RTCWEB   M. Perumal
Internet-Draft   Ericsson
Intended status: Standards Track   D. Wing
Expires: March 19, 2015   R. Ravindranath
                                                            T. Reddy
                                                       Cisco Systems
                                                         M. Thomson
                                                            Mozilla
                                                 September 15, 2014

STUN Usage for Consent Freshness
draft-ietf-rtcweb-stun-consent-freshness-07

Abstract

   To prevent sending excessive traffic to an endpoint, periodic consent
   needs to be obtained from that remote endpoint.

   This document describes a consent mechanism using a new Session
   Traversal Utilities for NAT (STUN) usage. This same mechanism can
   also determine connection loss ("liveness") with a remote peer.

Status of This Memo

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

   Internet-Drafts are working documents of the Internet Engineering
   Task Force (IETF). Note that other groups may also distribute
   working documents as Internet-Drafts. The list of current Internet-
   Drafts is at http://datatracker.ietf.org/drafts/current/.

   Internet-Drafts are draft documents valid for a maximum of six months
   and may be updated, replaced, or obsoleted by other documents at any
   time. It is inappropriate to use Internet-Drafts as reference
   material or to cite them other than as "work in progress."

   This Internet-Draft will expire on March 19, 2015.

Copyright Notice

Table of Contents

1. Introduction

To prevent attacks on peers, RTP endpoints have to ensure the remote
peer wants to receive traffic. This is performed both when the
session is first established to the remote peer using Interactive
Connectivity Establishment ICE [RFC5245] connectivity checks, and
periodically for the duration of the session using the procedures
defined in this document.

When a session is first established, ICE implementations obtain
initial consent by performing STUN connectivity checks as part of
ICE. That initial consent is not described further in this document
and it is assumed that ICE is being used for that initial consent.

Related to consent is loss of connectivity ("liveness"). Many
applications want notification of connection loss to take appropriate
actions (e.g., alert the user, try switching to a different
interface).

This document describes a new STUN usage with exchange of request and
response messages to verify the remote peer's consent to receive

traffic, and the absence of which for a period of time indicates a
loss of liveness.

When a (full) ICE implementation interworks with an ICE-lite
implementation the ICE-lite implementation will not generate consent
checks, but will just just respond to consent checks it receives.

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].

Consent: It is the mechanism of obtaining permission to send traffic
   to a certain transport address. This is the initial consent to
   send traffic, which is obtained by ICE or a TCP handshake.

Consent Freshness: Permission to continue sending traffic to a
   certain transport address. This is performed by the procedure
   described in this document.

Session Liveness: Detecting loss of connectivity to a certain
   transport address. This is performed by the procedure described
   in this document.

Transport Address: The remote peer's IP address and (UDP or TCP)
   port number.

3. Design Considerations

Although ICE requires periodic keepalive traffic to keep NAT bindings
alive (Section 10 of [RFC5245], [RFC6263]), those keepalives are sent
as STUN Indications which are send-and-forget, and do not evoke a
response. A response is necessary both for consent to continue
sending traffic, as well as to verify session liveness. Thus, we
need a request/response mechanism for consent freshness. ICE can be
used for that mechanism because ICE implementations are already
required to continue listening for ICE messages, as described in
section 10 of [RFC5245].

4. Solution

There are two ways consent to send traffic is revoked: expiration of
consent and immediate revocation of consent, which are discussed in
the following sections.

4.1. Expiration of Consent

   A WebRTC implementation [I-D.ietf-rtcweb-overview], which implements
   ICE, MUST perform a combined consent freshness and session liveness
   test using STUN request/response as described below:

   An endpoint MUST NOT send application data (e.g., RTP, RTCP, SCTP,
   DTLS), over any transport protocol (e.g., UDP, TCP) on an ICE-
   initiated connection unless the receiving endpoint consents to
   receive the data. After a successful ICE connectivity check on a
   particular transport address, subsequent consent MUST be obtained
   following the procedure described in this document. The consent
   expires after a fixed amount of time. During ICE restart consent
   checks MUST continue to be sent on previously validated pair, and
   MUST be responded to on the previously validated pair, until ICE
   restart completes.

      Note: Although TCP has its own consent mechanism (TCP
      acknowledgements), consent is necessary over a TCP connection
      because it could be translated to a UDP connection (e.g.,
      [RFC6062]).

