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The WebSocket Protocol as a Transport for the Session Initiation Protocol (SIP)

Abstract

The WebSocket protocol enables two-way real-time communication between clients and servers in web-based applications. This document specifies a WebSocket subprotocol as a reliable transport mechanism between Session Initiation Protocol (SIP) entities to enable use of SIP in web-oriented deployments.

Status of This Memo

This is an Internet Standards Track document.

This document is a product of the Internet Engineering Task Force (IETF). It represents the consensus of the IETF community. It has received public review and has been approved for publication by the Internet Engineering Steering Group (IESG). Further information on Internet Standards is available in Section 2 of RFC 5741.

Information about the current status of this document, any errata, and how to provide feedback on it may be obtained at http://www.rfc-editor.org/info/rfc7118.

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1. Introduction

The WebSocket protocol [RFC6455] enables message exchange between clients and servers on top of a persistent TCP connection (optionally secured with Transport Layer Security (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 make a WebSocket client stack available. The specification in this document enables use of SIP in these scenarios.

This specification defines a WebSocket subprotocol (as defined in Section 1.9 of [RFC6455]) for transporting SIP messages between a WebSocket client and server, a reliable and message-boundary-preserving transport for SIP, and DNS Naming Authority Pointer (NAPTR) [RFC3403] service values and procedures for SIP entities implementing the WebSocket transport. Media transport is out of the scope of this document.

Section 3 in this specification relaxes the requirement in [RFC3261] by which the SIP server transport MUST add a "received" parameter in the top Via header in certain circumstances.

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 to WebSocket servers and communicating using the WebSocket SIP subprotocol as defined by this document.

SIP WebSocket Server: A SIP entity capable of listening for inbound connections from WebSocket clients and communicating using the WebSocket SIP subprotocol as defined by this document.

3. The WebSocket Protocol

The 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 subprotocol 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] and utilizes the HTTP GET method with an "Upgrade" request. This is sent by the client and then answered by the server (if the negotiation succeeded) with an HTTP 101 status code. Once the handshake is completed, the connection upgrades from HTTP to the WebSocket protocol. This handshake procedure is designed to reuse the existing HTTP infrastructure. During the connection handshake, the client and server agree on the application protocol to use on top of the WebSocket transport. Such an application protocol (also known as a "WebSocket subprotocol") defines the format and semantics of the messages exchanged by the endpoints. This could be a custom protocol or a standardized one (as defined by the WebSocket SIP subprotocol in this document). Once the HTTP 101 response is processed, both the client and server reuse the underlying TCP connection for sending WebSocket messages and control frames to each other. Unlike plain HTTP, this connection is persistent and can be used for multiple message exchanges.

WebSocket defines message units to be used by applications for the exchange of data, so it provides a message-boundary-preserving transport layer. These message units can contain either UTF-8 text or binary data and can be split into multiple WebSocket text/binary transport frames as needed by the WebSocket stack.

> The WebSocket API [WS-API] for web browsers only defines callbacks to be invoked upon receipt of an entire message unit, regardless of whether it was received in a single WebSocket frame or split across multiple frames.

4. The WebSocket SIP Subprotocol

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

4.1. Handshake

The SIP WebSocket Client and SIP WebSocket Server negotiate usage of the WebSocket SIP subprotocol 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.

> The WebSocket client initiates a WebSocket connection when attempting to send a SIP request (unless there is an already established WebSocket connection for sending the SIP request). In case there is no HTTP 101 response during the WebSocket handshake, it is considered a transaction error as per [RFC3261], Section 8.1.3.1., "Transaction Layer Errors".

Below is an example of a WebSocket handshake in which the client requests the WebSocket SIP subprotocol 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 accepting the WebSocket SIP subprotocol 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 has been completed, the WebSocket connection is established and can be used for the transport of SIP requests and responses. Messages other than SIP requests and responses MUST NOT be transmitted over this connection.

4.2. SIP Encoding

WebSocket messages can be transported in either UTF-8 text frames or binary frames. SIP [RFC3261] allows both text and binary bodies in SIP requests and responses. Therefore, SIP WebSocket Clients and SIP WebSocket Servers MUST accept both text and binary frames.

> If there is at least one non-UTF-8 symbol in the whole SIP message (including headers and the body), then the whole message MUST be sent within a WebSocket binary message. Given the nature of

JavaScript and the WebSocket API, it is RECOMMENDED to use UTF-8 encoding (or ASCII, which is a subset of UTF-8) for SIP messages carried over a WebSocket connection.

