716: Opus Audio Codec
   - (d) RTP Payload Format for Opus Speech and Audio Codec
   - (e) RFC3551: RTP Profile for Audio and Video Conference
   - (f) RFC4733: RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
   - (g) WebRTC: Additional WebRTC audio codecs for interoperability with legacy networks
   - (h) WebRTC: Video Processing and Codec Requirements
   - (i) RFC6386: VP8 Data Format and Decoding Guide
   - (j) ITU H.264: Advanced video coding for generic audiovisual services
   - (k) RTP Payload Format for VP8 Video
   - (l) RFC6236: Negotiation of Generic Image Attributes in the Session Description Protocol (SDP)

5. Connection management
   - (a) RFC3264: An Offer/Answer Model with the Session Description Protocol (SDP)
   - (b) RFC2327: SDP: Session Description Protocol
   - (c) WebRTC: Javascript Session Establishment Protocol (JSEP)
   - (d) RFC5245: Interactive Connectivity Establishment (ICE)
   - (e) RFC5763: Trickle ICE: Incremental Provisioning of Candidates for the Interactive Connectivity Establishment (ICE) Protocol

Note: the signaling transport itself is not defined as part of WebRTC. This is up to the application designer.

3.4 PeerJS

PeerJS is a project that provides a simplified abstraction for using WebRTC. The first component of PeerJS is a node.js server library that implements a WebRTC signaling server. The second component is a JavaScript library that interacts with that signaling server and also provides a simpler interface for using the WebRTC APIs.

The JavaScript interface use for establishing the initial connection (signaling) in the WebRTC 1.0 standard is somewhat complicated. A competing standard called Object Real-time Communication (ORTC) is working to define a simpler and more modular JavaScript API for WebRTC signaling. However, at the time of this work, the ORTC interface was not yet widely supported and still in a state of flux. The use of PeerJS provides a suitable abstraction in the meantime (and will likely be modified to support ORTC).

3.4.1 PeerJS Server

The PeerJS project provides a simple WebRTC signaling server called PeerServer. This server uses HTTP and WebSockets transport protocols to perform WebRTC signaling on behalf of browser clients. The PeerServer server is also extensible so that more advanced services can be built with it. The PeerJS project includes an extension to PeerServer called ExpressPeerServer that combines a PeerJS signaling server with the Node.js based Express web application framework.

3.4.2 PeerJS Client Library

The PeerJS client library provides an simple abstraction over the native browser WebRTC APIs. The WebRTC API standards are not yet finalized and so different browsers and browser releases may support different versions of the WebRTC draft APIs. The PeerJS client library abstracts over these differences and provides a common interface so that the application developer does not have to deal directly with browser differences.

The PeerJS client library also provides a default signaling mechanism that is designed to operate with a PeerJS signaling server. This is particularly helpful because much of the WebRTC signaling transport and protocol is not defined as part of the WebRTC standardization effort. The PeerJS organization provides a cloud-hosted version of a PeerJS compatible signaling server, however an application developer can also run their own PeerJS signaling service using the PeerServer server described above.

4 Design and Implementation

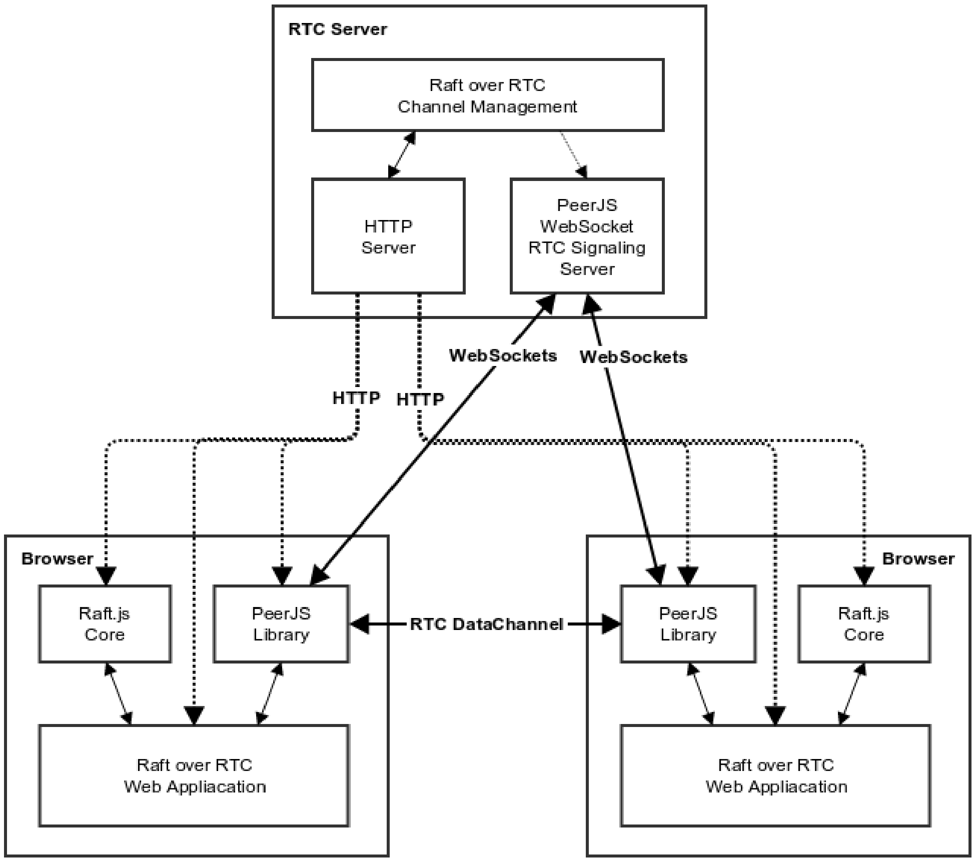


Figure 1: Raft over RTC Architecture

4.1 Incremental Steps

During the planning phase, the project was broken down into ten incremental steps:

1. Simple WebRTC messaging: implement a small JavaScript application that uses the PeerJS client library and an unmodified PeerJS signaling server. Demonstrate communication over WebRTC DataChannel using two browser instances (or browser tabs).

