# Standards SOTA

IETF Rtcweb Working Group

Why this was created in the IETF?

There are a number of proprietary implementations that provide direct interactive rich communication using audio, video, collaboration,  
games, etc. between two peers' web-browsers. These are not interoperable, as they require non-standard extensions or plugins to  
work. There is a desire to standardize the basis for such communication so that interoperable communication can be established  
between any compatible browsers. The goal is to enable innovation on top of a set of basic components. One core component is to enable  
real-time media like audio and video, a second is to enable data transfer directly between clients.  
  
This work will be done in collaboration with the W3C. The IETF WG will produce architecture and requirements for selection and profiling of the on the wire protocols. The architecture needs to be coordinated with W3C. The IETF WG work will identify state information and events that need to be exposed in the APIs as input to W3C. The W3C will be responsible for defining APIs to ensure that application developers can control the components. We will reach out to the developer community for consultation and early feedback on implementation.  
  
The security and privacy goals and requirements will be developed by the WG. The security model needs to be coordinated with the W3C. The work will also consider where support for extensibility is needed. RTP functionalities, media formats, security algorithms are example of things that commonly need extensions, additions or replacement, and thus some support for negotiation between clients is required.

Task of the Rtcweb Working Group

The WG will perform the following work:

Define the communication model in detail, including how session management is to occur within the model.

Define a security model that describes the security and privacy goals and specifies the security protocol mechanisms necessary to achieve those goals.

Define the solution - protocols and API requirements - for firewall and NAT traversal.

Define which media functions and extensions shall be supported in the client and their usage for real-time media, including media adaptation to ensure congestion safe usage.

Define what functionalities in the solution, such as media codecs, security algorithms, etc., can be extended and how the extensibility mechanisms works.

Define a set of media formats that must or should be supported by a client to improve interoperability.

Define how non media data is transported between clients in a secure and congestion safe way.

Provide W3C input for the APIs that comes from the communication model and the selected components and protocols that are part of the solution.

The group will consider options for interworking with legacy VoIP equipment.

This work will be done primarily by using already defined protocols or functionalities. If there is identification of missing protocols or  
functionalities, such work can be requested to be done in another working group with a suitable charter or by requests for chartering it  
in this WG or another WG. The following topics will be out of scope for the initial phase of the WG: RTSP, RSVP, NSIS, Location services, Resource Priority, and IM & Presence specific features.  
  
The products of the working group will support security and keying as required by BCP 61 and be defined for IPv4, IPv6, and dual stack deployments. The Working Group will consider the possibility of defining a browser component that implements an existing session negotiation and management protocol. The working group cannot explicitly rule out the possibility of adopting encumbered technologies; however, the working group will try to avoid encumbered technologies that require royalties or other encumbrances that would prevent such technologies from being easy to use in web browsers.

Goals and Milestones

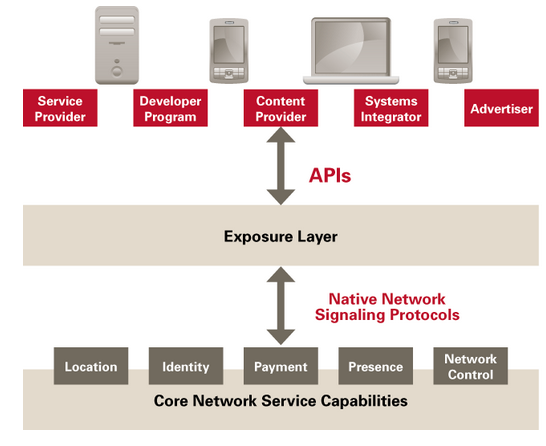
Feb 2014 - Complete Overview (and hold for dependency resolution) (draft-ietf-rtcweb-overview) Done - Send Use Cases document (draft-ietf-rtcweb-use-cases-and- requirements) to IESG for publication as Informational Mar 2014 - Send Security and Privacy Problem Statement (draft-ietf- rtcweb-security) to IESG for publication as Informational Apr 2014 - Send Media Transport (draft-ietf-rtcweb-rtp-usage) to IESG for publication as Proposed Standard Apr 2014 - Audio Processing and Audio Codecs (draft-ietf-rtcweb-audio) to IESG for publication as Proposed Standard May 2014 - Send Data Stream Transport for non-media data (draft-ietf- rtcweb-data-channel) to IESG for publication as Proposed Standard Jun 2014 - Send Security Solution (draft-ietf-rtcweb-security-arch) to IESG for publication as Proposed Standard Jun 2014 - Send Specification of Transport Protocols and their NAT Traversal to IESG for publication as Proposed Standard Sep 2014 - Send Signalling Negotiation and NAT Traversal (draft-ietf- rtcweb-jsep) to IESG for publication as Proposed Standard Sep 2014 - Send STUN Usage for Consent Freshness to IESG for publication as proposed standard Dec 2014 - Video Processing and Video Codecs (draft-ietf-rtcweb-video) to IESG for publication as Proposed StandardVideo Processing and Video Codecs (draft-ietf-rtcweb-video) to IESG for publication as Proposed Standard Dec 2014 - Send Overview (after dependencies are ready) to IESG for publication as Applicability Statement

### Overview

The Open Mobile Alliance (OMA) is a Mobile Operator driven industry forum for the definition of interoperable mobile service enablers. OMA defines APIs to offer functionalities and resources of Operator networks to developers.

#### OMA API Program

The OMA API program provides standardized interfaces to the service infrastructure residing within communication networks and on devices. Focused primarily between the service access layer and generic network capabilities, OMA API specifications allow operators and other service providers to expose device capabilities and network resources in an open and programmable way—to any developer community independent of the development platform.



image

OMA APIs exposes the network assets that developers need—no matter what protocols, platforms or other APIs they use. The Core network assets must be made available in order to deploy the wide variety of new applications and services that enter the market every day. These APIs are the tool that Operator offers to developer to make its services accessible to massive markets in a standard way. The OMA set of APIs increases the portability of applications and services in order to reach the subscriber base of operators and service providers that deploy OMA APIs.

As the number of APIs that perform the same functionality proliferate, fragmentation occurs. This limits developer access to subscribers, and operator and service providers’ choices of development platforms and communities. The OMA API Program, through standardization, solves this problem. A full OMA API inventory can be found at: <http://technical.openmobilealliance.org/Technical/technical-information/oma-api-program/oma-api-inventory>

### Selected APIs with potential relevance for reTHINK

There are several OMA APIs which can be potentially adopted by reTHINK project:

1. [RESTful Network API for WebRTC Signaling V1.0](http://technical.openmobilealliance.org/Technical/technical-information/oma-api-program/oma-api-inventory/api-details?API_ID=141). This API was released in February 2014 and it is a comprehensive REST-API for the WebRTC offer/answer signaling model. The payload that is transferred in the requests and responses is defined here as XML.
2. [Authorization Framework for Network APIs](http://technical.openmobilealliance.org//Technical/Release_Program/docs/Autho4API/V1_0-20131120-C/OMA-ER-Autho4API-V1_0-20131120-C.pdf). It was released in November 2013.The Authorization Framework for Network APIs enables a Resource Owner owning network resources exposed by Network APIs and RESTful APIs in particular, to authorize third-party Applications (desktop, mobile and web Applications)to access these resources via that API on the Resource Owner’s behalf.

There are also APIs for: Converged Address Book, Customer Profile, Network Message Store, Notification Channel, OneAPI, Payment, Presence and Quality of Service.

### OMA protocols with potential relevance for reTHINK

#### The Lightweight M2M

OMA Lightweight M2M is a protocol from the Open Mobile Alliance for M2M or IoT device management. Lightweight M2M enabler defines the application layer communication protocol between a LWM2M Server and a LWM2M Client, which is located in a LWM2M Device. The OMA Lightweight M2M enabler includes device management and service enablement for LWM2M Devices. It is normally used with CoAP. This protocol can be used in reTHINK for several components such as the Catalogue as porposed in [Delivery 2 from WP2].

### W3C WebRTC API

The Web Real-Time Communications Working Group was created in May 2011 within the W3C to define client-side APIs to enable Real-Time Communications in Web browsers.

It has defined a functional WebRTC 1.0 API which is implemented by major browser vendors to build real-time media applications in the browser without the need of installing any additional plugin. Additionaly to real-time media, WebRTC also supports the exchange of generic peer-to-peer data thanks to the Datachannel feature. This API is currently supported and production-ready in Firefox, Chrome and Opera.

Together with WebRTC 1.0 API the W3C is working in a series of drafts for which can be used toghether with the WebRTC API to create real-time media Web applications:

* Media Capture and Streams (getUserMedia): set of JavaScript APIs that allow local media, including audio and video, to be requested from a platform. This API allows to capture real time audio and video from the device which is running the web browsers. It is used by all the WebRTC applications which require capturing audio or video.
* MediaStream Recording: a recording API for use with MediaStreams as defined in [GETUSERMEDIA]
* MediaStream Image Capture: specific the takePhoto() and grabFrame() methods, and corresponding camera settings for use with MediaStreams as defined in Media Capture and Streams [GETUSERMEDIA]
* Media Capture Depth Stream Extensions: extends the Media Capture and Streams specification [GETUSERMEDIA] to allow a depth stream to be requested from the web platform using APIs familiar to web authors.
* Media Capture from DOM Elements: defines how a stream of media can be captured from a DOM element, such as a , , or
* element, in the form of a MediaStream [GETUSERMEDIA].
* Audio output devices API: defines a set of JavaScript APIs that let a Web application manage how audio is rendered on the user audio output devices.
* Identifiers for WebRTC's Statistics API: defines a set of Javascript APIs that allow access to the statistical information about a PeerConnection
* Screen Capture: defines how a user's display, or parts thereof, can be used as the source of a media stream using getOutputMedia, an extension to the Media Capture API [GETUSERMEDIA].

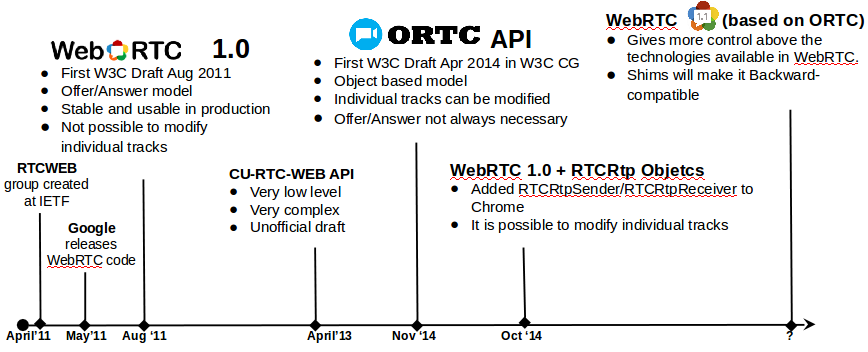
#### Applicability in reTHINK

The WebRTC is going to be intensively used in reTHINK to implement hyperties for Human-to-Human scenarios and also M2M scenarios where WebRTC Datachannel is used.

ORTC

Introduction

ORTC is an alternative to the current WebRTC API 1.0 to write WebRTC Applications to be executed in Web browser.

The protocols on the wire are exactly the same so it is compatible with aaplications written using the current API.  


Differences between ORTC and WebRTC 1.0.

Those are the main differences between ORTC and WebRTC 1.0:

ORTC specifically tailored to provide the direct control needed to enable advanced multimedia and conferencing features.

Limitations of the WebRTC API 1.0 have been addressed in the new version.

many of the parameters which are automatically handled by the browser in WebRTC 1.0 can be now modified by using native methods.

SDP is not the mandatory format to exchange information between browsers.

Advantages of ORTC over WebRTC 1.0.

An example of WebRTC 1.0 limitation it is that is not possible to modify parameters on individual media tracks since the SDPs must contain all the mediatracks of a mediastream. ORTC includes the RTCRtpSender object which associated to a sending MediaStreamTrack which provides methods to tweak its parameters.

An example of application of this new element is the ability to change the bitrate used for a video in-progess session which is being sent over a bad-quality connection keeping the parameters of the audio mediatrack. In WebRTC 1.0 this changes requires a SDP re-negotiation but with ORTC this is not nececesary.

Although RTCRtpSender was not included in the WebRTC 1.0 definition, this element was planned to be supported by Chrome from version 39.

Compatibility between ORTC and WebRTC 1.0.

Although SDP is not mandatory but WebRTC 1.0 applications would make it compatible thanks to Javasrcipts shims. All the use cases that ORTC would enable are already possible but they are more complex to implement as they require manual SDP manipulations which are error-prone and may require several Offer/Answer SDP exchanges to update the multimedia sessions.

ORTC implementations.

ORTC is the WebRTC API which Microsoft will implement in Internet Explorer and Skype will be supported in the Web Browser supporting ORTC along as other browsers which supports WebRTC 1.0.

The W3C SysApps working group was originally chartered to provide a runtime and security model however, it has not been possible to reach such objective. In the meanwhile we may find proprietary web runtimes, FFOS and Chrome, which provide a security model for installed packaged web runtimes.

The WHATWG was formed by individuals of Apple, the Mozilla Foundation, and Opera Software in 2004, in response to the slow development of World Wide Web Consortium (W3C) Web standards and W3C's decision to abandon HTML in favor of XML-based technologies.

On 10 April 2007, the Mozilla Foundation, Apple, and Opera Software proposed that the new HTML working group of the W3C adopt the WHATWG’s HTML5 as the starting point of its work and name its future deliverable as "HTML5". On 9 May 2007, the new HTML working group resolved to do that

### W3C Application Lifecycle and Events

The W3C Application Lifecycle and Events draft (last version from 16 May 2014) extends the [Service Worker](w3c-service-workers.md) global execution context, to allow web developers to author applications that manage the application lifecycle and react to system events e.g. email or voip application. These capabilities allow application developers to create applications that integrate closely with the underlying system.

The following functionalities are provided: \* A background App or Service can run without a visible user interfaces \* An application is able to decide when to show the user interface \* The Application can be terminated without user’s consent, and that is able to restore to its previous state. \* The application is able to show a different user interface given how the app was launched. For example, if launched as a photo picker, the application will not show the default application window, but instead creates a special purpose user interface. \* The Application is only launched for a specific set of events eg the runtime uses somekind of events pre-filtering mechanism. For example, if an application listens to a "USB plugged" event, it can additionally ask to only listen to a specific device connected or a specific port. \* The application is able to enumerate windows associated with it, and create new windows.

This API extends the ServiceWorkerGlobalScope interfaces in the following way:

partial interface ServiceWorkerGlobalScope {  
 attribute EventHandler onlaunch;  
 attribute EventHandler onterminate;  
 attribute EventHandler onterminatecanceled;  
 readonly attribute TaskScheduler taskScheduler;  
};

#### Applicability in reTHINK

Similar to Service Workers, this extension can facilitate the development of some Runtime features notably to govern the runtime life-cycle of Hyperty instances. However, it seems this draft has not much support by the industry. However, [Chrome Packaged App lifecycle](https://developer.chrome.com/apps/app_lifecycle) looks similar. [Firefox Add-ons](https://developer.mozilla.org/en-US/Add-ons) should also support some kind of App life-cycle.

#### References

* http://www.w3.org/2012/sysapps/app-lifecycle/
* https://lists.w3.org/Archives/Public/public-sysapps/2015Apr/0001.html
* https://www.w3.org/community/trustperms/
* https://whatwg.org/

### Service Workers

Service workers are based on previous [Web Worker](http://www.w3.org/TR/workers/) W3C work and they essentially act as proxy servers that sit between web applications, and the browser and network (when available.) They are intended to (amongst other things) enable the creation of effective offline experiences, intercepting network requests and taking appropriate action based on whether the network is available and updated assets reside on the server.

A service worker is an event-driven worker registered against an origin and a path. It takes the form of a JavaScript file that can control the web page/site it is associated with, intercepting and modifying navigation and resource requests, and caching resources in a very granular fashion to give you complete control over how your app behaves in certain situations (the most obvious one being when the network is not available.)

A service worker is run in its context: it therefore has no DOM access, and runs on a different thread to the main JavaScript that supports the web app, so it is not blocking. It is designed to be fully async;

A service worker is first registered and, if successful, it will be downloaded to the client and attempt installation/activation for URLs accessed by the user inside the whole origin, or inside a subset specified by you.

Service Worker registration Example:

if ('serviceWorker' in navigator) {  
 navigator.serviceWorker.register('/my-app/sw.js', {  
 scope: '/my-app/'  
 }).then(function(reg) {  
 console.log('Yey!', reg);  
 }).catch(function(err) {  
 console.log('Boo!', err);  
 });  
}

Where /my-app/sw.js is the location of the ServiceWorker script, and it controls pages whose URL begins /my-app/.

At this point, your service worker will observe the following lifecycle: \* Download \* Install \* Activate

The service worker is immediately downloaded when a user first accesses a server worker–controlled site/page. Installation is attempted when the downloaded file is found to be new — either different to an existing service worker (byte-wise compared), or the first service worker encountered for this page/site.

The user agent may terminate service workers at any time it has no event to handle or detects abnormal operation such as infinite loops and tasks exceeding imposed time limits, if any, while handling the events.

Service Workers can be used to intercept network messages. Example:

self.addEventListener('fetch', function(event) {  
 console.log(event.request);  
});

Service Workers provides the basis for other features including: \* [Push](http://w3c.github.io/push-api/) \* [Background sync](https://github.com/slightlyoff/BackgroundSync) \* [Geofencing](https://github.com/slightlyoff/Geofencing)

Service Workers are still an experimental technology only supported in Desktop Chrome and Firefox.

#### Applicability in reTHINK

Service Workers provides features that can facilitate the development of some Runtime features including Event BUS, ProtOfly engine, Policy Engine. Its usage to support the Hyperty instance itself should also be evaluated. However it seems this technology is only available in Browsers and not in server side javascript runtime like node.js.

#### References

* http://www.w3.org/TR/workers/
* https://developer.mozilla.org/en-US/docs/Web/API/ServiceWorker\_API
* https://github.com/slightlyoff/ServiceWorker/blob/master/explainer.md
* http://www.w3.org/TR/service-workers/
* https://jakearchibald.github.io/isserviceworkerready/

### Content Security Policy Level 2

Content Security Policy (CSP) is an added layer of security that helps to detect and mitigate certain types of attacks, including Cross Site Scripting (XSS) and data injection attacks. These attacks are used for everything from data theft to site defacement or distribution of malware.

Defines a policy language used to declare a set of content restrictions for a web resource, and a mechanism for transmitting the policy from a server to a client where the policy is enforced.

CSP provides a standard HTTP header that allows website owners to declare approved sources of content that browsers should be allowed to load on that page — covered types are JavaScript, CSS, HTML frames, fonts, images and embeddable objects such as Java applets, ActiveX, audio and video files.

The following header names are in use as part of an experimental CSP implementations:

Content-Security-Policy — standard header name proposed by the W3C document. Google Chrome supports this as of version 25. Firefox supports this as of version 23, released on 6 August 2013. X-WebKit-CSP — experimental header introduced into Google Chrome and other WebKit-based browsers (Safari) in 2011. X-Content-Security-Policy — experimental header introduced in Gecko 2 based browsers (Firefox 4 to Firefox 22, Thunderbird 3.3, SeaMonkey 2.1). Support for the sandbox directive is also available in Internet Explorer 10 and Internet Explorer 11 using the experimental X-Content-Security-Policy header.

There's initial support for CSP in some web frameworks such as AngularJS and Django.

Example: script-src 'self'; object-src 'none'

Security policies contain a set of security policy directives (script-src and object-src in the example above), each responsible for declaring the restrictions for a particular resource type, or manipulating a specific aspect of the policy’s restrictions.

The server delivers a policy to the user agent via an HTTP response header or an HTML meta element. The Content-Security-Policy header field is the preferred mechanism for delivering a policy. The grammar is as follows:

"Content-Security-Policy:" 1#policy-token

For example, a response might include the following header field: Content-Security-Policy: script-src 'self'

A Content Security Policy consists of a U+003B SEMICOLON (;) delimited list of directives.

#### Applicability in reTHINK

In a preliminary analysis CSP seems too limited to be applied for the runtime policy engine but it may be useful to improve security in the protOfly engine.

#### References

* http://www.w3.org/TR/CSP2/
* https://developer.mozilla.org/en-US/docs/Web/Security/CSP/Introducing\_Content\_Security\_Policy
* http://en.wikipedia.org/wiki/Content\_Security\_Policy

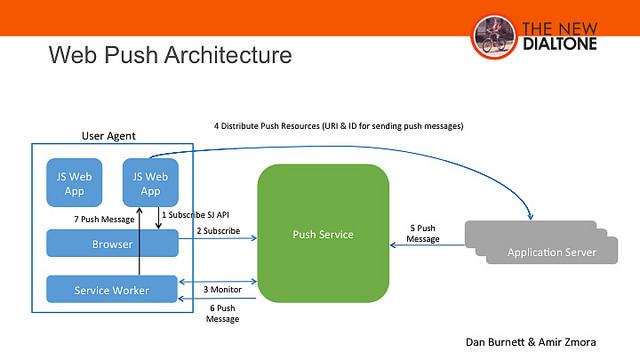
### W3C Push API

W3C Push API work began in the W3C Web Applications (WEBAPPS) Working Group to provide very simple JavaScript APIs for push client registering and message receipt. It was in large part, driven by the urgent WebRTC use case of waking up an application for a real-time incoming call. The API has two logical pieces:

* a pushRegistrationManager that consults the user agent about which push service to use and returns information (a registration id and a URI) that can be sent to the application server for it to know how to reach the push service. In general, the URI is where the application server can send push messages and the registration id is to be provided to the push service to indicate the delivery target for the messages.
* a Service Worker that is used to catch, store if necessary, and ultimately deliver push messages to the application.

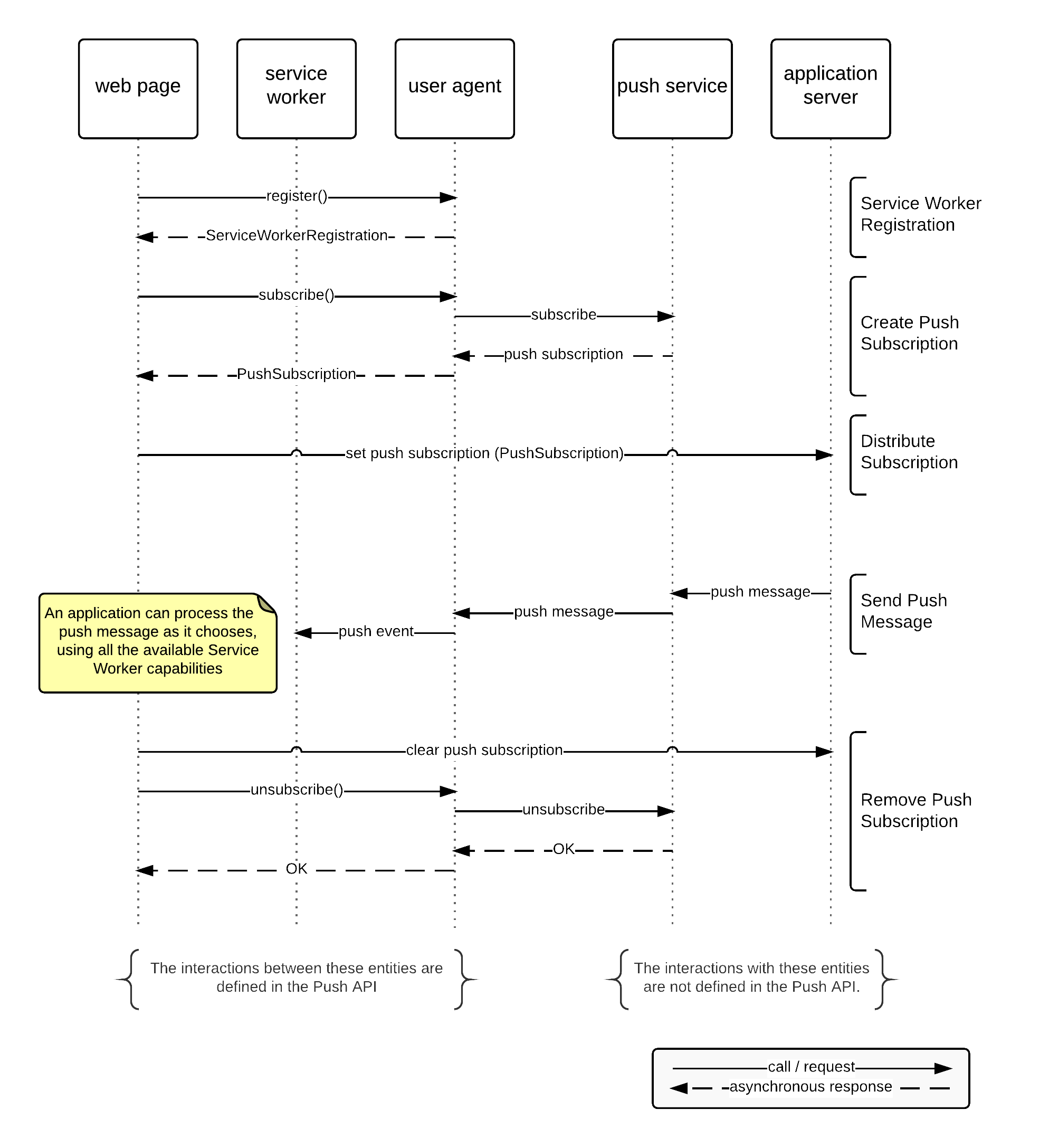
The following general entities and example operation for delivery of push messages are the following (see figure below):

* Application servers request delivery of a push message to a webapp via a RESTful API exposed by a push service
* The push service delivers the message to a specific user agent
* The user agent delivers the push message to the specific webapp intended to receive it.



Web Push Architecture

The main data flows involved in the PUSH API usage are described below:



Main flows of events for subscription, push message delivery, and unsubscription

The W3C Push API is a client API that does not provide any standard for how the application server sends messages to the push server. This part is covered by the IETF WEBPUSH protocol.

### IETF Web Push

In order to provide a standard way for push notification to be used on the Internet, the IETF WEBPUSH working group was created in late 2014. The WEBPUSH WG is developing an HTTP2-based mechanism for applications “to request the delivery of data to a device using a consolidated push notification service. This protocol will include the ability to push the same message to multiple subscribed devices.

Expected clients are both web applications and field gateways that consolidate and forward messages to embedded devices. Several models have been discussed, and there currently seems to be most interest in a publication-subscription model where each device subscribes individually, but with no requirement for a separate registration in advance.

This working group will create an HTTP based protocol that will allow applications to request delivery of data to applications through a consolidated service. The working group will work in cooperation with W3C’s Web Push API.

### Applicability to reTHINK

The ability to push notifications towards Hyperty runtime is an essential feature that must be supported according to these standards.

### References

* http://w3c.github.io/push-api/
* http://thenewdialtone.com/webrtc-browser-push-notification/
* http://datatracker.ietf.org/doc/draft-thomson-webpush-protocol/?include\_text=1

# HTTP/2

## Introduction

HTTP/1.1 has served the Web for more than 15 years, but its age is starting to show. Web application has evolved a lot from the beginning but the protocol which transport it has not evolved at the same pace.

Loading a Web page is more resource intensive than ever because HTTP practically only allows one outstanding request per TCP connection: \* Browsers have used multiple TCP connections to issue parallel requests. This is counter-productive (TCP congestion control is effectively negated leading to congestion events), and unfair (browsers take more network resources). \* At the same time, the large number of requests means a lot of duplicated data “on the wire”. \* These factors mean that HTTP/1.1 requests have a lot of overhead associated with them; the more requests are made, the worse performance we get.

This problems leaded the industry to consider “Best Practices” things like: spriting, data inlining, domain sharding and concatenation which are just workarounds which improves the user experience.These hacks are indications of underlying problems in the protocol itself, and cause a number of problems on their own when used. On the other side, the way in which the Web is accessed has also change a lot, mobile devices has become the main point of entry to the web. Characteristics of wireless connections (high latency, jitter and packet loss) may prevent Web applications served with HTTP over TCP from being responsive and even usable.

[HTTP/2](https://tools.ietf.org/html/rfc7540), which is already a definitive RFC, was designed to be adapted to the new conditions of the WWW and its main goal is to improve the user experience. HTTP/2 is an evolution of SPDY, experimental protocol mainly developed by Google which is currenlty being used in production in many Google applications. To take advantage of HTTP/2 new features a new transport protocol, QUIC, has been designed. Both protocols combined will be extensily used in Internet in the next years.

## Main differences from HTTP1.1

|  |  |
| --- | --- |
| HTTP/1.1 | HTTP/2 |
| textual | binary |
| ordered and blocking | fully multiplexed |
| several connections for parallelism | one connection for parallelism |
| only content compression | header compression |
| not proactive push | allows servers to “push” responses proactively into client caches |

### Binary

HTTP/2 is a binary protocol, it means that no human-understable ASCII chars are sent on the wire. The main advantages of being binary are: \* Binary protocols are more efficient to parse by applications which does not have to handle with issues related to text-protocol parsing. \* It is more compact “on the wire” since no extra information needs to be sent. \* It is much less error-prone, compared to textual protocols like HTTP/1.x with whitespace handling, capitalization, line endings, blank links... HTTP/1.1 defines five different ways to parse a message; in HTTP/2, there’s just one code path.

### Fully multiplexed

HTTP/1.x suffers “head-of-line blocking” only one request can be outstanding on a connection at a time. Pipelining of request is not a solution since a large or slow response will block others behind it. Additionally, it has been found very difficult to deploy, because many intermediaries and servers do not process it correctly.

