SIPCORE Working Group
Internet-Draft
Intended status: Standards Track
Expires: October 18, 2012

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April 16, 2012

The WebSocket Protocol as a Transport for the Session Initiation Protocol (SIP) draft-ibc-sipcore-sip-websocket-02

Abstract

The WebSocket protocol enables two-way realtime communication between clients and servers. This document specifies a new WebSocket sub-protocol as a reliable transport mechanism between SIP (Session Initiation Protocol) entities and enables usage of the SIP protocol in new scenarios.

Status of this Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on October 18, 2012.

Table of Contents

1. Introduction

The WebSocket [RFC6455] protocol enables messages exchange between clients and servers on top of a persistent TCP connection (optionally secured with TLS [RFC5246]). The initial protocol handshake makes use of HTTP [RFC2616] semantics, allowing the WebSocket protocol to reuse existing HTTP infrastructure.

Modern web browsers include a WebSocket client stack complying with The WebSocket API [WS-API] as specified by the W3C. It is expected that other client applications (those running in personal computers and devices such as smartphones) will also run a WebSocket client stack. The specification in this document enables usage of the SIP protocol in those new scenarios.

This specification defines a new WebSocket sub-protocol (section 1.9 in [RFC6455]) for transporting SIP messages between a WebSocket client and server, a new reliable and message boundary transport for the SIP protocol, new DNS NAPTR [RFC3403] service values and procedures for SIP entities implementing the WebSocket transport. Media transport is out of the scope of this document.

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

2.1. Definitions

SIP WebSocket Client: A SIP entity capable of opening outbound connections with WebSocket servers and speaking the WebSocket SIP Sub-Protocol as defined by this document.

SIP WebSocket Server: A SIP entity capable of listening for inbound connections from WebSocket clients and speaking the WebSocket SIP Sub-Protocol as defined by this document.

3. The WebSocket Protocol

This section is non-normative.

WebSocket protocol [RFC6455] is a transport layer on top of TCP (optionally secured with TLS [RFC5246]) in which both client and server exchange message units in both directions. The protocol defines a connection handshake, WebSocket sub-protocol and extensions

negotiation, a frame format for sending application and control data, a masking mechanism, and status codes for indicating disconnection causes.

The WebSocket connection handshake is based on HTTP [RFC2616] protocol by means of a specific HTTP GET method with Upgrade request sent by the client which is answered by the server (if the negotiation succeeded) with HTTP 101 status code. Once the handshake is done the connection upgrades from HTTP to the WebSocket protocol. This handshake procedure is designed to reuse the existing HTTP infrastructure. During the connection handshake, client and server agree in the application protocol to use on top of the WebSocket transport. Such application protocol (also known as the "WebSocket sub-protocol") defines the format and semantics of the messages exchanged between both endpoints. It may be a custom protocol or a standarized one (as the WebSocket SIP Sub-Protocol proposed in this document). Once the HTTP 101 response is processed both client and server reuse the underlying TCP connection for sending WebSocket messages and control frames to each other in a persistent way.

WebSocket defines message units as application data exchange for communication endpoints, becoming a message boundary transport layer. These messages can contain UTF-8 text or binary data, and can be split into various WebSocket text/binary frames.

> However, the WebSocket API [WS-API] for web browsers just includes callbacks that are invoked upon receipt of an entire message, regardless of whether it was received in a single or multiple WebSocket frames.

4. The WebSocket SIP Sub-Protocol

The term WebSocket sub-protocol refers to the application-level protocol layered on top of a WebSocket connection. This document specifies the WebSocket SIP Sub-Protocol for carrying SIP requests and responses through a WebSocket connection.

4.1. Handshake

The SIP WebSocket Client and SIP WebSocket Server need to agree on the WebSocket SIP Sub-Protocol during the WebSocket handshake procedure as defined in section 1.3 of [RFC6455]. The client MUST include the value "sip" in the Sec-WebSocket-Protocol header in its handshake request. The 101 reply from the server MUST contain "sip" in its corresponding Sec-WebSocket-Protocol header.