2. Raft.js over WebRTC: combine the existing Raft.js implementation with the small JavaScript application above to demonstrate Raft over WebRTC. This step uses a statically configured three node cluster (three browser tabs). The PeerJS signaling server is polled to retrieve the list of other clients and the Raft cluster is started when the number of PeerJS clients reaches three.

3. Update Raft.js protocol: update the Raft.js implementation to match the latest description of the Raft protocol (as described in "Consensus", Ongaro dissertation). This includes the implementation of the simplified membership change algorithm (sequential changes) and the separation of the log commit and state machine apply concepts. Additionally, the new

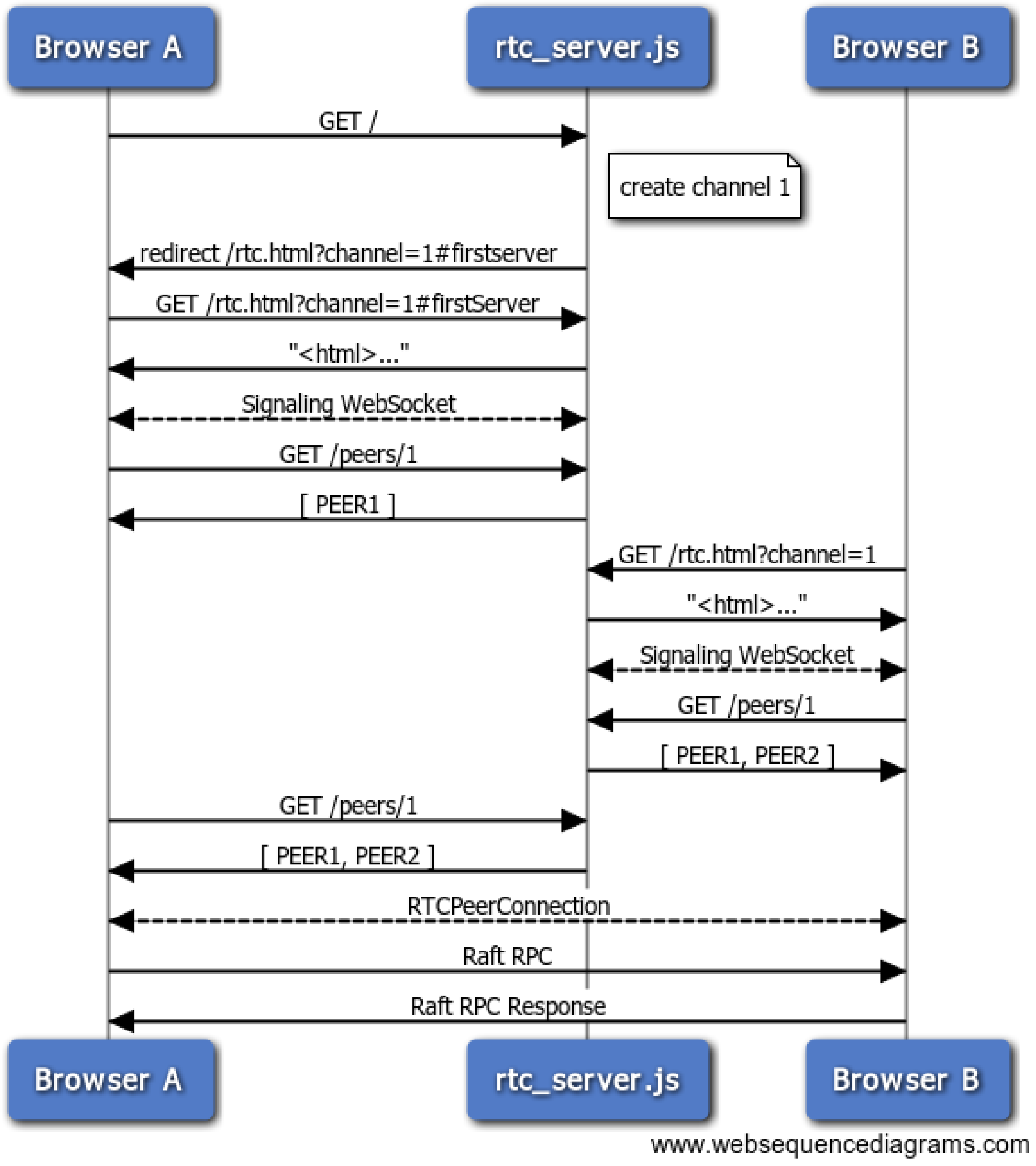


Figure 2: Raft over RTC Sequence Diagram

Raft.js implementation always starts a new Raft cluster as a cluster with a single node and additional nodes are dynamically added to the cluster (as suggested in section 4.4 of the paper).

4. Dynamic Raft.js over WebRTC: enhance the signaling server with support for multiple sessions/channels to enable separate Raft clusters with the same signaling server. Start each cluster with a single node and automatically add and remove Raft nodes as they come and go.

5. Visual representation: display Raft node state and RPC statistics in addition to the internal Raft log messages.

6. Sessionless/Datagram RPC messages: change Raft RPC requests and response messages so that RPC responses have enough context to be handled separately from RPC requests. Specifically this enables Raft to support datagram (sessionless) network transports like WebRTC.

7. Atomic state machine operations: implement a state machine update operation that only succeeds if no other operation has been applied to the same key since the last time this client read from that key.

8. Automatic client request forwarding: extend the client request RPC to also be sessionless/datagram based. Implement an asynchronous JavaScript function that determines which node is the leader and forwards client request RPCs to the leader. This function must also correlate responses to the client request and invoke the callback for the original request.

9. Chat application: add the ability to send a message to the chat log using the client request forwarding func-

tionality together with atomic state machine operation. Show the current chat history based on the chat entries in the state machine.