   Explicit consent to send is obtained by sending an ICE binding
   request to the remote peer's Transport Address and receiving a
   matching, authenticated, non-error ICE binding response from the
   remote peer's Transport Address. These ICE binding requests and
   responses are authenticated using the same short-term credentials as
   the initial ICE exchange. Implementations MUST cease sending data if
   their consent expires. To prevent expiry of consent, a STUN binding
   request MUST be sent every N milliseconds, where N is chosen randomly
   with each consent check in the interval [.8N, 1.2N] (to prevent
   network synchronization), where N SHOULD be 5000. Using the value
   5000 milliseconds and that 20% randomization range, N would be a
   value between 4000 and 6000. These STUN binding requests for consent
   are not re-transmitted. Each STUN binding request for consent re-
   calculates a new random value N and a new cryptographically-random
   [RFC4086] STUN transaction ID.

   The initial Consent to send traffic is obtained by ICE. Consent
   expires after 30 seconds. That is, if a valid STUN binding response
   corresponding to one of the STUN requests sent in the last 30 seconds
   has not been received from the remote peer's Transport Address, the
   endpoint MUST cease transmission on that 5-tuple.

   To meet the security needs of consent, an untrusted application
   (e.g., JavaScript) MUST NOT be able to obtain or control the STUN
   transaction ID, because that enables spoofing STUN responses,
   falsifying consent.

While TCP affords some protection from off-path attackers ([RFC5961],
[RFC4953]), there is still a risk an attacker could cause a TCP
sender to send packets forever by spoofing ACKs. To prevent such an
attack, consent checks MUST be performed over all transport
connections, including TCP. In this way, an off-path attacker
spoofing TCP segments can not cause a TCP sender to send packets
longer than the consent timer (30 seconds).

An endpoint that is not sending any application traffic does not need
to obtain consent which can slightly conserve its resources.
However, the endpoint needs to ensure its NAT or firewall mappings
persist which can be done using keepalive or other techniques (see
Section 10 of [RFC5245] and see [RFC6263]). If the endpoint wants to
send application traffic, it needs to first obtain consent if its
consent has expired.

4.2. Immediate Revocation of Consent

The previous section explained how consent expires due to a timeout.
In some cases it is useful to signal a connection is terminated,
rather than relying on a timeout. This is done by immediately
revoking consent.

Consent for sending traffic on the media or data channel is
immediately revoked by receipt of an authenticated message that
closes the connection (e.g., a TLS fatal alert) or receipt of a valid
and authenticated STUN response with error code Forbidden (403).
Those consent revocation messages can be lost on the network, so an
implementation wanting to immediately revoke consent needs to
remember those credentials until consent expiry (30 seconds).

Receipt of an unauthenticated message that closes a connection (e.g.,
TCP FIN) does not indicate revocation of consent. Thus, an endpoint
receiving an unauthenticated end-of-session message SHOULD continue
sending media (over connectionless transport) or attempt to re-
establish the connection (over connection-oriented transport) until
consent expires or it receives an authenticated message revoking
consent.

Note that an authenticated SRTCP BYE does not terminate consent; it
only indicates the associated SRTP source has quit.

5. Connection Liveness

A connection is considered "live" if packets are received from a
remote endpoint within an application-dependent period. An
application can request a notification when there are no packets
received for a certain period (configurable).

Similarly, if packets haven't been received within a certain period, an application can request a consent check (heartbeat) be generated. These two time intervals might be controlled by the same configuration item.

Sending consent checks (heartbeats) at a high rate could allow a malicious application to generate congestion, so applications MUST NOT be able to send heartbeats at an average rate of more than 1 per second.

6. DiffServ Treatment for Consent packets

It is RECOMMENDED that STUN consent checks use the same Diffserv Codepoint markings as the ICE connectivity checks described in section 7.1.2.4 of [RFC5245] for a given 5-tuple.

Note: It is possible that different Diffserv Codepoints are used by different media over the same transport address [I-D.ietf-tsvwg-rtcweb-qos]. Such a case is outside the scope of this document.

7. W3C API Implications

For the consent freshness and liveness test the W3C specification should provide APIs as described below:

1. Ability for the browser to notify the JavaScript that consent freshness has failed for a 5-tuple and the browser has stopped transmitting on that 5-tuple.