Below is an example of the WebSocket handshake in which the client

requests the WebSocket SIP Sub-Protocol support from the server:

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

The handshake response from the server supporting the WebSocket SIP Sub-Protocol would look as follows:

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

Once the negotiation is done, the WebSocket connection is established with SIP as the WebSocket sub-protocol. The WebSocket messages to be transmitted over this connection MUST conform to the established application protocol.

4.2. SIP encoding

WebSocket messages are carried on top of WebSocket UTF-8 text frames or binary frames. The SIP protocol [RFC3261] allows both text and binary bodies in SIP messages. Therefore SIP WebSocket Clients and SIP WebSocket Servers MUST accept both WebSocket text and binary frames.

5. SIP WebSocket Transport

5.1. General

WebSocket [RFC6455] is a reliable protocol and therefore the WebSocket sub-protocol for a SIP transport defined by this document is also a reliable transport. Thus, client and server transactions using WebSocket transport MUST follow the procedures and timer values for reliable transports as defined in [RFC3261].

Each complete SIP message MUST be carried within a single WebSocket message, and a WebSocket message MUST NOT contain more than one SIP message. Therefore the usage of the Content-Length header field is optional.

This makes parsing of SIP messages easier on client side (typically web-based applications with an strict and simple API for receiving WebSocket messages). There is no need to establish boundaries (using Content-Length headers) between different messages. Same advantage is present in other message-based SIP transports such as UDP or SCTP [RFC4168].

5.2. Updates to RFC 3261

5.2.1. Via Transport Parameter

Via header fields carry the transport protocol identifier. This document defines the value "WS" to be used for requests over plain WebSocket protocol and "WSS" for requests over secure WebSocket protocol (in which the WebSocket connection is established using TLS [RFC5246] with TCP transport).

The updated RFC 3261 augmented BNF (Backus-Naur Form) [RFC5234] for this parameter reads as follows:

```
transport  =  "UDP" / "TCP" / "TLS" / "SCTP" / "TLS-SCTP"
              / "WS" / "WSS"
              / other-transport
```

5.2.2. SIP URI Transport Parameter

This document defines the value "ws" as the transport parameter value for a SIP URI [RFC3986] to be contacted using WebSocket protocol as transport.

The updated RFC 3261 augmented BNF (Backus-Naur Form) for this parameter reads as follows:

```
transport-param  =  "transport="
                    ( "udp" / "tcp" / "sctp" / "tls" / "ws"
                    / other-transport )
```

5.2.3. Sending Responses

This specification updates the section 18.2.2 "Sending Responses" in [RFC3261] by adding the following:

o If the Via "sent-protocol" is "WS" or "WSS" the response MUST be sent using the existing WebSocket connection to the source of the original request that created the transaction, if that connection is still open. If that connection is no longer open, the server SHOULD NOT attempt to open a WebSocket connection for sending the response.

This is due to the nature of the WebSocket protocol in which just the WebSocket client can establish a connection with the WebSocket server. Typically a WebSocket client does not listen for inbound connections and WebSocket servers do not open outbound connections.

5.3. Locating a SIP Server

RFC 3263 [RFC3263] specifies the procedures which should be followed by SIP entities for locating SIP servers. This specification defines the NAPTR service value "SIP+D2W" for SIP WebSocket Servers that support plain WebSocket transport and "SIPS+D2W" for SIP WebSocket Servers that support secure WebSocket transport.

Unfortunately neither JavaScript stacks nor WebSocket stacks running in current web browsers are capable of performing DNS NAPTR/SRV queries.

In the absence of an explicit port and DNS SRV resource records, the default port for a SIP URI with "ws" transport parameter is 80 in case of SIP scheme and 443 in case of SIPS scheme.

6. Connection Keep Alive

This section is non-normative.

It is RECOMMENDED that the SIP WebSocket Client or Server keeps the WebSocket connection open by sending periodic WebSocket Ping frames as described in [RFC6455] section 5.5.2.

Note however that The WebSocket API [WS-API] does not provide a mechanism for web applications running in a web browser to decide whether or not to send periodic WebSocket Ping frames to the server. The usage of such a keep alive feature is a decision of each web browser vendor and may depend on the web browser configuration.