10. Server as Raft.js peer (aspirational): enable server nodes (Raft.js running on Node.js) to fully participate as Raft cluster nodes.

Steps 1-9 were successfully implemented for this paper. Step 10 was an aspirational goal because this functionality is not yet supported in PeerJS although it is being tracked by issue report: https://github.com/peers/peerjs/issues/103

4.2 Server

The file 'rtc_server.js' implements the signaling server for the Raft over WebRTC application. The server extends the PeerJS 'ExpressPeerServer' object in order to provide WebRTC signaling. In addition to the signaling function, the server also provides the following service endpoints:

1. Static web service endpoint: this serves the static HTML, CSS and JavaScript that make up the Raft over WebRTC web application.

2. New channel endpoint: when a browser loads this endpoint, a new PeerJS channel is created and the browser is redirected to a URL for loading the main application (via static file service above). The redirect URL has a query string appended that contains the channel ID. In addition the redirect URL has a fragment identifier (hash/history) that communicates to the web application that this is the first server in the channel.

3. Peer list endpoint: this endpoint takes a channel ID and returns the list of WebRTC peers (browser user agents) that are currently registered in that channel.

   Refer to the "4.3 Client" section for a description of how these endpoints interact.

4.3 Client

The Raft over WebRTC testing application is designed as a single web page ('rtc.html'). The application has different starting behavior depending on the value of the URL query string (the URL component following the "?" and up to the "#") and the fragment identifier (the URL component after the "#").

When a new Raft cluster is created, the browser (user agent) connects to the server new channel endpoint. This endpoint causes the server to create a new channel for PeerJS signaling. Once this is complete the server responds to the browser with a redirect message that contains the new channel identifier in the query parameter and an indication that this is the very first Raft cluster node in the fragment identifier.

The browser follows the redirect and loads the static web application at the static file endpoint provided by the server. This includes the main application page ('rtc.html'), the core raft algorithm ('base.js' and 'local.js'), the PeerJS client library ('peer.js'), the actual Raft over WebRTC implementation ('rtc.js'), and a layout stylesheet ('web/demo.css').

If the URL fragment identifier is set to "firstServer", then this Raft node is initialized as the leader of a single node cluster and all normal timers are started. The fragment identifier is then unset. The query parameter contains the ID of this cluster (the PeerJS channel).

Subsequent nodes are added to the same Raft cluster by starting a new browser context (separate browser, browser window, or browser tab) and loading the same URL (which no longer has the "firstServer" fragment identifier). There is a convenience link provided in the application page that opens a new window with a new cluster node.

When the page first loads, a PeerJS signaling connection is established with the server. Once this connection is established, the application will be notified when any PeerJS clients connect to or disconnect from the server. After the PeerJS signaling connection is established, the client immediately queries the Peer list endpoint to discover any clients that were connected prior to when this node connected to the server.

The web application also begins a periodic async polling function ('addRemoveServersAsync') when the page loads. When the node is a Raft leader, the function compares the current active list of PeerJS clients to the Raft cluster node list. If there are any differences between the two lists, then any new clients are added to the Raft cluster and missing clients are removed from the Raft cluster one at a time using the 'addServer' and 'removeServer' RPC calls. Servers additions are prioritized before removals (as recommended in Ongaro 4.4) in order to maximize availability.

When an instance of the web application starts that is not a "firstServer", the normal Raft timers are not automatically enabled until the server receives an RPC. The prevents the server from possibly becoming a leader before it is incorporated into the cluster by an appendEntries RPC from the "true" leader.

In addition to the low level Raft message log, the following statistics are shown in the application:

1. *Term*: the current Raft leadership term.

2. *State*: the current Raft node state (follower, candidate, or leader)

3. *Cluster Size*: a count of the current Raft cluster membership (number of nodes currently participating in the cluster)

4. *Log Length*: number of entries in the transaction log.

5. *Request Votes Sent/Received*: the number of 'requestVote' RPCs that have been sent from and received by this node.

6. *Append Entries Sent/Received*: the number of 'appendEntries' RPCs that have been sent from and received by this node.

4.3.1 Client Request Redirect Implementation

In the Raft over WebRTC implementation each local browser is a client of the Raft cluster. However in Raft, the leader node is the only node that adds entries to the log so clients on follower nodes must have a way to communicate with the leader. This is implemented as pair of functions named clientRequest and clientRequestResponse. clientRequest is called by the client with the arguments of a client request RPC and a callback function to be called once the request is completed. The clientRequest/clientRequestResponse functions work as follows:

1. the clientRequestResponse is registered with the local Raft.js implementation as the callback function anytime clientRequestResponse RPCs are received.

2. the clientRequest function keeps track of the current leader ID. If the current leader ID is not set or this node is the leader then the client's request is sent directly to the local Raft node. If the current leader ID is set then the request is sent to that node as a normal RPC request over the WebRTC channel.

3. when the clientRequestResponse function is called and the status is "NOT_LEADER" then the current leader ID being tracked by the clientRequest function is set to the leader ID returned in the response. Then the original request is re-issued over the normal WebRTC channel to the leader.

4. when the clientRequestResponse function is called with a successful status, the original callback argument of the clientRequest function is called with the results from the RPC response.

4.4 Chat Application

The chat application extends the basic Raft.js over WebRTC client. The chat application provides a large text area containing the chat history and a text entry field below it. When the user clicks on the *Send* button (or pressed enter), the current node ID is prepended to the beginning of the text line and it is added to a pending send queue and the text entry field is blanked. A function to flush sends runs periodically (currently every 100ms) and does the following:

1. If there is line of text that has been sent but not yet added (applied) to the state machine or there are no lines of text on the pending send queue, then the function does nothing.

2. A text line is removed from the head of the pending send queue and stored in *curSend* to indicate that a send is in progress. The a client request is made using an atomic *seqAppend* command to add the new text to the state machine. The *seqAppend* operation attempts to append the text line to a list at the *history* key in the state machine.