5. SIP WebSocket Transport

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

Each SIP message MUST be carried within a single WebSocket message, and a WebSocket message MUST NOT contain more than one SIP message. Because the WebSocket transport preserves message boundaries, the use of the Content-Length header in SIP messages is not necessary when they are transported using the WebSocket subprotocol.

This simplifies the parsing of SIP messages for both clients and servers. There is no need to establish message boundaries using Content-Length headers between messages. Other SIP transports, such as UDP and the Stream Control Transmission Protocol (SCTP) [RFC4168], also provide this benefit.

5.1. Via Transport Parameter

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

The updated augmented BNF (Backus-Naur Form) [RFC5234] for this parameter is the following (the original BNF for this parameter can be found in [RFC3261], which was then updated by [RFC4168]):

```
transport  =/  "WS" / "WSS"
```

5.2. SIP URI Transport Parameter

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

The updated augmented BNF for this parameter is the following (the original BNF for this parameter can be found in [RFC3261]):

```
transport-param  =/  "transport=" "ws"
```

5.3. Via "received" Parameter

The following is stated in [RFC3261], Section 18.2.1, "Receiving Requests":

> 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" field 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 the WebSocket protocol design. The WebSocket connection handshake reuses the existing HTTP infrastructure in which there could be an unknown number of HTTP proxies and/or TCP load balancers between the SIP WebSocket Client and Server, so the source address the server would write into the Via "received" parameter would be the address of the HTTP/TCP intermediary in front of it. This could reveal sensitive information about the internal topology of the server's network to the client.

Given the fact that SIP responses can only be sent over the existing WebSocket connection, the Via "received" parameter is of little use. Therefore, in order to allow hiding possible sensitive information about the SIP WebSocket Server's network, this document updates [RFC3261], Section 18.2.1 by stating:

> When a SIP WebSocket Server receives a request, it MAY decide not to add a "received" parameter to the top Via header. Therefore, SIP WebSocket Clients MUST accept responses without such a parameter in the top Via header regardless of whether the Via "sent-by" field contains a domain name.

5.4. SIP Transport Implementation Requirements

The following is stated in [RFC3261], Section 18, "Transport":

> All SIP elements MUST implement UDP and TCP. SIP elements MAY implement other protocols.

The specification of this transport enables SIP to be used as a session establishment protocol in scenarios where none of the other transport protocols defined for SIP can be used. Since some

environments do not enable SIP elements to use UDP and TCP as SIP transport protocols, a SIP element acting as a SIP WebSocket Client is not mandated to implement support of UDP and TCP.

5.5. Locating a SIP Server

[RFC3263] specifies the procedures that 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 connections and "SIPS+D2W" for SIP WebSocket Servers that support secure WebSocket connections.

> At the time this document was written, DNS NAPTR/Service Record (SRV) queries could not be performed by commonly available WebSocket client stacks (in JavaScript engines and web browsers).

In the absence of DNS SRV resource records or an explicit port, the default port for a SIP URI using the "sip" scheme and the "ws" transport parameter is 80, and the default port for a SIP URI using the "sips" scheme and the "ws" transport parameter is 443.

6. Connection Keep-Alive

SIP WebSocket Clients and Servers may keep their WebSocket connections open by sending periodic WebSocket "Ping" frames as described in [RFC6455], Section 5.5.2.

> The WebSocket API [WS-API] does not provide a mechanism for applications running in a web browser to control whether or not periodic WebSocket "Ping" frames are sent to the server. The implementation of such a keep-alive feature is the decision of each web browser manufacturer and may also depend on the configuration of the web browser.

The indication and use of the CRLF NAT keep-alive mechanism defined for SIP connection-oriented transports in [RFC5626], Section 3.5.1 or [RFC6223] are, of course, usable over the transport defined in this specification.

7. Authentication

This section describes how authentication is achieved through the requirements in [RFC6455], [RFC6265], [RFC2617], and [RFC3261].

The WebSocket protocol [RFC6455] does not define an authentication mechanism; instead, it exposes the following text in Section 10.5, "WebSocket Client Authentication":

> This protocol doesn't prescribe any particular way that servers can authenticate clients during the WebSocket handshake. The WebSocket server can use any client authentication mechanism available to a generic HTTP server, such as cookies, HTTP authentication, or TLS authentication.