### One connection for parallelism

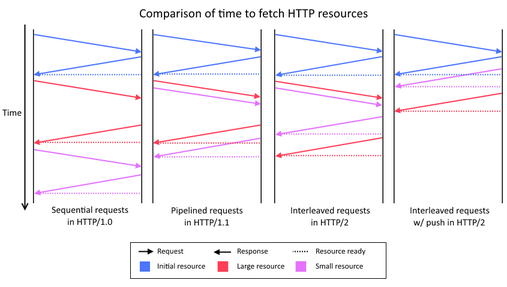
With HTTP/1, browsers open between 4 and 8 connections per origin. Since many sites use multiple origins, this could mean that a single page load opens more than 30 connections. One application opening so many connections simultaneously breaks a lot of the assumptions that TCP was built upon. Since each connection will start a flood of data in the response, there is a real risk that buffers in the intervening network will overflow, causing a congestion event and retransmits. Additionally, using so many connections unfairly monopolizes network resources, “stealing” them from other, better-behaved applications (e.g., VoIP).

### Header compression

HTTP/1 supported compression for content but not for headers. Assuming an average of 80 assets per page and each request has 1400 bytes of headers, it takes at least 7-8 round trips just to get the headers out “on the wire.” That is not counting response time - that is just to get them out of the client. Headers add a lot of overhead traffic and increase latency.

### Server push

In HTTP/2 the server can push resources to the client before receiving a request to server that resource. This reduce the load time as the browser does not have to send GET request to ask for all the resources avoiding RTT delays.



HTTP/2 Push

## Features of HTTP/2

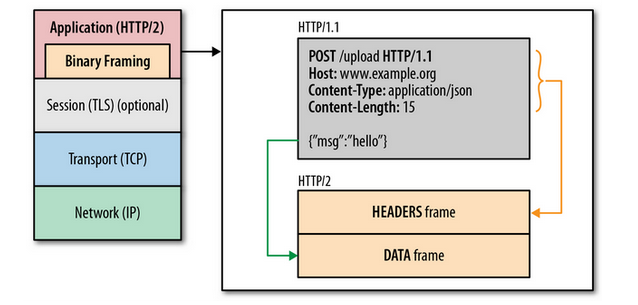
### HTTP/2 key concepts

1. All communication is performed over a single TCP connection that can carry any number of bidirectional streams.
2. Each stream has a unique identifier and optional priority information that is used to carry bidirectional messages.
3. Each message is a logical HTTP message, such as a request, or response, which consists of one or more frames.
4. The frame is the smallest unit of communication that carries a specific type of data—e.g., HTTP headers, message payload, and so on.
5. Frames from different streams may be interleaved and then reassembled via the embedded stream identifier in the header of each frame.

### Framing

The frame is the smallest unit of communication in HTTP/2, each containing a frame header, which at a minimum identifies the stream to which the frame belongs.

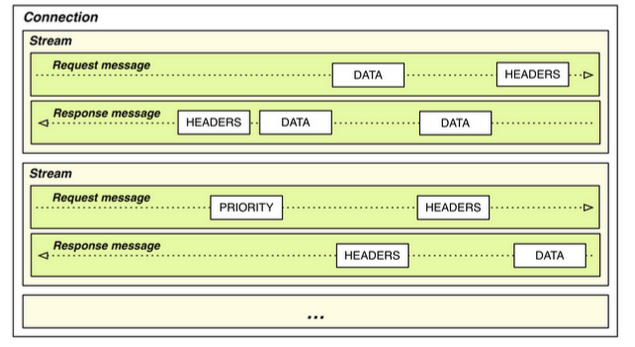
The most relevant frames are the used to transport Headers and Data, but there are also frames for other functions like stream control, push content and set the priority of the streams. DATA frames are used to transport HTTP message bodies and HEADERS frames are used to communicate header fields for a stream. The rest of frames are used for control tasks and they are called Control Frames.



HTTP/2 Framing

### Multiplexing

HTTP/2 interleave multiple requests and responses in parallel without blocking any one using a single connection to deliver and receive them. This allow to remove unnecessary HTTP/1.x workarounds such as concatenated files, image sprites, and domain sharding. It also resultes in lower page load times by eliminating unnecessary latency and improving utilization of available network capacity.



HTTP/2 Streams

### Priority

Not all the elements of a Web page are equally important for the functionality and visualization. An user can start consuming the content before loading all the elements. HTTP/2 allows to set different priorities to each stream and priority can be specified by the client side and it can be changed in runtime.

### Flow control

HTTP/2 applies flow control mechanism to the different streams. The flow control is directional with overall control provided by the receiver and the frame type determines whether flow control applies to a frame. Only DATA frames are subject to flow control (the most important part part of the traffic will transported in DATA frames) so control frames are not blocked by flow control. On the other side, flow control cannot be disabled according to the HTTP/2 RFC draft.

## HTTP/2 encryption

The RFC allows HTTP/2 to be sent over unencrypted transport protocols like TCP. However Google and Mozilla have officially said that HTTP/2 is not going to be allowd to be sent over plain TCP but over TLS connection to avoid issues with intermediates element between the browser and the server. This has an impact in ReTHINK project as if a web site is served over TLS or a secure protocol, the rest of elements of that site must be served using an encrypted protocol.

## Why using for HTTP/2 in ReTHINK project.

HTTP/2 is aimed to make the Web more efficient and responsive and it advantages are more notable for web sites. However the following points should be considered when discussing the use of HTTP/2 in the ReTHINK project: \* HTTP/2 is going to be massively adopted by Internet companies following Google and Twitter movements. \* The most relevant projects (Apache, NGINX, ) already supports or plan to support HTTP/2. ReTHINK Javscript libraries are expected to be used also in web applications so supporting HTTP/2 would avoid mixing HTTP/1.1 and HTTP/2. \* ReTHINK will take advantage of the benefits of HTTP/2. \* HTTP/2 is normally going to be used over QUIC so we will take advantage of using QUIC as the transport protocol. \* Despite the fact of not being used to transport media, its characteristics would make it possible.

All the available implementations of HTTP/2 are gathered in the official HTTP/2 WG Github repository: https://github.com/http2/http2-spec/wiki/Implementations

## Websocket over HTTP/2

The WebSocket protocol enables two-way communication between a client running untrusted code in a controlled environment to a remote host that has opted-in to communications from that code. Since it requires one TCP connection for every WebSocket connection, having multiple WebSocket connections between the same client and the same server is inefficient. On the other hand, HTTP/2 specifies a fast, secure, multiplexed framing protocol. The [draft!](http://tools.ietf.org/html/draft-hirano-httpbis-websocket-over-http2) document provides bi-directional multiplexed communication by layering WebSocket on top of HTTP/2. The document is still a draft but this could be used in ReTHINK project as it allows to get a bidirectional communication over an existing HTTP connection without the need of creating additional socket connections.

# QUIC (Quick UDP Internet Connection)

## Introduction

Internet has evolved a lot since the first version of TCP was designed 40 years ago. Many RFCs have added functionality to TCP since then and it played a vital role being the transport of the WWW for decades. UDP was designed to be used in real-time scenarios where the realiability is not so important as getting a very low delay and being tolerant to packet losses.

More flexible protocols which combine feature from TCP and UDP were also designed. The most remarkable effort was SCTP which was first designed to transport by the Signaling Transport Working Group to transport Signaling System 7 (SS7) and ISDN communications protocols over IP networks. Amongst other features, SCTP creates a tunnel over UDP and to multiplex a connection into multiple streams. It also provides notification of duplicated or missing data chunks, creating a reliable transport over UDP.  
The fact of allowing multiple streams over a single connections prevents Head-of-Line blocking from hapenning. It means that a packet loss which affect to a stream will not imply a delay in the rest of streams as it would happen if several streams are multiplexed over a TCP connection.

## Transport layer replacement

Google started the definition, implementation and real filed testing of a new transport protocol called QUIC in 2013. QUIC stands for "Quick UDP Internet Connection" and it was designed as a replacement for TCP as response of the new needs and the evolutions of Internet: streaming-based services are more and more demanded, many connections are established over wireless connections and they are established from mobile devices.

The features the new protocol must support to meet the today's Internet requirements: \* it is necessary to make the web faster and the low layer protocols must adapt to the evolution of the web applications. \* connection establishment latency must be reduced to improve the user experience and to make the web and Internet more usable. \* the web must be secyre by default, TCP does not secure the traffic and an optional upper layer must be added to provide this security.

QUIC introduces improvement in many aspects of the TCP protocol: \* TCP needs a three-handshake to create a connection. If the connection is established over TLS more RTT are necessary. \* TCP provides reliable, ordered and error-checked delivery based re-tx, acknowledgments and checksum. The cost of having this features is that any packet loss will trigger retransmissions which will normaly delay the delivery of the rest of packets. \* TCP allows to stablish a single full duplex byte stream and all the data over that stream will be treated processed indistinctly. \*TCP requires an extra protocol on top of it (TLS) to provide encryptin and authentication. It adds an overhead and delays in the connection setup.

### Why not to use SCTP for this?

SCTP can be considered as an alternative for TCP. It has two main features which could make it an atarctive choice:  
\* SCTP also provides stream multiplexing over a single connection. \* DTLS provides SSL quality encryption and authentication over UDP in the same way as TLS over TCP.

These features made that SCTP had been chosen to be used internally by WebRTC Datachannels however it presents some problems which forced the QUIC developer to design a new alternative:

* SCTP requires 4 roundtrips are necessary to establish a SCTP over DTLS connection. This means an unacceptable delay in many applications and degrades the user experience.
* It was not designed to reduce latency, SCTP connections were designed to be persistent between two peers.

In contrast the goal of QUIC is to have to perform a connection establishment with zero RTT overhead.

## QUIC current use

QUIC is currently being used by Chrome to interact with Google Apps and it is going to be used by more services in short-term. HTTP/2 is going to become a definitive RFC in a few months and it is being used in production by many relevant Internet companies such as Twitter and Facebook. HTTP/2 performs much better over QUIC since it allows to leverage it stream-based designed so QUIC isvery likely to be adopted by the IETF in short-term.

## QUIC main features

QUIC can be deployed in today's internet, actually it is been used in many real deployments without having to apply any modification in any intermediate node. This is a key point which makes its adoption much easier than IPv6.

It provides a very low latency in connections and responses. That is a critical point as many services are cloud -based and must be accessed from mobile and transoceanic connections which high RTTs.

It has a reliable-stream support and the packet losses which affect one strem does not affect the rest of streams. This reduces the Head Of Line (HOL) blocking due to packet loss.

It provides a better congestion avoidance than TCP which implcitly considered that any packet loss was due to a network congestion which is not normally true for wireless connections.

The privacy and Security it provides is comparable to TLS but with a much lower delay in the connection setup which can from 0 in the most optmistic cas to 2 RTTs in the worst case. This is a key feature of QUIC as it allows to improve the user experience.

Connection are identified by a Connection Identifier not by layer 3 and layer 4 elements (IPs and ports). This fits very well in mobile scenarios where the IP of the clients may change due to handover mechanism.

### Is it a good choice to use UDP for QUIC?

QUIC can be considered as an intelligent layer over UDP which provides enhanced features. UDP is intensively used by VoIP and gamers for years in very latency-sensitive applications. On the other side, 91-94% of the users which had TCP connectivity with Google can make outbound UDP connections so it is possible to build a transport in today's internet over UDP. It is also necesary to consider NAT unbinding which does not happen to TCP. This problem has been addressed internally by QUIC designers through the use of keepalives packets.

## Applicability to reThink project

Including the use of QUIC as a requirement or a recommendation could help to support more reliably mobility scenarios where the End-User IP may be changed during a connection with the Signaling service. Additionaly thee transport layer connectivity provided by QUIC is more suitable for wireless connections (longer RTTs, packet lost and changes at IP level) than TCP. QUIC has been designed bearing HTTP/2 in mind as it improves its performance a lot, however the use of QUIC by other protocols can also be very advantageous.

During a media session, the change of an IP requires an SDP re-negotiation when a media sessions is ongoing, so we can't leverage QUIC features for media. However QUIC would be helpful in all the scenarios at signaling level.

### Existing QUIC implementations

The QUIC reference library is libquic (https://github.com/devsisters/libquic). It has been mainly developed by Google. This repository and its sources and dependencies were extracted from Chromium's QUIC Implementation with a few modifications and patches to minimize dependencies needed to build QUIC library. This code can be used to be integrated with HTTP Server like Apache and nginx but this has not been done so far. This library implementes

In the Chromium repository it is available a standalone client and server which can be used as a reference for ReThink project implementations. http://src.chromium.org/viewvc/chrome/trunk/src/net/tools/quic/

In all the imeplementations QUIC is used with SPDY and HTTP/2 so its use separated from those protocols has to be investigated.

## Drawbacks of using QUIC as transport protocol

QUIC is a very new protocol so it is still not widely used. It means that many existing systems and projects does still not support it so testing and implementation which requires an additional effrot compared to TCP.

On the other side, despite the fact that is a protocol likely to become an RFC darft in short-term it has not been formally specified by the IETF. This is the official definition document mantained by Google: https://docs.google.com/document/d/1RNHkx\_VvKWyWg6Lr8SZ-saqsQx7rFV-ev2jRFUoVD34/edit Any implementation made today may not be completely compliant with the final protocol.

The reTHINK project describes a framework that provides solutions to manage real time communication capabilities. To implement this framework the project team tried to use the most suitable existing standards which provides compability which existing technoligies. Using consolidated and widely used standards also make the development more efficient since Open Source libraries can be used in the developments. Addtionally to well-known standards, the project team has also tried to find emerging standards which can be adapted for ReTHINK requirements. In those cases, a tradeoff analysis has been made to determine if the choice of a not consolidated standard is optimal in terms of cost of use due to the lack of existing libraries and projects which use them.

The IETF has been creating and promoting the Internet standards since 1986. The IETF is organized in a large number of Working Groups (WG) which works on specific areas. For ReTHINK project, the team has focused on standards delivered by several WG (namely Rtcweb, TRAM, HTTP/2 and Network). The Rtcweb WG has defined a set of RFCs (many of them are still drafts) which are used in WebRTC, it defines how WebRTC works on the wire. Many of the used protocols already existed but many of them were created ad-hoc to meet WebRTC requirements. Other RFCs are informational and hes been released to gather the WG knowledge in a formal way. The TRAM (TURN Revised and Modernized) working group is carrying out a modernization of the protocols used to transport real-time media over Internet which is the final function of ReTHINK framework.

HTTP/2 is the new version of HTTP/1.1 which has been used in the web for the last 16 years. It provides a new low level design to optimize current Web applications keeping the semmantic of HTTP/1.1 which is still valid. HTTP/1.1 has been historically transported over TCP, however to take advantage of all the new features of HTTP/2 a new transport protocol build over UDP has been designed: QUIC. HTTP/2 draft is based on SPDY but it includes new features and will soon become a definitive RFC. The draft belongs to the HTTP WG. QUIC was developed by Google but it has been recently become an IETF Draft taking over the last changes in the protocol until close the defintiive RFC. HTTP/2 over QUIC has been considered as an alternative for messaging in the ReTHINK framework as it is optimized to be used over wireless connection and minimizes the delay in every communication.

The IETF is in charge of standarizes all the protocols on the wire in Internet. In turn, the W3C (WWW Consortium) is the main international standards organization for the World Wide Web. It standarizes how the browser behave (e.g. WebRTC 1.0 API exposed by the browsers) and and the lenguages (e.g. HTML and Javascript) which can be executed by a standar browser. It is main role is to promote and homogenize the evolution of the Web. During the state of the Art research work we focused on the standards susceptible of being used by any element within the ReTHINK framework.

The WebRTC 1.0 API has been standarized by the W3C is the way in which a Javascript application interacts with the browser to establish real-time sessions with other WebRTC endpoints. A comprehensive knowledge of this API was necessary to make design decissions and to define the architecture and the data model of the framework.

A Community group has been created within the W3C to promote an alternative WebRTC API called ORTC (Object Real-Time Communications) which gives more control to the WebRTC developer making easier to implement some scenarios. There are still not implementations of ORTC in production-ready browser, however the features introduced by this standard which is likely to become the base of the WebRTC 2.0 API have been considered during the design phase.

Another relevant W3C API is the Push API which allows a push service to send "push messages" to a webapp regardless of whether the webapp is currently active on the user agent. This is specially usefull for webapps running on mobile devices where the webapp may need to receive a notification while the browser is not in foreground.

The use of another feature supported by browser called Service Workers has been already evaluated to be used to implement different parts of the Runtime environment. Despite the fact that this specification is still a Working Draft of the W3C it is already supported by the most important browsers. However, this is feature is not supported by server side Javascript-based runtime environment, it only can be used when the Runtime is executed by a browser.

There is another interesting W3C Draft called "Application Lifecycle and Events" which extends the Service Workers with APIs for managing the lifecycle of an application and associated events. This Draft allows web developers to create applications that manage the application lifecycle and react to system events e.g. email or VoIP application. However, this Draft has been not been adopted by many vendors so far.

In this section the standars released by the Open Mobile Alliance (OMA) were also reviewed. The OMA is a Mobile Operator driven industry forum for the definition of interoperable mobile service enablers. OMA defines APIs to offer functionalities and resources of Operator networks to developers. Amongst the API and protocols standarized by the OMA the team decided to reviewed those which are relevant for the project such as the Authorization Framework for Network APIs, the RESTful Network API for WebRTC Signaling, Quality of Service API and Notification Channel. The OMA LWM2M protocol for endpoint management is based on CoAP, designed to be supported by constrained devices has also been considered as a suitable alternative to interact with the Catalogue, Registry and Discovery services.

Finally, a recent standard the Smart Device Template (SDT) released by the HGI (Home Gateway Iniative) has been reviewed. It provides a framework to create a consistent representation of Smart Home devices. This makes easier the integration of new devices in Home Gateway or in the cloud being specially interesting to implement M2M within the ReTHINK framework.

# Projects SOTA

## WONDER Project

### Overview

The main motivation of the [OpenLab WONDER experimentation project](http://hypercomm.github.io/wonder/) [32] was to experiment and evaluate some of the P2252 Eurescom Study recommendations. WONDER evaluated whether to use IMS to deliver services to WebRTC endpoints or to use a more disruptive pure Web based approach to deliver services to WebRTC endpoints.

For the second option WONDER has enlightened some paths to be followed in a post-IMS era dominated by Web technologies and large eclectic cooperative eco-systems. The novel [Signalling On-the-fly (SigOfly)](http://ieeexplore.ieee.org/xpl/articleDetails.jsp?arnumber=7073799&filter%3DAND(p_IS_Number%3A7073795)) concept was conceived and successfully demonstrated to address the lack of a standard WebRTC signalling protocol. The SigOfly concept enables seamless interoperability between different WebRTC service domains avoiding NNI interfaces by using peer side APIs and restful push notification services.

WONDER experimentation involved Portugal Telecom and Deutsche Telekom.

### SigOfly Concept

The SigOfly concept leverages the use of scripts (JavaScript) by WebRTC Applications to implement signalling protocol stacks. This means, the signalling protocol stack can be selected, loaded and instantiated during runtime. Such characteristic enables signalling protocols to be selected per WebRTC Conversation to ensure full signalling interoperability among peers using Triangle based Network topologies. The SigOfly procedures should be applied at the end-user client to benefit from WebRTC model. However, the concept is also feasible between Messaging Servers supporting Javascript execution engines (e.g. nodejs or vertx.io).

Before the SigOfly concept is described in detail, some terms require a definition:

**Messaging Server:** the server that supports the exchange of signalling messages required for the establishment of WebRTC sessions. Each Messaging Server belongs to a domain;

**Domain Channel:** the signalling channel that is established with domain’s messaging server as soon as a domain‘s user is registered and is online;

**Transient Channel:** the signalling channel that is established, typically with a foreign messaging server (i.e. from another domain) in scope of a inter-domain conversation;

**Messaging Stub:** the script containing the protocol stack and all the logic needed to establish a channel to a certain Messaging Server;

**Conversation Hosting Messaging Server:** is the Messaging Server that is used to support the exchange of all signalling messages among peers belonging to different domains. The Hosting peer uses the Domain Channel to exchange signalling messages, while other peers use Transient Channels that connect to the Hosting Messaging Server.

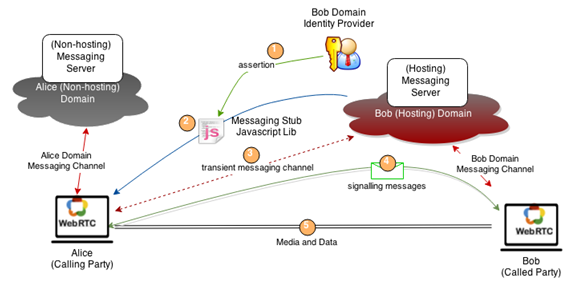


Figure (**???**) signalling on-the-fly concept

The classic Alice and Bob example is used to explain the SigOfly concept. We assume that Alice and Bob are registered in different Service Provider domains having each one a Domain Channel established with their own Messaging Server (see Fig. below). In case Alice wants to talk to Bob by using Bob’s WebRTC identity e.g. bob@domain.com, the following steps will be performed:

Step 1: Information about the Identity of Bob, including Bob’s Messaging Stub provider, is provided and asserted by Bob’s Identity Provider (IdP).

Step 2: Alice downloads and instantiates Bob’s Messaging Stub in her browser to setup a Transient Channel with Bob’s domain Messaging Server.

Step 3: As soon as the Transient Channel is established, Alice can send an Invitation message to Bob containing her Session Description Protocol (SDP) based communication offer.

Step 4: Since Bob is connected in the same Message Server via his Domain Channel, he will receive Alice’s invitation in his Browser. If Bob accepts the invitation, an Accepted message containing Bob SDP response will be send to Alice.

Step 5: As soon as Alice’s browser receives Bob’s SDP, the media and/or data streams can be directly connected between the two browse

It should be noted that SigOfly does not directly address identity management aspects but aims to be compliant with ongoing WebRTC Identity Management work from W3C and IETF, mainly by extending RTCIdentityAssertion to also include the assertion of MessagingStubs. This means, Alice and Bob authentication is done outside SigOfly procedures, described above, which are agnostic of the IdP and authentication protocols used.

The SigOfly concept is also applicable in use cases where conversations are hosted by calling parties, in multiparty conversations or to support interoperability with legacy networks (e.g. IMS and PSTN).

#### Data Codec On-the-fly

In addition, the SigOfly principle is also applied to address Data Channel Services Interoperability, to conceive another novel concept called “Data Codec On-the-fly”. the “Data Codec On-the-fly” concept ensure all peers are using the same protocol on top of the Data Channel. A Data Codec is a JavaScript library that implements a data communication processing algorithm to code and send data to the Data Channel and to receive and decode data from the Data Channel. As with Messaging Stubs in the SigOfly concept, Data Codecs are downloaded and instantiated by peers according to URLs identified in Conversation setup or update signalling messages

Further information about SigOfly and Data Codec On-the-fly is provided in [34](https://github.com/hypercomm/wonder/wiki/Signalling-on-the-fly).

### WONDER Library

A JavaScript framework, the WONDER lib, was designed and implemented to validate the SigOfly and Data Codec On-the-fly concepts.

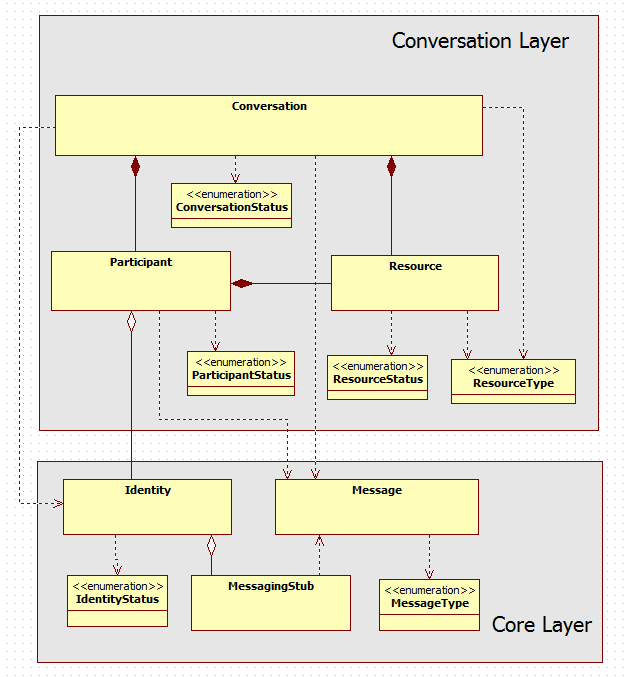


Figure (**???**) Wonder Library Main Classes

Main WONDER library classes are:

* The Identity, representing a user and containing all information needed to support Conversation services. This also includes the service endpoint to retrieve the protocol stack (Messaging Stub) which will be used to establish a signalling channel with the Messaging Server of the Identity domain. The Identity entity extends the current Identity concept defined by W3C to support seamless interoperability by using the SigOfly mechanism.
* The MessagingStub implementing the protocol Stack used to communicate with a certain Messaging server.
* The Conversation class managing all participants including the setup, update or close of media and data connections.
* The Participant class, handling all operations needed to manage the participation of an Identity (User) in a conversation including the WebRTC PeerConnection functionalities. The Local Participant is associated with the Identity that is using the Browser while the Remote Participant is associated to remote Identities (users) involved in the conversation.
* The Resource class representing the digital assets that are shared among participants in the conversation including participants’ voice, video, screens, photos, video Clips, music clips, documents, etc. These assets are usually managed by the Participant that owns it. For local participants assets are sent (e.g. WebRTC outgoing stream tracks) while for remote participants assets are received (e.g. WebRTC incoming stream tracks). Some Resource types like Chat are not managed by a Participant but by the Conversation.
* The Data Codec, which is used by Resources that are shared on top of the Data Channel, like file sharing and Textual Chat, to decode and encode the data in a consistent way by all the peers. The Data Codec may also be downloaded on-the-fly by the peers.
* The Message, which is used to exchange all data needed to setup, update and close media and data connection between peers via the Messaging Server. It may also be used for other purposes e.g. presence information management. Each message is comprised by a Header and a Body.

The [API documentation](https://raw.githack.com/hypercomm/wonder/master/docs/api/index.html) and the [source code](https://github.com/hypercomm/wonder/tree/master/src/libs) were published in a public github repository [35][36] .

### Input to reTHINK

WONDER Library can be used in reTHINK in different ways that are detailed in the following sections.

#### Runtime Messaging API

The MessagingStub API is a good starting point to design the Javascript API to support Hyperty Message communication which is comprised by the following functions:

connect(ownRtcIdentity, credentials, callbackFunction) {  
// connects to Msg Server  
};  
  
addListener(StubEvtHandler, URI, contextId){  
//Adds a listener for a certain context.  
};  
  
removeListener(StubEvtHandler, URI, contextId){  
//Removes a listener from a certain context.  
};  
  
getListeners(){  
//Gets the list of listeners.  
};  
  
sendMessage = function(message) {  
 // send Message  
};  
  
disconnect = function() {  
 // disconnects from server  
};

Check WONDER MessagingStub API documentation in [37](https://raw.githack.com/hypercomm/wonder/master/docs/api/symbols/MessagingStub.html).

#### Messages Format

The WONDER Message class provides good input for the design of Hyperty Messages. Wonder Message is a JSON structure and it is comprised by a Header and a Body. The following Message Header attributes are defined:

type Type of the Message   
from Sender of the message  
to Recipients of the message  
context identifies a certain context for the message eg the Id of the conversation

The following Message Types are defined:

* INVITATION - Message to invite a peer to a conversation.
* ACCEPTED - Answer for conversation accepted or Context subscription accepted
* CONNECTIVITY\_CANDIDATE - Message contains ICE connectivity candidate
* NOT\_ACCEPTED - Answer for conversation not accepted or Context subscription not accepted
* BYE - Message to finish the communication with a peer
* UPDATE - Message to Update conversation by adding or removing a Resource
* UPDATED - Answer to Message UPDATE
* CONTEXT - Message used to publish the context and status of an Identity.
* SUBSCRIBE - Message to request to receive CONTEXT notifications from a certain Identity
* MESSAGE - Mainly used to support Pager Mode Chat. But it can be used for other use cases instead of Data Channel eg small files.
* CRUD\_OPERATION - Messages to handle data persistence in a resource tree

The Message body will depend on the Message Type. Detailed description of WONDER Messages are provided [here](wonder-messages.md).

#### Runtime Identity API

The Identity and IDP classes could also provide good input for the support of Identity Management functionalities by reTHINK Runtime.

The WONDER IDP is a singleton object which handles WONDER Identities creation and retrieval from IDP Servers. IDP is agnostic of the protocol used by IDP Server. IDP main function is:

/\*This method takes either a single rtcIdentity or an array of rtcIdentities and creates Identity objects from them. The successfully created Identities are then returned in an Array in the success callback. If one or more rtcIdentities can't be created then the returned array is shorter than the given array.\*/  
  
createIdentities(rtcIdentities, onSuccessCallback, onErrorCallback)

The WONDER Identity class represents a user and contains all information needed to support a Conversation service including the service endpoint to retrieve the protocol stack (Messaging Stub) that will be used to establish a signalling channel with the Identity domain messaging server. Identities are only created by using the corresponding create-methods of the IDP class.

Identity is comprised by the following attributes:

context; // including Identity presence status;  
sessionId // identification of the session established with the domain through the Messaging Server.  
username;  
messagingStubLibUrl; // the service URL from where Identity domain messagingStub can be downloaded  
notificationAddress; // to support notifications when the user is not connected  
credentials; // to be used to connect to the domain  
avatar; //

Some of these attributes that are more sensitive in terms of security should be handled by the Runtime itself (e.g. credentials / tokens, sessionId, username, messagingStubLibUrl) while the others could be handled by an Hyperty representing the Identity eg avatar, context.

