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Implementation of Websocket over p2p Network As Review Paper

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Abstract - This paper introduces an implementation of Websocket protocol where it enables two way communication between one or more no of clients running untrusted code into a controlled environment to a remote host that has opted into communication from that of The code. Where this application work as a client as well as server. p2p network which is use in case of The distribution of the videos. where goal of this technology is to provide a mechanism for browser based applications that need two way communication with servers that does not relay on opening multiple of http connections. where proposed design which enables flexible customization of video streams to support heterogeneous of receivers, highly utilizes upload bandwidth of peers, and quickly adapts to network and peer dynamics.

Keywords - *Peer to peer coding, scalable video coding, network coding, websocket.*

1. Introduction

From past of the time, creating web applications that need bidirectional communication between a client and a server (e.g., instant messaging and gaming applications) has required an abuse of HTTP to poll the server for updates while sending upstream notifications as distinct HTTP calls .This can be provided by WebSocket Protocol. Combined with the WebSocket API , it provides an alternative to HTTP polling for two-way communication from a web page to a remote server.Then same technique can be The WebSocket Protocol is designed to supersede the existing bidirectional communication technologies that use HTTP as a transport layer to benefit from existing infrastructure (proxies, filtering ,authentication).Such technologies were implemented as trade-offs between efficiency and reliability because HTTP was not initially meant to be used for bidirectional communication[2]. The Protocol attempts to The address the goals of existing bidirectional HTTP technologies in the context of the existing HTTP infrastructure dedicated port without reinventing the entire protocol. This last point is important because of the traffic patterns of interactive messaging do not closely match standard HTTP traffic and can induce unusual loads on some components. it will significantly improve their performance. it present the design of a P2P streaming system that employs both scalable video coding and network coding where design is modular and can be used as an improvement plug- in other P2P streaming systems. The p2p mechanism can potentially achieve a very high efficiency of data exchange between end devices,its very useful in particular network infrastructure[1].In addition, we quantitatively show the expected performance gain from the proposed design using actual scalable video traces in realistic P2P streaming environments with high churn rates, heterogeneous peers, and flash crowd scenarios.In particular, our results show that the proposed system can achieve (i) significant improve-ment in the visual quality perceived by peers (several dBs are observed), (ii) smoother and more sustained streaming rates (up to 100% increase in the average streaming rate is obtained), (iii) higher streaming capacity by serving more requests from peers, and (iv) more robustness against high churn rates and flash crowd arrivals of peers[1].

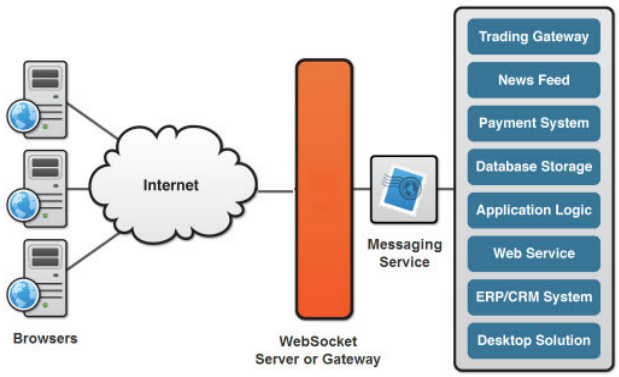

Fig1.architecture of websocket server

2. Related Work

A. Protocol Overview

The protocol has two parts: a handshake and the data transfer. The handshake from the client looks as follows:

GET /chat HTTP/1.1
Host: server.example.com
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Key: dGhlIHNhbXBsZSBub25jZQ==
Origin: http://example.com
Sec-WebSocket-Protocol: chat, superchat
Sec-WebSocket-Version: 13
The handshake from the server looks as follows:
HTTP/1.1 101 Switching Protocols
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Accept:
s3pPLMBiTxaQ9kYGzzhZRbK+xOo=
Sec-WebSocket-Protocol: chat