Any future WebSocket protocol extension providing a keep alive mechanism could also be used.

The SIP stack in the SIP WebSocket Client MAY also use Network Address Translation (NAT) keep-alive mechanisms defined for SIP connection-oriented transports, such as the CRLF Keep-Alive Technique mechanism described in [RFC5626] section 3.5.1 or [RFC6223].

Implementing these techniques would involve sending a WebSocket message to the SIP WebSocket Server whose content is a double

> CRLF, and expecting a WebSocket message from the server containing a single CRLF as response.

7. Authentication

This section is non-normative.

Prior to sending SIP requests, the SIP WebSocket Client connects to the SIP WebSocket Server and performs the connection handshake. As described in Section 3 the handshake procedure involves a HTTP GET request replied with HTTP 101 status code by the server.

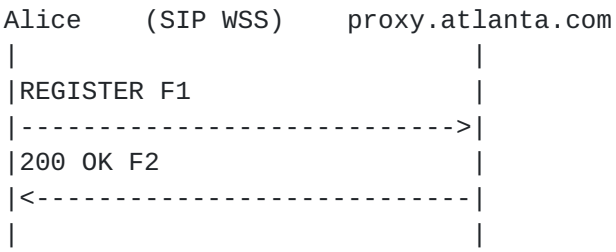
In order to authorize the WebSocket connection, the SIP WebSocket Server MAY inspect the Cookie [RFC6265] header in the HTTP GET request (if present). In case of web applications the value of such a Cookie is usually provided by the web server once the user has authenticated itself with the web server by following any of the multiple existing mechanisms. As an alternative method, the SIP WebSocket Server could request HTTP authentication by replying with a HTTP 401 status code. The WebSocket protocol [RFC6455] covers this usage in section 4.1:

> If the status code received from the server is not 101, the client handles the response per HTTP [RFC2616] procedures, in particular the client might perform authentication if it receives 401 status code.

Regardless whether the SIP WebSocket Server requires authentication during the WebSocket handshake or not, authentication MAY be requested at SIP protocol level. Therefore it is RECOMMENDED for a SIP WebSocket Client to implement HTTP Digest [RFC2617] authentication as stated in [RFC3261].

8. Examples

8.1. Registration



Alice loads a web page using her web browser and retrieves a JavaScript code implementing the WebSocket SIP Sub-Protocol defined in this document. The JavaScript code (a SIP WebSocket Client) establishes a secure WebSocket connection with a SIP proxy/registrar (a SIP WebSocket Server) at proxy.atlanta.com. Upon WebSocket connection, Alice constructs and sends a SIP REGISTER by requesting Outbound and GRUU support. Since the JavaScript stack in a browser has no way to determine the local address from which the WebSocket connection is made, this implementation uses a random ".invalid" domain name for the Via sent-by and for the URI hostpart in the Contact header (see Appendix A.1).

Message details (authentication and SDP bodies are omitted for simplicity):