The main WONDER Identity functions are:

/\*\*  
 \* This method downloads a messaging stub and keeps a reference to it in a local  
 \* attribute, if not already done before. That means the download will only be performed once.  
 \* After download it invokes the given callback with a reference to the downloaded MessagingStub.  
 \*   
 \* @param callback {callback(MessagingStub)} callback that is invoked with messagingStub as param; if download failed then the stub param is empty  
 \*/  
Identity.prototype.resolve = function( callback ) {  
};  
  
/\*\*   
 \* This method subscribes to add a listener to receive status information (CONTEXT message type) from the user associated to this Identity.   
 \* The Signalling on the fly concept is also used to ensure cross domain Presence management interoperability  
 \* by calling the Identiy.resolve() function  
 \* @param subscriber :  
 \* Identity ... The identity of the subscriber  
 \* @param type :  
 \* SubscriptionType ... The subscription type  
 \*  
 \*/  
  
Identity.prototype.subscribe = function(subscriber) {  
};  
  
/\*\*   
 \*   
 \* This method removes a listener previously added with "subscribe()" function to receive status information   
 \* (CONTEXT message type) from the user associated to this Identity  
 \* @param subscriber :  
 \* Identity ... The identity of the subscriber  
 \* @param type :  
 \* SubscriptionType ... The subscription type  
 \*  
 \*/  
Identity.prototype.unsubscribe = function(subscriber, type) {  
};  
  
/\*\*  
 \* To set Identity context and to publish it by sending a CONTEXT message to address "rtcIdentity.context"  
 \*   
 \* @param context :  
 \* String ... The context to set  
 \*   
 \*/  
   
Identity.prototype.setContext = function(context) {  
};  
  
/\*\*  
 \* getContext  
 \*   
 \* @returns ContextData ... gets the context attribute for this Identity  
 \*   
 \*/  
Identity.prototype.getContext = function() {  
};  
  
/\*Handler to receive incoming messages eg context update\*/  
  
Identity.prototype.onMessage = function(message){  
}  
  
/\*\*  
 \* addListener function usually implemented by the App  
 \* @param.. listener  
 \*/  
  
Identity.prototype.addListener = function( listener, rtcIdentity ){  
}  
  
/\*\*  
 \* removeListener function  
 \* @param.. listener  
 \*/  
  
Identity.prototype.removeListener = function( listener, rtcIdentity ){  
}

The "resolve(..)" function should be handled by the Runtime itself but all the others could be implemented by an Hyperty representing the Identity. For further discussion.

#### Javascript Framework

WONDER library can provide some input for the design and implementation of reTHINK Javascript framework that should facilitate the development of Hyperties, namely: \* Conversation \* Participant \* Resource \* Identity \* MessageFactory

#### Javascript Shim Layer for non-compliant reTHINK Runtime

WONDER library can provide some input for the design and implementation of reTHINK Javascript Shim Layer to be used in non-compliant reTHINK Runtime, namely:

* MessagingStub
* Idp
* Identity
* DataBroker
* DataCodec

## CoAP/ Well-known CoRE Projects

If Hyperties, Codecs, Protostub and other artifacts to be provisioned on the end devices are to be regarded as resources with attributes describing: capabilities (audio, video, text), running platform (OS), configuration (DNS name of the messaging node, DNS server), implementation (code/script, codecs), these artifacts can be organized as resources in the Repository/Catalogue component.

### LibCoap Project

#### Overview

The implementation includes features for receiving and sending CoAP requests. It also supports the [CoRE-link format RFC 6690] (https://tools.ietf.org/html/rfc6690) to organize the CoAP resources as a well-known CORE. It has support for Linux but also Contiki Operating Systems.

The implementation is using C as programming language.

Link: https://gitlab.informatik.uni-bremen.de/bergmann/libcoap/tree/master

The library is published as open-source software without any warranty of any kind. Use is permitted under the terms of the GNU General Public License (GPL), Version 2 or higher, OR the revised BSD license.

#### How to use

For starting a CoAP server based on a configuration file, a main program has to be written. Handlers for CRUD operations triggered by CoAP requests: Post, Get, Put and Delete can be registered to the main information named coap\_context. Callbacks will be generated to the registered handlers when the requests or replies are received. When creating resources, attributes can be associated. The attributes are then XML encoded when Get messages are received. A command line application, example of code and ETSI tests are included.

### Copper (Cu) CoAP user-agent Project

#### Overview

The CoAP User Agent is a JavaScript implemention of [Constrained Application Protocol (CoAP) RFC 7252] (http://tools.ietf.org/html/rfc7252) with support for DTLS, Observe and blockwise transfers. A plugin for Mozilla is also included. The project is available on github at: https://github.com/mkovatsc/Copper

The license is 3-Clause BSD with the text available at: http://opensource.org/licenses/BSD-3-Clause, and permits redistribution.

#### How to use

The JavaScript code can be used directly in other JavaScript components.

### Californium Project

#### Overview

The project implements CoAP RFC 7152 with DTLS, a CoAP-HTTP translator. The implementation is using Java as programming language and is designed for IoT Cloud services with the focus on scalability and usability instead of resource-efficiency like for embedded devices.

The project is available on github at: https://github.com/eclipse/californium The license is business-friendly and of type Eclipse Distribution License, available at http://www.eclipse.org/org/documents/edl-v10.html.

#### How to use

It can be used as a CoAP server that supports all CRUD operations and Observe/Notification mechanism. For example for the Catalogue it would make sense to use it.

## OMA Device Management Projects

For exchanging information on the device properties and also monitor/manage connectivity of the device, [OMA LWM2M standard] (http://member.openmobilealliance.org/ftp/Public\_documents/DM/LightweightM2M/) can be used, as an energy efficient and scalable evolution from OMA DM standard.

Several projects have been analyzed in terms of features, flexibility and license in order to be able to choose the most suitable for the Rethink project.

### Leshan Project

The project is supported by Sierra Wireless and hosted by the Eclipse foundation.

#### Overview

The implementation supports all the interfaces: Bootstrap, Registration, Device Management and Service Enablement, Information Reporting.

The project uses Java as programming language.

The project is hosted on github at: https://github.com/eclipse/leshan

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#### How to use

The project can be further extended with some new Management Objects, if necessary. The server is to be compiled and run according to a configuration file.

### OMA LWM2M Dev Kit Project

The project can act as multiple virtual OMA LWM2M clients by connecting to a remote OMA LWM2M server.

#### Overview

The supported features include interfaces: Registration, Device Management and Service Enablement Interface, Information Reporting Interface.

The programming language is Javascript.

The homepage can be found on: https://github.com/OpenMobileAlliance/OMA-LWM2M-DevKit

The license is [BSD-like](https://github.com/OpenMobileAlliance/OMA-LWM2M-DevKit/blob/master/LICENSE), the name of the project has to be mentioned in the redistribution.

#### How to use

The project provides a Web GUI as an addon for Firefox to get the user/developer familiarize. The core functionality can be included in any software package like a Javascript library and also, the Core Framework of the client side for the Rethink project.

## ETSI and oneM2M Projects

### OM2M Project

#### Overview

The project is developed under the Eclipse umbrella and is described at: http://www.eclipse.org/proposals/technology.om2m/ and available at: http://www.eclipse.org/om2m/

The project is a Java implementation of the ETSI M2M standard in version 0.8.x, available at the moment. It aims to implement also oneM2M in version 1.x.x, the compatibility with ETSI M2M will not be included.

The current version has support for both CoAP and HTTP. For the OMA-DM support it uses SyncML files and can interoperate with [Funambol](http://sourceforge.net/projects/funambol/files/) as server and [Koneki](http://www.eclipse.org/koneki/omadm-simulator/), the OMA-DM simulator for firmware update operation.

The code is licensed using [Eclipse Public License](https://www.eclipse.org/legal/epl-v10.html)

#### How to use

The project could be used to store data coming from sensors or smart devices and expose it to applications.

### WONDER Messages Format

The WONDER Message class provides good input for the design of Hyperty Messages. Wonder Message is a JSON structure and it is comprised by a Header and a Body. The following Message Header attributes are defined:

type Type of the Message   
from Sender of the message  
to Recipients of the message  
context identifies a certain context for the message eg the Id of the conversation

The following Message Types are defined:

* INVITATION - Message to invite a peer to a conversation.
* ACCEPTED - Answer for conversation accepted or Context subscription accepted
* CONNECTIVITY\_CANDIDATE - Message contains ICE connectivity candidate
* NOT\_ACCEPTED - Answer for conversation not accepted or Context subscription not accepted
* BYE - Message to finish the communication with a peer
* UPDATE - Message to Update conversation by adding or removing a Resource
* UPDATED - Answer to Message UPDATE
* CONTEXT - Message used to publish the context and status of an Identity.
* SUBSCRIBE - Message to request to receive CONTEXT notifications from a certain Identity
* MESSAGE - Mainly used to support Pager Mode Chat. But it can be used for other use cases instead of Data Channel eg small files.
* CRUD\_OPERATION - Messages to handle data persistence in a resource tree

The Message body will depend on the Message Type. Some of these messages and associated bodies are more detailed below.

##### Invitation Message Type

Invitation for a new conversation to be hosted by the inviting identity ie to use Messaging Server of the inviting identity which is provided in the message body as well as the connection description of the inviting identity.

**Invitation Message Body**

conversationURL;  
 connectionDescription; // SDP  
 subject;  
 hosting; // Identity of who is hosting the conversation  
 agenda;  
 peers;  
 constraints; // To describe media and data constraints for each resource including Audio, Video constraints and direction (in,out,inout)

##### Accepted Message Type

To accept eg Invitations, Conversation updates or Context Subscription. Similar to SIP 200 OK

**Accepted Message Body**

connectionDescription; // SDP  
 hosting; // Identity of who is hosting the conversation  
 constraints; // To describe media and data constraints for each resource including Audio, Video constraints and direction (in,out,inout)

##### Not Accepted

Eg Busy, Reject, No\_answer to: - Invitation requests - Update requests - Subscription requests

This information will go in the message body as a String

##### CONNECTIVITY\_CANDIDATE Message Type

Messages used to exchange ICE connectivity candidates between peers

**Message Body**

label - The label of the candidate.  
id - The id of the candidate.  
candidate - The ICE candidate string.  
lastCandidate - Boolean indicating if the candidate is the last one. If true, include the full SDP in the candidate parameter for compatibility with domains that don't support trickling.

##### CRUD\_OPERATION

These Messages are used to handle data persistence in a resource tree by using the four basic functions create, read, update and delete.

**Message Body**

operation // create, read, update or delete.  
syntax // syntax used for CRUD operation field "criteria" examples: mongoDB, SQL  
criteria // some filtering expression used in read and update operations  
doc // Contains data for CREATE and UPDATE operations  
resource; // Resource URI where the operation is applied

# Runtime SOTA

## WebRTC.org

[WebRTC.org](http://www.webrtc.org/)[18] is an open-source project aiming at allowing developers to write applications bringing real-time communication capabilities to browsers, mobile platforms and Internet of Things (IoT) devices, without installing proprietary plugins or extensions. These challenge of integrating these different systems is leveraged by the definition of simple cross-platform APIs.

WebRTC comes with a native code package for developers to work over. This package features audio, video and network transport components. The audio component comes with a complete software stack for voice communications that includes not only codecs, but also software to help in communications' noise reduction, echo cancellation, automatic gain control, between others. The video component is built over the VP8 codec and comes with software for cleaning up noisy images, leveraging packet loss in transmissions and also record/playback functionality. Finally, the network package features components to establish P2P connections using ICE/Turn/STUN/RTP-over-TCP, and also software for error stashing on audio and video communications. Also, WebRTC provides browser developers the ability to choose their own audio, video and network protocols, to work with the packaged software.

### Architecture

WebRTC architecture offers two different layers, one for browser developers and other for third-party application developers. The first one is a C++ API intended to enable the proposed Web API for video/audio capture and render, making it possible for application developers to make use of it. The second one is the Web API for developers to produce applications to interact with WebRTC-powered browsers. Currently, several JavaScript APIs are in process of standardization, like [WebRTC 1.0](http://w3c.github.io/webrtc-pc/)[16] and [Media Capture and Streams](http://w3c.github.io/mediacapture-main/)[17]. In fact, there is another abstract layer responsible for session management and signalling, leaving the signalling protocol implementation up to the application developer, who has to choose between currently existing alternatives.

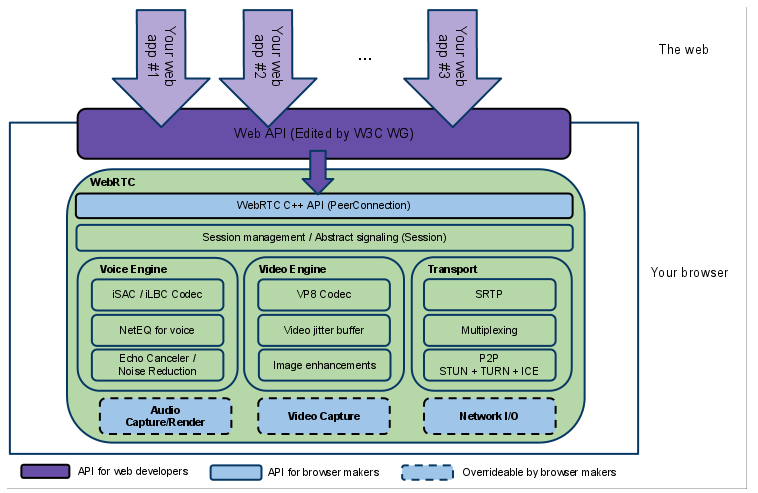


Figure (**???**) WebRTC.org architecture scheme

### Software stack organization

As explained before, WebRTC.org comes with a software stack that splits itself into a network package, an audio package and a video package.

#### Packages identification

The different packages can be easily identifiable on the WebRTC source code tree. The network package is under src/net and both the audio and video packages are under src/webrtc/, with some mixed up classes. Also, there is not a class diagram which helps developers to get the big picture on this code's organization.

### Code documentation

In the audio/video package almost every file is well-documented. However, the network package doesn't, and it even comes with a README file whose content just states that code documentation is a TODO task on the network package.

### WebRTC.org and runtime requirements

1. WebRTC is intended to be used on latest browser like Google Chrome, Mozilla Firefox, mobile platforms like Android and iOS and also IoT devices like Raspberry Pi.
2. WebRTC.org implements the W3C WebRTC APIs.
3. Yes, both the WebRTC 1.0 and Media Capture and Streams APIs use ECMAScript.
4. The WebRTC 1.0 API, and concretely its Peer-to-peer Data API for sending and receiving data models the behaviour of WebSockets
5. Yes, WebRTC 1.0 supports Web Messaging Notifications.
6. TODO
7. TODO
8. The effort to perform changes in the runtime like protocols for network I/O, signalling, session management, video capture and audio capture/render depends on the package these changes are meant to be inserted. The audio and video package is well-documented, despite not having a class diagram. The network package, by its turn, is not documented, increasing the effort to understand the functionality and to perform changes in the runtime.

## OpenWebRTC

[OpenWebRTC](http://www.openwebrtc.org/)[19] in another open source reference implementation of WebRTC standard [16][17] that can be used to build native WebRTC apps that communicate with browsers that supports the WebRTC standard, such as Chrome, Firefox and Bowser. OpenWebRTC is especially focused on mobile platforms, with powerful features such as hardware accelerated video coding and OpenGL-based video rendering.

OpenWebRTC architecture is highly modular, enabling easy modifications and possible extensions. Since the WebRTC standard is still evolving, that flexibility is a very desirable trait.

OpenWebRTC is built to support cross-platform operation, supporting IOS, Android, MAC OS and Linux. It is expected that Windows will be supported soon.

It is built on the expectation that several browsers will be able to support its operation. The bulk of the API layer is implemented in JavaScript, making it super fast to modify and extend with new functionality. This is expecially important on mobile platforms with its rapid developping application environment.

With support for both H.264 (OpenH264) and VP8 (libvpx) video codecs, OpenWebRTC is compatible with most video communication services.

The OpenWebRTC project is free and Open Source with a permissive BSD-2 license.

Figure (**???**) OpenWebRTC Architecture

Figure (**???**) OpenWebRTC Architecture

OpenWebRTC is built on top of widely used and powerful [GStreamer multimedia framework](http://gstreamer.freedesktop.org/)[20]. GStreamer is a set of libraries and plugins that can be used to implement various multimedia applications ranging from desktop players, audio/video recorders, multimedia servers, transcoders, etc. Applications are built by constructing a pipeline composed of elements. An element is an object that performs some action on a multimedia stream such as:

* read a file
* decode or encode between formats
* capture from a hardware device
* render to a hardware device
* mix or multiplex multiple streams

Unfortunately, OpenWebRTC has very few available information making it hard to use and extend it.

## V8 Javascript Engine Evaluation

### Overview

The [V8 JavaScript Engine](https://developers.google.com/v8/) [21] is an open source JavaScript engine developed by Google for the Google Chrome web browser.

V8 compiles JavaScript to native machine code (IA-32, x86-64, ARM, or MIPS ISAs)before executing it, instead of more traditional techniques such as interpreting bytecode or compiling the whole program to machine code and executing it from a filesystem. The compiled code is additionally optimized (and re-optimized) dynamically at runtime, based on heuristics of the code's execution profile. Optimization techniques used include inlining, elision of expensive runtime properties, and inline caching, among many others

### Architecture

Figure (**???**) V8 Architecture

Figure (**???**) V8 Architecture

**Handles & Garbage Collection**

Handles represent a reference for a Javascript object location on the process heap. The Garbage collector deletes any object on the heap with no valid reference on the process. The Garbage collector besides deleting objectws on the heap frequently moves objects and updates all references to those objects. Obviously the Garbage Collector does not operate often, but from time to time it deletes all obsolete objects. Handles may come in different flavours inside v8, ranging from local handles, which have limited scope and terminate when the scope finishes, therefore susceptible to garbage collection, to persistent and even eternal scopes. In fact we can look on scopes as handle containers. Each time a scope terminates the objects refered by the handlers, in it residing, are flagged for collection. We always have to be in mind that an handle cannot survive its default scope, unless we predetermine its scope to be a special one (EscapableHandleScope ).

**Contexts**

Figure (**???**) V8 Multiple Contexts

Figure (**???**) V8 Multiple Contexts

Contexts are different execution environments that allow separate even unrelated Javascript applications to run concurrently on v8. In fact, the context in which a Javascript code is run must be explicitly specified. This happens because Javascript provides functions and objects that may be changed globally and that may turn into unexpected results. One of the advantages of V8 is that it gives you an extensive cache, so in the first time a context may be expensive in time and resources, subsequente times will be substantialy less. Additionally v8 has a snapshot feature that by default has pre-compiled Javascript code on the heap, diminishing time procedures on first context initialization.

**Templates**

Templates are blueprints for Javascript functions and objects in a context. Templates may be used to wrap c++ code onto Javascript objects permiting its manipulation. One can only have one instance of a template on any given context. There are two types of templates:

* function templates - blueprint for a function;
* object templates - each template has associated an object

**Accessors**

Accessors are c++ callbacks that obtain and return a value when an object property is accessed by Javascript. Obviously then can be used to set or read these values. The complexity of them depends on the data being manipulated (Static Global Variables or Dynamic Variables).

**Interceptors**

Interceptors are callbacks used to permit access to an object property. They can be: named property interceptors - when accessing by string names; indexed property interceptors - when the access is made by index.

**Exceptions**

v8 throws exceptions when an error occurs. In fact v8 returns an empty handle on an unsuccessfull call.

**Inheritance**

While Javascript is a class free language, c++ has classes and instances. It is important to take this in consideration because Javascript only has objects, it is a prototype based language. To adapt both we have to refer to templates in v8.

**V8 Code provided for Javascript processing**

process.cc - this code provides the capability to extend the proccess of an HTTP request. The Javascript argument must provide a method named Process() for the execution to succed. This provides an interface for HTTP Javascript introduction on V8 and runtime execution.

shell.cc - this code takes as argument a filename with a Javascript code inside and executes it. It extendes several functionalities to Javascript including a shell capability to run Javascript snipets and their disponibilization to other Javascript code in runtime.

### Requirements Analysis

Analysis against [Hyperty Runtime Requirements](https://github.com/reTHINK-project/core-framework/labels/Runtime%20Requirement) (section ?)

#### [Runtime Performance](https://github.com/reTHINK-project/core-framework/issues/6)

Its aparently clear that V8 provides a significant improvement over previously adopted JavaScript interpretation engines like:

* JScript from IExplorer;
* SpiderMonkey (in Firefox);
* JavaScriptCore (in Safari).

The amount of the improvements will depend on the multiplicity of the calls made to implemented methods. If the methods are made to be run only once the gains would be minimal, otherwise the gains will improve exponentially.

The reasons for these obtained improvements are:

* Fast Property Access - unlike strong type languages like C# and Java, Javascript like Python is a dynamic programing language. This means that properties can be added to and deleted from objects on the fly, so likelly to change over time. Most Javascript engines use a dictionary-like data structure as storage for the object properties. The fetching of each property, on access case, involves a dynamic lookup of the property memory location. This approach turns these accesses much slower than accesses in strong type languages. In these languages, the instance variables are located at fixed offsets determined at compile time due to the fixed layout of objects defined by the object's class. In fact, objects are obtained and stored frequently with only a single instruction. V8 does not use dynamic lookup of properties. It creates hidden classes behind the scenes. Each time a change of property occurs in an object a new hidden class is created and the object changes its representative class for the new hidden class. The hierarchy of hidden classes is mantained and shared each time a new object of the refered type is used again.This type of behaviour promotes reuse by sharing off the hierarchy of hidden classes therefore avoiding dictionary lookups and eficiency by the inline caching of the classes in use.
* Dynamic Machine Code Generation - V8 generates machine code directly from source code the first time the script is executed. A current Javascript engine usually creates intermediate byte code and interpreter. The consequence thus is an object property access is handled with inline cache code in execution that may be patched with further instructions on execution. It may be explained by the execution of an access to an object property, V8 retrieves its associated hidden class and optimizes all future property accesses using this template, providing they share the same scope. This information is used in code patching of the inline cach code. If the V8 has gessed right, the property value is fetched in one operation, otherwise V8 patches the code to remove the optimization. This kind of optimization mirrors the beneficts of static languages and achieves most benefits the more accesses to properties from an object in an wider scope.
* Efficient Garbage Collection - V8, like most garbage collecting languages, reclaims memory used by objects that are no longer used in a process. Obviously garbage collection has known problems like memory fragmentation, pauses for garbage collection and fast object allocation. To avoid those problems as much as possible:
* stops the program execution when in a cycle of garbage collection;
* slices the object heap and only operates on part of it during a collecting cycle - lesses the time the application is stopped;
* correct identification of objects and pointers in memory, avoiding memory leaks by wromg identification.

The V8 separates the heap in two distinct parts. The new space is where new objects are created and the old space where objects surviving a garbage collection cycle are promoted. V8 actualizes references when each cycle finishes.

#### [How to extend and to introduce new Features](https://github.com/reTHINK-project/core-framework/issues/8)

It is possible to extend the functionalities of V8 by adding new modules in c++. These new functionalities would be available to any programer in Javascript where this particular v8 engine resides. V8 provides functions that permit accessing c++ methods and classes, handling errors and enabling security checks. It provides full duality, in which it permits access from javascript scripts to c++ structures an vice-versa.

Code to add a new Javascript code to V8

Handle<Value> Include(const Arguments& args) {  
 for (int i = 0; i < args.Length(); i++) {  
 String::Utf8Value str(args[i]);  
  
 std::string js\_file = load\_file(\*str);  
  
 if(js\_file.length() > 0) {  
 Handle<String> source = String::New(js\_file.c\_str());  
 Handle<Script> script = Script::Compile(source);  
 return script->Run();  
 }  
 }  
 return Undefined();  
}  
  
Handle<ObjectTemplate> global = ObjectTemplate::New();  
  
global->Set(String::New("include"), FunctionTemplate::New(Include));

Obviouly we also have to implement load\_file(). It obtains in string format the content of a file.

#### [Runtime Security](https://github.com/reTHINK-project/core-framework/labels/Runtime%20Requirement)

The "Same Origin Policy" is applied and in fact prevents one document from changing the properties of another. This means one document has the same origin when protocol, domain name and port are the same. This provides a usefull protection against malicious alterations. In v8 origin is defined as its context. To access other context it is necessary to use security tokens and callbacks. The security token are generated by v8 for each context created. when security tokens are not equal a callback must be made to challenge acceptable access.

#### Using Sandboxes with Node.js

[Node.js](https://nodejs.org/en/) [22] is a platform built on Chrome's JavaScript runtime V8 for easily building fast, scalable network applications. Node.js uses an event-driven, non-blocking I/O model that makes it lightweight and efficient, perfect for data-intensive real-time applications that run across distributed devices.

It is open source and specialized for server-side networking applications. Node.js operates on a single thread, using non-blocking I/O calls, allowing it to support tens of thousands of concurrent connections without incurring the cost of thread context-switching. The design of sharing a single thread between all the requests means it can be used to build highly concurrent applications. The design goal of a Node.js application is that any function performing I/O must use a callback.

Used to execute untrusted code. Has support for timeouts preventing infinite loops. Handles errors gracefully. Provides limited access to node.js methods, Supports print and console.log.

**Using Docker to run unsafe code on Node.js**

[Docker](https://www.docker.com/) [23] has a following on its own. Obviously security over docker is a well addressed issue.

#### [Web Messaging Notifications](https://github.com/reTHINK-project/core-framework/issues/5)

Implemented on WebKit on Chromium, so not applied directly on v8.

#### [Web Sockets](https://github.com/reTHINK-project/core-framework/issues/4)

V8 does not have an implementation of web sockets per si. There is an implementation of Web sockets in Chromium, and node.js seems to have an implementation also. It is not considered an issue for v8.

## Firefox OS

### Overview

[Firefox OS](https://www.mozilla.org/en-US/firefox/os/2.0/) (project name: Boot to Gecko, also known as B2G) [24] is a Linux kernel-based open-source operating system for smartphones, tablet computers and smart TVs, developed by Mozilla.

Firefox OS is designed to provide a complete, community-based alternative system for mobile devices, using open standards and approaches such as HTML5 applications, JavaScript, a robust privilege model, open web APIs to communicate directly with cellphone hardware, and application marketplace.

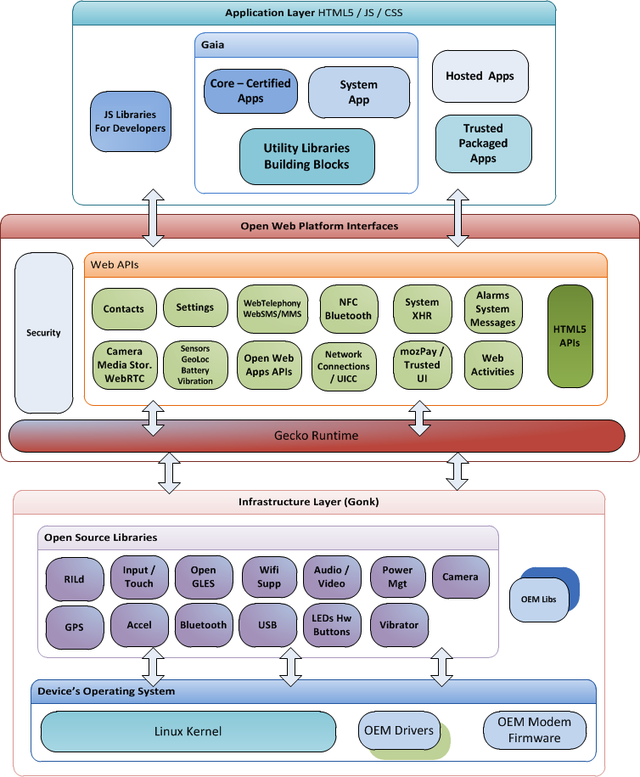


Figure (**???**) Firefox OS Architecture

The initial development work involves three major software layers:

* Gonk – platform denomination for a combination of the Linux kernel and the HAL from Android
* Gecko – the web browser engine and application run-time services layer;
* Gaia – an HTML5 layer and user-interface system.

[FXOS Web-API status page](https://wiki.mozilla.org/WebAPI) [25].

### Requirements Analysis

Analysis against [Hyperty Runtime Requirements](https://github.com/reTHINK-project/core-framework/labels/Runtime%20Requirement) (section ?)

* [The Runtime should be deployable in the most used Devices and Operating Systems](https://github.com/reTHINK-project/core-framework/issues/1)
* NO
* Firefox OS is made for a certain set of FXOS devices (phones, tables, smart TVs)
* [The Runtime should support W3C WebRTC APIs](https://github.com/reTHINK-project/core-framework/issues/2)
* YES
* Since FXOS version 2.1 this is officially stated as done.
* Tests with 2.0, showed that it basically worked there already
* tested basic A/V calls + separate apps that used the DataChannel to transport arbitrary files
* [The runtime must support standard Javascript (ECMAScript)](https://github.com/reTHINK-project/core-framework/issues/3)
* YES
* "Gecko" is the Javascript interpreter
* provides Javascript access to a lot of Web APIs (even non-standardized)
* Whole UI (Gaia) is based on HTML, Javascript, CSS
* [The Runtime should support Web Socket](https://github.com/reTHINK-project/core-framework/issues/4)
* YES, client side
* Websockets clients are supported
* Websocket servers are not supported
* --> same situation as in a browser runtime
* [The Runtime should support Web Messaging Notifications](https://github.com/reTHINK-project/core-framework/issues/5)
* NO
* according to Web-API status, no indication of planned support
* must be double-checked with practical tests
* what they have is a "Simple Push" API
* [The Runtime must have a good performance](https://github.com/reTHINK-project/core-framework/issues/6)
* very subjective, depends on device hardware it is running on
* tested 3 different devices with rather different experience in terms of performance
* [The Runtime must be secured](https://github.com/reTHINK-project/core-framework/issues/7)
* this would require much more analysis and expertise in attacking the device or the running applications
* general assumptions is that the security is comparable to a browser
* but because the browser IS the middle layer of the OS a potential breakout of the sandbox might have stronger consequences
* [The effort to introduce new capabilities in the runtime should be reasonable](https://github.com/reTHINK-project/core-framework/issues/8)
* YES
* extension with Javascript libraries is possible very easy
* due to the open source nature of the Gecko and Gonk layers it is also possible to add low- and medium-level capabilities there
* The effort for low-level extensions will be relatively high.