The following list exposes mandatory-to-implement and optional mechanisms for SIP WebSocket Clients and Servers in order to get interoperability at the WebSocket authentication level:

- A SIP WebSocket Client MUST be ready to add a session cookie when it runs in a web browser (or behaves like a browser navigating a website) and has previously retrieved a session cookie from the web server whose URL domain matches the domain in the WebSocket URI. This mechanism is defined by [RFC6265].

- A SIP WebSocket Client MUST be ready to be challenged with an HTTP 401 status code [RFC2617] by the SIP WebSocket Server when performing the WebSocket handshake.

- A SIP WebSocket Client MAY use TLS client authentication (when in a secure WebSocket connection) as an optional authentication mechanism.

  > Note, however, that TLS client authentication in the WebSocket protocol is governed by the rules of the HTTP protocol rather than the rules of SIP.

- A SIP WebSocket Server MUST be ready to read session cookies when present in the WebSocket handshake request and use such a cookie value for determining whether the WebSocket connection has been initiated by an HTTP client navigating a website in the same domain (or subdomain) as the SIP WebSocket Server.

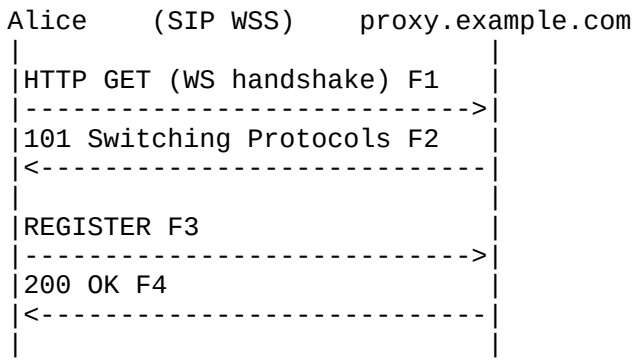
- A SIP WebSocket Server SHOULD be able to reject a WebSocket handshake request with an HTTP 401 status code by providing a Basic/Digest challenge as defined for the HTTP protocol.

Regardless of whether or not the SIP WebSocket Server requires authentication during the WebSocket handshake, authentication MAY be requested at the SIP level.

Some authentication use cases are exposed in Appendix A.

8. Examples

8.1. Registration



Alice loads a web page using her web browser and retrieves JavaScript code implementing the WebSocket SIP subprotocol 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.example.com. Upon WebSocket connection, Alice constructs and sends a SIP REGISTER request, including Outbound [RFC5626] and Globally Routable User Agent URI (GRUU) [RFC5627] support. Since the JavaScript stack in a browser has no way to determine the local address from which the WebSocket connection was made, this implementation uses a random ".invalid" domain name for the Via header "sent-by" parameter and for the hostport of the URI in the Contact header (see Appendix B.1).

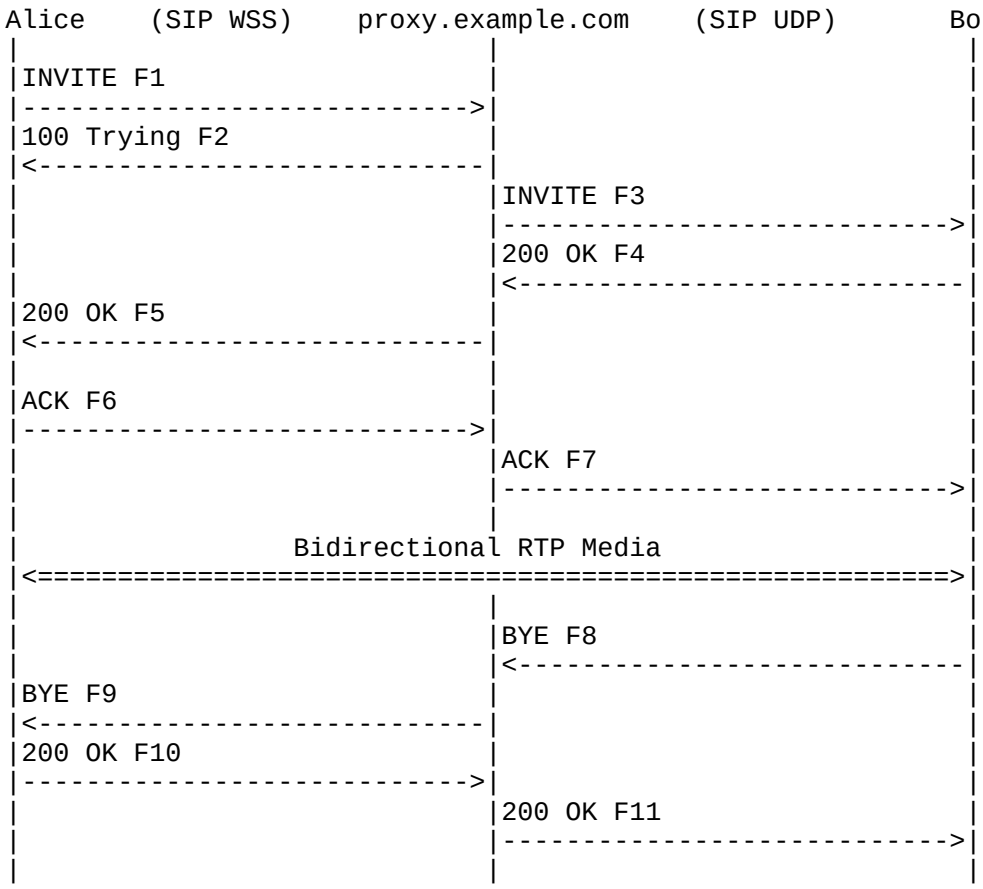
Message details (authentication and Session Description Protocol (SDP) bodies are omitted for simplicity):