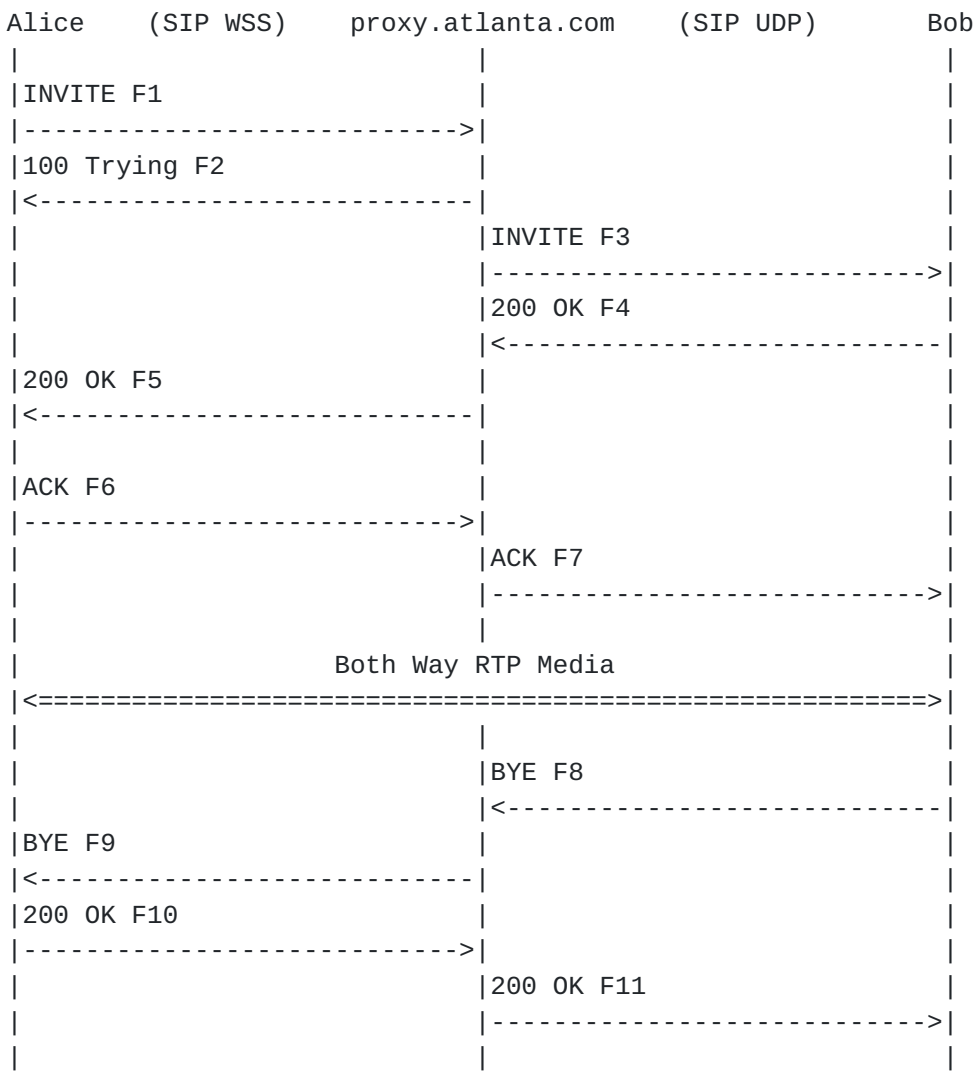
F1 REGISTER Alice -> proxy.atlanta.com (transport WSS)

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

F2 200 OK proxy.atlanta.com -> Alice (transport WSS)

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

8.2. INVITE dialog through a proxy



In the same scenario Alice places a call to Bob's AoR. The WebSocket SIP server at proxy.atlanta.com acts as a SIP proxy routing the INVITE to the UDP location of Bob, who answers the call and terminates it later.

Message details (authentication and SDP bodies are omitted for simplicity):