## Jitsi Videobridge

[Jitsi Videobridge](https://jitsi.org/Projects/JitsiVideobridge) [26] is a WebRTC compatible Selective Forwarding Unit (SFU) that allows for multiuser video communication.

Jitsi Video bridge supports RTP Relay, audio mixing, Call encryption with DTLS/SRTP and ICE.

### Architecture

JItsi Video bridge is a [XMPP component](http://xmpp.org/) [27] and can be integrated with any compliant XMPP Server like eJabberd or Openfire. In addition there is another XMPP component, the Jicofo, that uses an XMPP extension protocol called COLIBRI (COnferences with LIghtweight BRIdging) to provide conferencing focus functionalities including channels allocation and add / remove participants from each call. Finally, SIP interoperability is provided by a third XMPP component called Jigasi. There is an OpenSource WebRTC JavaScript application, called Jitsi Meet, that uses Jitsi Videobridge to provide high quality, scalable video conferences.

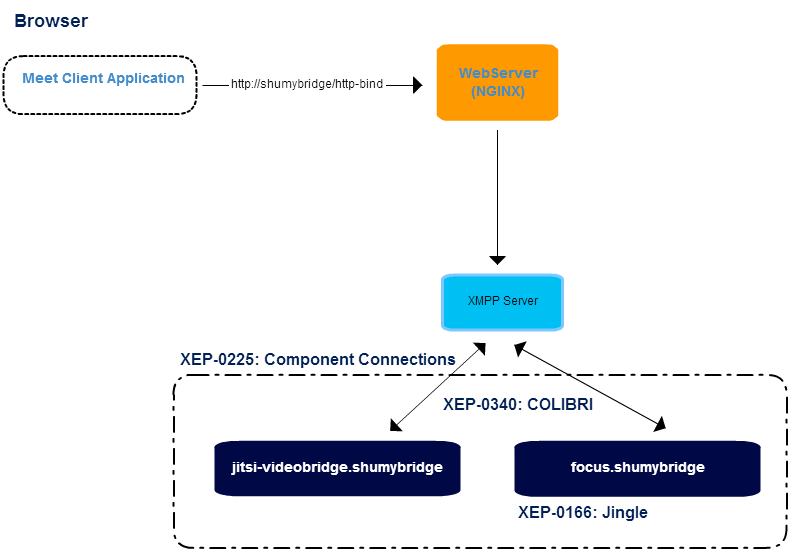


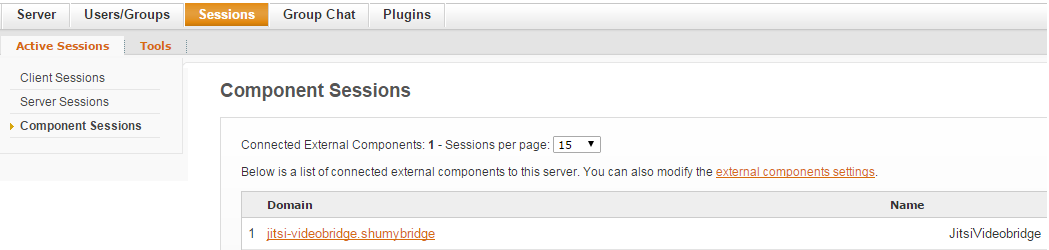
Figure (**???**) Jitsi Videobridge Architecture

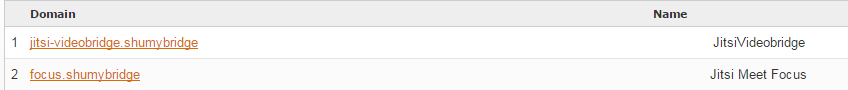
### Installation Procedures

**Required software** \* JVM (select the latest version) \* XMPP Server (openfire, prosody.im, Tigase ...) \* Jitsi VideoBridge (stream XMPP component) \* Jicofo (Session handler XMPP component) \* NGINX (web server and proxy) \* Jitsi Meet App

**Domain selection** \* Select a name for a domain, "shumybridge" will be use for this example. \* Add an entry in DNS hosts file "127.0.0.1 shumybridge".

**XMPP Server (using Openfire)** \* Download and install openfire from http://www.igniterealtime.org/downloads/index.jsp \* Access admin console at http://localhost:9090 \* For the machine name and XMPP domain is important that you use "shumybridge", server certificates will be generated for the domain. \* Select embedded SQLLite database, and an admin user account. Just enough for testing. \* On config "Server -> Server Settings -> HTTP Binding", enable "Script Syntax -> BOSH" and "Provides support for XFF (X-Forwarded-For) headers" \* On config "Server -> Server Settings -> External Components" enable and set the password, ex: xpassword

**Jitsi VideoBridge** \* Download and install Jitsi Videobridge from https://jitsi.org/Projects/JitsiVideobridge \* Run videobridge with: jvb --host=shumybridge --secret=xpassword \* You should see an entry in XMPP components like: 

**Jicofo Session** \* Clone from "git clone https://github.com/jitsi/jicofo.git" \* Ant build with "ant dist.{os-name}" \* Add lines "org.jitsi.impl.neomedia.transform.srtp.SRTPCryptoContext.checkReplay=false" and "org.jitsi.jicofo.auth.URL=XMPP:shumybridge" to the file sip-communicator.properties. In Windows this is located at "C:"user"\.sip-communicator-communicator.properties" or in linux "/usr/share/jicofo/.sip-communicator/sip-communicator.properties" \* Run videobridge with: jicofo --host=shumybridge --port=5275 --secret=xpassword \* You should see an entry in XMPP components like: 

**NGINX** \* Download and install from http://nginx.org/en/download.html \* Change nginx.conf file with:

server {  
 listen 80;  
 server\_name shumybridge;  
  
 location ~ ^/([a-zA-Z0-9]+)$ {  
 rewrite ^/(.\*)$ / break;  
 }  
   
 location / {  
 root srv/jitsi.example.com;  
 index index.html;  
 }  
  
 # BOSH  
 location /http-bind {  
 proxy\_pass http://shumybridge:7070/http-bind/;  
 proxy\_set\_header X-Forwarded-For $remote\_addr;  
 proxy\_set\_header Host $http\_host;  
 }  
   
 # redirect server error pages to the static page /50x.html  
 error\_page 500 502 503 504 /50x.html;  
 location = /50x.html {  
 root html;  
 }  
}

**Jitsi Meet App** \* Clone or download Meet App from https://github.com/jitsi/jitsi-meet.git \* Copy Meet App to NGINX folder ./srv/jitsi.example.com \* Change config.js file with:

var config = {  
 hosts: {  
 domain: 'shumybridge',  
 muc: 'conference.shumybridge',  
 bridge: 'jitsi-videobridge.shumybridge',  
 focus: 'focus.shumybridge'  
 },  
  
 ...  
 bosh: '//shumybridge/http-bind',  
 clientNode: 'http://shumybridge/jitsimeet',  
 ...  
};

### Evalusation of Jitsi Meet Application

Jitsi Meet uses strophe.js internally, but it's clustered with UI dependencies and other non wanted stuff. **Strophe.js** is an XMPP library for JavaScript. Its primary purpose is to enable web-based, real-time XMPP applications that run in any browser. There are Jingle plugins for strophe.js. You need to include the following files in your application from projects [jingle](https://github.com/estos/strophe.jingle) and [strophe](https://github.com/strophe/strophejs):

<!--add jQuery lib-->  
 <script src='strophe/strophe.js'></script><!-- strophe-->  
 <script src='strophe/strophe.disco.js'></script><!-- strophe.disco, optional -->  
 <script src='strophe/strophe.jingle.js' charset='utf-8'></script><!-- strophe jingle connection plugin -->  
 <script src='strophe/strophe.jingle.session.js' charset='utf-8'></script><!-- strophe jingle connection plugin -->  
 <script src='strophe/strophe.jingle.sdp.js' charset='utf-8'></script><!-- sdp library -->  
 <script src='strophe/strophe.jingle.adapter.js' charset='utf-8'></script><!-- getusermedia cross browser compat layer -->

Starting the XMMP session is normaly made with:

var BOSH\_SERVICE = '/http-bind';  
var ICE\_CONFIG = {iceServers: [{url: 'stun:stun.l.google.com:19302'}]};  
  
var DOMAIN = window.location.hostname;  
var CONFERENCEDOMAIN = 'conference.' + DOMAIN;  
  
var connection = null;  
var rtc = null;  
var localStream = null;  
  
var myroomjid = null;  
var roomjid = null;  
var listMembers = [];  
  
$(document).ready(function () {  
 rtc = setupRTC();  
 connection = new Strophe.Connection(BOSH\_SERVICE);  
 connection.jingle.ice\_config = ICE\_CONFIG;  
 connection.jingle.pc\_constraints = rtc.pc\_constraints;  
   
 //nice for debug purposes...  
 connection.xmlInput = function (data) { console.log('RECV: ', data); };  
 connection.xmlOutput = function (data) { console.log('SEND: ', data); };  
});  
  
//call this on a click button (connect)  
getUserMediaWithConstraints(['audio', 'video']);

**getUserMediaWithConstraints** will fire an event configured with jQuery.

$(document).bind('mediaready.jingle', function (event, stream) {  
 localStream = stream;  
 connection.jingle.localStream = stream;  
 RTC.attachMediaStream($(<video-tag>), localStream);  
   
 //connect to videobridge  
 connection.connect(<user>, <pasword>, function (event) {  
 //TODO: handle other connection states Strophe.Status  
 if (status == Strophe.Status.DISCONNECTED) {  
 if (localStream) {  
 localStream.stop();  
 localStream = null;  
 }  
 } else if (status == Strophe.Status.CONNECTED) {  
 connection.jingle.getStunAndTurnCredentials();  
   
 // disco stuff  
 if (connection.disco) {  
 connection.disco.addIdentity('client', 'web');  
 connection.disco.addFeature(Strophe.NS.DISCO\_INFO);  
 }  
   
 //CONNECTED:  
 roomjid = <hash> + '@' + CONFERENCEDOMAIN; //select room id  
 myroomjid = roomjid + '/' + Strophe.getNodeFromJid(connection.jid);  
  
 //config XMPP presence event handlers...  
 connection.addHandler(onPresence, null, 'presence', null, null, roomjid, {matchBare: true});  
 connection.addHandler(onPresenceUnavailable, null, 'presence', 'unavailable', null, roomjid, {matchBare: true});  
 connection.addHandler(onPresenceError, null, 'presence', 'error', null, roomjid, {matchBare: true});  
  
 var pres = $pres({to: myroomjid }).c('x', {xmlns: 'http://jabber.org/protocol/muc'});  
 connection.send(pres);  
 }  
 });  
});

and define presence handlers:

function onPresence(pres) {  
 var from = pres.getAttribute('from');  
 var type = pres.getAttribute('type');  
   
 if (type !== null) {  
 return true;  
 }  
   
 if ($(pres).find('>x[xmlns="http://jabber.org/protocol/muc#user"]>status[code="201"]').length) {  
 // http://xmpp.org/extensions/xep-0045.html#createroom-instant  
 var create = $iq({type: 'set', to: roomjid})  
 .c('query', {xmlns: 'http://jabber.org/protocol/muc#owner'})  
 .c('x', {xmlns: 'jabber:x:data', type: 'submit'});  
 connection.send(create); // fire away  
 }  
   
 //manage list members  
 if (from == myroomjid) {  
 for (i = 0; i < listMembers.length; i++) {  
 connection.jingle.initiate(listMembers[i], myroomjid);  
 }  
 } else {  
 listMembers.push(from);  
 }  
   
 return true;  
}  
  
function onPresenceUnavailable(pres) {  
 connection.jingle.terminateByJid($(pres).attr('from'));  
  
 //manage list members  
 for (var i = 0; i < listMembers.length; i++) {  
 if (listMembers[i] == $(pres).attr('from')) {  
 listMembers.splice(i, 1);  
 break;  
 }  
 }  
   
 return true;  
}  
  
function onPresenceError(pres) {  
 //TODO: process error  
 return true;  
}

Handle add/remove video/audio streams:

$(document).bind('remotestreamadded.jingle', function (event, data, sid) {  
 var el = $("<video autoplay='autoplay' style='display:none'/>").attr('id', 'largevideo\_' + sid);  
 RTC.attachMediaStream(el, data.stream);  
 });  
   
 $(document).bind('remotestreamremoved.jingle', function (event, data, sid) {  
 //TODO: remove video element  
 });

## Docker

[Docker](https://www.docker.com/) [23] is an open platform for developers and sysadmins to build, ship, and run distributed applications. Consisting of Docker Engine, a portable, lightweight runtime and packaging tool, and Docker Hub, a cloud service for sharing applications and automating workflows, Docker enables apps to be quickly assembled from components.

Docker containers are lightweight and fast. Containers have sub-second launch times, reducing the cycle time of development, testing, and deployment.

Docker consists of:

* The Docker Engine - lightweight and powerful open source container virtualization technology combined with a work flow for building and containerizing your applications.
* Docker Hub - SaaS service for sharing and managing your application stacks.

Docker is a standard container format that lets developers care about their applications inside containers while sysadmins and operators can work on running the container in the deployment environment. This separation of duties streamlines and simplifies the management and deployment of code.

Docker containers run (almost) everywhere including desktops, physical servers, virtual machines, into data centers, and up to public and private clouds. Since Docker runs on so many platforms, it's easy to move applications around including moving an application from a testing environment into the cloud and back. Docker's lightweight containers also make scaling up and down fast and easy. More containers can be launched when needed and then shut them down easily when they're no longer needed.

### Architecture

Figure (**???**) Docker Architecture

Figure (**???**) Docker Architecture

Docker uses a client-server architecture. The Docker client talks to the Docker daemon, which does the heavy lifting of building, running, and distributing Docker containers. Both the Docker client and the daemon can run on the same system, or can connect a Docker client to a remote Docker daemon. The Docker client and daemon communicate via sockets or through a RESTful API.

#### How to obtain security on standalone components using Docker.

To understand how to obtain security using Docker we have to look at its architecture:

The Docker daemon - the Docker daemon runs on a host machine. The user does not directly interact with the daemon, but instead through the Docker client.

The Docker client - The Docker client, in the form of the docker binary, is the primary user interface to Docker. It accepts commands from the user and communicates back and forth with a Docker daemon.

To understand Docker, we need to understand its three components:

Docker images - A Docker image is a read-only template. For example, an image could contain an Android minimal operating system with a minimal HTTP demon and a web application installed. Images are used to create Docker containers. Docker provides a simple way to build new images or update existing images, or we can download Docker images that other people have created

Docker registries - Docker registries hold images. These are public or private stores from which we upload or download images. These can be images we create ourselves or we can use images that others have previously created.

Docker containers - Docker containers are similar to a directory. A Docker container holds everything that is needed for an application to run. Each container is created from a Docker image. Docker containers can be run, started, stopped, moved, and deleted. Each container is an isolated and secure application platform.

Obviously the containers are the base for our security architecture. Each container is absolutely independent and each interface for comunication must be stricly observed. There is no other way to access the container and its content in operation.

### How does a Docker image works

We've already seen that Docker images are read-only templates from which Docker containers are launched. Each image consists of a series of layers. Docker makes use of union file systems to combine these layers into a single image. Union file systems allow files and directories of separate file systems, known as branches, to be transparently overlaid, forming a single coherent file system.

One of the reasons Docker is so lightweight is because of these layers. When we change a Docker image—for example, update an application to a new version— a new layer gets built. Thus, rather than replacing the whole image or entirely rebuilding, as we may do with a virtual machine, only that layer is added or updated. Now we don't need to distribute a whole new image, just the update, making distributing Docker images faster and simpler.

Every image starts from a base image, for example an Android base image. Or we can also use images of our own as the basis for a new image, for example if we have a base HTTPD image we could use this as the base of all our web application images.

### How to build a Docker image

Docker images are then built from base images using a set of steps we call instructions. Each instruction creates a new layer in our image. Instructions include actions like:

* Run a command.
* Add a file or directory.
* Create an environment variable.
* What process to run when launching a container from this image.

These instructions are stored in a file called a Dockerfile. Docker reads this Dockerfile when we request a build of an image, executes the instructions, and returns a final image.

### How does a Docker registry work

The Docker registry is the store for our Docker images. Once we build a Docker image we can push it to a public registry Docker Hub or to our own registry running behind our firewall.

Using the Docker client, we can search for already published images and then pull them down to our Docker host to build containers from them.

Docker Hub provides both public and private storage for images. Public storage is searchable and can be downloaded by anyone. Private storage is excluded from search results and only we and our users can pull images down and use them to build containers.

### How does a container work

A container consists of an operating system, user-added files, and meta-data. As we've seen, each container is built from an image. That image tells Docker what the container holds, what process to run when the container is launched, and a variety of other configuration data. The Docker image is read-only. When Docker runs a container from an image, it adds a read-write layer on top of the image in which our application can then run.

Either by using the docker binary or via the API, the Docker client tells the Docker daemon to run a container.

$ sudo docker run -i -t ubuntu /bin/bash

Let's break down this command. The Docker client is launched using the docker binary with the run option telling it to launch a new container. The bare minimum the Docker client needs to tell the Docker daemon to run the container is:

What Docker image to build the container from, here ubuntu, a base Ubuntu image; The command we want to run inside the container when it is launched, here /bin/bash, to start the Bash shell inside the new container.

when we run this command, the following actions are performed:

1 - Pulls the ubuntu image. Docker checks for the presence of the ubuntu image and, if it doesn't exist locally on the host, then Docker downloads it from Docker Hub. If the image already exists, then Docker uses it for the new container.

2 - Creates a new container. Once Docker has the image, it uses it to create a container.

3 - Allocates a filesystem and mounts a read-write layer. The container is created in the file system and a read-write layer is added to the image.

4 - Allocates a network / bridge interface. Creates a network interface that allows the Docker container to talk to the local host.

5 - Sets up an IP address. Finds and attaches an available IP address from a pool.

6 - Executes a process that we specify. Runs our application.

7 - Captures and provides application output. Connects and logs standard input, outputs and errors for we to see how our application is running.

### Dockerizing a Node.js Web App

The goal of this example is to show how to build your Docker images from a parent image using a Dockerfile. We will do that by making a simple Node.js hello world web application running on CentOS.

Create Node.js app

First, create a directory src where all the files would live. Then create a package.json file that describes your app and its dependencies:

{  
 "name": "docker-centos-hello",  
 "private": true,  
 "version": "0.0.1",  
 "description": "Node.js Hello world app on CentOS using docker",  
 "author": "Miguel Mesquita <mesquita@av.it.pt>",  
 "dependencies": {  
 "express": "3.2.4"  
 }  
}

Then we need to create an index.js file that defines a web app using the Express.js framework:

var express = require('express');  
  
// Constants  
  
var PORT = 8080;  
  
// App  
  
var app = express();  
  
app.get('/', function (req, res) {  
 res.send('Hello world\n');  
});  
  
app.listen(PORT);  
  
console.log('Running on http://localhost:' + PORT);

We'll look at how to run an application inside a CentOS container using Docker. First we need to build a Docker image of our app.

**Creating a Dockerfile**

Create an empty file called Dockerfile:

$ touch Dockerfile

Open the Dockerfile in your favorite text editor (I'm an old fashioned guy, I use vi)

Define the parent image we want to use to build your own image on top of. Here, we'll use CentOS (tag: centos6) available on the Docker Hub:

FROM centos:centos6

Since we're building a Node.js app, we have to install Node.js as well as npm on your CentOS image. Node.js is required to run our app and npm to install our app's dependencies defined in package.json. To install the right package for CentOS, we'll use the instructions from the Node.js wiki:

# Enable EPEL for Node.js  
  
RUN rpm -Uvh http://download.fedoraproject.org/pub/epel/6/i386/epel-release-6-8.noarch.rpm  
  
# Install Node.js and npm  
  
RUN yum install -y npm

To bundle our app's source code inside the Docker image, we use the COPY command:

# Bundle app source  
  
COPY . /src

Install our app dependencies using the npm command:

# Install app dependencies  
  
RUN cd /src; npm install

Our app binds to port 8080 so we use the EXPOSE command to have it mapped by the docker daemon:

EXPOSE 8080

Define the command to run our app using CMD which defines our runtime, i.e. node, and the path to our app src/index.js (see the step where we added the source to the container):

CMD ["node", "/src/index.js"]  
  
Our Dockerfile should now look like this:  
  
  
FROM centos:centos6  
  
# Enable EPEL for Node.js  
  
RUN rpm -Uvh http://download.fedoraproject.org/pub/epel/6/i386/epel-release-6-8.noarch.rpm  
# Install Node.js and npm  
  
RUN yum install -y npm  
  
# Bundle app source  
  
COPY . /src  
# Install app dependencies  
  
RUN cd /src; npm install  
  
EXPOSE 8080  
CMD ["node", "/src/index.js"]

**Building our image**

Go to the directory that has our Dockerfile and run the following command to build a Docker image. The -t flag adds a tag to our image so it's easier to find later using the docker images command:

$ sudo docker build -t <your username>/centos-node-hello .

Our image will now be listed by Docker:

$ sudo docker images  
  
# Example  
  
REPOSITORY TAG ID CREATED  
centos centos6 539c0211cd76 8 weeks ago  
<your username>/centos-node-hello latest d64d3505b0d2 2 hours ago

**Run the image**

Running our image with -d runs the container in detached mode, leaving the container running in the background. The -p flag redirects a public port to a private port in the container. Run the image we previously built:

$ sudo docker run -p 49160:8080 -d <your username>/centos-node-hello

To print the output of our app:

# Get container ID  
  
$ sudo docker ps  
  
# Print app output  
  
$ sudo docker logs <container id>  
  
# Output

Running on http://localhost:8080

**Test**

To test our app, get the port of our app that Docker mapped:

$ sudo docker ps  
  
# Example  
  
ID IMAGE COMMAND ... PORTS  
ecce33b30ebf <your username>/centos-node-hello:latest node /src/index.js 49160->8080

In the example above, Docker mapped the 8080 port of the container to 49160.

Is our application working? Lets test it with curl (ok install it with sudo apt-get install curl)

$ curl -i localhost:49160  
  
HTTP/1.1 200 OK  
  
X-Powered-By: Express  
  
Content-Type: text/html; charset=utf-8  
  
Content-Length: 12  
  
Date: Sun, 02 Jun 2013 03:53:22 GMT  
  
Connection: keep-alive   
  
Hello world

Yes!!! It's working.

Every application must connect through the port. The code is absolutely isolated from misuse. We implicitly have created an internal virtual net using docker from a internal pool of IPs docker has assumed on instalation. Every new application has a brand new on initiation which is relented on finishing the app. For the external user, well, it is invisible.

## Janus Gateway

[Janus Gateway](https://janus.conf.meetecho.com/) [29] is a WebRTC gateway implemented in C, which means it doesn't provide any functionality for itself, apart from establishing a WebRTC media connection between browsers and server-side applications they might be using (e.g. echo tests, conference bridges, media recorders). This communication consists in exchanging JSON messages and relaying RTP/RTCP packets between a browser and the application logic. Janus is intended to be a light component to cover a big range of use-cases. It can be used to deploy a full-fledged WebRTC gateway on a cloud provider or just a small nettop/box to handle a specific use case, looking at applications as pluggable modules that a client can connect to through this gateway.

### Architecture and APIs

Janus Gateway splits itself into three software modules. The Core module implements the WebRTC protocols (SDP, ICE, DTLS-SRTP, RTP/RTCP), which are themselves listed in the Protocols module, in a web server that interacts with browsers. This Core module takes care of aspects like session handling, media signalling, negotiation and plugins availability. The Plugins module provides some out-of-the-box plugins to be integrated or extended into new applications developed on top of Janus core. The already available plugins perform media record and play and establish streaming, voice mail or SIP connections, just to mention some useful examples. There is also a Plugin API which offers potential plugin developers an overview of how these plugins are implemented and also instructions of how to develop a new one.

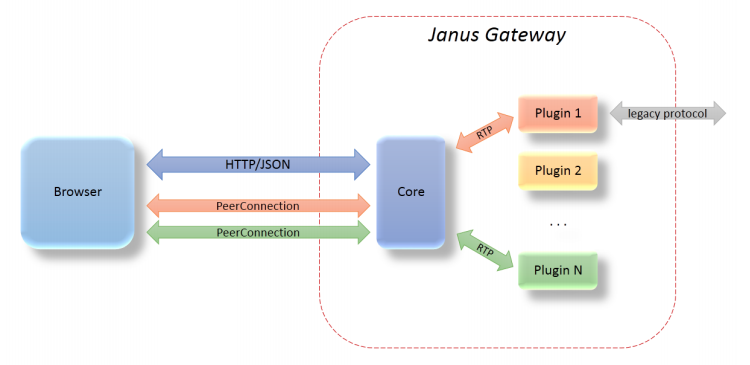


Figure (**???**) Janus Gateway architecture

### Janus Gateway and runtime requirements

#### [The Runtime should be deployable in the most used Devices and Operating Systems](https://github.com/reTHINK-project/core-framework/issues/1)

Janus Gateway web servers can currently be deployed on Linux systems only, and a cross-platform version is not going to be developed for now. For client-side, Google Chrome and Mozilla Firefox are the supported browsers.

#### [The Runtime should support W3C WebRTC APIs including](https://github.com/reTHINK-project/core-framework/issues/2)

Janus Gateway implementation of the WebRTC stack supports the W3C WebRTC APIs.

#### [The runtime must support standard Javascript (ECMAScript)](https://github.com/reTHINK-project/core-framework/issues/3)

Janus Gateway supports standard JavaScript in its implementation. Concretely, it offers a JavaScript library (janus.js) which allows developers to access both the REST and WebSocket interfaces. This library facilitates the task of establishing sessions between clients and the gateway, attaching plugins to clients, exchange events between them and so on.

#### [The Runtime should support Web Socket](https://github.com/reTHINK-project/core-framework/issues/4)

As said above, Janus Gateway offers a WebSocket API which developers may choose instead of the default plain HTTP REST API to interact with the gateway.

#### [The Runtime should support Standardised Messaging Notifications](https://github.com/reTHINK-project/core-framework/issues/5)

Janus can send events and notifications at any time through the long poll channel or the related push mechanisms made when using WebSockets. On plain HTTP, a user has to explicitly subscribe to notifications, repeating that as soon as an event has been received. However, the push mechanisms used by the WebSockets makes this task easier: as soon as a client creates a session through a WebSocket, that channel becomes automatically subscribed for events related with that session, and these events of interest will be then received through the same channel.

#### [The Runtime must have a good performance](https://github.com/reTHINK-project/core-framework/issues/6)

Measuring the performance of Janus is a complicated task, since it is just a gateway. Thus, the performance of plugins and applications written by third-party developers and working with Janus takes an important role on the measurements.There was a [recent study](http://dl.acm.org/citation.cfm?id=2749223) [30] on the performance of the Janus Gateway, when applied to several use-cases. Also, when [comparing the Video MCU conferencing plugin](http://www.rtc-conference.com/wp-content/uploads/gravity_forms/2-2f7a537445fa703985ab4d2372ac42ca/2014/09/Romano_Janus.pdf)[31] with other similar systems like [Jitsi](https://jitsi.org/) [26] and [Licode](http://lynckia.com/licode/), it revealed strong improvements on CPU and memory usage, both on client and server sides.

#### [The effort to introduce new capabilities in the runtime should be reasonable](https://github.com/reTHINK-project/core-framework/issues/8)

Due to its modular architecture, in which plugins can be seen as "bricks" in an application, introducing new features like a policy engine or hyperty registry should not be a very hard task.

## Kurento Media Server

### Overview

[Kurento](http://www.kurento.org/) is an Open Source Software WebRTC media server, that can be used to manage media flows :

* Send / Receive
* Recording
* Transcoding
* Augmented reality
* Mixing
* broadcasting

It can be used to handle different type of communications applications : 1 to 1, N to N, 1 to N (Real time exchanges or streaming)

### Architecture

Kurento is mainly composed of the two elements : - Kurento media server - Kurento Application

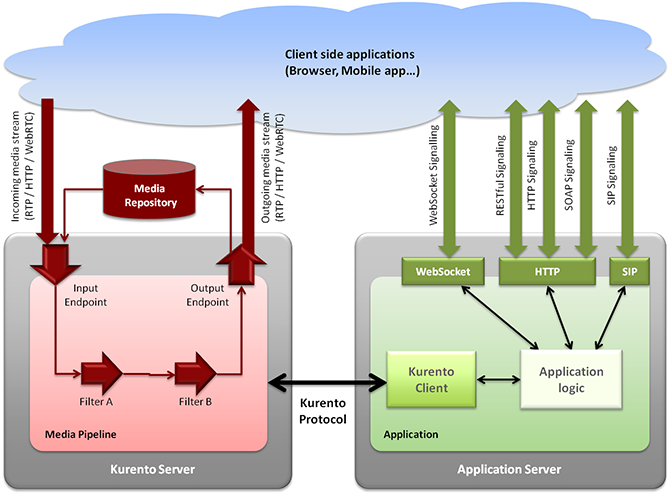


Figure (**???**) Kurento Architecture

Application developers can use Kurento Clients or Kurento API directly for creating their multimedia enabled applications. Developpers can use Javascript clients, Java Client or Kurento Protocol. This is interesting as it can easily be integrated with NodeJs

### APIs

*Available APIs for developers*

Media server API is describe here : http://www.kurento.org/docs/current/mastering/kurento\_API.html

Kurento Clients are also available for application developers :

JAVA : http://www.kurento.org/docs/current/langdoc/javadoc/index.html

JavaScript : http://www.kurento.org/docs/current/langdoc/jsdoc/kurento-client-js/index.html http://www.kurento.org/docs/current/langdoc/jsdoc/kurento-utils-js/index.html

### Integration in Rethink

Multiparty conversations supported with MCU/SFU for Star topologies can be supported with server side Hyperties running in the MCU/SFU ie there would be protofly in the MCU/SFU.