F1 HTTP GET (WS handshake) Alice -> proxy.example.com (TLS)

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

```
   F2 101 Switching Protocols  proxy.example.com -> Alice (TLS)

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


   F3 REGISTER  Alice -> proxy.example.com (transport WSS)

   REGISTER sip:proxy.example.com SIP/2.0
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bKasudf
   From: sip:alice@example.com;tag=65bnmj.34asd
   To: sip:alice@example.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>"


   F4 200 OK  proxy.example.com -> Alice (transport WSS)

   SIP/2.0 200 OK
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bKasudf
   From: sip:alice@example.com;tag=65bnmj.34asd
   To: sip:alice@example.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@example.com;gr=urn:uuid:f81-7dec-14a06cf1"
     ;temp-gruu="sip:87ash54=3dd.98a@example.com;gr"
     ;expires=3600
```

8.2. INVITE Dialog through a Proxy



In the same scenario, Alice places a call to Bob's Address of Record (AOR). The SIP WebSocket Server at proxy.example.com acts as a SIP proxy, routing the INVITE to Bob's contact address (which happens to be using SIP transported over UDP). Bob answers the call and then terminates it.

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

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

INVITE sip:bob@example.com SIP/2.0
Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bK56sdasks
From: sip:alice@example.com;tag=asdyka899
To: sip:bob@example.com
Call-ID: asidkj3ss
CSeq: 1 INVITE
Max-Forwards: 70
Supported: path, outbound, gruu
Route: <sip:proxy.example.com:443;transport=ws;lr>
Contact: <sip:alice@example.com
 ;gr=urn:uuid:f81-7dec-14a06cf1;ob>
Content-Type: application/sdp


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

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


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

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

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

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP proxy.example.com;branch=z9hG4bKhjhjqw32c
     ;received=192.0.2.10
   Via: SIP/2.0/WSS df7jal23ls0d.invalid;branch=z9hG4bK56sdasks
   Record-Route: <sip:proxy.example.com;transport=udp;lr>,
     <sip:h7kjh12s@proxy.example.com:443;transport=ws;lr>
   From: sip:alice@example.com;tag=asdyka899
   To: sip:bob@example.com;tag=bmqkjhsd
   Call-ID: asidkj3ss
   CSeq: 1 INVITE
   Contact: <sip:bob@203.0.113.22:5060;transport=udp>
   Content-Type: application/sdp


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

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


   F6 ACK  Alice -> proxy.example.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.example.com:443;transport=ws;lr>,
     <sip:proxy.example.com;transport=udp;lr>,
   From: sip:alice@example.com;tag=asdyka899
   To: sip:bob@example.com;tag=bmqkjhsd
   Call-ID: asidkj3ss
   CSeq: 1 ACK
   Max-Forwards: 70
```

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

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

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