3. When the atomic *seqAppend* operation completes, it may have failed (if another text line was appended first), in which case the flush function puts the failed text line back on the pending send queue. Whether the send was successful or failed, the *curSend* variable is set to nil/null to indicate that no send is in progress (the next time the flush function runs it will attempt to send again).

The chat application also extends the default set of state machine operations to add a seqAppend operation. This is an atomic operation that is implemented as follows:

1. the seqAppend operation operates on a specially structured key/value pair in the state machine map. The value is a associative map that contains a counter and list structure that contains the actual data. seqAppend operations are sent with both a count and then new value to append.

2. When a seqAppend operation is applied to the state machine and the count value in the command matches the counter in the state machines value, then the new value in the request is added to the list and the counter is incremented. If the count value does not match then counter then the the operation is not applied. In either case, a list is returned that contains two values: whether the seqAppend operation succeeded and what the current value of the counter.

3. the first time the seqAppend operation is used on a non-existent key, the structure described above is created with the counter set to 0 and the list structure is created with the single value passed in the seqAppend operation.

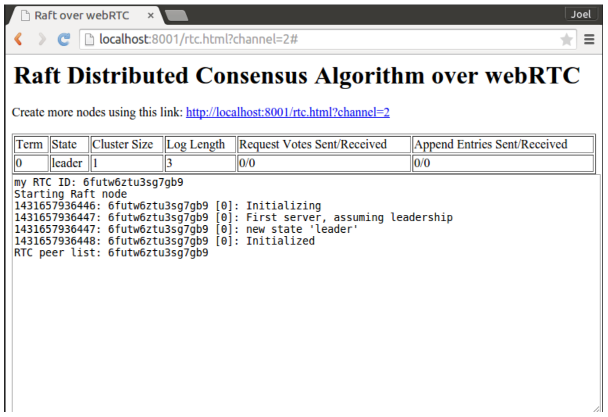
4. clients that call the seqAppend operation should check the status value in the result list. If the status is false, then the client should retry the operation using the counter value as the count in the operation.

5 Results

5.1 Manual Testing

5.1.1 Setup

The initial way that the application was tested was by loading the initial landing page to create a Raft over WebRTC cluster with a single leader. Here is a screenshot of the browser after the first node is created:



Raft Distributed Consensus Algorithm over webRTC

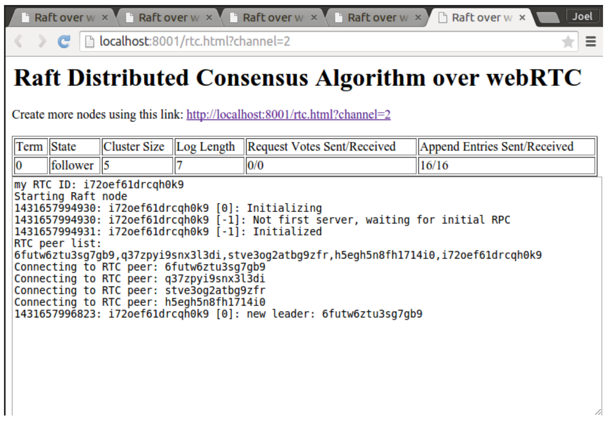
Create more nodes using this link: http://localhost:8001/rtc.html?channel=2

Term	State	Cluster Size	Log Length	Request Votes Sent/Received	Append Entries Sent/Received
0	leader	1	3	0/0	0/0

```
my RTC ID: 6futw6ztu3sg7gb9
Starting Raft node
1431657936446: 6futw6ztu3sg7gb9 [0]: Initializing
1431657936447: 6futw6ztu3sg7gb9 [0]: First server, assuming leadership
1431657936447: 6futw6ztu3sg7gb9 [0]: new state 'leader'
1431657936448: 6futw6ztu3sg7gb9 [0]: Initialized
RTC peer list: 6futw6ztu3sg7gb9
```

Figure 3: Basic app with one Node/Tab

Then four more Raft nodes were then added to the cluster by clicking on the embedded "create a node" link in the application. Here is a screenshot of the browser once four new nodes are added to the cluster (5 total):


Raft Distributed Consensus Algorithm over webRTC

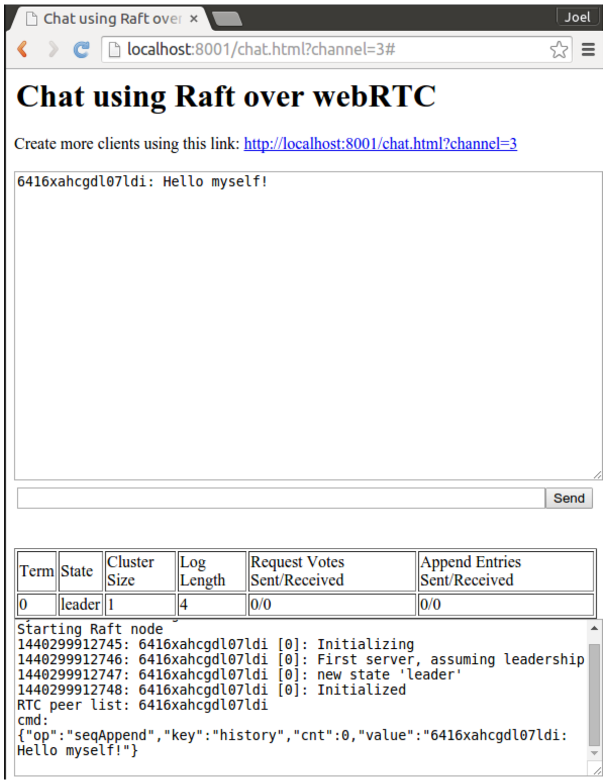
Create more nodes using this link: http://localhost:8001/rtc.html?channel=2