Kurento Media Server can be connected through a NodeJs Client : it will be possible to add protOfly interface on nodeJs to then connect to the MCU.

How to obtain security on standalone components using Docker.

To understand how to obtain security using Docker we have to look at its architecture:

The Docker daemon - the Docker daemon runs on a host machine. The user does not directly interact with the daemon, but instead through the Docker client.

The Docker client - The Docker client, in the form of the docker binary, is the primary user interface to Docker. It accepts commands from the user and communicates back and forth with a Docker daemon.

Inside Docker

To understand Docker , we need to understand its three components:

Docker images - A Docker image is a read-only template. For example, an image could contain an Android minimal operating system with a minimal HTTP demon and a web application installed. Images are used to create Docker containers. Docker provides a simple way to build new images or update existing images, or we can download Docker images that other people have created

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Docker containers - Docker containers are similar to a directory. A Docker container holds everything that is needed for an application to run. Each container is created from a Docker image. Docker containers can be run, started, stopped, moved, and deleted. Each container is an isolated and secure application platform.

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How does a Docker image works

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One of the reasons Docker is so lightweight is because of these layers. When we change a Docker image—for example, update an application to a new version— a new layer gets built. Thus, rather than replacing the whole image or entirely rebuilding, as we may do with a virtual machine, only that layer is added or updated. Now we don't need to distribute a whole new image, just the update, making distributing Docker images faster and simpler.

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How to build a Docker image

Docker images are then built from base images using a set of steps we call instructions. Each instruction creates a new layer in our image. Instructions include actions like:

Run a command. Add a file or directory. Create an environment variable. What process to run when launching a container from this image.

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How does a Docker registry work

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Using the Docker client, we can search for already published images and then pull them down to our Docker host to build containers from them.

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Either by using the docker binary or via the API, the Docker client tells the Docker daemon to run a container.

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Let's break down this command. The Docker client is launched using the docker binary with the run option telling it to launch a new container. The bare minimum the Docker client needs to tell the Docker daemon to run the container is:

What Docker image to build the container from, here ubuntu, a base Ubuntu image; The command we want to run inside the container when it is launched, here /bin/bash, to start the Bash shell inside the new container.

when we run this command:

In order, Docker does the following:

1 - Pulls the ubuntu image. Docker checks for the presence of the ubuntu image and, if it doesn't exist locally on the host, then Docker downloads it from Docker Hub. If the image already exists, then Docker uses it for the new container.

2 - Creates a new container. Once Docker has the image, it uses it to create a container.

3 - Allocates a filesystem and mounts a read-write layer. The container is created in the file system and a read-write layer is added to the image.

4 - Allocates a network / bridge interface. Creates a network interface that allows the Docker container to talk to the local host.

5 - Sets up an IP address. Finds and attaches an available IP address from a pool.

6 - Executes a process that we specify. Runs our application.

7 - Captures and provides application output. Connects and logs standard input, outputs and errors for we to see how our application is running.

Why Docker?

Docker containers are lightweight and fast. Containers have sub-second launch times, reducing the cycle time of development, testing, and deployment.

Docker containers run almost everywhere. We can deploy containers on desktops, physical servers, virtual machines, into data centers, and up to public and private clouds. Since Docker runs on so many platforms, it's easy to move our applications around. We can easily move an application from a testing environment into the cloud and back.

Dockerizing a Node.js Web App

The goal of this example is to show how to build your Docker images from a parent image using a Dockerfile. We will do that by making a simple Node.js hello world web application running on CentOS.

Create Node.js app

First, create a directory src where all the files would live. Then create a package.json file that describes your app and its dependencies:

{ "name": "docker-centos-hello", "private": true, "version": "0.0.1", "description": "Node.js Hello world app on CentOS using docker", "author": "Miguel Mesquita [mesquita@av.it.pt](mailto:mesquita@av.it.pt)", "dependencies": { "express": "3.2.4" } }

Then we need to create an index.js file that defines a web app using the Express.js framework:

var express = require('express');

// Constants

var PORT = 8080;

// App

var app = express();

app.get('/', function (req, res) { res.send('Hello world'); });

app.listen(PORT);

console.log('Running on http://localhost:' + PORT);

We'll look at how to run an application inside a CentOS container using Docker. First we need to build a Docker image of our app.

Creating a Dockerfile

Create an empty file called Dockerfile:

$ touch Dockerfile

Open the Dockerfile in your favorite text editor (I'm an old fashioned guy, I use vi)

Define the parent image we want to use to build your own image on top of. Here, we'll use CentOS (tag: centos6) available on the Docker Hub:

FROM centos:centos6

Since we're building a Node.js app, we have to install Node.js as well as npm on your CentOS image. Node.js is required to run our app and npm to install our app's dependencies defined in package.json. To install the right package for CentOS, we'll use the instructions from the Node.js wiki:

# Enable EPEL for Node.js

RUN rpm -Uvh http://download.fedoraproject.org/pub/epel/6/i386/epel-release-6-8.noarch.rpm

# Install Node.js and npm

RUN yum install -y npm

To bundle our app's source code inside the Docker image, we use the COPY command:

# Bundle app source

COPY . /src

Install our app dependencies using the npm command:

# Install app dependencies

RUN cd /src; npm install

Our app binds to port 8080 so we use the EXPOSE command to have it mapped by the docker daemon:

EXPOSE 8080

Define the command to run our app using CMD which defines our runtime, i.e. node, and the path to our app src/index.js (see the step where we added the source to the container):

CMD ["node", "/src/index.js"]

Our Dockerfile should now look like this:

FROM centos:centos6

# Enable EPEL for Node.js

RUN rpm -Uvh http://download.fedoraproject.org/pub/epel/6/i386/epel-release-6-8.noarch.rpm # Install Node.js and npm

RUN yum install -y npm

# Bundle app source

COPY . /src # Install app dependencies

RUN cd /src; npm install

EXPOSE 8080 CMD ["node", "/src/index.js"]

Building our image

Go to the directory that has our Dockerfile and run the following command to build a Docker image. The -t flag adds a tag to our image so it's easier to find later using the docker images command:

$ sudo docker build -t /centos-node-hello .

Our image will now be listed by Docker:

$ sudo docker images

# Example

REPOSITORY TAG ID CREATED centos centos6 539c0211cd76 8 weeks ago /centos-node-hello latest d64d3505b0d2 2 hours ago

Run the image

Running our image with -d runs the container in detached mode, leaving the container running in the background. The -p flag redirects a public port to a private port in the container. Run the image we previously built:

$ sudo docker run -p 49160:8080 -d /centos-node-hello

To print the output of our app:

# Get container ID

$ sudo docker ps

# Print app output

$ sudo docker logs

# Output

Running on http://localhost:8080

Test

To test our app, get the port of our app that Docker mapped:

$ sudo docker ps

# Example

ID IMAGE COMMAND ... PORTS ecce33b30ebf /centos-node-hello:latest node /src/index.js 49160->8080

In the example above, Docker mapped the 8080 port of the container to 49160.

Is our application working? Lets test it with curl (ok install it with sudo apt-get install curl)

$ curl -i localhost:49160

HTTP/1.1 200 OK

X-Powered-By: Express

Content-Type: text/html; charset=utf-8

Content-Length: 12

Date: Sun, 02 Jun 2013 03:53:22 GMT

Connection: keep-alive

Hello world

Yes!!! It's working.

Every application must connect through the port. The code is absolutely isolated from misuse. We implicitly have created an internal virtual net using docker from a internal pool of IPs docker has assumed on instalation. Every new application has a brand new on initiation which is relented on finishing the app. For the external user, well, it is invisible.

# Messaging SOTA

## Vert.x 3 Evaluation (Draft)

**Note:** to be reviewed for [v3](http://vert-x3.github.io/) by identifying differences with version 2.x

### Overview

*Overview of functionalities and type of WP3 component that the asset can be used for ie Messaging Node, Runtime, Network QoS and Framework*

This SOTA will evidence differences between version 2 and 3 of vert.x. It will not describe all the architecture as in the version 2 evaluation.

The concept of the framework is summarized as follows: \* **Polyglot (supports several languages)**: Vert.x framework runs on the JVM. However, Java is not required to run a Verticle. Main languages supported in version 3 are Java, JavaScript, Groovy and Ruby. \* **Concurrency model**: Concurrency model has not changed between versions. \* **Provides Event Bus**: Event bus is still available and is an essential part of vert.x engine for communication between programs, even when written in different languages. The event bus even penetrates into in-browser JavaScript allowing you to create effortless so-called real-time web applications. \* **Module System & Public Module Repository**: It seems that this feature is dropped in version 3. There is no more methods like deployModule(..) available. Now you create your verticles and package them into standard java jars. These jars can be pushed to Maven repositories just like any Maven artifact. For static module imports this is enough, however if we need dynamic module maintenance, an OSGi container could be used instead.

### Architecture

This subsection highlights the main building blocks of the Vert.x architecture. (diagram not supplied)

Figure 1. Vert.x Architecture

### Vert.x Runtime (Java 8 only)

Vert.x 3.0 is Java 8 only. This is to take advantage of new language features in Java 8, the most important of which is Lambdas which make developing against event based APIs so much nicer than in previous versions of Java. And is also chosen so that it can use Nashorn - the new high performance JavaScript engine that it contains.

### Addressing

Messages are sent on the event bus to an address. Vert.x instances are not bound to any addressing schemes. An address is simply a string, any string is valid. Some examples of valid addresses are europe.news.feed1, acme.games.pacman, sausages, and X. As a convention the names of the packages that implement certain functionalities should also be represented on the event bus and should be combined with a meaningful event/operation name, e.g. org.acme.MyPackage.MyClass.doSomething

### Handlers

A handler is an entity that receives messages from the event bus. You register a handler at an address. Many different handlers from the same or different modules can be registered at the same address. A single handler can be registered at many different addresses at the same time.

### Messaging Schemes

The Event Bus supports the following modes of operation: \* *Publish / subscribe messaging*: Publishing means delivering the message to all handlers that are registered at that address. This is the familiar publish/subscribe messaging pattern. \* *Point to point and Request-Response messaging*: Messages are routed to just one of the handlers registered at an address. They can optionally be replied to. \* *Remote Procedure Call (RPC)*: This mode of operation is implemented on top of the Request-Response model, basically by enforcing certain conventions on requests and responses

This example shows the Event Bus (in version 3) can be instantiated, how a Handler can be defined and registered on the Event Bus and how the Event Bus can subsequently publish a message for the defined Handler:

final EventBus eb = vertx.eventBus();  
  
/\*Create a message consumer against the specified address.  
The returned consumer is not yet registered at the address, registration will be effective when MessageConsumer.handler(..) is called.  
\*/  
final MessageConsumer<JsonObject> msgConsumer = eb.consumer("test.address");  
  
//register handler for the MessageConsumer  
msgConsumer.handler(message -> {  
 System.out.println("I just recieved a message "+ message.body());  
});  
  
...  
//publishing a message. The message will be delivered to all handlers registered against the address  
eb.publish("test.address", "hello world");  
  
//point-2-point sending of message.   
//Only one handler registered at the address receiving the message.   
//The handler is chosen in a non strict round-robin fashion  
eb.send("test.address", "hello world");  
  
...  
//unregister handler for the MessageConsumer  
msgConsumer.unregister();

### Types of Messages

Messages that you send on the event bus can be as simple as a string, a number or a boolean. It is also possible to send Vert.x buffers or JSON messages. It's highly recommended to use JSON messages to communicate between verticles. JSON is easy to create and parse in all the languages that Vert.x supports. For RPC messages, JSON is enforced.

## Verticle

The unit of execution for Vert.x applications is called a Verticle. Verticles can be written in multiple languages (Java, JavaScript, Groovy and Ruby). Many verticles can be executed concurrently in the same Vert.x instance. An application might be composed of multiple verticles deployed on different nodes of the network communicating by exchanging messages over the Vert.x Event Bus. You can now (version 3) instantiate verticles and deploy verticle instances programmatically. The platform manager API has been removed, and methods for deploying verticles have moved to the Vertx interface. The API for deploying verticles is much simpler, so this should simplify things when embedding.

## Module

The Vert.x 3 module system has gone. It's advisable to use already available options like Maven or Gradle. Non Java verticles (e.g. JavaScript) can also be packaged in jars and pushed as Maven artifacts. It will also support resolving in other ways and from other places (e.g. at run-time and from npm modules) before 3.0.final.

## Event Loop

By default, all verticles run in an asynchronous event loop. When developing a verticle, it is essential not to block the event loop. Blocking here means either doing any kind of blocking I/O or even doing any kind of computational intensive work. Modules that do either of these should indicate that they are so called worker modules by setting "worker": true in their *mod.json* file. The advantage of an event loop is that it is enormously scalable. Instead of waiting for I/O operations to complete, the executing thread will rather do other stuff (e.g. servicing the next request) in the meantime. This is achieved by using a callback driven style of programming. Imagine the following scenario: *We want to read some data in an I/O intensive operation (function readData)* We want to do something with that data (function doSomething) *We want to do something completely different (function doSomethingUnrelated)* In the traditional blocking world we would do something like the following:

def doSomething(data):  
 # do something with data  
data = readData()  
doSomething(data)  
doSomethingUnrelated()

What happens here is the following:

After the data is read, the program waits for the operation (readData) to complete (which is consuming the event loop thread lifetime). As soon as readData returns, we have our data and can go on to do something with it (doSomething(data)). Finally, when that is done, we can go on and do other stuff (doSomethingUnrelated).

In the asynchronous world, we do something like this:   
def doSomething(data):  
 # do something with data  
readData(callback = doSomething)  
doSomethingUnrelated()

As can be seen, the result of readData is not received in the functions return value. Instead doSomething is passed in the handler method as a callback. The framework will take care that this handler is called asynchronously as soon as the data is available

### APIs

Vert.x provides the different APIs which are implemented in various languages. The two main modules are "Core API" and "Apex API":

**Core API** \* Deploy and undeploy verticles \* Logging \* TCP client/Server API \* HTTP client/Server API \* File System Access \* DNS client API \* Shared Data \* Event Bus API \* JSON API

**Apex API** \* Routing (based on method, path, etc) \* Event-bus bridge \* SockJS support \* Session support - both local (for sticky sessions) and clustered (for non sticky) \* Basic Authentication and Redirect based authentication \* User/role/permission authorisation

### Requirements Analysis

*According to Component Type addressed by the solution ie Messaging Node, Runtime, Network QoS and Framework*

#### Messaging Node Requirements Analysis

##### [Autentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10) (PTIN)

External Authentication and Authorisation are supported through Maven artifacts: vertx-apex and vertx-auth-service

The Authorisation module can be the front-end to interact with an external vertx service eg with restful APIs or could be attached to the vertx-io event bus.

**Authorisation to Send/publish a Message** \* SockJSHandler, where we need a bridge configuration \* Inbound and outbound options have specific bridge configuration classes. java final BridgeOptions options = new BridgeOptions(); options.addInboundPermitted(new PermittedOptions().setAddress("chat.to.server").setRequiredPermission("tim")); options.addOutboundPermitted(new PermittedOptions().setAddress("chat.to.client")); Inboundpermitted with "setRequiredPermission" or "setRequiredRole" will force an authenticated session to send into that address.

* Configure authentication handler and provider java final AuthProvider authProvider = //implement this interface for authentication and authorization control final AuthHandler basicAuthHandler = BasicAuthHandler.create(authProvider); AuthProvider is similar to a SPI (Service Provider Interface) with 3 basic methods: login(..), hasRole(..), hasPermission(..). It's available for custom implementations, so that it's possible to interop with other parts of the system (like a database). AuthHandler can also be rewritten, but in this case we use simple browser authentication.