The leading line from the client follows the Request-Line format..The Request-Line and Status-Line productions are defined into An unordered set of header fields comes after the leading line in the both cases. The meaning of these header fields is specified in the Section 4 of this document. Additional header fields may also be present, such as cookies. The format and parsing of headers is as defined in the client and server have both sent their handshakes, and if the handshake was successful, then the data transfer part starts. This is a two-way communication channel where each side can, independently from the other, send data at will.[1] After a successful handshake, clients and servers transfer data back and forth in conceptual units referred to in that of specification as "messages". On the wire, a message is composed of one or more frames. The WebSocket messages does not necessarily correspond to a particular network layer framing, as a fragmented message may be coalesced or split by an intermediary. A frame has an associated types. Each frame belonging to the same message contains the same type of data. Broadly speaking, there are types for textual data , binary data (whose interpretation is left up to the application), and control frames (which are not intended to carry data for the application but instead for protocol-level signaling, such as to signal that the connection should be closed). This version of the protocol defines six frame types and leavesit in ten reserved for future use.

B. Opening Handshake

The opening handshake is intended to be compatible with HTTP-based server-side software and intermediaries, so that a single port can be used by both HTTP clients talking to that of The server and WebSocketclients talking to that of The server. To this end, the WebSocket client's handshake is an HTTP Upgrade request:

GET /chat HTTP/1.1
Host: server.example.com
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Key: dGhlIHNhbXBsZSBub25jZQ==
Origin: http://example.com
Sec-WebSocket-Protocol: chat, superchat
Sec-WebSocket-Version: 13

In compliance with, header fields in the handshake may be sent by the client in any order, so the order in which different header fields are received is not significant. The "Request-URI" of the GET method is used to identify the endpoint of the WebSocket connection, both to allow multiple domains to be served from one IP address and to allow multiple WebSocketendpoints to be served by a single server. The client includes the hostname in the |Host| header field of its handshake as per , so that both the client and the server can verify that they agree on which host is in use.The WebSocket Protocol in December 2011 The Additional header fields are used to select options into the WebSocketProtocol. Typical options available in this version are the subprotocol selector (|Sec-WebSocket-Protocol|), list of extensions support by the client (|Sec-WebSocket-Extensions|), |Origin| header field, etc.

The |Sec-WebSocket-Protocol| request-header field it can be used to indicate what a subprotocols (application-level protocols layered over the WebSocket Protocol) are acceptable to the client.The server selects one or none of the acceptable protocols and echoes that value in its handshake to indicate that it has selected that protocol. Sec-WebSocket-Protocol: chat The |Origin| header field is used to protect against unauthorized cross-origin use of a WebSocket server by scripts using the WebSocket API in a web browser. The server is informed of the script origin generating the WebSocket connection request. If the server does not wish to accept connections from this origin, it can choose to reject the connection by sending an appropriate HTTP error code. This header field is sent by browser clients; for non-browser clients, this header field may be sent if it makes sense in the context of those clients. Finally, the server has to prove to the client that it

received the client's WebSocket handshake, so that the server doesn't accept connections that are not WebSocket connections. This prevents an attacker from tricking a WebSocket server by sending it carefully crafted packets using XMLHttpReques or a form submission.To prove that the handshake was received, the server has to take two pieces of information and combine them to form a response. The first piece of information comes from the |Sec-WebSocket-Key| header field in the client handshake:
Sec-WebSocket-Key: dGhlIHNhbXBsZSBub25jZQ=
For this header field, the server has to take the value (as present in the header field, e.g., the base64-encoded version minus any leading and trailing whitespace) and concatenate this with the Globally Unique Identifier"258EAFA5-E914-47DA-95CA-C5AB0DC85B11" in string form, which is unlikely to be used by network endpoints that do not understand the WebSocket Protocol. A SHA-1 hash (160 bits), base64-encoded , of this concatenation is then returned in the server's handshake.

C. Closing Handshake

The closing handshake is far simpler than the opening handshake. Either peer can send a control frame with data containing a specified control sequence to begin the closing handshake (detailed in Section 5.5.1). Upon receiving such a frame, the other peer sends a Close frame in response, if it hasn't already sent one. Upon receiving that control frame, the first peer then closes the connection, safe in the knowledge that no further data is forthcoming. After sending a control frame indicating the connection should be closed, a peer does not send any further data; after receiving a control frame indicating the connection should be closed, a peer discards any further data received. It is safe for both peers to initiate this handshake simultaneously. The closing handshake is intended to complement the TCP closing handshake (FIN/ACK), on the basis of TCP closing handshake is not always reliable end-to-end, especially in the presence of intercepting proxies and other intermediaries. By sending a no of Close frames and waiting for a Close frames in response, certain cases are avoided where data may be unnecessarily lost. For instance, on some platforms, if a socket is closed with The data in the receive queue, a RST packet is sent, which will then cause recv() to fail for the party that received the RST, even if there were data waiting to be read.