Term	State	Cluster Size	Log Length	Request Votes Sent/Received	Append Entries Sent/Received
0	follower	5	7	0/0	16/16

```
my RTC ID: i72oef61drcqh0k9
Starting Raft node
1431657994930: i72oef61drcqh0k9 [0]: Initializing
1431657994930: i72oef61drcqh0k9 [-1]: Not first server, waiting for initial RPC
1431657994931: i72oef61drcqh0k9 [-1]: Initialized
RTC peer list:
6futw6ztu3sg7gb9,q37zpyi9snx3l3di,stve3og2atbg9zfr,h5egh5n8fh1714i0,i72oef61drcqh0k9
Connecting to RTC peer: 6futw6ztu3sg7gb9
Connecting to RTC peer: q37zpyi9snx3l3di
Connecting to RTC peer: stve3og2atbg9zfr
Connecting to RTC peer: h5egh5n8fh1714i0
1431657996823: i72oef61drcqh0k9 [0]: new leader: 6futw6ztu3sg7gb9
```

Figure 4: Basic app with Five Nodes/Tabs

In addition, the full chat application was also tested manually by loading the initial chat application landing page to create a Raft over WebRTC cluster with a single leader. Then one or more lines of text were entered into the text line each followed by clicking the Send button. Here is a screenshot of the browser after the first node/tab of the chat application is created and a line of text was sent:


Chat using Raft over webRTC

Create more clients using this link: http://localhost:8001/chat.html?channel=3

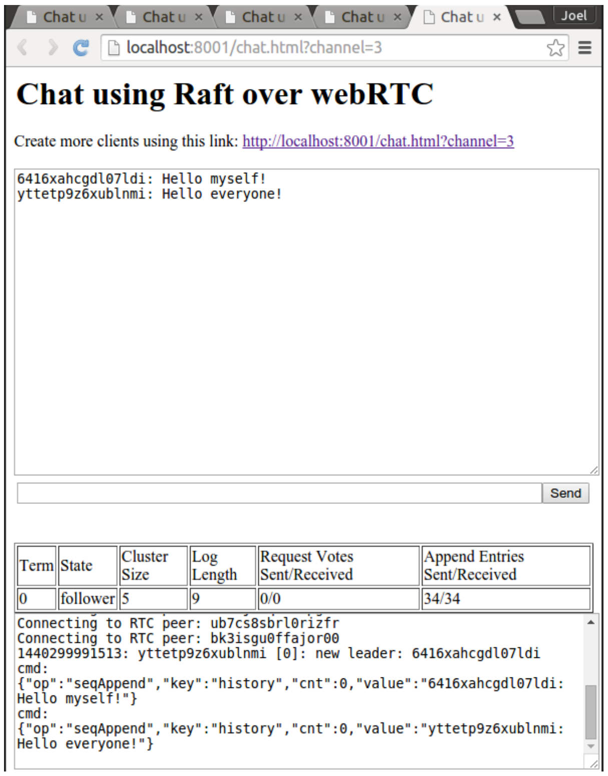
```
6416xahcgdl07ldi: Hello myself!
```

Term	State	Cluster Size	Log Length	Request Votes Sent/Received	Append Entries Sent/Received
0	leader	1	4	0/0	0/0

```
Starting Raft node
1440299912745: 6416xahcgdl07ldi [0]: Initializing
1440299912746: 6416xahcgdl07ldi [0]: First server, assuming leadership
1440299912747: 6416xahcgdl07ldi [0]: new state 'leader'
1440299912748: 6416xahcgdl07ldi [0]: Initialized
RTC peer list: 6416xahcgdl07ldi
cmd:
{"op":"seqAppend","key":"history","cnt":0,"value":"6416xahcgdl07ldi:
Hello myself!"}
```

Figure 5: Chat app with one Node/Tab

Then four more Raft nodes were then added to the cluster by clicking on the embedded "create more clients" link in the application. Finally, on a non-leader node more text lines were entered and sent by clicking the Send button. Here is a screenshot of the browser once four new nodes/tabs are added to the cluster (5 total) and another text line was sent:


Chat using Raft over webRTC

Create more clients using this link: http://localhost:8001/chat.html?channel=3

```
6416xahcgdl07ldi: Hello myself!
yttetp9z6xublnmi: Hello everyone!
```

Term	State	Cluster Size	Log Length	Request Votes Sent/Received	Append Entries Sent/Received
0	follower	5	9	0/0	34/34

```
Connecting to RTC peer: ub7cs8sbrl0rizfr
Connecting to RTC peer: bk3isgu0ffajor00
1440299991513: yttetp9z6xublnmi [0]: new leader: 6416xahcgdl07ldi
cmd:
{"op":"seqAppend","key":"history","cnt":0,"value":"6416xahcgdl07ldi:
Hello myself!"}
cmd:
{"op":"seqAppend","key":"history","cnt":0,"value":"yttetp9z6xublnmi:
Hello everyone!"}
```

Figure 6: Chat app with Five Nodes/Tabs

After every change was made to the cluster all the nodes were checked to ensure that the following steady state properties were observed:

1. only a single leader
2. all nodes have the same term and transaction log count
3. all non-leader nodes are receiving 'appendEntries' RPCs from the leader
4. all nodes have the same perception of the cluster size
5. the transaction log entries are identical on all nodes
6. the state machine is identical on all nodes

The current leader was then terminated. The other nodes were checked for the steady state properties. This was repeated until only two nodes remained in the cluster.

After the five node cluster was brought up, state machine commands were manually sent to leader to add to the transaction log. This was done using the JavaScript console as there is not yet a way to do this via the interface. Again the nodes were checked for the steady state properties.

Then the application was tested was by loading the initial cluster creation landing page in a browser tab. Then nodes were randomly added to the cluster (by clicking on the link embedded in the page) or removed from the cluster (by closing a tab). After each change the steady state properties were checked.

Similar testing to the above was done with browser instances on two separate computers. In addition, cross-browser functionality was verified by running nodes from the same cluster running simultaneously in Chrome 42 and Firefox 37.