**Receive a Message** \* SockJS handler is needed with bridge options.

java final SockJSHandler sockJSHandler = SockJSHandler.create(vertx); sockJSHandler.bridge(options); \* Configure Apex Router with SockJS handler for "/eventbus/\*" uri

```java //required Cookie and Session handlers for every address final Router router = Router.router(vertx); router.route().handler(CookieHandler.create()); router.route().handler(SessionHandler.create(LocalSessionStore.create(vertx)));

//apply AuthHandler handler to an address (order of this handler is important) router.route("/eventbus/\*").handler(basicAuthHandler);

//apply SockJS handler to an address router.route("/eventbus/*").handler(sockJSHandler); ```*  EventBus handler for "chat.to.server" address, every message sent to this address will be processed in this handler

java final EventBus eb = vertx.eventBus(); eb.consumer("chat.to.server").handler(message -> { //user code... });

**Example: simple communication send/receive in the same client connected via SockJS** \* Register handler to receive messages in "chat.to.client" address

javascript eb.registerHandler('chat.to.client', function(msg) { console.log('Message Received: ' + msg); });

* Send message to "chat.to.server" address javascript eb.send('chat.to.server', {name: 'tim', age: 35});

**Subscribe / register handlers to be notified about published messages**

EventBusBridgeHook is not yet available in version 3, however it's possible to override the SockJSHandlerImpl class and bypass this limitation.

ServerHook takes some keyword arguments for example: \* pre-register: Called before a client handler registration is processed.

java public boolean handlePreRegister(SockJSSocket sock, String address) { out.println("handlePreRegister, sock = " + sock + ", address = " + address); return true; } \* message-handler: it's possible in this version to discovery the user that has sent the message (available in apex Session)

java public boolean handleSendOrPub(SockJSSocket sock, boolean send, JsonObject msg, String address) { msg.put("principal", sock.apexSession().getPrincipal()); return true; }

In this way handlers registration can be controlled, and the user information can be sent to the EventBus.

##### [Unstable Connections](https://github.com/reTHINK-project/core-framework/issues/15)(PTIN)

Hint from Fokus: Since vertx is based on http://hazelcast.org/ we can use it to cache some info including the sessionId

##### [Carrier grade deployment features (Resilience, DoS and DDoS protection, Service Assurance)](Messaging%20Node%20with%20carrier%20grade%20deployment%20features) (FOKUS)

* Resilience: Vert.x provides resilience through the "automatic failover" and "HA group" options. When a module is run with HA, if the Vert.x instance where it is running fails, it will be re-started automatically on another node of the cluster. An HA group denotes a logical grouping of nodes in the cluster. Only nodes with the same HA group will failover onto one another.
* DoS and DDoS Protection: Vert.x 2.x. has no support for this, BUT Vert.x 3.0 provides built-in core functiionality for this core
* Service Assurance: Modules can be deployed in clusters, and Vert.x provides an internal Load Balancer for routing messages within the cluster. Also the above mentioned "auomatic failover" and "HA group" options contribute to enforce service assurance.

##### [Scalability] (https://github.com/reTHINK-project/core-framework/issues/16) (FOKUS)

Verticle instances, except advanced multi-threaded worker verticles are almost always single threaded. what this implies is that, a single verticle instance can at most utilise one core of the server. In order to scale across cores, several verticles which are responsible for the same task can be instantiated and the runtime will distribute the workload among them (load balancing), this way taking full advantage of all SPU cores without much effort. Verticles can also be distributed between several machines. This will be transparent to the application code. The Verticles use the same mechanisms to communicate as if they would run on the same machine. This makes it very easy to scale applications.

##### [Messaging Transport Protocols] (https://github.com/reTHINK-project/core-framework/issues/20)(FOKUS)

* Websockets - Yes supported
* SockJS - Yes supported
* HTTP Long-Polling - Yes
* HTTP Streaming - ? (Not sure what this means, clarification needed)

##### [Message delivery reliability] (https://github.com/reTHINK-project/core-framework/issues/17)(FOKUS)

No. Vert.x uses the Event Bus to send messages through pub/sub mechanism or point-2-point mechanism. In both cases, there is no feedback to the sender if the message was recieved and processed or if it was not recieved at all. In the end reliability will boil down to the application logic service build on top of vert.x.

#### Runtime Requirements Analysis

From a runtime perspective the Vert.x is transparent to JVM8 (nashorn). Nashorn supports the full ECMAScript 5.1 specification. In that regard there are no browser API's such as: HTML5 canvas, HTML5 audio, WebWorkers, WebSockets, WebGL...

Initialization of the runtime engine:

final ClassLoader classLoader = TestClass.class.getClassLoader();  
final NashornScriptEngineFactory factory = new NashornScriptEngineFactory();

Nashorn is built on top of Java and takes advantage of standard Java security measures. Fine-grained security is enabled within applications. We can control the class load mechanism, effectively building a sandbox: ```java final ScriptEngine engine = factory.getScriptEngine(name -> { if(name.equals(TestClass.class.getName())) { return true; //OK, Java TestClass available from JavaScript }

return false; //everything else fails...

}); ```

Binding variables to the JavaScript scope is just one line of code. We can expose the Vert.x EventBus and use it like if we were in the JVM:

engine.getBindings(ScriptContext.ENGINE\_SCOPE).put("eb", vertx.eventBus());

Running a JavaScript file is also just a line of code:

engine.eval(new FileReader(classLoader.getResource("myjs.js").getFile()));

Although there are no WebSockets in the Nashorn runtime, it's possible to simulate a WS interface connecting directly through the EventBus, delegating the actual connection with the Vert.x.

Found some performance measures on: http://ariya.ofilabs.com/2014/03/nashorn-the-new-rhino-on-the-block.html

Realtime backends (aka noBackend or BackendAsAService(BaaS)) is a concept related to real time databases. It is a way to build web architectures without necessarily defining and standardizing data structures or interworking protocols if they are really needed. The backend and its remote framework is taking into account all low level mechanism of client-server dialogue, allowing developer to concentrate in service logic, in its local runtime. The realtime backend concept would allow to define and manage interworking with other services, in a way that entirely belongs to each application. It can also be a solution to manage and maintain user preferences, endpoints registration status … and also to manage discovery.

For instance, in IoT domain, nobody is able now to identify every kind of objects that will be available in a close future, the way they will communicate, their need of security, level of authentication … It’s the reason why a solution that still allows in the future to define or modify data structures is the best way to have an evolving solution, well understood and adopted by a large number of developers.

“Real-Time Web Technologies Guide”, from Phil Leggetter gives an overview of the different tools that are currently offered [http://www.leggetter.co.uk/real-time-web-technologies-guide/]. Among them we can site PubNub, Firebase, recently acquired by Google, and many others. \* Here is an example of code given by the firebase site:

// Use YOUR Firebase URL (not the one below)

var fb = new Firebase("https://.firebaseio.com");

/\* Remember to include firebase JS Library

\*/

* Save data:

fb.set({ name: "Alex Wolfe" });

* Update in real time

fb.on("value", function(data) {

var name = data.val() ? data.val().name : "";

alert("My name is " + name);

});

Ref: http://en.wikipedia.org/wiki/Real-time\_database

http://www.leggetter.co.uk/real-time-web-technologies-guide/

http://www.leggetter.co.uk/2013/12/09/choosing-realtime-web-app-tech-stack.html

## Matrix.org

From matrix.org spec: > *The end goal of Matrix is to be a ubiquitous messaging layer for synchronising arbitrary data between sets of people, devices and services - be that for instant messages, VoIP call setups, or any other objects that need to be reliably and persistently pushed from A to B in an interoperable and federated manner.*

### Overview

In the scope of the reTHINK project, matrix.org is a candidate technology for the **Messaging Node**

### Architecture

Following picture shows the main data flow in a matrix architecture.



image

The core components are the Home Servers (HS) which can federate to sync and maintain the history of shared communication sessions among domains. Home Servers resolv each other via DNS.

Some general points: \* **Every** communication requires a room. Even for a simple chat message to a dedicated receiver a room MUST be created first and the receiver MUST be invited and join it \* Rooms are persistent. They can be re-entered after successive login sessions. \* A set of event/message types is defined in the API, own extensions can be made. \* Focus of matrix is on federation and consistency and history of session states.

### APIs

Matrix.org defines 3 types of APIs with different scope. All these APIs are REST APIs using JSON Objects as payload.

* [Client Server API v1](http://www.matrix.org/docs/spec/#client-server-api-v1)
* for implementation of application frontends
* functions for:
  + Registration and Login
  + sending and receiving of Events
  + management of communication rooms
* well documented, easy to implement
* version 2 is currently in development - will be backward compatible
* [Federation API (Server-Server API)](http://www.matrix.org/docs/spec/#id100)
* API for the inter-domain communication between Home Servers
* uses HTTPS GET and PUT requests
* transaction based
* requests are authenticated by PK signatures
* [Application Service API](http://www.matrix.org/docs/spec/#id79)
* API to provide custom server-side behaviour (e.g. gateways, filters, extensible hooks etc)
* so far only "Passive Application Services" are specified, i.e. services that can only monitor but not block or modify events

some words about Identifiers: \* Users are identified as: **(???)** (with an optional **:port** suffix) \* Rooms are identified as: **#roomalias:host.domain** (with an optional **:port** suffix) \* Home servers are identified and resolved by their FQDN like: **https://host.domain:port**

The Matrix concept includes the concept of an "Identity Server", which is intended to map 3rd party entities to matrix ids: \* no documentation \* testing didn't work in our lab environment

### Requirements Analysis

Analysis against **Messaging Node** Requirements

* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* Yes
* the Client Server API could be wrapped in a protocol stub, that can be downloaded at runtime
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* Partially
* matrix is based on REST, example client uses long-polling to receive Events
* wrapping of Events/message in other transport protocols would require potential changes on HomeServer side
* (since "synapse", the ref-impl of Matrix HS, is implemented on top of Twisted, it should be possible to do that for e.g. Websockets)
* [Message Caching](https://github.com/reTHINK-project/core-framework/issues/19)
* Yes
* this is kind of a core feature
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* Yes
* could be done via an attached passive Application Service
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* Yes
* transmission errors are returned to clients
* messages are re-delivered to clients from internal history of HS
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* No/(Perhaps via load balancers)
* from spec: "HS SHOULD implement rate limiting ..."
* experiments showed that already simple forwarding of every individual candidate of a WebRTC call can trigger this rate limitiation
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* Yes
* since all communication is initiated by the client and HS addressing happens via DNS or static IP:port it is up to the client to re-connect after network interruptions
* HS always has the complete history of the session - nothing is lost
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* Yes
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* No
* HTTP(S) is not the most efficient protocol in terms of latency
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* Yes
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* rather Yes (not tested on embedded device yet)
* current ref-impl is based on Twisted (Python)
* overall installation size 94MB (including python and webclient)
* ==> environments supporting python should also support a Matrix HS
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* No
* matrix includes concept of external "Identity Server", but only for mapping of 3rd party IDs to internal IDs
* AA is performed internally
* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* Yes, via room-IDs

## RabbitMQ Evaluation

It is defined as a robust and easy to use messaging platform that can work synchronously an asynchronously.

From rabbitmq.com: > \*RabbitMQ is a messaging broker - an intermediary for messaging. It gives your applications a common platform to send and receive messages, and your messages a safe place to live until received.

### Overview

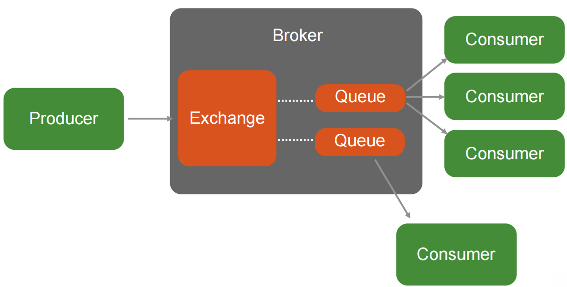
In the scope of the reTHINK project, RabbitMQ is a candidate technology for the Messaging Node

##### Highlights:

* **Reliability** - RabbitMQ offers a variety of features to let you trade off performance with reliability, including persistence, delivery acknowledgements, publisher confirms, and high availability.
* **Flexible Routing** - Messages are routed through exchanges before arriving at queues. RabbitMQ features several built-in exchange types for typical routing logic. For more complex routing you can bind exchanges together or even write your own exchange type as a plugin.
* **Clustering** - Several RabbitMQ servers on a local network can be clustered together, forming a single logical broker.
* **Federation** - For servers that need to be more loosely and unreliably connected than clustering allows, RabbitMQ offers a federation model.
* **Highly Available Queues** - Queues can be mirrored across several machines in a cluster, ensuring that even in the event of hardware failure your messages are safe.
* **The Polyglot Broker** - RabbitMQ supports messaging over a variety of messaging protocols including STOMP, MQTT, or AMQP.
* **Tracing** - If your messaging system is misbehaving, RabbitMQ offers tracing support to let you find out what's going on.

### Architecture

The core component is the **Broker** that routes the messages from the **producers** to the **consumers** as depicted in the following image.



image

The **Broker** has two main componentes: \* **Exchange**: accepts and routes messages from producer to clients based on the message information such as keys, bindings, filtering or broadcast.

* **Queue**: a FIFO queue of messages

Message routing can be based on topic / wild-card, as shown in the following image. 

Image Source: [www.rabbitmq.com](https://www.rabbitmq.com/getstarted.html)

### APIs and Documentation

[AMQP Protocol Specification](https://www.rabbitmq.com/protocol.html)

RabbitMQ provides the following APIs: \* [Java and JVM](https://www.rabbitmq.com/devtools.html#java-dev) \* Java \* Spring Framework \* Scala \* Groovy \* Grails \* Clojure \* JRuby \* [Ruby](https://www.rabbitmq.com/devtools.html#ruby-dev) \* [Python](https://www.rabbitmq.com/devtools.html#python-dev) \* [.NET / C#](https://www.rabbitmq.com/devtools.html#dotnet-dev) \* [PHP](https://www.rabbitmq.com/devtools.html#php-dev) \* [PERL](https://www.rabbitmq.com/devtools.html#perl-dev) \* [C / C++](https://www.rabbitmq.com/devtools.html#c-dev) \* [Node.js](https://www.rabbitmq.com/devtools.html#node-dev) \* [Go](https://www.rabbitmq.com/devtools.html#go-dev) \* [Erlang](https://www.rabbitmq.com/devtools.html#erlang-dev) \* [Haskell](https://www.rabbitmq.com/devtools.html#haskell-dev) \* [**Web Messaging**](https://www.rabbitmq.com/devtools.html#web-messaging) \* [SockJS-erlang](https://github.com/sockjs/sockjs-erlang) SockJS-erlang is compatible with SockJS client \* [rabbitmq-chat](https://github.com/videlalvaro/rabbitmq-chat) A Web chat implemented with RabbitMQ and Websockets \* [rabbithub](https://github.com/tonyg/rabbithub) RabbitHub provides an HTTP-based interface to RabbitMQ. \* [VorpalBunny](https://github.com/myYearbook/VorpalBunny) PHP for talking to RabbitMQ's JSON-RPC-Channel Plugin

### Requirements Analysis

Analysis against **Messaging Node** Requirements

* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* Yes
* the Client Server API could be wrapped in a protocol stub, that can be downloaded at runtime
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* Partially
* Has support for :
  + AMQP 0.9.1, 0.9, 0.8
  + STOMP via plugin
  + MQTT via plugin
  + AMQP 1.0 via plugin
  + Web-STOMP using WebSockets (SockJS) via plugin
  + Non-Reliable HTTP API to send and receive messages via the management plugin
  + JSON-RPC via plugin (synchronous)
* [Message Caching](https://github.com/reTHINK-project/core-framework/issues/19)
* Yes
* Core feature
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* Yes
* RabbitMQ has a built-in tracer feature that is able to see every message that is published, and every message that is delivered on a per-node.
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* Yes
* For AMQP:
  + Acknowledgements can be used in both directions - to allow a consumer to indicate to the server that it has received/processed a message and to allow the server to indicate the same thing to the producer.
  + Use of acknowledgements guarantees at-least-once delivery. Without acknowledgements, message loss is possible during publish and consume operations and only at-most-once delivery is guaranteed.
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* Yes
* In a RabbitMQ cluster all data/state required for the operation of a broker is replicated across all nodes, for reliability and scaling, with full ACID properties. Queues may be located on a single node, or mirrored across multiple nodes. A client connecting to any node in a cluster can see all queues in the cluster, even if they are not located on that node.
* Brokers tolerate the failure of individual nodes
* 1 million messages/second was benchmarked using a cluster of 30 Nodes [[Link1]](http://blog.pivotal.io/pivotal/products/rabbitmq-hits-one-million-messages-per-second-on-google-compute-engine)[[Link2]](http://googlecloudplatform.blogspot.pt/2014/06/rabbitmq-on-google-compute-engine.html)
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* No
* The client should use heartbeats for detecting dead TCP connections that have a timeout interval.
* Business logic must be implemented on the application layer in order to deal with this case.
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* Yes
* Using the HTTP management API
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* Yes
* Has low latency derived from its Erlang message oriented implementation. [[Link1]](http://www.rabbitmq.com/blog/2012/04/17/rabbitmq-performance-measurements-part-1/)[[Link2]](http://www.rabbitmq.com/blog/2012/04/25/rabbitmq-performance-measurements-part-2/)
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* Yes
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* Depends on the protocol used:
  + AMQP,STOMP, HTTP - depends (some computation is needed) (>256K RAM)
  + MQTT - yes (Designed especially for IoT)
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* Yes
* Using SSL client certificates via plugin
* Using LDAP and Federation via pugins
* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* Yes
* Core functionality, including topic/wildcard based filtering and broadcasting.

## ZeroMQ Evaluation

### Overview

In the scope of the reTHINK project ZeroMQ is a candidate technology for the Messaging Node.

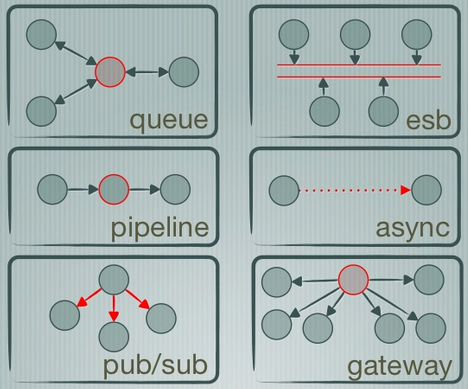
It is a high-performance, low level, asynchronous messaging library originally written in C++, that now has multiple native Implementations. It is used as a thin layer between the application and transport layers.

##### Highlights:

* Connect your code in any language, on any platform. 40+ Language Bindings
* Carries messages across inproc, IPC, TCP, TIPC, multicast.
* Smart patterns like pub-sub, push-pull, and router-dealer can be combined together to form powerful architectures.
* High-speed asynchronous I/O engines, in a tiny library ((20k lines of C++)).
* Backed by a large and active open source community.
* Supports every modern language and platform.
* Any architecture: centralized, distributed, small, or large.
* Multicore Optimized
* Automatic TCP (re)connect
* Fast: 8M msg/sec, usec latency
* Faster than TCP, for clustered products and supercomputing
* Fast for development thanks to many useful abstraction layers, native languages, bindings and huge open source community.

### Architecture

The following figure represents the six basic types of communication patterns that ZeroMQ supports.



image

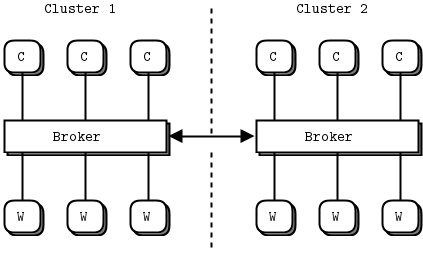
Image Source: [http://www.slideshare.net](http://www.slideshare.net/IanBarber/zeromq-is-the-answer)

#### Built-in core ZeroMQ patterns

* Request-reply: connects a set of clients to a set of services. This is a remote procedure call and task distribution pattern.
* Pub-sub: connects a set of publishers to a set of subscribers. This is a data distribution pattern.
* Pipeline: connects nodes in a fan-out/fan-in pattern that can have multiple steps and loops. This is a parallel task distribution and collection pattern.
* Exclusive pair: connects two sockets exclusively. This is a pattern for connecting two threads in a process, not to be confused with "normal" pairs of sockets.

On top of the built-in core ZeroMQ patterns high-level messaging patterns are defined [here](http://zguide.zeromq.org/page:all). They are not part of the core library, do not come with the ZeroMQ package, and exist in their own space as part of the ZeroMQ community. For example the Majordomo pattern ([Reliable Request-Reply Patterns](http://zguide.zeromq.org/page:all#reliable-request-reply)), sits in the GitHub Majordomo project in the ZeroMQ organization. One of the things that ZeroMQ also aims to provide is a set of high-level patterns, both small (how to handle messages sanely) and large (how to make a reliable pub-sub architecture).

These can be used as "fabric" to make very powerful architectures. The next image shows an example of this modularity. [A pub/sub multi-cluster Architecture](http://zguide.zeromq.org/page:all#Scaling-to-Multiple-Clusters)



image

Image Source: [http://zguide.zeromq.org](http://zguide.zeromq.org/page:all#Scaling-to-Multiple-Clusters)

The internal Architecture in more detail can be found [here.](http://zeromq.org/whitepapers:architecture)

### APIs and Bindings

#### Native Implementations of the library

* [C/C++](https://github.com/zeromq/libzmq) represents the State of the Art.
* [Java](https://github.com/zeromq/jeromq) JeroMQ. Fully compatible at both API and protocol level but sans encryption or PGM. *what is PGM?*
* [.NET](https://github.com/zeromq/netmq) same constraints as JeroMQ
* [Erlang](https://github.com/zeromq/ezmq)
* [Python](https://github.com/caedesvvv/zmqproto)
* [C](https://github.com/zeromq/libzmtp) designed for small devices you would not normally expect messaging technology to fit
* [Netty](https://github.com/spotify/netty-zmtp)

#### Language Bindings

* [Ada, C, Chicken Scheme, Common Lisp, C#(.NET & Mono), C++, D, delphi binding, Eiffel, Erlang, F#, Felix, Flex (ActionScript), Fortran77, Go, Guile, Haskell, Haxe, Java binding, JavaScript (Flash), Julia, LabVIEW, Lua bindings, Nimrod, Node.js, Objective-C, Objective Caml binding, ooc, Perl s, PHP binding, Python binding, Q, Racket, R, RE, RE, Red, Ruby, Ruby(FFI), Scala, Smalltalk, Tcl, Twisted (Python), XPCOM](http://zeromq.org/bindings:_start), and more [on github.com](https://github.com/search?utf8=%E2%9C%93&q=zmq&type=Repositories&ref=searchresults)

#### Web Clients/Servers

* [JSMQ](https://github.com/zeromq/JSMQ) Javascript client for ZeroMQ/NetMQ over WebSockets
* [NullMQ](https://github.com/progrium/nullmq) ZeroMQ semantics in the browser [Link1](http://www.slideshare.net/progrium/nullmq-pdx) [Link2](http://avalanche123.com/blog/2012/02/25/interacting-with-zeromq-from-the-browser/)
* [ZmqSocket.js](http://zeromq.org/bindings%3ajavascript) talk to zmq sockets from your JavaScript code.
* [SockJSProxy](https://bitbucket.org/vladev/sockjsproxy/) a simple proxy server that proxies message from SockJS to a ZeroMQ.
* [Zerogw](https://github.com/tailhook/zerogw) HTTP to zeromq gateway
* [XARP](http://rfc.zeromq.org/spec:40) (Draft) Extensible Resource Access Protocol (XRAP), a RESTful protocol built over ZeroMQ
* [Zato](https://zato.io/docs/index.html) ESB, SOA, REST, APIs and cloud integrations
* [Malamute](https://github.com/miska/malamute) All the enterprise messaging patterns in one box.

[ZeroMQ API Reference](http://api.zeromq.org/)

[Other Projects](http://zeromq.org/docs:labs)

### Requirements Analysis

Analysis against **Messaging Node** Requirements

* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* Yes
* the Client Server API could be wrapped in a protocol stub, that can be downloaded at runtime
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* Yes
* Has support for Javascript and WebSockets using:
  + [JSMQ](https://github.com/zeromq/JSMQ)
  + [NullMQ](https://github.com/progrium/nullmq)
  + [ZmqSocket.js](http://zeromq.org/bindings%3ajavascript)
  + [SockJSProxy](https://bitbucket.org/vladev/sockjsproxy/)
  + [Zerogw](https://github.com/tailhook/zerogw)
  + [XRAP](http://rfc.zeromq.org/spec:40)
  + [Zato](https://zato.io/docs/index.html)
* [Message Caching](https://github.com/reTHINK-project/core-framework/issues/19)
* Yes
* Using the [Titanic Service Protocol](http://rfc.zeromq.org/spec:9)
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* Yes
* Using the [Titanic Service Protocol](http://rfc.zeromq.org/spec:9)
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* Yes
* Using [Reliable Patterns](http://zguide.zeromq.org/page:all#Chapter-Reliable-Request-Reply-Patterns)
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* Yes
* Using scalable patterns such as a [brokerless design](http://zeromq.org/whitepapers:brokerless)
* 9,5 Million Messages / second were benchmarked on a 16 core machine[Link1](http://zeromq.org/results:0mq-tests-v03) [Other Tests](http://zeromq.org/results:_start)
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* Partial
* Business logic can be developed to deal with this issue
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* Yes
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* Yes
* Very low usec latency
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* Yes
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* Yes
* ZeroMQ runs on everything of interest, from 32KB embedded chips to z/OS mainframes running IBM dialects of Unix. [Link](http://zeromq.org/docs:features)
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* Yes
* Implemented:
  + [ZAP - ZeroMQ Authentication Protocol (PAM, LDAP, Kerberos, passwd, etc)](http://rfc.zeromq.org/spec:27)
  + CurveZMQ Authentication and Encryption Protocol [Link1](http://curvezmq.org/) [Link2](http://rfc.zeromq.org/spec:26)
  + [GSSAPI (Kerberos)](http://rfc.zeromq.org/spec:38) or [CURVE](http://curvezmq.org/)
  + [SRP](http://rfc.zeromq.org/spec:34)
* Supports the ability to be extended
* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* Yes
* Using the [ZeroMQ Publish-Subscribe Pattern](http://rfc.zeromq.org/spec:29)

## Redis Evaluation

### Overview

In the scope of the reTHINK project Redis is a candidate technology for the Messaging Node.

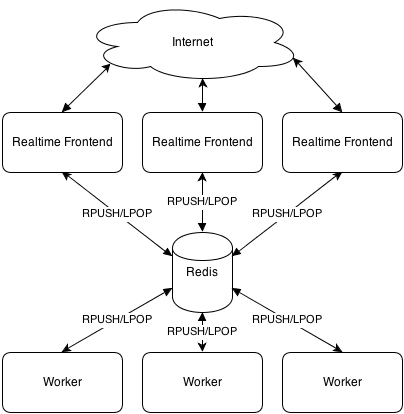
From http://redis.io/: > Redis is an open source, BSD licensed, advanced key-value cache and store. It is often referred to as a data structure server since keys can contain strings, hashes, lists, sets, sorted sets, bitmaps and hyperloglogs.

##### Highlights:

* Very Fast in memory data-store
* Usually used as Cache or MQ
* Scalable
* Supports strings, hashes, lists, sets, sorted sets, bitmaps and hyperloglogs as data structures.

### Architecture

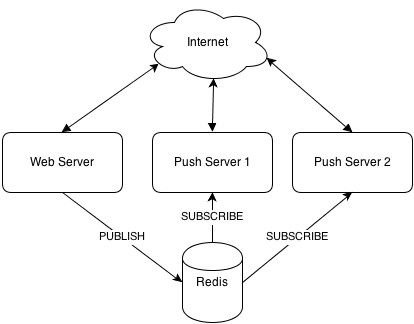
* The main Redis architecture can be resumed as a in memory key-value data-store for service decoupling.



image

Image Source: [http://mrjoes.github.io](http://mrjoes.github.io/2013/06/21/python-realtime.html)

* Another possibility is a PubSub model with message filtering based on prefix matching.



image

Image Source: [http://mrjoes.github.io](http://mrjoes.github.io/2013/06/21/python-realtime.html)

### APIs and Bindings

#### Language Bindings

* [C, C#, C++, Clojure, Common Lisp, D Dart, Elixir, emacs, lisp, Erlang, Fancy, GNU Prolog, Go Haskell, haXe, Io, Java, Lua, Matlab, Nimrod, Node.js, Objective-C, OCaml, Perl, PHP, Pure Data Python, Rebol, Ruby, Rust, Scala, Scheme, Smalltalk, Tcl, VCL](http://redis.io/clients)

#### Other

* [webdis](https://github.com/nicolasff/webdis) A Redis HTTP interface with JSON output
* [BankersBox](https://github.com/twilio/BankersBox) A redis-like wrapper for javascript data storage
* [Bone](https://github.com/solutious/bone) Rudimentary Redis over HTTP(S)

### Requirements Analysis

Analysis against **Messaging Node** Requirements

* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* Yes
* the Client Server API could be wrapped in a protocol stub, that can be downloaded at runtime
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* No
* An external wrapper should be used
* [Message Caching](https://github.com/reTHINK-project/core-framework/issues/19)
* Yes
* Core feature
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* No
* Should be implemented externally
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* Yes
* Atomic message processing
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* Yes
* Using:
  + Clustering [Link1](http://redis.io/topics/cluster-tutorial) [Link2](http://redis.io/topics/cluster-spec)
  + Partitioning [Link1](http://redis.io/topics/partitioning)
  + Sentinel provides high availability [Link1](http://redis.io/topics/sentinel)
  + Replication [Link1](http://redis.io/topics/replication)
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* Partial
* Business logic can be developed to deal with this issue
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* Yes
* Built-in
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* Yes
* Very low usec latency [Link1](http://redis.io/topics/benchmarks)
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* Yes
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* Yes
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* No
* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* Yes
* Core functionality, including prefix filtering

## XMPP Evaluation

From http://xmpp.org/:

The Extensible Messaging and Presence Protocol (XMPP) is an open technology for real-time communication, which powers a wide range of applications including instant messaging, presence, multi-party chat, voice and video calls, collaboration, lightweight middleware, content syndication, and generalized routing of XML data.

### Overview

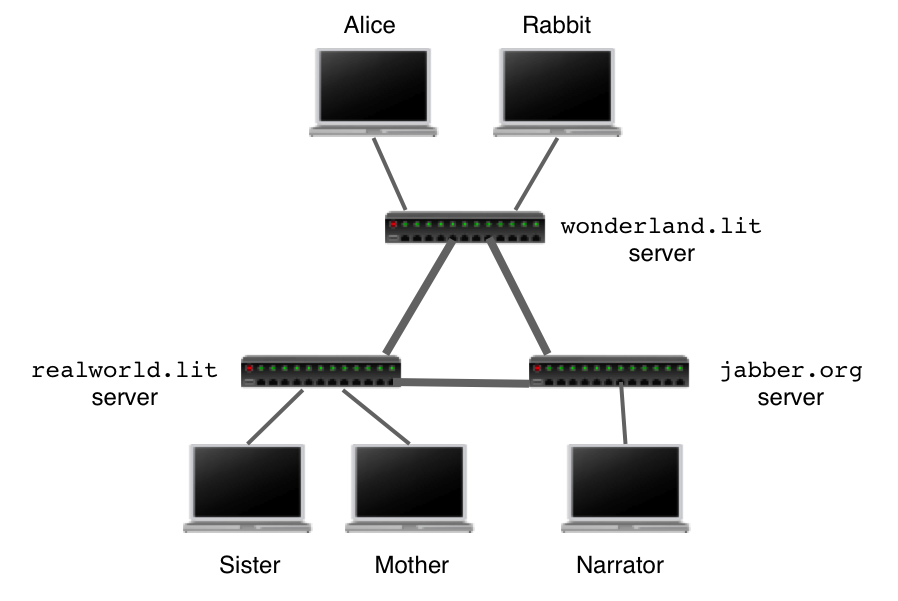
In the scope of the reTHINK project XMPP is a candidate technology for the Messaging Node.

##### Highlights:

* **Open** - the XMPP protocols are free, open, public, and easily understandable; in addition, multiple implementations exist in the form clients, servers, server components, and code libraries.
* **Standard** - the Internet Engineering Task Force (IETF) has formalized the core XML streaming protocols as an approved instant messaging and presence technology. The XMPP specifications were published as RFC 3920 and RFC 3921 in 2004, and the XMPP Standards Foundation continues to publish many XMPP Extension Protocols. In 2011 the core RFCs were revised, resulting in the most up-to-date specifications (RFC 6120, RFC 6121, and RFC 6122).
* **Proven** - the first Jabber/XMPP technologies were developed by Jeremie Miller in 1998 and are now quite stable; hundreds of developers are working on these technologies, there are tens of thousands of XMPP servers running on the Internet today, and millions of people use XMPP for instant messaging through public services such as Google Talk and XMPP deployments at organizations worldwide.
* **Decentralized** - the architecture of the XMPP network is similar to email; as a result, anyone can run their own XMPP server, enabling individuals and organizations to take control of their communications experience.
* **Secure** - any XMPP server may be isolated from the public network (e.g., on a company intranet) and robust security using SASL and TLS has been built into the core XMPP specifications. In addition, the XMPP developer community is actively working on end-to-end encryption to raise the security bar even further.