F1 INVITE Alice -> proxy.atlanta.com (transport WSS)

```
INVITE sip:bob@atlanta.com SIP/2.0
Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bK56sdasks
From: sip:alice@atlanta.com;tag=asdyka899
```

```
   To: sip:bob@atlanta.com
   Call-ID: asidkj3ss
   CSeq: 1 INVITE
   Max-Forwards: 70
   Supported: path, outbound, gruu
   Route: <sip:proxy.atlanta.com:443;transport=ws;lr>
   Contact: <sip:alice@atlanta.com
    ;gr=urn:uuid:f81-7dec-14a06cf1;ob>"
   Content-Type: application/sdp


   F2 100 Trying  proxy.atlanta.com -> Alice (transport WSS)

   SIP/2.0 100 Trying
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bK56sdasks
   From: sip:alice@atlanta.com;tag=asdyka899
   To: sip:bob@atlanta.com
   Call-ID: asidkj3ss
   CSeq: 1 INVITE


   F3 INVITE  proxy.atlanta.com -> Bob (transport UDP)

   INVITE sip:bob@203.0.113.22:5060 SIP/2.0
   Via: SIP/2.0/UDP proxy.atlanta.com;branch=z9hG4bKhjhjqw32c
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bK56sdasks
   Record-Route: <sip:proxy.atlanta.com;transport=udp;lr>,
     <sip:h7kjh12s@proxy.atlanta.com:443;transport=ws;lr>
   From: sip:alice@atlanta.com;tag=asdyka899
   To: sip:bob@atlanta.com
   Call-ID: asidkj3ss
   CSeq: 1 INVITE
   Max-Forwards: 69
   Supported: path, outbound, gruu
   Contact: <sip:alice@atlanta.com
     ;gr=urn:uuid:f81-7dec-14a06cf1;ob>"
   Content-Type: application/sdp


   F4 200 OK  Bob -> proxy.atlanta.com (transport UDP)

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP proxy.atlanta.com;branch=z9hG4bKhjhjqw32c
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bK56sdasks
   Record-Route: <sip:proxy.atlanta.com;transport=udp;lr>,
     <sip:h7kjh12s@proxy.atlanta.com:443;transport=ws;lr>
   From: sip:alice@atlanta.com;tag=asdyka899
   To: sip:bob@atlanta.com;tag=bmqkjhsd
```

```
   Call-ID: asidkj3ss
   CSeq: 1 INVITE
   Max-Forwards: 69
   Contact: <sip:bob@203.0.113.22:5060;transport=udp>
   Content-Type: application/sdp


   F5 200 OK  proxy.atlanta.com -> Alice (transport WSS)

   SIP/2.0 200 OK
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bK56sdasks
   Record-Route: <sip:proxy.atlanta.com;transport=udp;lr>,
     <sip:h7kjh12s@proxy.atlanta.com:443;transport=ws;lr>
   From: sip:alice@atlanta.com;tag=asdyka899
   To: sip:bob@atlanta.com;tag=bmqkjhsd
   Call-ID: asidkj3ss
   CSeq: 1 INVITE
   Max-Forwards: 69
   Contact: <sip:bob@203.0.113.22:5060;transport=udp>
   Content-Type: application/sdp


   F6 ACK  Alice -> proxy.atlanta.com (transport WSS)

   ACK sip:bob@203.0.113.22:5060;transport=udp SIP/2.0
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bKhgqqp090
   Route: <sip:h7kjh12s@proxy.atlanta.com:443;transport=ws;lr>,
     <sip:proxy.atlanta.com;transport=udp;lr>,
   From: sip:alice@atlanta.com;tag=asdyka899
   To: sip:bob@atlanta.com;tag=bmqkjhsd
   Call-ID: asidkj3ss
   CSeq: 1 ACK
   Max-Forwards: 70


   F7 ACK  proxy.atlanta.com -> Bob (transport UDP)

   ACK sip:bob@203.0.113.22:5060;transport=udp SIP/2.0
   Via: SIP/2.0/UDP proxy.atlanta.com;branch=z9hG4bKhwpoc80zzx
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bKhgqqp090
   From: sip:alice@atlanta.com;tag=asdyka899
   To: sip:bob@atlanta.com;tag=bmqkjhsd
   Call-ID: asidkj3ss
   CSeq: 1 ACK
   Max-Forwards: 69


   F8 BYE  Bob -> proxy.atlanta.com (transport UDP)
```

```
   BYE sip:alice@atlanta.com;gr=urn:uuid:f81-7dec-14a06cf1;ob SIP/2.0
   Via: SIP/2.0/UDP 203.0.113.22;branch=z9hG4bKbiuiansd001
   Route: <sip:proxy.atlanta.com;transport=udp;lr>,
     <sip:h7kjh12s@proxy.atlanta.com:443;transport=ws;lr>
   From: sip:bob@atlanta.com;tag=bmqkjhsd
   To: sip:alice@atlanta.com;tag=asdyka899
   Call-ID: asidkj3ss
   CSeq: 1201 BYE
   Max-Forwards: 70


   F9 BYE  proxy.atlanta.com -> Alice (transport WSS)

   BYE sip:alice@atlanta.com;gr=urn:uuid:f81-7dec-14a06cf1;ob SIP/2.0
   Via: SIP/2.0/WSS proxy.atlanta.com:443;branch=z9hG4bKmma01m3r5
   Via: SIP/2.0/UDP 203.0.113.22;branch=z9hG4bKbiuiansd001
   From: sip:bob@atlanta.com;tag=bmqkjhsd
   To: sip:alice@atlanta.com;tag=asdyka899
   Call-ID: asidkj3ss
   CSeq: 1201 BYE
   Max-Forwards: 69


   F10 200 OK  Alice -> proxy.atlanta.com (transport WSS)

   SIP/2.0 200 OK
   Via: SIP/2.0/WSS proxy.atlanta.com:443;branch=z9hG4bKmma01m3r5
   Via: SIP/2.0/UDP 203.0.113.22;branch=z9hG4bKbiuiansd001
   From: sip:bob@atlanta.com;tag=bmqkjhsd
   To: sip:alice@atlanta.com;tag=asdyka899
   Call-ID: asidkj3ss
   CSeq: 1201 BYE


   F11 200 OK  proxy.atlanta.com -> Bob (transport UDP)

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP 203.0.113.22;branch=z9hG4bKbiuiansd001
   From: sip:bob@atlanta.com;tag=bmqkjhsd
   To: sip:alice@atlanta.com;tag=asdyka899
   Call-ID: asidkj3ss
   CSeq: 1201 BYE
```