5.1.2 Outcome

The results of the manual testing above showed that Raft over WebRTC was able to perform leader elections correctly, the log and state machine was replicated correctly to all nodes, and the safety properties were maintained. During testing the membership of the cluster was changed numerous times (both node additions and removals) and the Raft properties were maintained throughout. In addition, client interaction was tested although the application does not provide a friendly interface for doing so.

One limitation of the system is that the if the cluster membership drops from two nodes to one node then the cluster will become unavailable without manual intervention. This is because when there are two nodes in the cluster, both nodes are required for for a majority vote and when one member is forcible shutdown then the remaining cluster node will be unable to commit transaction entries and the cluster will become unavailable. There are several options to get around this problem:

1. The node can ask the leader to be removed and then wait for the transaction to become effective before it disconnects. This may not always be an option especially in a browser context where the user may immediately kill the window/tab without giving the application a change to remove itself from the cluster.

2. The signaling server can serve as a tie-breaker. This would allow the remaining leader successfully commit the removal of the other node. In the case of a partition, the signaling server must only vote for one of the nodes.

3. The node with the lower ID could have an implicit tie-breaker vote. This would allow one node of the partition to continue in the case of a network partition. If a node dies or is killed, this would only allow the cluster to continue to operate if the node that dies happens to have the higher ID.

4. The system could allow manual user intervention. However, this is only likely to work correctly in the case where the user of the last remaining node can be certain that the other node will never return otherwise they is the risk of data-corruption.

Running a Raft cluster in a browser context leads to some level of timer instability. The JavaScript execution context is a fully event driven environment with a single thread of execution for a given JavaScript context. JavaScript code is run when browser (or Node.js) events trigger JavaScript code that is registered to that event. In order to implement logic that runs periodically, JavaScript code is register as a callback timer using the setTimer, setInterval or requestAnimationFrame APIs. However, none of these mechanisms is a precise timer. When a timer fires, the callback code they refer to is added to the JavaScript event queue and events on the queue are processed in the order they arrive. This means callbacks may be delayed by other code that is already running or queued first.

However, there is another browser mechanism that is less well known but that causes even larger delays in timer code execution. Modern browsers often have dozens or even hundreds of windows and/or tabs open simultaneously. In order to reduce CPU load, increase user responsiveness and improve battery life, most modern browsers will throttle the JavaScript callback timers for all windows and tabs that are hidden or have not been directly interacted with for some period of time. This throttling can cause timers to slow down by an order of magnitude or more.

In the context of Raft over WebRTC, the slowdown of background page timers means that whenever the leader node is residing on a page that is in the background, the propagation of transaction log entries slows down substantially. This also means that other nodes which are in the foreground are more likely to have an election timeout timer fire and they are more likely to become a leader because they are more likely to be able to send and receive votes quickly than nodes that are in the background. In this sense the problem is self-correcting because over the long term because higher performing nodes more are likely to start and win elections. This also indicates dynamic adjustment of timeout values may be a fruitful area of further study.

In the context of the full chat application the slowdown of background pages is even more pronounced because each addition to the chat history can take several messages to fully propagate the new date.

5.2 Automated Testing

5.2.1 Setup

Automated testing of the Raft over WebRTC system was performed by leveraging the SlimerJS scriptable browser environment. SlimerJS enables web applications to be scripted by loading pages in a special Mozilla rendering environment called XULRunner. The XULRunner instance is directed to a X virtual frame buffer (xvfb) which provides effectively headless testing.

To simplify the execution of the tests and to make the testing infrastructure more portable, a docker container environment was used to encapsulate and execute the SlimerJS environment. The container is built using *test/Dockerfile* which is in turn derived from the "fentas/slimerjs" docker image (https://github.com/fentas/docker/blob/master/slimerjs/Dockerfile). This custom Dockerfile was created because the upstream version of the Dockerfile used an older version of XULRunner that did not support more than 7 WebRTC clients communicating together simultaneously.

Writing tests using SlimerJS involves creating JavaScript programs that are run within the SlimerJS environment. These scripts can create browser instances (basically headless browser tabs), query the status of those pages and execute JavaScript within the context of the pages themselves.

Three different tests were created to test the Raft over WebRTC functionality and the chat application. Each of the tests takes a parameter indicating the number of nodes/pages to launch and test again. The tests scripts files are:

1. *test/common.js*: a module of common functions used by the other tests scripts.

2. *test/test_up.js*: launch and start an initial leader node/page, then start the rest of the nodes (if any) so that the leader will configure them into the cluster. Then test then waits for every node to be recorded

(applied) into the log of every other node of the cluster. The test reports the number of milliseconds for the cluster to reach this state.

3. *test/test_kill_nodes.js*: similar to *test/test_up.js* but once all the nodes are configured into the cluster, this script then kills the leader plus a configurable number of nodes. By default the script kills just under half the nodes of the cluster. The test then waits for a new leader to be elected and remove the dead nodes from the cluster. The test reports both the number of milliseconds for the cluster to come up and the recover from the failure.

4. *test/test_chat_propagate.js*: similar to *test/test_up.js* but once the cluster is up, the test connects to one of the non-leader nodes and and sends a chat text line. The test then waits for the chat message to propagate to every other node's state machine. The test reports both the number of milliseconds for the cluster to come up and for the messages to propagate to every node.

The tests were all run on a system with the following configuration:

1. *CPUs*: 8 Intel Core i7 processors running at 3.20GHz
2. *Memory*: 12 GB
3. *OS*: Ubuntu 14.04.1 LTS
4. *Docker*: version 1.3.1
5. *Docker image*:
   (a) *OS*: Ubuntu 12.04.5 LTS
   (b) *SlimerJS*: version 0.9.6
   (c) *XULRunner*: 38.0.5

Three tests were run at 10 different cluster sizes (odd cluster sizes from 3 to 21 nodes): *test/test_up.js* and *test/test_kill_nodes.js* in kill one (leader) node and kill half nodes. Each of these tests was run with three different Raft election timeout values: 100ms, 500ms, 1000ms. The *test/test_chat_propagate.js* test was run at 11 different cluster sizes (odd cluster sizes from 1 to 21 nodes). Each test mode and cluster size was run 7 times for a total of 707 test runs (3*10*7*3 + 11*7)

5.2.2 Outcome

Figure 7 shows the results of running the *test/test_chat_propagate.js* test to show the amount of time that a stable cluster takes to propagate, commit and apply a single state machine command to the state machines on every cluster node.