```
BYE sip:alice@example.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.example.com;transport=udp;lr>,
  <sip:h7kjh12s@proxy.example.com:443;transport=ws;lr>
From: sip:bob@example.com;tag=bmqkjhsd
To: sip:alice@example.com;tag=asdyka899
Call-ID: asidkj3ss
CSeq: 1201 BYE
Max-Forwards: 70
```

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

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

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

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

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

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

9. Security Considerations

9.1. Secure WebSocket Connection

It is RECOMMENDED that the SIP traffic transported over a WebSocket communication be protected by using a secure WebSocket connection (using TLS [RFC5246] over TCP).

When establishing a connection using SIP over secure WebSocket transport, the client MUST authenticate the server using the server's certificate according to the WebSocket validation procedure in [RFC6455].

> Server operators should note that this authentication procedure is different from the procedure for SIP domain certificates defined in [RFC5922]. Certificates that are appropriate for SIP over TLS over TCP will probably not be appropriate for SIP over secure WebSocket connections.

9.2. Usage of "sips" Scheme

The "sips" scheme in a SIP URI dictates that the entire request path to the target be secure. If such a path includes a WebSocket connection, it MUST be a secure WebSocket connection.

10. IANA Considerations

10.1. Registration of the WebSocket SIP Subprotocol

IANA has registered the WebSocket SIP subprotocol under the "WebSocket Subprotocol Name" registry with the following data:

Subprotocol Identifier: sip

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

Subprotocol Definition: [RFC7118]

10.2. Registration of New NAPTR Service Field Values

This document defines two new NAPTR service field values (SIP+D2W and SIPS+D2W) and IANA has registered these values under the "Registry for the Session Initiation Protocol (SIP) NAPTR Resource Record Services Field". The entries are as follows:

Services Field	Protocol	Reference
SIP+D2W	WS	[RFC7118]
SIPS+D2W	WS	[RFC7118]

10.3. SIP/SIPS URI Parameters Subregistry

IANA has added a reference to this document under the "SIP/SIPS URI Parameters" subregistry within the "Session Initiation Protocol (SIP) Parameters" registry:

Parameter Name	Predefined Values	Reference
transport	Yes	[RFC3261][RFC7118]

10.4. Header Fields Subregistry

IANA has added a reference to this document under the "Header Fields" subregistry within the "Session Initiation Protocol (SIP) Parameters" registry:

Header Name	compact	Reference
Via	v	[RFC3261][RFC7118]

10.5. Header Field Parameters and Parameter Values Subregistry

IANA has added a reference to this document under the "Header Field Parameters and Parameter Values" subregistry within the "Session Initiation Protocol (SIP) Parameters" registry:

Header Field	Parameter Name	Predefined Values	Reference
Via	received	No	[RFC3261][RFC7118]

10.6. SIP Transport Subregistry

This document adds a new subregistry, "SIP Transport", to the "Session Initiation Protocol (SIP) Parameters" registry. Its format and initial values are as shown in the following table:

Transport	Reference
UDP	[RFC3261]
TCP	[RFC3261]
TLS	[RFC3261]
SCTP	[RFC3261], [RFC4168]
TLS-SCTP	[RFC4168]
WS	[RFC7118]
WSS	[RFC7118]

The policy for registration of values in this registry is "Standards Action" [RFC5226].

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Appendix A. Authentication Use Cases

The sections below briefly describe some SIP over WebSocket scenarios in which authentication takes place in different ways.

A.1. Just SIP Authentication

SIP Private Branch Exchange (PBX) model A implements the SIP WebSocket transport defined by this specification. Its implementation is 100% website agnostic as it does not share information with the web server providing the HTML code to browsers, meaning that the SIP WebSocket Server (here, PBX model A) has no knowledge about web login activity within the website.

In this simple scenario, the SIP WebSocket Server does not inspect fields in the WebSocket handshake HTTP GET request such as the request URL, the Origin header value, the Host header value, or the Cookie header value (if present). However, some of those fields could be inspected for a minimal validation (i.e., PBX model A could require that the Origin header value contains a specific URL so just users navigating such a website would be able to establish a WebSocket connection with PBX model A).

Once the WebSocket connection has been established, SIP authentication is requested by PBX model A for each SIP request coming over that connection. Therefore, SIP WebSocket Clients must be provisioned with their corresponding SIP password.

A.2. Just Web Authentication

A SIP-to-PSTN (Public Switched Telephone Network) provider offers telephony service for clients logged into its website. The provider does not want to expose SIP passwords into the web for security/ privacy reasons.

Once the user is logged into the web, the web server provides him with a SIP identity (SIP URI) and a session temporary token string (along with the SIP WebSocket Client JavaScript application and SIP settings). The web server stores the SIP identity and session token into a database.