D. Design Philosophy

The WebSocket Protocol is to be designed on the principle that there should be minimal framing (the only framing that exists is to make the protocol frame-based instead of stream-based and to support a distinctions between Unicode text and binary frames). It is expected that metadata would be layered on top of the WebSocket by the application Fette&Melnikov Standards Track The WebSocket Protocol December 2011 layer, in the same way this metadata is layered on top of TCP by the application layer (e.g., HTTP). Conceptually, WebSocket is really just a layer on top of TCP that protocol frame-based instead of stream-based and to support a distinction between Unicode text and binary frames. It is expected that the metadata would be layered on top of WebSocket .

It designed in such a way that its servers can be share a port with HTTP servers, by having its handshake be a valid HTTP Upgrade request. One could conceptually use the other protocols to establish client-server messaging, but the intent of WebSockets is to be provide a relatively simple protocol that can coexist with HTTP and deployed HTTP infrastructure (such as proxies) and that is as close to TCP as is safe for use with such infrastructure given security considerations, with targeted additions to be simplify usage and keep simple things.The protocol is intended to be extensible; future versions will be likely introduce additional concepts such as multiplexing

E. Security Model

The WebSocket Protocol uses the origin model used by web browsers to restrict which web pages can contact a WebSocket server when the WebSocket Protocol is used from a web page. Naturally, when the WebSocket Protocol is used by a dedicated client directly (i.e., not from a web page through a web browser), the origin model is not useful, as the client can provide any arbitrary origin string. This protocol is intended to fail to establish a connection with servers of the pre-existing protocols like SMTP and HTTP, while allowing HTTP servers to opt-in to supporting this protocol if Fette&Melnikov Standards Track. The WebSocket Protocol December 2011 desired. This is achieved by having a strict and elaborate handshake and by limiting the data that can be inserted into the connection before the handshake is finished (thus limiting how much the server can be influenced). It is similarly intended to fail to establish a connection when data from other protocols, especially HTTP, is sent to a WebSocket server, for example, as might happen if an HTML "form" were submitted to a WebSocket server. This is primarily achieved by requiring that the server prove that it read the handshake, which it can only do if the handshake contains the appropriate parts, which can only be sent by a WebSocket client. In particular, at the time of writing of this specification, fields starting with

|Sec-| cannot be set by an attacker from a web browser using only HTML and JavaScript APIs such asXMLHttpRequest [XMLHttpRequest].

F. Relationship to TCP and HTTP

Relationship to TCP and HTTP The WebSocket Protocol is an independent TCP-based protocol. Its only relationship to HTTP is that its handshake is interpreted by HTTP servers as an Upgrade request. By default, the WebSocket Protocol uses port 80 for regular WebSocketconnections and port 443 for WebSocket connections tunneled over Transport
Layer Security (TLS).

G. Establishing a Connection

When a connection is to be made to a port that is shared by an HTTP server (a situation that isquite likely to occur with traffic to ports 80 and 443), the connection will appear to the HTTP server to be a regular GET request with an Upgrade offer. In relatively simple setups with just one IP address and a single server for all traffic to a single hostname, this might allow a practical way for systems based on the WebSocket Protocol to be deployed. In more elaborate setups (e.g., with load balancers and multiple servers), a dedicated set of hosts for WebSocket connections separate from the HTTP servers is probably easier to manage. At the time of writing of this specification, it should be noted that connections on ports 80 and 443 have significantly different success rates, with connections on port 443 being significantly more likely to succeed, though this may change with time.