* **Extensible** - using the power of XML, anyone can build custom functionality on top of the core protocols; to maintain interoperability, common extensions are published in the XEP series, but such publication is not required and organizations can maintain their own private extensions if so desired.
* **Flexible** - XMPP applications beyond IM include network management, content syndication, collaboration tools, file sharing, gaming, remote systems monitoring, web services, lightweight middleware, cloud computing, and much more.
* **Diverse** - a wide range of companies and open-source projects use XMPP to build and deploy real-time applications and services; you will never get “locked in” when you use XMPP technologies.

### Architecture

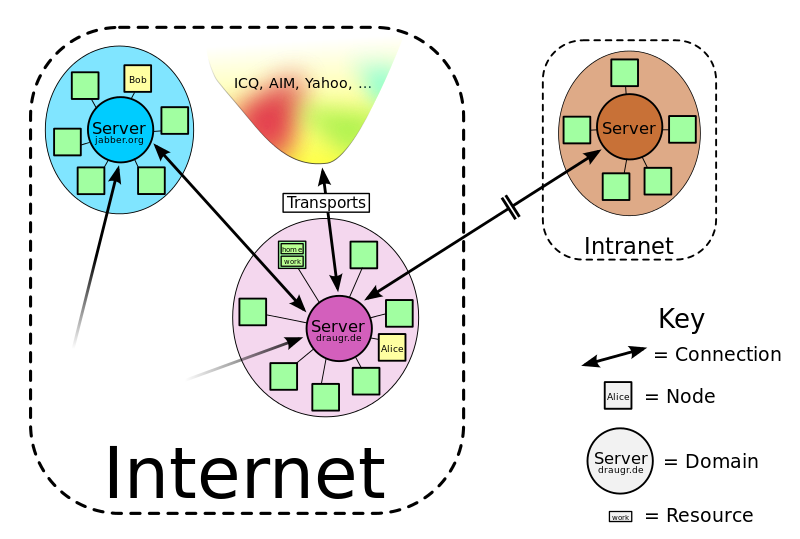
A typical XMPP network consists of several Servers/Domains connected together. The mechanism is similar to email where the servers are used as relays for the messages. Every entity on the XMPP network is addressed using a JabberID (JID). It has the form : username@domain/resource where domain is the domain name of the XMPP server, and username identifies an account on that server.



image

Image Source: [https://el-tramo.be](https://el-tramo.be/documents/beautiful-xmpp-testing/)

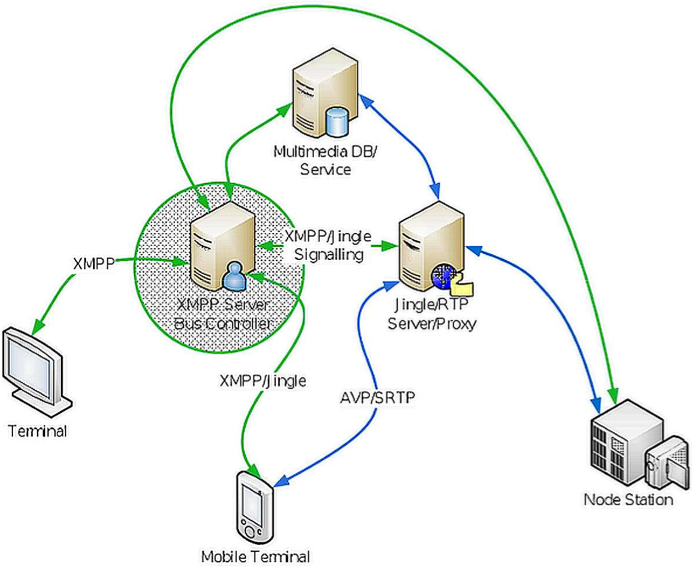
Server can have adapters/gateways to be able to talk to other protocols such as those used by ICQ, AIM, Yahoo and others.



image

Image Source: [https://en.wikipedia.org](https://en.wikipedia.org/wiki/XMPP)

Sometimes the servers are used only for signaling, in order to establish a media connection. [XEP-0166: Jingle](http://xmpp.org/extensions/xep-0166.html) is a protocol extension for initiating and managing peer-to-peer media sessions between two XMPP entities.



image

Image Source: [http://xmppjingle.blogspot.pt](http://xmppjingle.blogspot.pt/2010/09/jingle-nodes-proxy-and-eu-surveillance.html)

### APIs, Bindings and Extensions

#### APIs

|  |  |  |
| --- | --- | --- |
| Name | Language(s) | Details |
| agsXMPPSDK | C#/.NET/Mono | [Link](http://www.ag-software.net/agsxmpp-sdk/) |
| AnyEvent::XMPP | Perl | [Link](http://www.ta-sa.org/projects/net_xmpp2.html) |
| as3xmpp | Flash/ActionScript | [Link](http://code.google.com/p/seesmic-as3-xmpp/) |
| asmack | Java(Android) | [Link](https://github.com/flowdalic/asmack) |
| AXMPP | Ada | [Link](http://forge.ada-ru.org/axmpp) |
| Babbler | Java | [Link](http://babbler-xmpp.blogspot.de/) |
| Babylon | Ruby | [Link](https://github.com/bryanwoods/babylon) |
| Blather | Ruby | [Link](http://adhearsion.com/blather) |
| cl-xmpp | Lisp | [Link](http://common-lisp.net/project/cl-xmpp/) |
| CoversantSoapBoxSDKStudio | C#/.NET/Mono/C | [Link](http://www.coversant.com/products/soapbox-sdk) |
| dojox.xmpp | JavaScript | [Link](http://api.dojotoolkit.org/jsdoc/1.3/dojox.xmpp) |
| dxmpp | C | [Link](http://deusexmachinae.se/dxmpp) |
| EchomineFeridian | Java | [Link](http://freecode.com/projects/feridian) |
| Eiffel | PHP | [Link](https://github.com/jocelyn/bricabrac/wiki/Eiffel-XMPP) |
| emite | Java | [Link](https://github.com/EmiteGWT) |
| exmpp | Erlang | [Link](http://exmpp.org/) |
| frabjous | JavaScript | [Link](https://github.com/theozaurus/frabjous) |
| gloox | C | [Link](http://camaya.net/gloox) |
| goexmpp | Go | [Link](http://code.google.com/p/goexmpp/) |
| headstock | Python | [Link](https://github.com/Lawouach/headstock) |
| hsxmpp | Haskell | [Link](http://חנוך.se/hsxmpp/) |
| hxmpp | haXe | [Link](http://hxmpp.disktree.net/) |
| iksemel | C | [Link](http://code.google.com/p/iksemel/) |
| IP\*WorksInternetToolkit | ActiveX C C# .NET Mono Delphi Java | [Link](http://www.nsoftware.com/ipworks/) | | Iris | C | [Link](http://delta.affinix.com/iris/) | | jabber-net | C#/.NET/Mono | [Link](http://code.google.com/p/jabber-net/) | | jabber.py | Python | [Link](http://jabberpy.sourceforge.net/) | | JabberLib | Tcl | [Link](http://coccinella.im/jabberlib) | | JabberStreamObjects(JSO) | Java | [Link](http://java.net/projects/jso/) | | JAXL | PHP | [Link](http://code.google.com/p/jaxl/) | | jQuery-XMPP-plugin | JavaScript | [Link](http://github.com/maxpowel/jQuery-XMPP-plugin) | | Jreen | C/Qt | [Link](http://qutim.org/jreen) | | JSJaC | JavaScript | [Link](http://blog.jwchat.org/jsjac/) | | libstrophe | C | [Link](http://strophe.im/) | | Lightr | PHP | [Link](http://code.google.com/p/lightr/) | | Loudmouth | C | [Link](http://groups.google.com/group/loudmouth-dev) | | Loudmouth | Ruby | [Link](http://groups.google.com/group/loudmouth-dev/web/loudmouth-ruby) | | MatriX | C#/.NET/Mono | [Link](http://www.ag-software.net/matrix-xmpp-sdk/) | | Net::XMPP | Perl | [Link](http://search.cpan.org/dist/Net-XMPP/) | | node-xmpp | JavaScript | [Link](http://github.com/astro/node-xmpp) | | oajabber | C | [Link](http://www.openaether.org/oajabber.html) | | PontariusXMPP | Haskell | [Link](https://github.com/pontarius/pontarius-xmpp/) | | pyxmpp | Python | [Link](http://pyxmpp.jajcus.net/) | | QXmpp | C | [Link](http://code.google.com/p/qxmpp/) | | seesmic-as3-xmpp | Flash/ActionScript | [Link](http://code.google.com/p/seesmic-as3-xmpp/) | | SleekXMPP | Python | [Link](http://github.com/fritzy/SleekXMPP/wiki) | | Smack | Java | [Link](http://igniterealtime.org/projects/smack/) | | stanza.io | JavaScript | [Link](https://github.com/legastero/stanza.io) | | strophe.js | JavaScript | [Link](http://strophe.im/) | | StropheCappuccino | Objective-J | [Link](http://github.com/primalmotion/strophecappuccino) | | Swiften | C | [Link](http://swift.im/swiften/) | | Tinder | Java | [Link](http://igniterealtime.org/projects/tinder/) | | txmpp | C | [Link](http://github.com/silas/txmpp) | | TwistedWords | Python | [Link](http://twistedmatrix.com/trac/) | | Ubeity | C# | [Link](https://github.com/ubiety/xmpp) | | Verse | Lua | [Link](http://matthewwild.co.uk/projects/verse/home) | | XIFF | Flash/ActionScript | [Link](http://igniterealtime.org/projects/xiff/) | | xmpp-psn | Python | [Link](http://code.google.com/p/xmpp-psn/) | | jaxmpp2 | Java/Android/GoogleWebToolkit | [Link](https://projects.tigase.org/projects/jaxmpp2) | | xmpp4js | JavaScript | [Link](http://xmpp4js.sourceforge.net/) | | XMPP4R | Ruby | [Link](http://home.gna.org/xmpp4r/) | | xmpp4r-simple | Ruby | [Link](http://code.google.com/p/xmpp4r-simple/) | | xmppframework | ObjectiveC | [Link](https://github.com/robbiehanson/XMPPFramework) | | xmpphp | PHP | [Link](http://code.google.com/p/xmpphp/) | | xmppy | Python | [Link](http://xmpppy.sourceforge.net/) | | XMPP-FTW | JavaScript | [Link](https://github.com/lloydwatkin/xmpp-ftw) | | Z-XMPP | JavaScript | [Link](http://ivan.vucica.net/zxmpp/) | |  |  |

#### Relevant Extensions

* [XEP-0166: Jingle](http://xmpp.org/extensions/xep-0166.html) Protocol extension for initiating and managing peer-to-peer media sessions (e.g., voice chat, video chat, file transfer) with a wide variety of transport methods (e.g., TCP, UDP, ICE, application-specific transports)
* [XEP-0343: Signaling WebRTC datachannels in Jingle](http://xmpp.org/extensions/xep-0343.html) Defines how to use the ICE-UDP Jingle transport method to send media data using WebRTC DataChannels.
* [XEP-0060: Publish-Subscribe](http://www.xmpp.org/extensions/xep-0060.html) Protocol extension for generic publish-subscribe functionality
* [XEP-0045: Multi-User Chat](http://xmpp.org/extensions/xep-0045.html) Protocol extension for multi-user text chat.
* [XEP-0072: SOAP Over XMPP](http://www.xmpp.org/extensions/xep-0072.html) Defines methods for transporting SOAP messages over XMPP
* [XEP-0124: Bidirectional-streams Over Synchronous HTTP (BOSH)](http://xmpp.org/extensions/xep-0124.html) bidirectional TCP connection between two entities (such as a client and a server) by efficiently using multiple synchronous HTTP request/response pairs.
* [XMPP over WebSocket](http://tools.ietf.org/html/rfc7395) Binding for the XMPP protocol over a WebSocket transport layer. It provides higher performance than the current HTTP binding for XMPP.
* [XEP-0030: Service Discovery](http://xmpp.org/extensions/xep-0030.html) Protocol extension for discovering information about other XMPP entities.

### Requirements Analysis

Analysis against **Messaging Node** Requirements

* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* Yes
* the Client Server API could be wrapped in a protocol stub, that can be downloaded at runtime
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* Yes
* [Message Caching](https://github.com/reTHINK-project/core-framework/issues/19)
* Yes
* Using [XEP-0203: Delayed Delivery](http://xmpp.org/extensions/xep-0203.html)
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* Yes
* Using [XEP-0313: Message Archive Management](http://xmpp.org/extensions/xep-0313.html) and [XEP-0136: Message Archiving](http://xmpp.org/extensions/xep-0136.html)
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* Yes
* Using [XEP-0184: Message Delivery Receipts](http://xmpp.org/extensions/xep-0184.html) and/or [XEP-0079: Advanced Message Processing](http://xmpp.org/extensions/xep-0079.html)
* [More info](http://www.isode.com/whitepapers/reliable-xmpp.html)
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* Yes
* Using scalable Erlang-based servers [mongooseim](https://www.erlang-solutions.com/products/mongooseim-massively-scalable-ejabberd-platform) or [ejabberd](http://docs.ejabberd.im/architect/) clusters can handle tens of millions of users.
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* Yes
* If Using [XEP-0198: Stream Management](http://xmpp.org/extensions/xep-0198.html)
* [More info](http://www.isode.com/whitepapers/reliable-xmpp.html)
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* Yes
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* Yes
* [ms latency](https://www.ejabberd.im/benchmark)
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* Yes
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* Yes
* Clients based on C libraries can run on embedded systems
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* Yes
* Using [XEP-0178: Best Practices for Use of SASL EXTERNAL](http://xmpp.org/extensions/xep-0178.html)
* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* Yes
* Using [XEP-0060: Publish-Subscribe](http://www.xmpp.org/extensions/xep-0060.html)

## MQTT Evaluation

MQ Telemetry Transport (MQTT) is a lightweight broker-based publish/subscribe messaging protocol designed to be open, simple, lightweight and easy to implement. These characteristics make it ideal for use in constrained environments such as machine-to-machine (M2M)/"Internet of Things" connectivity scenarios.

MQTT-SN (MQTT for Sensor Networks) is a variation of the main protocol aimed at embedded devices on non-TCP/IP networks, such as ZigBee.

### Overview

In the scope of the reTHINK project, MQTT is a candidate technology for the Messaging Node.

#### Highlights:

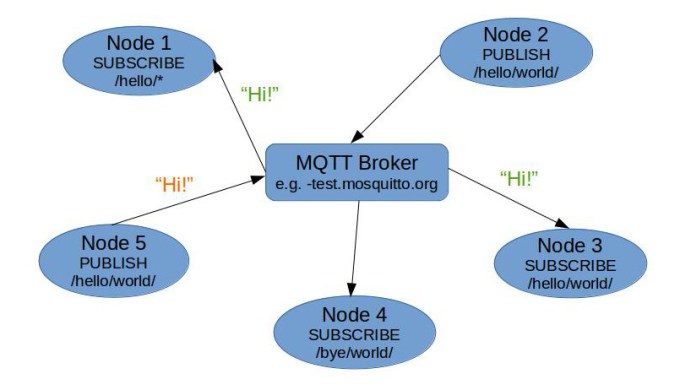
MQTT was designed for low-bandwidth, high latency networks in the late 1990s/early 2000s. As a result, the designers made a number of key choices which influenced the way it "looks and feels".

* Simplicity, simplicity, simplicity! Don't add too many "bells and whistles" but provide a solid building block which can easily be integrated into other solutions. Be simple to implement.
* Publish/subscribe messaging. Useful for most sensor applications, and enables devices to come online and publish "stuff" that hasn't been previously known about or predefined.
* Zero administration (or as close as possible). Behave sensibly in response to unexpected actions and enable applications to "just work", e.g. by dynamically creating topics when needed.
* Minimise the on-the-wire footprint. Add an absolute minimum of data overhead to any message. Be lightweight and bandwidth efficient.
* Expect and cater for frequent network disruption (for low bandwidth, high latency, unreliable, high cost-to-run networks)... -> [Last Will and Testament](http://www.hivemq.com/mqtt-essentials-part-9-last-will-and-testament/)
* Continuous session awareness -> [Last Will and Testament](http://www.hivemq.com/mqtt-essentials-part-9-last-will-and-testament/)
* Expect that client applications may have very limited processing resources available.
* Provide traditional messaging qualities of service where the environment allows. Provide "quality of service"
* Data agnostic. Don't mandate content formats, remain flexible.

## Architecture

MQTT has a client/server model, where every node is a client that connects to a server, know as a broker, over TCP. MQTT is message oriented. Messages are published to a topic. Clients may subscribe to multiple topics or use a wildcard matching based subscription.

MQTT supports three quality of service levels, "Fire and forget", "delivered at least once" (QoS1) and "delivered exactly once" (QoS2).



image

Image Source: [https://sakshambhatla.wordpress.com](https://sakshambhatla.wordpress.com/2014/08/11/simple-mqtt-broker-and-client-in-python/)

## APIs and Bindings

#### Specification

The current formal protocol specification can be found at:

[MQTT v3.1.1 specification](http://docs.oasis-open.org/mqtt/mqtt/v3.1.1/mqtt-v3.1.1.html)

#### Device-Specific

* [Arduino](https://github.com/knolleary/pubsubclient) ([more information](http://knolleary.net/arduino-client-for-mqtt/))
* [mbed](https://github.com/yilun/MQTT-client-on-mbed) ([more information](http://ceit.uq.edu.au/content/mqttclient-mbed-version-20))
* [mbed (simple port of the Arduino pubsubclient)](http://mbed.org/users/jwende/code/MQTT/)
* [mbed (native implementation)](http://mbed.org/users/Nim65s/code/niMQTT/)
* [mbed (Paho Embedded C++ port)](http://developer.mbed.org/teams/mqtt/code/MQTT/) ([more information] (https://www.eclipse.org/paho/clients/c/embedded/))
* [mbed (Paho Embedded C port)](http://developer.mbed.org/teams/mqtt/code/MQTTPacket/) ([more information](https://www.eclipse.org/paho/clients/c/embedded/))
* [Nanode](http://github.com/njh/NanodeMQTT/)
* [Netduino](https://github.com/danielan/NetduinoMQTT)
* [M2MQTT (works with .Net Micro Framework)](https://m2mqtt.codeplex.com/)

(see also [devices](things) page for more on hardware with built-in support)

#### Actionscript

* [as3MQTT](https://github.com/yangboz/as3MQTT)

#### Bash

* see [Shell Script](#shell-script), below

#### C

* [Eclipse Paho C] (https://www.eclipse.org/paho/clients/c/)
* [Eclipse Paho Embedded C] (https://www.eclipse.org/paho/clients/c/embedded/)
* [libmosquitto](http://mosquitto.org)
* [libemqtt](https://github.com/menudoproblema/libemqtt) - an embedded C client

#### C++

* [Eclipse Paho C++] (https://www.eclipse.org/paho/clients/cpp/)
* [libmosquittopp](http://mosquitto.org)
* [Eclipse Paho Embedded C++] (https://www.eclipse.org/paho/clients/c/embedded/)

#### Clojure

* [Machine Head](http://clojuremqtt.info)
* [Clojure MQTT Codec for Netty](https://github.com/xively/clj-mqtt/)

#### Dart

* [mqtt.dart](http://pub.dartlang.org/packages/mqtt)

#### Delphi

* [TMQTTClient](http://jamiei.com/code/TMQTTClient.zip) ([more information](http://jamiei.com/blog/code/mqtt-client-library-for-delphi/))

#### Erlang

* [erlmqtt](https://github.com/squaremo/erlmqtt)
* [emqttc](https://github.com/emqtt/emqttc) - Erlang MQTT Client
* [mqtt4erl](http://code.google.com/p/mqtt4erl/)
* [my-mqtt4erl](http://code.google.com/p/my-mqtt4erl/) - updated fork of mqtt4erl

#### Elixir

* [hulaaki](https://github.com/suvash/hulaaki)

#### Go

* [Eclipse Paho Go](http://git.eclipse.org/c/paho/org.eclipse.paho.mqtt.golang.git/)

#### Haskell

* [mqtt-hs](http://hackage.haskell.org/package/mqtt-hs)

#### Java

* [Eclipse Paho Java](http://git.eclipse.org/c/paho/org.eclipse.paho.mqtt.java.git/)
* [Xenqtt](http://xenqtt.sf.net) Includes a client library, mock broker for unit/integration testing, and applications to support enterprise needs like using a cluster of servers as a single client, an HTTP gateway, etc.
* [MeQanTT](https://github.com/AlbinTheander/MeQanTT)
* [Fusesource mqtt-client](https://github.com/fusesource/mqtt-client)
* [moquette](http://code.google.com/p/moquette-mqtt/)
* ["MA9B" zip of 1/2 dozen mobile clients source code. Includes Android-optimized Java source that works with Android notifications, based on Paho](http://www-933.ibm.com/support/fixcentral/swg/selectFix?product=ibm%2FWebSphere%2FWebSphere+MQ&fixids=1.0.0.1-WS-MQCP-MA9B&source=dbluesearch&function=fixId&parent=ibm/WebSphere)
* [IA92](http://www-01.ibm.com/support/docview.wss?rs=171&uid=swg24006006&loc=en_US&cs=utf-8&lang=en) - *deprecated* IBM IA92 support pack, use Eclipse Paho GUI client instead. A useful MQTT Java swing GUI for publishing & subscribing. The Eclipse Paho GUI is identical but uses newer client code

#### Javscript / Node.js

* [Eclipse Paho HTML5 JavaScript over WebSocket.](http://git.eclipse.org/c/paho/org.eclipse.paho.mqtt.javascript.git/)
* [mqtt.js](https://github.com/adamvr/MQTT.js)
* [node\_mqtt\_client](https://github.com/yilun/node_mqtt_client) ([more information](http://ceit.uq.edu.au/content/simple-mqtt-cient-nodejs))
* [IBM-provided PhoneGap / Apache Cordova MQTT plug-in for Android](http://www-01.ibm.com/support/docview.wss?rs=171&uid=swg24033580&loc=en_US&cs=utf-8&lang=en) - JavaScript API is identical to Eclipse Paho HTML5 JavaScript
* [Ascoltatori](https://github.com/mcollina/ascoltatori) - a node.js pub/sub library that allows access to Redis, AMQP, MQTT and ZeroMQ with the same API.

#### LotusScript

* [MQTT From LotusScript](https://tingenek.wordpress.com/2011/11/30/mqtt-with-lotus-notes/)

#### Lua

* [Eclipse Paho Lua](http://git.eclipse.org/c/paho/org.eclipse.paho.mqtt.lua.git/)

####.NET / dotNET

* [MqttDotNet](http://sourceforge.net/projects/mqttdotnet/)
* [nMQTT](https://github.com/markallanson/nmqtt)
* [M2MQTT](https://m2mqtt.codeplex.com/)
* [KittyHawkMQ] (http://www.kittyhawkmq.com/)

#### Objective-C

* [mqttIO-objC](https://github.com/m2mIO/mqttIO-objC)
* [libmosquitto](https://mosquitto.org) - via wrappers ([example](https///github.com/njh/marquette))
* [MQTTKit](https://github.com/jmesnil/MQTTKit) ([sample app](https///github.com/jmesnil/MQTTExample))
* ["MA9B" zip of 1/2 dozen mobile clients source code including Objective-C](http://www-933.ibm.com/support/fixcentral/swg/selectFix?product=ibm%2FWebSphere%2FWebSphere+MQ&fixids=1.0.0.1-WS-MQCP-MA9B&source=dbluesearch&function=fixId&parent=ibm/WebSphere)

#### OCaml

* [mqtt\_client](https://github.com/philtomson/mqtt_client)

#### Perl

* [net-mqtt-perl](https://github.com/beanz/net-mqtt-perl)
* [anyevent-mqtt-perl](https://github.com/beanz/anyevent-mqtt-perl)
* [WebSphere-MQTT-Client](http://search.cpan.org/dist/WebSphere-MQTT-Client/)
* Net::MQTT::Simple [cpan](https://metacpan.org/pod/Net::MQTT::Simple) [github](https://github.com/Juerd/Net-MQTT-Simple)

#### PHP

* [phpMQTT](http://github.com/bluerhinos/phpMQTT)
* [Mosquitto-PHP](https://github.com/mgdm/Mosquitto-PHP)
* [sskaje's MQTT library](http://github.com/sskaje/mqtt)

#### Python

* [Eclipse Paho Python](http://git.eclipse.org/c/paho/org.eclipse.paho.mqtt.python.git/) - originally the mosquitto Python client
* [nyamuk](https://github.com/iwanbk/nyamuk)
* [MQTT for twisted python](https://github.com/adamvr/MQTT-For-Twisted-Python)

#### REXX

* [REXX MQTT](https://github.com/DougieLawson/REXX_MQTT)

#### Ruby

* [ruby-mqtt](https://github.com/njh/ruby-mqtt)
* [em-mqtt](https://rubygems.org/gems/em-mqtt)
* [mosquitto](https://github.com/xively/mosquitto)

#### Shell Script

* [bish-bosh](https://github.com/raphaelcohn/bish-bosh), supports bash, ash (including BusyBox), pdksh and mksh.

#### Tcl

* [tcl-mqtt](https://github.com/Tingenek/tcl-mqtt)

### Brokers

Server | QoS 0 | QoS 1 | QoS 2 | auth | [bridge](bridge_protocol) | [$SYS](conventions#$sys) | SSL | [dynamic topics](are_topics_dynamic) | cluster | websockets | plugin system ------ | ----- | ----- | ----- | ---- | ------------------------- | ------------------------ | --- | ------------------------------------ | ------- | ---------- | ------------- | [mosquitto](mosquitto_message_broker) | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✘ | ✔ | ✔ | [RSMB](http://mqtt.org/wiki/doku.php/really_small_message_broker) | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✘ | ✔ | ✘ | ✘ | ? | [WebSphere MQ](http://www-03.ibm.com/software/products/en/wmq/) | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ? | ? | ? | [HiveMQ](http://www.hivemq.com) | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | [Apache Apollo](http://activemq.apache.org/apollo) | ✔ | ✔ | ✔ | ✔ | ✘ | ✘ | ✔ | ✔ | ? | ✔ | ? | [Apache ActiveMQ](http://activemq.apache.org/) | ✔ | ✔ | ✔ | ✔ | ✘ | ✘ | ✔ | ✔ | ✔ | ✔ | ✔ | [my-Channels Nirvana Messaging](http://www.my-channels.com/products/nirvana.html) | ✔ | ✔ | ✔ | § | ✘ | ✘ | ✔ | ✘ | ? | ? | ? | [RabbitMQ](http://www.rabbitmq.com/blog/2012/09/12/mqtt-adapter/) | ✔ | ✔ | ✘ | ✔ | ✘ | ✘ | ✔ | ✔ | ? | ? | ? | [MQTT.js](https///github.com/adamvr/MQTT.js) | ✔ | ✘ | ✘ | § | ✘ | ✘ | ✔ | ✔ | ✘ | ? | ✘ | [moquette](http://code.google.com/p/moquette-mqtt/) | ✔ | ✔ | ✘ | ? | ? | ? | ? | ? | ✘ | ✘ | ✘ | <mosca> | ✔ | ✔ | ✘ | ✔ | ? | ? | ? | ? | ✘ | ✔ | ✘ | [IBM MessageSight](http://www-03.ibm.com/software/products/en/messagesight/) | ✔ | ✔ | ✔ | ✔ | ✘ | ✔ | ✔ | ✔ | § | ✔ | ✘ | [2lemetry](http://2lemetry.com/platform/) | ✔ | ✔ | ✔ | ✔ | ✔ | § | ✔ | ✔ | ✔ | ✔ | ✘ | [GnatMQ](http://mqttbroker.codeplex.com/) | ✔ | ✔ | ✔ | ✔ | ✘ | ✘ | ✘ | ✔ | ✘ | ✘ | ✘ | [JoramMQ](http://mqtt.jorammq.com) | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | [ThingMQ](https://thingmq.com) | ✔ | ✔ | ✔ | ✔ | ✔ | ✘ | ✔ | ✔ | ✔ | ✔ | ✔ | [VerneMQ](https://verne.mq) | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | ✔ | Key: ✔ supported ✘ not supported ? unknown § see limitations

## Requirements Analysis

Analysis against **Messaging Node** Requirements

* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* Yes
* the Client Server API could be wrapped in a protocol stub, that can be downloaded at runtime
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* Yes
* Using [MQTT over WebSockets](http://www.hivemq.com/mqtt-essentials-special-mqtt-over-websockets/)
* [Message Caching](https://github.com/reTHINK-project/core-framework/issues/19)
* Yes
* Using [Persistence Sessions](http://www.hivemq.com/mqtt-essentials-part-7-persistent-session-queuing-messages/)
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* Yes
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* Yes
* Using a QoS1 or QoS2 level.
* [More Info](http://www.hivemq.com/mqtt-essentials-part-6-mqtt-quality-of-service-levels/)
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* Yes
* Some Brokers can be clusters
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* Yes
* Using [Persistence Sessions](http://www.hivemq.com/mqtt-essentials-part-7-persistent-session-queuing-messages/)
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* Yes
* Called : [Last Will and Testament](http://www.hivemq.com/mqtt-essentials-part-9-last-will-and-testament/)
* And Using [MQTT Keep Alive and Client Take-Over](http://www.hivemq.com/mqtt-essentials-part-10-alive-client-take-over/)
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* Yes, because of the small overhead and simple architecture
* [In the usec range and 273M mobile messages/sec per rack](https://mobilebit.wordpress.com/2013/05/03/rest-is-for-sleeping-mqtt-is-for-mobile/)
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* Yes
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* Yes
* Core Feature
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* [Yes](http://docs.oasis-open.org/mqtt/mqtt/v3.1.1/os/mqtt-v3.1.1-os.html#_Toc398718111)
* Depends on the Broker, can be implemented on a higher abstraction layer [Link1](https://auth0.com/docs/scenarios/mqtt) [LInk2](http://www.hivemq.com/mqtt-security-fundamentals-oauth-2-0-mqtt/)
* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* Yes
* Core Feature

# PSYC

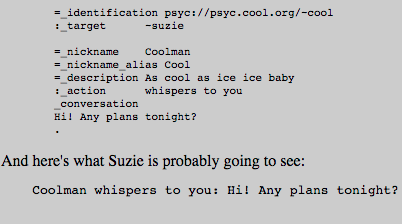
PSYC is a mostly text-based protocol, aiming at providing a decentralized global messaging infrastructure for unicast/multicast chatting and social media exchanging. Its goal is to replace the popular IRC protocol, which currently suffers from unscalability, lack of security and requires complex bureaucracy. This is because in IRC each server node may take responsibility over control aspects of a given network channel he participates in, adding this way unneccessary complexity and also raising security issues, like when an evil server takes the authority of the channel for malicious purposes. PSYC solves this by implementing a context master node (or well-known distributed set of nodes), that the protocol recognizes as the authority node(s). All other server nodes distributing this context to their members may perform jobs on behalf of the context master, but authority questions like allowing/banning users to/from entering the channel is always up to the master. Also, unlike current protocols like talk or msend, PSYC intends to be more people-oriented, identifying them through Uniform Network Identifications (UNI) (e.g. psyc://psyc.kanzlerant.de/~gerhard) or Uniform Network Locations (UNL) in case of a shared messaging room or conference.

## Architecture and main functions

On a top-level view, PSYC combines an IRC-like topology with the concepts of context, logical targets and packet IDs. Multiple PSYC nodes can be deployed in a tree to help in routing messages from the source to the destination, as it happens in IRC. Context slaves allow better routing options to be automatically discovered when multiple recipients ask to receive data from a given source (UNI/UNL), lowering the amount of traffic. Logical targets are the end-users to whom a given message is targeted. Finally, packet IDs allow PSYC to use redundant multicast strategies, when more than one may fit our needs. This way, duplicate packets due to multiple strategies can be caught and ignored. On a conference server, the minimalistic control module can be also used to deliver group messages in a peer to peer manner, by maintaining on each member a list of other members and how to reach them.

## Evolution to PSYC2

Due to the deprecated state of some key-concepts PSYC relies on, such as its uniform-based routing layer and federation architecture, the PSYC project is moving onto its second version: PSYC2. This new version combines the PSYC old message syntax with a pseudonymous routing technology which defines that an entity's address is its public key itself instead of a string in a uniform, and these public keys should be looked up in a Distributed Hash Table (DHT), avoiding this way the misunderstanding and spoofing problems that the uniform-based solution presented. Apart from that, the federation concept stated that each entity should run its own server and applications had the responsibility of connecting to the appropriate servers. This brings privacy and trust issues, since two entities exchanging messages on a server would need to trust the server owner, which generally is one of the reasons why users continued using the centralized messaging services offers instead of these free solutions. Also, there is a scalability problem due to the lack of resources on entity-owned servers. For example, an average linux-server could not efficiently distribute a multicast message to millions of recipients, in contrary to the powerful servers supporting big companies' services. To overcome these problems, PSYC developers are working on a fully-distributed end-to-end privacy-enabled solution, [secushare](http://secushare.org), a distributed social network operating on top of the public-key routing method explained above combined with PSYC messaging logic.



image

Figure 1. PSYC message example

### Requirements Analysis

Analysis against **Messaging Node** Requirements

* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* The PSYC2 project implementation, Secushare, comes with an [API for pubsub](http://secushare.org/pubsub)
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* PSYC has its own authentication method, inherited in PSYC2 specification. A request\_authentication\_method is used to query a UNI if a given network entity is actually a linked location of that UNI. This method can have different arguments (\_location, \_host\_IP, \_nonce, \_password) to help the querying entity to take a decision. However, there is no evidence in the documentation that PSYC is able to accept external authentication/authorisation methods other than this.
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* The performance of PSYC has been studied, and its wiki relies on benchmarks to say that PSYC is the fastest, yet extensible text-based protocol they are aware of. However, the benchamrk results are not available at this time.
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* PSYC server (psyced) can currently be deployed on Linux, Mac OSX and Windows (on a Cygwin environment) systems.
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* PSYC applies some techniques in order to reduce message delivery latency. First, by avoiding negotiations between nodes "talking" the same protocol between them. Since PSYC supports IRC and XMPP, if two nodes are exchanging messages through XMPP protocol, PSYC suggests the protocol switch in order to reduce latency. Also, PSYC avoids resource discovery (disco on XMPP) by pushing information to possibly interested recipients in advance. However, by applying TLS for encrypted PSYC and techniques for DoS prevention on psyced, a certain degree of latency is, therefore, inevitable.
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* PSYC currently features notification interfaces for software versioning systems (CVS and Git), syslog daemon events, MediaWiki page edits, phpBB forum events and IRC chat messages. Currently, there is no reference to notifications on clients' connection to and disconnection from messaging nodes.
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* TODO
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* TODO
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* PSYC provides three message families to inform clients about problems on message delivery. The \_error method family features methods like \_error\_invalid, \_error\_illegal, \_error\_duplicate and informs the client of a problem occuring on his side, rather on the server side. Basically, it is the server telling a client "It's your fault, not mine". The \_failure method family informs a message sender about a problem on the receiving side. This method family features methods like \_failure\_deliver or \_failure\_redirect (when a given destination changed its address). Finally, the \_warning method family means that a message was processed and sent, but maybe not as intended. An example is \_warning\_usage, which indicates a possible mistake on the message syntax, presented on Fig. 1, and has a single variant for each of the syntax fields.
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* On PSYC, each server running psyced implements the concept of "log of last messages" for every UNI registered on that server. It is used to store messages received by the server, and to let every user have its last session backlog whenever he logs in. It is possible to tune the log size for each UNI and to export the chat history of a room to a webpage.
* [Message Caching](https://github.com/reTHINK-project/core-framework/issues/19)
* PSYC does not have any reference to Store and Forward, probably because it goes against real time communication. And about caching in general, it states it's oriented towards using push events instead of caching. So, whenever a push event regarding a given resource is received by the server, anyone accessing the resource at that time will see it refresh, in order to present always the most recent version.
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* Current implementations of PSYC do not support WebSockets nor HTTP Live Streaming. About HTTP Long-Polling, it does not make much sense in the context of PSYC, since it models all data distribution based on an event push system. So, whenever some potentially interesting information for a recipient is available at the server, it is automatically sent, overcoming this way the need of something like HTTP Long-Polling or even REST.
* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* The psyced implementation of PSYC has a negotiation feature of protocol switch advertising. This way, each node has information about supported protocols on all the nodes it is communicating with. However, this could be achieved, since the Client-Server API could be wrapped in a protocol stub, that can be downloaded at runtime.

## Node.js

## Asset Evaluation

### Overview

In the scope of the reTHINK project NodeJs is a candidate technology for the Messaging Node.

From https://nodejs.org/ :

Node.js® is a platform built on Chrome's JavaScript runtime for easily building fast, scalable network applications. Node.js uses an event-driven, non-blocking I/O model that makes it lightweight and efficient, perfect for data-intensive real-time applications that run across distributed devices.

Several additional packages are available to be used with nodeJs : https://www.npmjs.com/

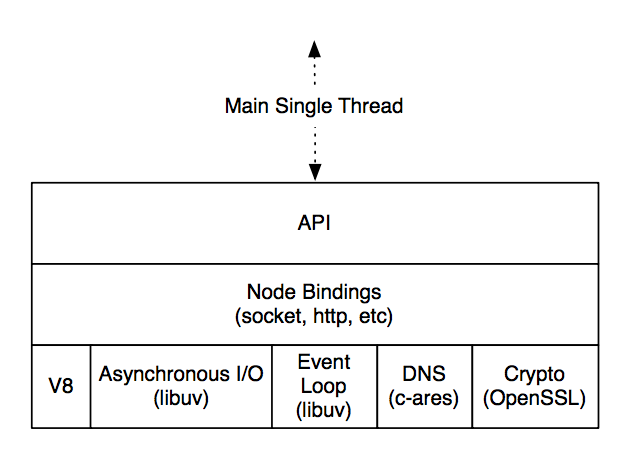
It can be used with component like Redis, socket.io to enhance connectivity and redundancy

A fork of nodejs exist : iojs but a merge seems to be ongoing.

NodeJs is already used in several WebRTC platform or product. (NodeJs can be used in front of Kurento)