9. Security Considerations

9.1. Secure WebSocket Connection

It is recommended to protect the privacy of the SIP traffic through the WebSocket communication by using a secure WebSocket connection (tunneled over TLS [RFC5246]).

9.2. Usage of SIPS Scheme

SIPS scheme within a SIP request dictates that the entire request path to the target be secured. If such a path includes a WebSocket node it MUST be a secure WebSocket connection.

10. IANA Considerations

10.1. Registration of the WebSocket SIP Sub-Protocol

This specification requests IANA to create the WebSocket SIP Sub-Protocol in the registry of WebSocket sub-protocols with the following data:

Subprotocol Identifier: sip

Subprotocol Common Name: WebSocket Transport for SIP (Session Initiation Protocol)

Subprotocol Definition: TBD, it should point to this document

10.2. Registration of new Via transports

This specification registers two new transport identifiers for Via headers:

WS: MUST be used when constructing a SIP request to be sent over a plain WebSocket connection.

WSS: MUST be used when constructing a SIP request to be sent over a secure WebSocket connection.

10.3. Registration of new SIP URI transport

This specification registers a new value for the "transport" parameter in a SIP URI:

ws: Identifies a SIP URI to be contacted using a WebSocket connection.

10.4. Registration of new NAPTR service field values

This document defines two new NAPTR service field values (SIP+D2W and SIPS+D2W) and requests IANA to register these values under the "Registry for the SIP SRV Resource Record Services Field". The resulting entries are as follows:

Services Field	Protocol	Reference
SIP+D2W	WS	TBD: this document
SIPS+D2W	WSS	TBD: this document

11. Acknowledgements

Special thanks to the following people who participated in discussions on the SIPCORE and RTCWEB WG mailing lists and contributed ideas and/or provided detailed reviews (the list is likely to be incomplete): Hadriel Kaplan, Paul Kyzivat, Adam Roach, Ranjit Avasarala, Xavier Marjou, Kevin P. Fleming.

Special thanks also to Alan Johnston, Christer Holmberg and Salvatore Loreto for their reviews.

Special thanks to Saul Ibarra Corretge for his contribution and suggestions.

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Appendix A. Implementation Guidelines

This section is non-normative.

Let us assume a scenario in which the users access with their web browsers (probably behind NAT) to an intranet, perform web login by entering their user identifier and credentials, and retrieve a JavaScript code (along with the HTML code itself) implementing a SIP WebSocket Client.

Such a SIP stack connects to a given SIP WebSocket Server (an outbound SIP proxy which also implements classic SIP transports such as UDP and TCP). The HTTP GET request sent by the web browser for the WebSocket handshake includes a Cookie [RFC6265] header with the value previously retrieved after the successful web login procedure. The Cookie value is then inspected by the WebSocket server for authorizing the connection. Once the WebSocket connection is established, the SIP WebSocket Client performs a SIP registration and common SIP stuf begins. The SIP registrar server is located behind the SIP outbound proxy.