H. Subprotocols using the WebSocket Protocol

The client can request that the server use a specific subprotocol by including the |Sec-WebSocket-Protocol| field in its handshake. If it is specified, the server needs to include the same field and one of the selected subprotocol values in its response for the connection to be established. to avoid potential collisions, it is use to be recommended names that contain the ASCII version of the domain name of this subprotocol'soriginator. for example carporation were to create a Chat subprotocol to be implemented by many servers around a Web, they could name it "chat.example.com". If the Example Organization called the competing subprotocol "chat.example.org", then the two subprotocols could be implemented by servers simultaneously, with that server dynamically selecting which subprotocol to be use based on the value sent by the client. These subprotocols would be considered completely separate by WebSocket clients. Backward-compatible versioning can be implemented by reusing the same subprotocol string but carefully designing the actual subprotocol to support this kind of extensibility

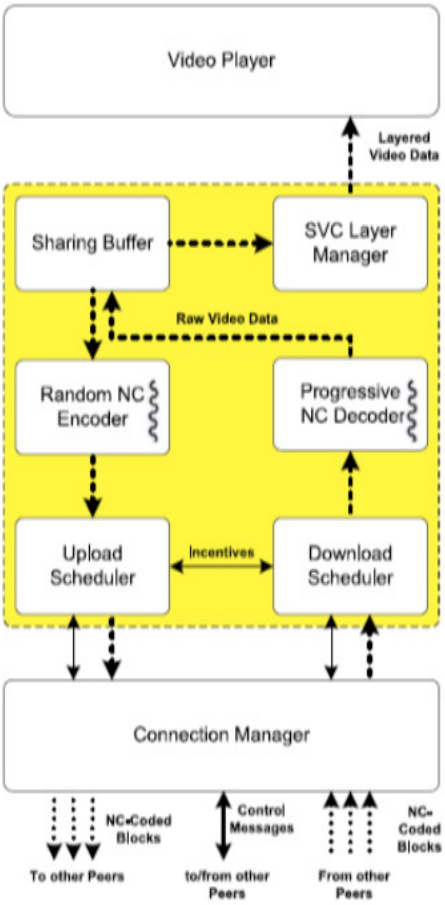

Fig:1Peer Software Architecture. Dashed arrows denote video data, and solid arrows denote control messages

3. Proposed Work

PROPOSED P2P STREAMING SYSTEM

In this section, we describe the proposed P2P live streaming system this employs network coding and scalable video coding. We start with a high-level of overview, followed by more details. network-coded the data received from others and process this data to be create proper scalable video streams and to ensure smooth videos quality. As senders, peers encodes the videos of data using network coding positions of with parameters based on their own upload capacity as well as the characteristics of the receiving peers. A simplified model for the software architecture of a peer in our system is shown in the Fig. 1. A similar model is used for source nodes, but with some

of differences as elaborated later. We do not address the design or optimization of trackers; the function of the tracker is orthogonal to the work to be presented in this paper. We also do not to be addresses other problems in mesh-based P2P streaming systems, including neighbor selection, gossip protocols (for exchanging data availability), incentive schemes, and overlay optimization which all have been heavily researched in the literature. All of the above issues are abstracted in the Connection Manager component in a Fig. 1, while our work is focused on the components in the shaded box in that of figure. The separation and abstraction of functions enable us to the support different P2P stream-ing systems with minimal changes in our design and code. Therefore, our work is fairly general.

4. Advantages

1) It is easy to install so the configuration of computers on of that network,
2) All the resources and contents are shared by all the peers, unlike server-client architecture where Server shares all the contents and resources.
3) .P2P is more reliable as central dependency is eliminated. The Failurity of one peer doesn't affect on functioning of other peers. In case of Client –Server network, if server goes down whole network gets affected. here is no need for full-time System Administrator.
4) Every user is the administrator of his machine. User can control their shared resources.
5) The over-all cost of building and maintaining this type of network is comparatively very less.

5. Conclusions and Future Work

In this paper, we showed an implementation of websocket protocol which is a application work as client as well as server. where next planning is to designing of web socket client application over p2p network .we will also implementing the algorithm of Adaptive queue based chunk scheduling where it provide full band width utilization in p2p network. Then designing of P2P streaming systems with scalable video coding and network coding can solve both of the above problems. The evaluation study confirms the significant potential performance gain, in terms of visual quality perceived by peers, average streaming rates, streaming capacity, and adaptation to higher peer dynamics. The work in this paper can be extended in multiple directions.

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