2. Ability for the JavaScript to start and stop liveness test and set the liveness test interval.

3. Ability for the browser to notify the JavaScript that a liveness test has failed for a media stream.

8. Security Considerations

This document describes a security mechanism.

The security considerations discussed in [RFC5245] should also be taken into account.

SRTP is encrypted and authenticated with symmetric keys; that is, both sender and receiver know the keys. With two party sessions, receipt of an authenticated packet from the single remote party is a strong assurance the packet came from that party. However, when a session involves more than two parties, all of whom know each others

keys, any of those parties could have sent (or spoofed) the packet. Such shared key distributions are possible with some MIKEY [RFC3830] modes, Security Descriptions [RFC4568], and EKT [I-D.ietf-avtcore-srtp-ekt]. Thus, in such shared keying distributions, receipt of an authenticated SRTP packet is not sufficient to verify consent.

9. IANA Considerations

This document does not require any action from IANA.

10. Acknowledgement

Thanks to Eric Rescorla, Harald Alvestrand, Bernard Aboba, Magnus Westerland, Cullen Jennings, Christer Holmberg, Simon Perreault, Paul Kyzivat, Emil Ivov, and Jonathan Lennox for their valuable inputs and comments.

11. References

11.1. Normative References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

[RFC4086] Eastlake, D., Schiller, J., and S. Crocker, "Randomness Requirements for Security", BCP 106, RFC 4086, June 2005.

[RFC5245] Rosenberg, J., "Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols", RFC 5245, April 2010.

[RFC6263] Marjou, X. and A. Sollaud, "Application Mechanism for Keeping Alive the NAT Mappings Associated with RTP / RTP Control Protocol (RTCP) Flows", RFC 6263, June 2011.

11.2. Informative References

[I-D.ietf-avtcore-srtp-ekt]
           McGrew, D. and D. Wing, "Encrypted Key Transport for Secure RTP", draft-ietf-avtcore-srtp-ekt-02 (work in progress), February 2014.

[I-D.ietf-rtcweb-overview]
           Alvestrand, H., "Overview: Real Time Protocols for Browser-based Applications", draft-ietf-rtcweb-overview-11 (work in progress), August 2014.

   [I-D.ietf-tsvwg-rtcweb-qos]
               Dhesikan, S., Jennings, C., Druta, D., Jones, P., and J.
               Polk, "DSCP and other packet markings for RTCWeb QoS",
               draft-ietf-tsvwg-rtcweb-qos-02 (work in progress), June
               2014.

   [RFC3830]   Arkko, J., Carrara, E., Lindholm, F., Naslund, M., and K.
               Norrman, "MIKEY: Multimedia Internet KEYing", RFC 3830,
               August 2004.

   [RFC4568]   Andreasen, F., Baugher, M., and D. Wing, "Session
               Description Protocol (SDP) Security Descriptions for Media
               Streams", RFC 4568, July 2006.

   [RFC4953]   Touch, J., "Defending TCP Against Spoofing Attacks", RFC
               4953, July 2007.

   [RFC5961]   Ramaiah, A., Stewart, R., and M. Dalal, "Improving TCP's
               Robustness to Blind In-Window Attacks", RFC 5961, August
               2010.

   [RFC6062]   Perreault, S. and J. Rosenberg, "Traversal Using Relays
               around NAT (TURN) Extensions for TCP Allocations", RFC
               6062, November 2010.

Authors' Addresses

   Muthu Arul Mozhi Perumal
   Ericsson
   Ferns Icon
   Doddanekundi, Mahadevapura
   Bangalore, Karnataka 560037
   India

   Email: muthu.arul@gmail.com

   Dan Wing
   Cisco Systems
   821 Alder Drive
   Milpitas, California 95035
   USA

   Email: dwing@cisco.com

Ram Mohan Ravindranath
Cisco Systems
Cessna Business Park
Sarjapur-Marathahalli Outer Ring Road
Bangalore, Karnataka 560103
India

Email: rmohanr@cisco.com

Tirumaleswar Reddy
Cisco Systems
Cessna Business Park, Varthur Hobli
Sarjapur Marathalli Outer Ring Road
Bangalore, Karnataka 560103
India

Email: tireddy@cisco.com

Martin Thomson
Mozilla
Suite 300
650 Castro Street
Mountain View, California 94041
US

Email: martin.thomson@gmail.com