This scenario is quite similar to the one in which SIP UAs behind NAT connect to an outbound proxy and need to reuse the same TCP connection for incoming requests. In both cases, the SIP clients are just reachable through the outbound proxy they are connected to.

Outbound [RFC5626] seems an appropriate solution for this scenario. Therefore these SIP WebSocket Clients and the SIP registrar implement both Outbound and Path [RFC3327], and the SIP outbound proxy becomes an Outbound Edge Proxy (as defined in [RFC5626] section 3.4).

SIP WebSocket Clients in this scenario receive incoming SIP requests via the SIP WebSocket Server they are connected to. Therefore, in some call transfer cases the usage of GRUU [RFC5627] (which should be implemented in both the SIP WebSocket Clients and SIP registrar) is valuable.

> If a REFER request is sent to a thirdy SIP user agent indicating the Contact URI of a SIP WebSocket Client as the target in the Refer-To header field, such a URI will be reachable by the thirdy SIP UA just in the case it is a globally routable URI. GRUU (Globally Routable User Agent URI) is a solution for those scenarios, and would enforce the incoming request from the thirdy

SIP user agent to reach the SIP registrar which would route the request via the Outbound Edge Proxy.

A.1. SIP WebSocket Client Considerations

The JavaScript stack in web browsers does not have the ability to discover the local transport address which the WebSocket connection is originated from. Therefore the SIP WebSocket Client creates a domain consisting of a random token followed by .invalid top domain name, as stated in [RFC2606], and uses it within the Via and Contact header.

The Contact URI provided by the SIP clients requesting Outbound support is not later used for routing purposes, thus it is safe to set a random domain in the Contact URI hostpart.

Both Outbound and GRUU specifications require the SIP client to indicate a Uniform Resource Name (URN) in the "+sip.instance" parameter of the Contact header during the registration. The client device is responsible for getting such a constant and unique value.

In the case of web browsers it is hard to get a URN value from the browser itself. This scenario suggests that value is generated according to [RFC5626] section 4.1 by the web application running in the browser the first time it loads the JavaScript SIP stack code, and then it is stored as a Cookie within the browser.

A.2. SIP WebSocket Server Considerations

The SIP WebSocket Server in this scenario behaves as a SIP Outbound Edge Proxy, which involves support for Outbound [RFC5626] and Path [RFC3327].

The proxy performs Loose Routing and remains in dialogs path as specified in [RFC3261]. Otherwise in-dialog requests would fail since SIP WebSocket Clients make use of their SIP WebSocket Server in order to send and receive SIP requests and responses.

Appendix B. HTTP Topology Hiding

This section is non-normative.

RFC 3261 [RFC3261] section 18.2.1 "Receiving Requests" states the following:

When the server transport receives a request over any transport, it MUST examine the value of the "sent-by" parameter in the top

> Via header field value. If the host portion of the "sent-by" parameter contains a domain name, or if it contains an IP address that differs from the packet source address, the server MUST add a "received" parameter to that Via header field value. This parameter MUST contain the source address from which the packet was received.

The requirement of adding the "received" parameter does not fit well into WebSocket protocol nature. The WebSocket handshake connection reuses existing HTTP infrastructure in which there could be certain number of HTTP proxies and/or TCP load balancers between the SIP WebSocket Client and Server, so the source IP the server would write into the Via "received" parameter would be the IP of the HTTP/TCP intermediary in front of it. This could reveal sensitive information about the internal topology of the provider network to the client.

Thus, given the fact that SIP responses can only be sent over the existing WebSocket connection, the meaning of the Via "received" parameter added by the SIP WebSocket Server is of little use. Therefore, in order to allow hiding possible sensitive information about the provider infrastructure, the implementer could decide not to satisfy the requirement in RFC 3261 [RFC3261] section 18.2.1 "Receiving Requests" and not add the "received" parameter to the Via header.

> However, keep in mind that this would involve a violation of the RFC 3261.

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