The web application adds the SIP identity and session token as URL query parameters in the WebSocket handshake request and attempts the connection. The SIP WebSocket Server inspects the handshake request and validates that the session token matches the value stored in the database for the given SIP identity. In case the value matches, the WebSocket connection gets "authenticated" for that SIP identity. The SIP WebSocket Client can then register and make calls. The SIP

WebSocket Server would, however, verify that the identity in those SIP requests (i.e., the From URI value) matches the SIP identity the WebSocket connection is associated to (otherwise, the SIP request is rejected).

When the user performs a logout action in the web, the web server removes the SIP identity and session token tuple from the database and notifies the SIP WebSocket Server, which revokes and closes the WebSocket connection.

No SIP authentication takes place in this scenario.

A.3. Cookie-Based Authentication

The Apache web server comes with a new module: mod_sip_websocket. In port 80, the web server is configured to listen for both HTTP common requests and WebSocket handshake requests. Therefore, both the web server and the SIP WebSocket Server are co-located within the same host and same domain.

Once the user is logged into the web, he is provided with the SIP WebSocket Client JavaScript application and SIP settings. The HTTP 200 response after the login procedure also contains a session cookie [RFC6265]. The web application then attempts a WebSocket connection against the same URL/domain of the website, and thus the session cookie is automatically added by the browser into the WebSocket handshake request (as the WebSocket protocol [RFC6455] states).

The web server inspects the cookie value (as it would do for a common HTTP request containing a session cookie so that the login procedure is not required again). If the cookie is valid, the WebSocket connection is authorized. And, as in the previous use case, the connection is also associated with a specific SIP identity that must be satisfied by every SIP request coming over that connection.

No SIP authentication takes place in this scenario but just common cookie usage as widely deployed in the World Wide Web.

Appendix B. Implementation Guidelines

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

Such a SIP stack connects to a given SIP WebSocket Server (an outbound SIP proxy that also implements classic SIP transports such as UDP and TCP). The HTTP GET method request sent by the web browser for the WebSocket handshake includes a Cookie [RFC6265] header with the value previously provided by the server after the successful login procedure. The cookie value is then inspected by the WebSocket server to authorize the connection. Once the WebSocket connection is established, the SIP WebSocket Client performs a SIP registration to a SIP registrar server that is reachable through the proxy. After registration, the SIP WebSocket Client and Server exchange SIP messages as would normally be expected.

This scenario is quite similar to ones in which SIP user agents (UAs) behind NATs connect to a proxy and must reuse the same TCP connection for incoming requests (because they are not directly reachable by the proxy otherwise). In both cases, the SIP UAs are only reachable through the proxy to which they are connected.

The SIP Outbound extension [RFC5626] seems an appropriate solution for this scenario. Therefore, these SIP WebSocket Clients and the SIP registrar implement both the Outbound and Path [RFC3327] extensions, and the SIP proxy acts as 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 to which they are connected. Therefore, in some call transfer cases, the use 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 third SIP user agent including the Contact URI of a SIP WebSocket Client as the target in its Refer-To header field, such a URI will be reachable by the third SIP UA only if it is a globally routable URI. GRUU (Globally Routable User Agent URI) is a solution for those scenarios and would cause the incoming request from the third SIP user agent to be sent to the SIP registrar, which would route the request to the SIP WebSocket Client via the Outbound Edge Proxy.

B.1. SIP WebSocket Client Considerations

The JavaScript stack in web browsers does not have the ability to discover the local transport address used for originating WebSocket connections. A SIP WebSocket Client running in such an environment can construct a domain name consisting of a random token followed by the ".invalid" top-level domain name, as stated in [RFC2606], and uses it within its Via and Contact headers.

The Contact URI provided by SIP UAs requesting (and receiving) Outbound support is not used for routing requests to those UAs, thus it is safe to set a random domain in the Contact URI hostport.

Both the Outbound and GRUU specifications require a SIP UA to include a Uniform Resource Name (URN) in a "+sip.instance" parameter of the Contact header in which they include their SIP REGISTER requests. The client device is responsible for generating or collecting a suitable value for this purpose.

In web browsers, it is difficult to generate or collect a suitable value to be used as an 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.

B.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 the path of dialogs as specified in [RFC3261]. If it did not do this, in-dialog requests would fail since SIP WebSocket Clients make use of their SIP WebSocket Server in order to send and receive SIP messages.

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