### Architecture

NodeJS is divided into two main components: the core and its modules. The core is built in C and C++. It combines Google’s V8 JavaScript engine with Node’s Libuv library and protocol bindings including sockets and HTTP.



image

More information are avaialble in this document : http://mcgill-csus.github.io/student\_projects/Submission2.pdf

### APIs

Available API are described on nodejs website : https://nodejs.org/api/

But this can be completed thanks to the different packages available on https://www.npmjs.com/

### Requirements Analysis

#### [Messaging Node Requirements](https://github.com/reTHINK-project/core-framework/labels/Messaging%20Node%20Requirement)\*\*

* [It should be possible to support Protocol on-the-fly](https://github.com/reTHINK-project/core-framework/issues/21)
* Yes
* ProtOFly connector can be developped. JS connector can be develop on top of NodeJs to enble protofly on server side. This connector will be for example reusable to connect Kurento Media Server
* [Messaging Transport Protocols](https://github.com/reTHINK-project/core-framework/issues/20)
* Yes (socket.io). Socket.io enables the usage of different transport protocol to establish connection between user and server. (Long polling, WebSocket ...)
* [Message Node logging](https://github.com/reTHINK-project/core-framework/issues/18)
* Yes - Several logging modules available : log4js, winston, bunyan ... Logs can be dispalyed in console, store in file with log rotate, send to a network entity ...
* [Message delivery reliability](https://github.com/reTHINK-project/core-framework/issues/17)
* Yes - Socket.io enables message acknowledgement
* [Messaging Node deployments with carrier grade scalability](https://github.com/reTHINK-project/core-framework/issues/16)
* Using:
  + Cluster Mode
  + Redis cluster : it is possible to use Redis Cluster with PUB/SUB mechanism : several NodeJs entities can be connected through the redis cluster : this can enable load balancing, redundancy
* [Messaging Node should be tolerant to unstable connections](https://github.com/reTHINK-project/core-framework/issues/15)
* Yes - socket.io can manage reconnection with different configurable parameters (timeout, retries ...)
* reconnection whether to reconnect automatically (true)
* reconnectionDelay how long to wait before attempting a new reconnection (1000)
* reconnectionDelayMax maximum amount of time to wait between reconnections (5000). Each attempt increases the reconnection by the amount specified by reconnectionDelay.
* timeout connection timeout before a connect\_error and connect\_timeout events are emitted (20000)
* [Events about clients connection / disconnection from Messaging Node](https://github.com/reTHINK-project/core-framework/issues/14)
* Yes - using socket.io different event are fired on connection status :
* connect. Fired upon connecting.
* error. Fired upon a connection error
* disconnect. Fired upon a disconnection.
* reconnect. Fired upon a successful reconnection.
* reconnect\_attempt. Fired upon an attempt to reconnect.
* reconnecting. Fired upon an attempt to reconnect.
* reconnect\_error. Fired upon a reconnection attempt error.
* reconnect\_failed. Fired when couldn’t reconnect within reconnectionAttempts
* [Messaging Node must support very low message delivery latency](https://github.com/reTHINK-project/core-framework/issues/13)
* Yes
* [Messaging Node must be deployable in the most used Virtual Machines](https://github.com/reTHINK-project/core-framework/issues/12)
* Yes - NodeJs is available on Linux, windows, mac
* [Messaging Node should require minimal computing resources](https://github.com/reTHINK-project/core-framework/issues/11)
* Yes
* [Messaging Node must support external authentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10)
* Yes. Module like Passport : http://passportjs.org/ enables to use external authentication like facebook, twitter, google .. (We will have to check if passport can be used as it seems to require Express which may not be relevant in rethink case)
* [Messaging Node must support pub/sub](https://github.com/reTHINK-project/core-framework/issues/9)
* No - Yes with Redis Pub/Sub mechanism : http://redis.io/topics/pubsub

#### [Runtime Requirements](https://github.com/reTHINK-project/core-framework/labels/Runtime%20Requirement)

**Web Sockets on node.js**

This suplemments the lack of implementation of Web Sockets inside v8.

npm install ws

IT is easy to use, fast and up-to-date against current HyBi protocol drafts. IT can send and receive typed arrays (ArrayBuffer, Float32Array) as binary data. IT passes the extensible Autobahn test suite: client report ( Autobahn WebSockets v0.4.5.), server report ( Autobahn WebSockets v0.4.5.).

It has extensive benchmarks relative to its overhall performance in node.js.

In fact there are extensive implementations of Web Sockets over node.js: websocket-node faye-websocket-node engine.io socket.io sockjs

The most performante is claimded to be ws.

Socket.io seems to be the most complete regarding feature (connection type, acknowledgement, configuration ..)

## Vert.x Evaluation

**Note:** to be reviewed for [v3](http://vert-x3.github.io/) by identifying differences with version 2.x

### Overview

*Overview of functionalities and type of WP3 component that the asset can be used for ie Messaging Node, Runtime, Network QoS and Framework*

Vert.x is an application framework developed by VMWare in 2011. The application framework provides possibilities to develope loosely coupled network service applications.

The concept of the framework is summarized as follows: \* **Polyglot (supports several languages)**: Vert.x framework runs on the Java Virtual Machine. However, Java is not required to use Vert.x. As well as languages based on JVM operation, such as Java or Groovy, Vert.x can be used with Ruby, Python, and even JavaScript. In addition, Scala and Closure are planned to be supported. \* **Super Simple Concurrency model**: When building an application by using Vert.x, users can write code as a single thread application. That means that the multi-thread programming effect can be achieved without synchronization, lock, or volatility. However, Vert.x allows to create multiple threads based on the number of CPU cores whlie only one process is executed. It handle the multi-threading so users can focus on implementing business logic. \* **Provides Event Bus**: The main concept of Vert.x is not only to produce a ‘one server process DAEMON'. Vert.x aims to make a variety of Vert.x-built server programs communicate well with each other. For this, Vert.x provides Event Bus. Therefore, functions such as Point to Point or Pub/Sub can be used (to provide Event Bus function, Vert.x uses Hazelcast, an In-Memory Data Grid). With this Event Bus, a server application built with different languages can easily communicate with each other. \* **Module System & Public Module Repository**: Vert.x has a module system. This module system can be understood as a type of component. That means the Vert.x-built server application project itself is modularized. It aims at reusability. Modules can be registered to Public Module Repository. Through the Public Module Repository, the module can be shared

### Architecture

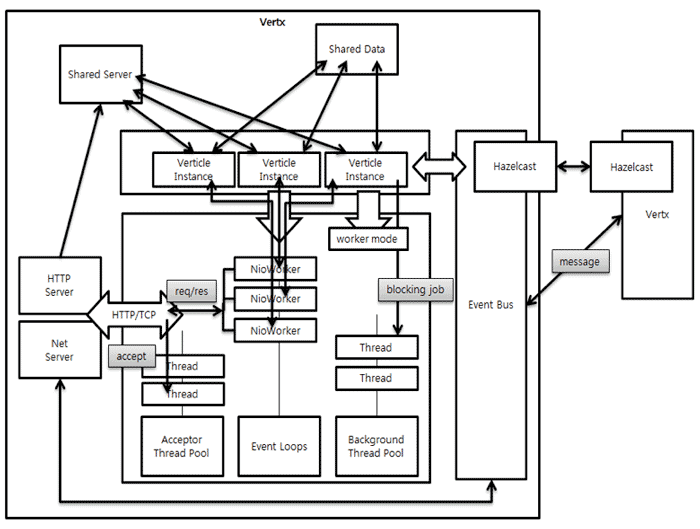
This subsection highlights the main building blocks of the Vert.x architecture. 

Figure 1. Vert.x Architecture

### Addressing

Messages are sent on the event bus to an address. Vert.x instances are not bound to any addressing schemes. An address is simply a string, any string is valid. Some examples of valid addresses are europe.news.feed1, acme.games.pacman, sausages, and X. As a convention the names of the packages that implement certain functionalities should also be represented on the event bus and should be combined with a meaningful event/operation name, e.g. org.acme.MyPackage.MyClass.doSomething

### Handlers

A handler is an entity that receives messages from the event bus. You register a handler at an address. Many different handlers from the same or different modules can be registered at the same address. A single handler can be registered at many different addresses at the same time.

### Messaging Schemes

The Event Bus supports the following modes of operation: \* *Publish / subscribe messaging*: Publishing means delivering the message to all handlers that are registered at that address. This is the familiar publish/subscribe messaging pattern. \* *Point to point and Request-Response messaging*: Messages are routed to just one of the handlers registered at an address. They can optionally be replied to. \* *Remote Procedure Call (RPC)*: This mode of operation is implemented on top of the Request-Response model, basically by enforcing certain conventions on requests and responses

This example shows the Event Bus can be instantiated, how a Handler can be defined and registered on the Event Bus and how the Event Bus can subsequently publish a message for the defined Handler:

EventBus eb = vertx.eventBus();  
  
Handler<Message> myHandler = new Handler<Message>(){  
  
 public void handle(Message message){  
 System.out.println("I just recieved a message "+ message.body);  
 }  
};  
//test.address is the address at which this handler will be registered  
eb.registerHandler("test.address", myHandler);  
  
...  
//publishing a message. The message will be delivered to all handlers registered against the address  
eb.publish("test.address", "hello world");  
//point-2-point sending of message.   
//Only one handler registered at the address receiving the message.   
//The handler is chosen in a non strict round-robin fashion  
eb.send("test.address", "hello world");  
  
...  
  
eb.unregisterHandler("test.address", myHandler);

### Types of Messages

Messages that you send on the event bus can be as simple as a string, a number or a boolean. It is also possible to send Vert.x buffers or JSON messages. It's highly recommended to use JSON messages to communicate between verticles. JSON is easy to create and parse in all the languages that Vert.x supports. For RPC messages, JSON is enforced.

## Verticle

The unit of execution for Vert.x applications is called a Verticle. Verticles can be written in multiple languages (JavaScript, Ruby, Java, Groovy or Python). Many verticles can be executed concurrently in the same Vert.x instance. An application might be composed of multiple verticles deployed on different nodes of the network communicating by exchanging messages over the Vert.x Event Bus. For trivial applications verticles can be run directly from the command line, but more usually they are packaged up into modules.

## Module

Applications within the framework comprise of one or more modules. The framework allows packaging of applications or other re-usable functionality into modules, which can be deployed or used by other code. Module can also by catalogue in the Vert.x module registry so others can discover and use it. The framework offers the possibility to automatically download and install modules from any repository given the module identifier. Each module has a unique identifier. The identifier is a string that is composed of three parts: A module can contain any number of (including zero) verticles and can depend on other modules (and their verticles) in turn. Creating a module with no verticles makes sense to provide only library support for other modules. Modules are described by a descriptor file: mod.json. A minimal descriptor looks like this:

{  
 "owner": "org.acmecorp",  
 "name": "myReThinkAdapterModule",  
 "version": "0.1"  
}

Additionally, three more fields are optionally recognized: \* worker indicates if this is a worker module. See below under event loop. \* main Indicates the startup routine for this module. \* includes Additional module dependencies as a comma-separated string.

## Event Loop

By default, all verticles run in an asynchronous event loop. When developing a verticle, it is essential not to block the event loop. Blocking here means either doing any kind of blocking I/O or even doing any kind of computational intensive work. Modules that do either of these should indicate that they are so called worker modules by setting "worker": true in their *mod.json* file. The advantage of an event loop is that it is enormously scalable. Instead of waiting for I/O operations to complete, the executing thread will rather do other stuff (e.g. servicing the next request) in the meantime. This is achieved by using a callback driven style of programming. Imagine the following scenario: *We want to read some data in an I/O intensive operation (function readData)* We want to do something with that data (function doSomething) *We want to do something completely different (function doSomethingUnrelated)* In the traditional blocking world we would do something like the following:

def doSomething(data):  
 # do something with data  
data = readData()  
doSomething(data)  
doSomethingUnrelated()

What happens here is the following:

After the data is read, the program waits for the operation (readData) to complete (which is consuming the event loop thread lifetime). As soon as readData returns, we have our data and can go on to do something with it (doSomething(data)). Finally, when that is done, we can go on and do other stuff (doSomethingUnrelated).

In the asynchronous world, we do something like this:   
def doSomething(data):  
 # do something with data  
readData(callback = doSomething)  
doSomethingUnrelated()

As can be seen, the result of readData is not received in the functions return value. Instead doSomething is passed in the handler method as a callback. The framework will take care that this handler is called asynchronously as soon as the data is available

### APIs

Vert.x provides the different APIs which are implemented in various languages:

**Core API** \* TCP client/Server API \* HTTP client/Server API \* Transport Protocol (Websocket, SockJS(provides websocket-like API through http), UDP, TCP) \* File System Access \* DNS client API \* Shared Data \* Event Bus API \* JSON API

**Container API** \* Deploy and undeploy verticles \* Deploy and undeploy modules \* Retrieve verticle configuration \* Logging

### Requirements Analysis

*According to Component Type addressed by the solution ie Messaging Node, Runtime, Network QoS and Framework*

##### [Autentication and Authorisation](https://github.com/reTHINK-project/core-framework/issues/10) (PTIN)

External Authentication and Authorisation are supported through the usage of an Authorisation module:

container.deployModule("io.vertx~mod-auth-mgr~2.0.0-final");

The Authorisation module can be the front-end to interact with an external vertx service eg with restful APIs or could be attached to the vertx-io event bus.

**Authorisation to Send/publish a Message**

* SockJSServer, where we need a bridge configuration
* InboudPermitted must have: vertx.basicauthmanager.login and clients handler ```java JsonArray inboundPermitted = new JsonArray();

JsonObject inboundPermitted1 = new JsonObject().putString("address", "vertx.basicauthmanager.login"); inboundPermitted.add(inboundPermitted1); JsonObject inboundPermitted2 = new JsonObject().putString("address", "aliceHandler").putBoolean("requires\_auth",true); inboundPermitted.add(inboundPermitted2); ``` Inboundpermitted allows the use of the "requires\_auth" flag. When it is true, messages will be first forwarded to the authorization module, where decisions to send or not the messages are taken.

**receive a Message**

* OutboundPermitted must have: clients handler. java outboundPermitted.add(new JsonObject().putString("address", "aliceHandler"));
* sockJSServer.bridge(new JsonObject().putString("prefix", "/eventbus"), inboundPermitted, outboundPermitted);

**Example: communication between two javascript clients connected via SockJS**

Both client applications perform log-in on the EventBus.

eb.login('alice','alice123', function(reply){console.log(reply);});

Then the client A(alice) wants to send messages to the client B(bob), so client B(bob) needs to register a handler. Before the client A(alice) can send a message to client B(bob), B must first register himself.

eb.registerHandler('bobHandler', function(reply){console.log(reply);});

After that client A (alice) can publish messages on client B (bob) handler

eb.publish('bobHandler','Hello bob from alice');

When client A publishes a message to client B handler, this message will be first forwarded to the authorization module because of inbounpermitted configuration.

**subscribe / register handlers to be notified about published messages**

In the SockJSServer configuration we can set a Hook (Registers functions to be called when certain events occur on an event bus bridge).

ServerHook hook = new ServerHook(logger);  
sockJSServer.setHook(hook);

ServerHook takes some keyword arguments for example:

* pre-register: Called before a client handler registration is processed.
* public boolean handlePreRegister(SockJSSocket sock, String address) {  
  logger.info("handlePreRegister, sock = " + sock + ", address = " + address);  
  return true;  
   }

In this way handlers registration can be controlled.

##### [Unstable Connections](https://github.com/reTHINK-project/core-framework/issues/15)(PTIN)

Hint from Fokus: Since vertx is based on http://hazelcast.org/ we can use it to cache some info including the sessionId

##### [Carrier grade deployment features (Resilience, DoS and DDoS protection, Service Assurance)](Messaging%20Node%20with%20carrier%20grade%20deployment%20features) (FOKUS)

* Resilience: Vert.x provides resilience through the "automatic failover" and "HA group" options. When a module is run with HA, if the Vert.x instance where it is running fails, it will be re-started automatically on another node of the cluster. An HA group denotes a logical grouping of nodes in the cluster. Only nodes with the same HA group will failover onto one another.
* DoS and DDoS Protection: Vert.x 2.x. has no support for this, BUT Vert.x 3.0 provides built-in core functiionality for this core
* Service Assurance: Modules can be deployed in clusters, and Vert.x provides an internal Load Balancer for routing messages within the cluster. Also the above mentioned "auomatic failover" and "HA group" options contribute to enforce service assurance.

##### [Scalability] (https://github.com/reTHINK-project/core-framework/issues/16) (FOKUS)

Verticle instances, except advanced multi-threaded worker verticles are almost always single threaded. what this implies is that, a single verticle instance can at most utilise one core of the server. In order to scale across cores, several verticles which are responsible for the same task can be instantiated and the runtime will distribute the workload among them (load balancing), this way taking full advantage of all SPU cores without much effort. Verticles can also be distributed between several machines. This will be transparent to the application code. The Verticles use the same mechanisms to communicate as if they would run on the same machine. This makes it very easy to scale applications.

##### [Messaging Transport Protocols] (https://github.com/reTHINK-project/core-framework/issues/20)(FOKUS)

* Websockets - Yes supported
* SockJS - Yes supported
* HTTP Long-Polling - Yes
* HTTP Streaming - ? (Not sure what this means, clarification needed)

##### [Message delivery reliability] (https://github.com/reTHINK-project/core-framework/issues/17)(FOKUS)

No. Vert.x uses the Event Bus to send messages through pub/sub mechanism or point-2-point mechanism. In both cases, there is no feedback to the sender if the message was recieved and processed or if it was not recieved at all. In the end reliability will boil down to the application logic service build on top of vert.x.

# QOS SOTA

## coturn

### Overview

The TURN [4] protocol is defined as an extension of the STUN [5] protocol.

TURN servers act as media relays and are directly placed in the media path. TURN server gets all the media packets from one endpoint and sends them towards the remote peer (remote endpoint) or to another TURN server, if it is used by the remote endpoint.

TURN server acts as a relay for the packets. In the most common case, a TURN server is in the public Internet and the hosts are behind NATs or restrictive firewalls.

The study about TURN servers is very up-to-date since there is an ongoing work on this subject in IETF. The recently formed IETF group – TRAM (TURN Revised and Modernized) focuses on improving TURN implementations and features in order to make STUN and TURN more suitable for WebRTC [6].

Coturn is an open source TURN server implementation [1]. It is a separate branch of the previous implementation rfc5766-turn-server-project, and is dedicated for testing new protocols. As a result, it supports more specifications than the previous version.

The asset can be used for network QoS.

### Architecture

Below, the list of supported specs is given, based on coturn codelab [2]. Based on discussion in discuss-webrtc google forum, information about support in Google Chrome is indicated [3].

TURN specs: - RFC 5766 - base TURN specs – supported in Chrome - RFC 6062 - TCP relaying TURN extension – not supported, but unnecessary - RFC 6156 - IPv6 extension for TURN – not supported, waiting on SSODA - DTLS support (http://tools.ietf.org/html/draft-petithuguenin-tram-turn-dtls-00) – not supported, but patch in flight - Mobile ICE (MICE) support (http://tools.ietf.org/html/draft-wing-tram-turn-mobility-01) – not a WG doc - TURN REST API (http://tools.ietf.org/html/draft-uberti-behave-turn-rest-00) – can be performed by an application - Origin field in TURN (Multi-tenant TURN Server) (http://tools.ietf.org/html/draft-johnston-tram-stun-origin-02) – not yet supported by Chrome, but there is a patch possible soon - TURN Bandwidth draft specs (http://tools.ietf.org/html/draft-thomson-tram-turn-bandwidth-00) - not a WG doc - SSODA (dual allocation) draft specs (http://tools.ietf.org/html/draft-martinsen-tram-ssoda-00) - supported, but not yet stable - Third-party authorization support (http://tools.ietf.org/html/draft-ietf-tram-turn-third-party-authz-05) – not supported yet, there is a need to add support to the W3C specs to pass in the access token first; it is on their list to get done soon though, not yet https://groups.google.com/forum/#!msg/discuss-webrtc/iLmbX8L5h4Q/IgoaYCIbTLUJ - ALPN support (http://tools.ietf.org/html/draft-ietf-tram-alpn-08) STUN specs: - RFC 3489 - "classic" STUN - RFC 5389 - base "new" STUN specs - RFC 5769 - test vectors for STUN protocol testing - RFC 5780 - NAT behavior discovery support

### APIs

Should be implemented and launched on a server.

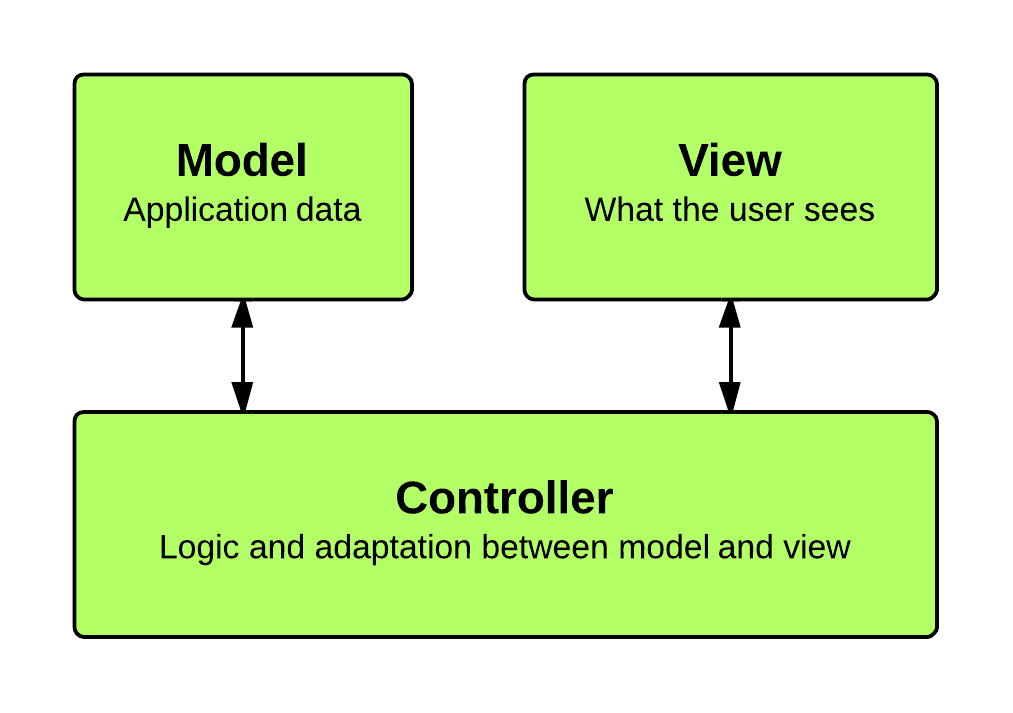
### Requirements Analysis

Download the latest official version, at least 4.4. TURN-based solution would allow a collaborative approach as it can be placed in a media path and be used for flow steering. It can impact different network types and segments and thanks to its compatibility with existing solutions, it can be introduces incrementally.

# Service Frameworks SOTA

# Overview of the Angular.js framework

AngularJS is a Javascript MVC Framework developed and promoted by Google, to build well architectured and maintainnable web applications. It is usually mistaken for a library due to its lightweight than normal frameworks. It is entirely based on Javascript and a client side framework. It is supported by multiple browsers.



image

## Main Concepts

* *Compiler* : Compiler is an angular service which traverses the DOM looking for attributes. It allows the developer to teach the browser new HTML syntax. The compiler allows you to attach behavior to any HTML element or attribute and even create new HTML elements or attributes with custom behavior. Angular calls these behavior extensions directives. The compilation process happens in two phases. Compile: traverse the DOM and collect all of the directives. The result is a linking function. Link: combine the directives with a scope and produce a live view. Any changes in the scope model are reflected in the view, and any user interactions with the view are reflected in the scope model. This makes the scope model the single source of truth.
* *Directive* : Directives can be placed in element names, attributes, class names, as well as comments.  Directives are a way to teach HTML new tricks. A directive is just a function which executes when the compiler encounters it in the DOM <input ng-model='name'>. It is possible to define custom derective <span draggable>Drag ME</span>.
* ng-app Directive: Angular uses this directive to auto-bootstrp an application. Only one ng-app directive can be used per HTML document. <html ng-app>
* Expression : Expressions are JavaScript-like code snippets that are usually placed in bindings such as {{ expression }} ```
* 1+2={{1+2}}

\* \*Form & Control\* : Forms and controls provide validation services, so that the user can be notified of invalid input. This provides a better user experience, because the user gets instant feedback on how to correct the error.``  
Modules: Modules declaratively specify how an application should be bootstrapped. There can be multiple modules in an app  
Those could be interdependent too. Modules are configured with routes, controllers, models etc.

var myAppModule = angular.module('myApp', [--here goes the dependent Modules--]); ```

* *Routing* : It Is used for deep-linking URLs to controllers and views (HTML partials). It watches $location.url() and tries to map the path to an existing route definition.
* *Scope* : Scope is an object that refers to the application model. It is an execution context for expressions. Scopes are arranged in hierarchical structure which mimic the DOM structure of the application. Scopes can watch expressions and propagate events.
* *Dependency Injection* : Dependency Injection (DI) is a software design pattern that deals with how code gets hold of its dependencies
* *Filters* : Angular filters format data for display to the user. Custon filters can be created as well

## Evaluation

### Pros

* Encourages MVC design pattern
* Two way data binding allows for automation synchronization of data bewteen model and view components
* Written for testability and with modularized code which enables clarity , extensibiliy and strong built-in-services
* Active project with a large eco-system, leading to higher rates of inquries answered and new developments
* End-to end integration testing and Unit testing
* Fast Development once familiar with structure and concept
* Very expressive, leading to less code for same result as with other libraries

###Cons \* Complex Directives API \* **Good for Single Page Apps (SPA), SO, is not the best option to go for Hyperty Developments** \* Complex Directives API \* Runtime configuration only only before Bootstrap procedure. No configuration possible after. \* Scopes are easy to use but difficult to debug

# Requirement Analysis

Analysis against **Service Framework** Requirements

* [Service Framework SHOULD support Model-View-Controller design pattern](https://github.com/reTHINK-project/core-framework/issues/36)
* Yes
* The Angular framework provides a powerful MVC design Pattern
* [Service Framework MUST be light weight and fast](https://github.com/reTHINK-project/core-framework/issues/37)
* YES
* Current minimized gzipped web version: 172KB
* Current minimized gzipped web app version: 146KB
* Can gzip compress down to one third and Data per page has small memory footprint
* [Service Framework should be Supported in all Devices and Operating Systems featuring Hyperty Runtime](https://github.com/reTHINK-project/core-framework/issues/38)
* Android (Smartphone and Tablet) - YES
* iOS (Smartphone and Tablet) - YES
* Raspberry PI - YES
* Linux VM - YES
* Windows VM - YES
* [Service Framework MUST be Modular in nature](https://github.com/reTHINK-project/core-framework/issues/42)
* YES

# Overview of the Backbone.js framework (taken from http://backbonejs.org)

## Models and Views

The model looks as follows: 

### Model

* Orchestrates data and business logic,
* Loads and saves from the server,
* Emits events when data changes.

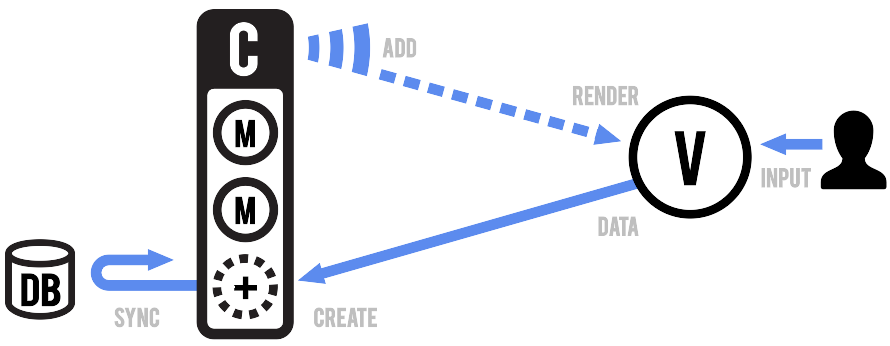
A Model manages an internal table of data attributes, and triggers "change" events when any of its data is modified. Models handle syncing data with a persistence layer — usually a REST API with a backing database. Design your models as the atomic reusable objects containing all of the helpful functions for manipulating their particular bit of data. Models should be able to be passed around throughout your app, and used anywhere that bit of data is needed.

### View

* Listens for changes and renders UI.
* Handles user input and interactivity.
* Sends captured input to the model.

A View is an atomic chunk of user interface. It often renders the data from a specific model, or number of models — but views can also be data-less chunks of UI that stand alone. Models should be generally unaware of views. Instead, views listen to the model "change" events, and react or re-render themselves appropriately.

### Collections



image

A Collection helps you deal with a group of related models, handling the loading and saving of new models to the server and providing helper functions for performing aggregations or computations against a list of models. Aside from their own events, collections also proxy through all of the events that occur to models within them, allowing you to listen in one place for any change that might happen to any model in the collection.

## Overall Evaluation

* No concept of separated Controller
* functionality is basic
* provides no "golden pattern" for structuring of an application
* leaves lots of structuring decisions to the developer
* well suited for creation of own frameworks
* **no two-way data binding --> lots of boilerplate code required**
* **views change DOM directly by looking up css class names**
* --> changing CSS or modifications in DOM (wrapping, nesting) requires updates in code

# Requirement Analysis

Analysis against **Service Framework** Requirements

* [Service Framework SHOULD support Model-View-Controller design pattern](https://github.com/reTHINK-project/core-framework/issues/36)
* NO
* The Backbone framework provides a MV Pattern with direct interaction between Models and Views.
* The controller part is mainly done in the code of the models and also the views.
* [Service Framework MUST be light weight and fast](https://github.com/reTHINK-project/core-framework/issues/37)
* YES
* minimized gzipped version is very small (approx 5.6kb)
* mandatory dependcies to underscore.js (5kb) and jQuery (32kb) or Zepto(9,1kb, a JQuery clone)
* small memory footprint
* [Service Framework should be Supported in all Devices and Operating Systems featuring Hyperty Runtime](https://github.com/reTHINK-project/core-framework/issues/38)
* rather YES
* every view is tight to its own root-DOM element and responsible for the tree below it
* Therefore Backbone.js relies on runtimes that provide a DOM tree.
* But this does NOT have to be a "real" DOM tree --> can be used with e.g. React virtual DOM (React has implemented a browser-independent events and DOM system)
* --> needs special additions in non-browser runtime environments
* [Service Framework MUST be Modular in nature](https://github.com/reTHINK-project/core-framework/issues/42)
* rather NO
* Backbone itself lacks a Controller concept and Views and Models are relatively tightly coupled
* therefore also resulting modules should be tightly coupled to view elements and not easily portable (would need further investigations)
* Models stand-alone and their synchronization capabilities with backend storages should be portable and fulfill this requirement

# StapesJS

## Introduction

Stapes.js is a little web framework designed to be agnostic about the setup and style of coding. It is very flexible and it allows to use MVC paradigm combined with libraries such as jQuery, Zepto, React and Rivets. Stapes provides the necessary building blocks to build a apps in a short-time.

## Advantages and main features

* It allows class creation, custom events, and data methods. It only has 20 methods compared to other web-frameworks like Backbone which has more than 75.
* It is very light so the loading time will be very short. It is just 2kb (minified and gzipped) so it is suitable for web sites designed for mobile devices.
* It has around 600 lines of codes, so if any issue is find or it is necessary to add some new feature it can be done at a small cost.

## Drawbacks

* It is less complete than other frameworks so some functionality may require some development.
* The last commit was done the Summer of 2014 so its contributors are not very active in Github. This may be a problem if any issue is found.

## Suitability for ReTHINK project

* It is an Open Source project. It has MIT license which is a permissive free software license. meaning that it permits reuse within proprietary software provided all copies of the licensed software include a copy of the MIT License terms and the copyright notice. Such proprietary software retains its proprietary nature even though it incorporates software under the MIT License. It can be used without restrictions in the project.
* ReTHINK develeopments are not expected to be very complex so StapesJS could be a right choice. Additionally its small size will reduce the overhead in the applications which will help to reduce the load time.