Figure 7 is a scatterplot with each test run as a blue "x" and a black line representing the average at each cluster size.

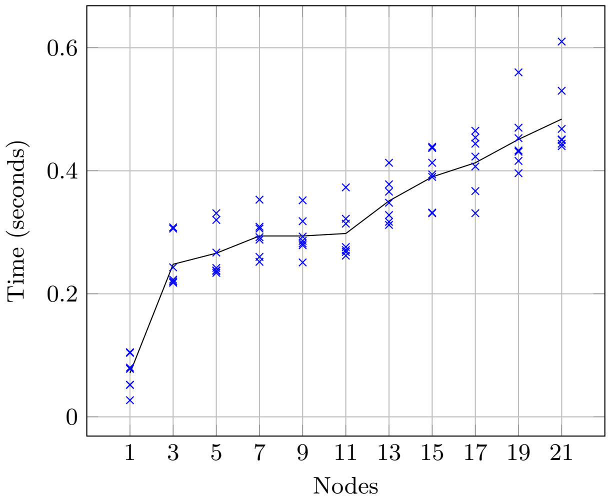


Figure 7: Propagate, Commit, and Apply Command

Figures 8, 9 and 10 show the results of running the *test/test_up.js* and *test/test_kill_nodes.js* in both modes (kill only leader and kill just under half the nodes). Three different scatterplots are shown for each test mode to test different values of the Raft election timeout value: 100ms, 500ms, and 1000ms. The election timeout value makes very little difference to the average time curve line. The reason for this is that the first leader tends to stay the leader as new nodes are added so for the most part the election timer does not have an impact.

Some specific notes about the figures:

1. *Figure 8 (a)*: The results show that once the cluster size reaches 17, the cluster sometimes does not activate before the test times out (the red "x" marks). Once the cluster reaches 21 nodes, all 7 test runs time out. The reason for this is because at 17 nodes, the leader is not always able to send out appendEntries RPCs to all the other nodes before their election timeout fires. More analysis is needed, but one possible explanation is that the single SlimerJS XULRunner process that is handling all the cluster nodes is not able to give enough CPU time to the leader node to reliable send appendEntries RPCs to all the other nodes within the election timeout value. Increasing the election timeout to 500ms or greater eliminates the failures without significantly changing the average cluster activation time curve.

2. *Figure 9 (a)*: The results show that when the cluster size reaches 15, then average amount of time for cluster recovery (after the leader is killed) increases dramatically (not all data points are shown on the plot for size 15). At cluster size 17 the cluster is no longer able to recover within the test timeout period. Increasing the election timeout to 500ms or greater eliminates the failures at the cost of slightly increased recovery time for smaller cluster sizes.

3. *Figure 10 (a)*: During this test cluster was unable to activate at cluster size 21 when the election timeout value is set to 100ms (see Figure 8 (a)). This means there was no opportunity for cluster recovery at cluster size 21 so there is no data for that column.

Each figure is a scatterplot with each test run as a blue "x" and a black line representing the average at each cluster size.

6 Next Steps

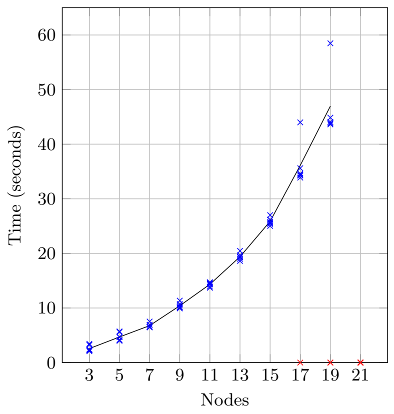
One of the useful outcomes of this project was that it brought to light many areas where further study of Raft, WebRTC, and Raft over WebRTC should be pursued.

Here are some interesting areas for further exploration:

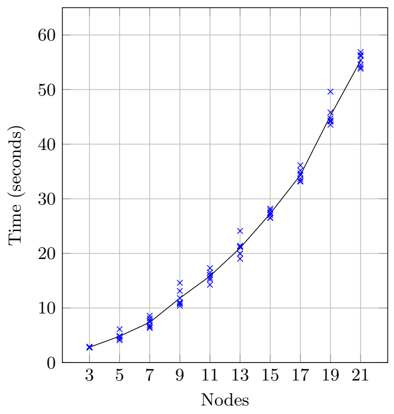
1. Explore the use of alternate WebRTC transport modes for relaxing ordering constraints and delivery guarantees. Since the Raft protocol is tolerant of out-of-order and dropped messages, it is possible that Raft over WebRTC could be more efficient with these one or both of these constraints relaxed.
2. Test the survivability of the cluster when the signaling server is becomes completely unresponsive or just very slow.
3. Explore models for dynamically adjusting Raft timeout values to account for changing network and browser conditions.
4. Test the ability of the system to catch up stale nodes by disconnecting one or more nodes, run the cluster for some period time with transactions created periodically, and then reconnect the stale clients.
5. Implement a user interface for sending commands to the transaction log and for viewing the current state of the Raft state machine.
6. Explore models for dealing with the case where the cluster goes from two nodes down to one node. Can usefulness and availability be improved without sacrificing Raft safety constraints.
7. Test the system in more various network environments. Determine the reliability of establishing direct browser-to-browser communication. Determine if corporate firewalls make this as difficult as is implied in some of the documentation and standards texts.
8. Implement and test forwarding client requests rather than default redirecting the client to the leader. In other words, allow client requests to originate from any node of the cluster and be redirected automatically to the leader (rather than the current manual process).
9. Perform quantitative and real-world testing to better characterize the performance, availability and scalability characteristics of the system.
10. Explore models for timing out and automatically removing nodes when they become unresponsive without needing to be notified by the signaling server that the nodes have disconnected.
11. Test larger scale deployments across multiple systems. Determine the factors that affect scalability and characterize scaling limits.
12. Perform wider cross-browser testing including Internet Explorer and Opera.
13. Implement Raft log compaction.
14. Explore models that allow for a combination of strict shared state and high performance bulk data such as media or object blobs. For example, the data with full consensus might contain hashes that to the refer to bulk data.
15. Implement a Node.js based Raft over WebRTC application. This would allow non-browser contexts to participate in the Raft cluster together with the browser clients. PeerJS issue [103](https://github.com/peers/peerjs/issues/103) must be resolved for this to be feasible.
16. Explore a design where some nodes of the cluster are non-voting and are not counted as part of transaction commit quorum. However, these nodes would still be included in the log replication algorithm. One model this would enable would be a configuration where a small group of voting nodes serve as the hub for a larger group of non-voting nodes. This may enable higher scaling of the system. It also enables interesting applications such as a logging or backup nodes.

Acknowledgments

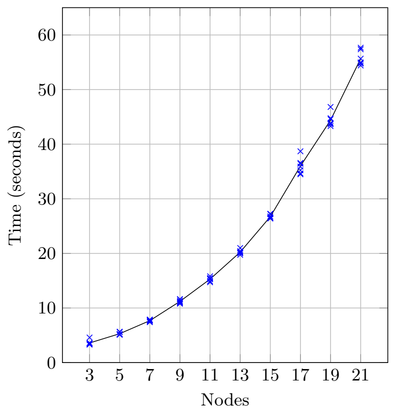
I would like to thank my academic supervisor, David Levine, for his excellent input and oversight of this project. I would also like to thank Diego Ongaro for his amazing work on the Raft protocol and paper which made implementing a Raft implementation a pleasure. There are too many people involved in the creation and standardization of WebRTC but I would like to thank them for their tireless work to make browser-to-browser communication possible. Last but not least, I would like to thank my family for putting up with the long nights and weekends I spent on this project. Thanks!



(a) 100ms election timer

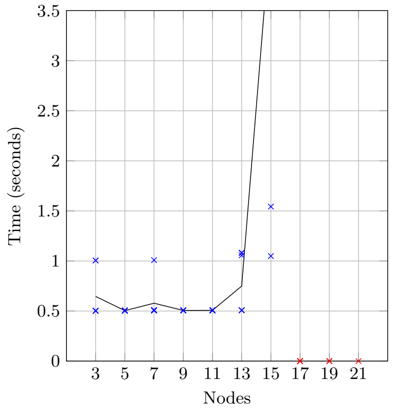


(b) 500ms election timer

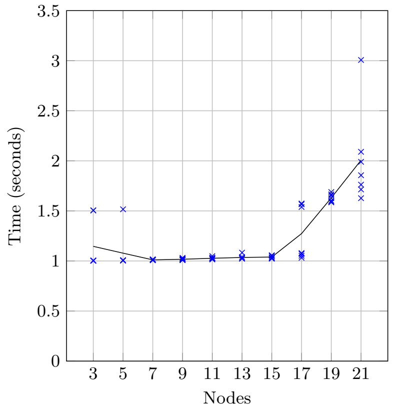


(c) 1000ms election timer

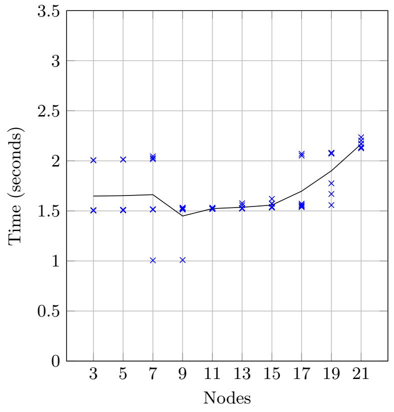
Figure 8: Initial Cluster Activation



(a) 100ms election timer

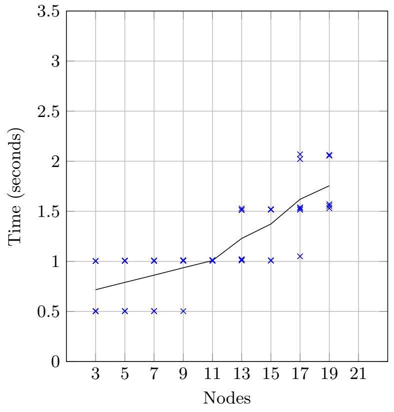


(b) 500ms election timer

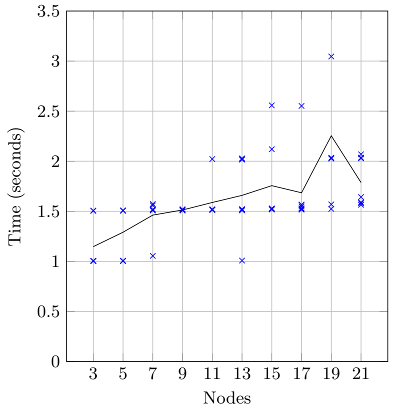


(c) 1000ms election timer

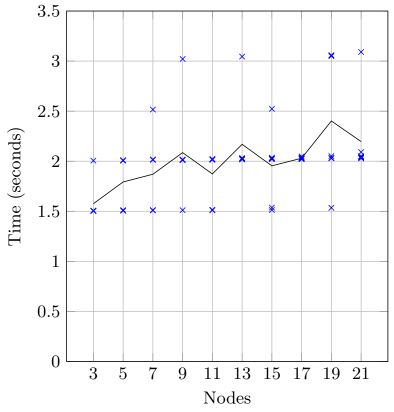
Figure 9: Recovery from Leader Failure



(a) 100ms election timer



(b) 500ms election timer



(c) 1000ms election timer

Figure 10: Recovery from Half Cluster Failure