Stream:	Internet Engineering Task Force (IETF)
RFC:	8723
Category:	Standards Track
Published:	January 2020
ISSN:	2070-1721
Authors:	C. Jennings	P. Jones	R. Barnes	A.B. Roach
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	Cisco Systems	*Cisco Systems*	*Cisco Systems*	*Mozilla*

RFC 8723
Double Encryption Procedures for Secure Real-Time Transport Protocol (SRTP)

Abstract

In some conferencing scenarios, it is desirable for an intermediary to be able to manipulate some parameters in Real-time Transport Protocol (RTP) packets, while still providing strong end-to-end security guarantees. This document defines a cryptographic transform for the Secure Real-time Transport Protocol (SRTP) that uses two separate but related cryptographic operations to provide hop-by-hop and end-to-end security guarantees. Both the end-to-end and hop-by-hop cryptographic algorithms can utilize an authenticated encryption with associated data (AEAD) algorithm or take advantage of future SRTP transforms with different properties.

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 7841.

Information about the current status of this document, any errata, and how to provide feedback on it may be obtained at https://www.rfc-editor.org/info/rfc8723.

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1. Introduction

Cloud conferencing systems that are based on switched conferencing have a central Media Distributor (MD) device that receives media from endpoints and distributes it to other endpoints, but does not need to interpret or change the media content. For these systems, it is desirable to have one cryptographic key that enables encryption and authentication of the media end-to-end while still allowing certain information in the header of a Real-time Transport Protocol (RTP) packet to be changed by the Media Distributor. At the same time, a separate cryptographic key provides integrity and optional confidentiality for the media flowing between the Media Distributor and the endpoints. The framework document [PRIVATE-MEDIA-FRAMEWORK] describes this concept in more detail.

This specification defines a transform for the Secure Real-time Transport Protocol (SRTP) that uses 1) the AES Galois/Counter Mode (AES-GCM) algorithm [RFC7714] to provide encryption and integrity for an RTP packet for the end-to-end cryptographic key and 2) a hop-by-hop cryptographic encryption and integrity between the endpoint and the Media Distributor. The Media Distributor decrypts and checks integrity of the hop-by-hop security. The Media Distributor **MAY** change some of the RTP header information that would impact the end-to-end integrity. In that case, the original value of any RTP header field that is changed is included in an "Original Header Block" that is added to the packet. The new RTP packet is encrypted with the hop-by-hop cryptographic algorithm before it is sent. The receiving endpoint decrypts and checks integrity using the hop-by-hop cryptographic algorithm and then replaces any parameters the Media Distributor changed using the information in the Original Header Block before decrypting and checking the end-to-end integrity.

One can think of the double transform as a normal SRTP transform for encrypting the RTP in a way such that things that only know half of the key, can decrypt and modify part of the RTP packet but not other parts, including the media payload.

2. Terminology

The key words "**MUST**", "**MUST NOT**", "**REQUIRED**", "**SHALL**", "**SHALL NOT**", "**SHOULD**", "**SHOULD NOT**", "**RECOMMENDED**", "**NOT RECOMMENDED**", "**MAY**", and "**OPTIONAL**" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

Terms used throughout this document include:

Media Distributor: A device that receives media from endpoints and distributes it to other endpoints, but does not need to interpret or change the media content (see also [PRIVATE-MEDIA-FRAMEWORK]).

end-to-end: The path from one endpoint through one or more Media Distributors to the endpoint at the other end.

hop-by-hop: The path from the endpoint to or from the Media Distributor.

Original Header Block (OHB): An octet string that contains the original values from the RTP header that might have been changed by a Media Distributor.

3. Cryptographic Context

This specification uses a cryptographic context with two parts:

- An inner (end-to-end) part that is used by endpoints that originate and consume media to ensure the integrity of media end-to-end, and
- An outer (hop-by-hop) part that is used between endpoints and Media Distributors to ensure the integrity of media over a single hop and to enable a Media Distributor to modify certain RTP header fields. RTCP is also handled using the hop-by-hop cryptographic part.

The **RECOMMENDED** cipher for the hop-by-hop and end-to-end algorithms is AES-GCM. Other combinations of SRTP ciphers that support the procedures in this document can be added to the IANA registry.

The keys and salt for these algorithms are generated with the following steps:

- Generate key and salt values of the length required for the combined inner (end-to-end) and outer (hop-by-hop) algorithms.
- Assign the key and salt values generated for the inner (end-to-end) algorithm to the first half of the key and the first half of the salt for the double algorithm.
- Assign the key and salt values for the outer (hop-by-hop) algorithm to the second half of the key and second half of the salt for the double algorithm. The first half of the key is referred to as the inner key while the second half is referred to as the outer key. When a key is used by a cryptographic algorithm, the salt that is used is the part of the salt generated with that key.
- the synchronization source (SSRC) is the same for both the inner and outer algorithms as it cannot be changed.
- The sequence number (SEQ) and rollover counter (ROC) are tracked independently for the inner and outer algorithms.

If the Media Distributor is to be able to modify header fields but not decrypt the payload, then it must have a cryptographic key for the outer algorithm but not the inner (end-to-end) algorithm. This document does not define how the Media Distributor should be provisioned with this information. One possible way to provide keying material for the outer (hop-by-hop) algorithm is to use [DTLS-TUNNEL].

3.1. Key Derivation

Although SRTP uses a single master key to derive keys for an SRTP session, this transform requires separate inner and outer keys. In order to allow the inner and outer keys to be managed independently via the master key, the transforms defined in this document **MUST** be used with

the following pseudorandom function (PRF), which preserves the separation between the two halves of the key. Given a positive integer `n` representing the desired output length, a master key `k_master`, and an input `x`:

```
PRF_double_n(k_master,x) = PRF_(n/2)(inner(k_master),x) ||
                           PRF_(n/2)(outer(k_master),x)
```

Here `PRF_double_n(k_master, x)` represents the AES_CM PRF Key Derivation Function (KDF) (see Section 4.3.3 of [RFC3711]) for DOUBLE_AEAD_AES_128_GCM_AEAD_AES_128_GCM algorithm and AES_256_CM_PRF KDF [RFC6188] for DOUBLE_AEAD_AES_256_GCM_AEAD_AES_256_GCM algorithm. The term `inner(k_master)` represents the first half of the key; `outer(k_master)` represents the second half of the key.

4. Original Header Block

The Original Header Block (OHB) contains the original values of any modified RTP header fields. In the encryption process, the OHB is included in an SRTP packet as described in Section 5. In the decryption process, the receiving endpoint uses it to reconstruct the original RTP header so that it can pass the proper additional authenticated data (AAD) value to the inner transform.

The OHB can reflect modifications to the following fields in an RTP header: the payload type (PT), the sequence number (SEQ), and the marker bit. All other fields in the RTP header **MUST** remain unmodified; since the OHB cannot reflect their original values, the receiver will be unable to verify the end-to-end (E2E) integrity of the packet.

The OHB has the following syntax (in ABNF [RFC5234]):

```
OCTET = %x00-FF

PT = OCTET
SEQ = 2OCTET
Config = OCTET
OHB = [ PT ] [ SEQ ] Config
```

If present, the PT and SEQ parts of the OHB contain the original payload type and sequence number fields, respectively. The final "Config" octet of the OHB specifies whether these fields are present, and the original value of the marker bit (if necessary):

```
+-+-+-+-+-+-+-+-+
|R R R R B M P Q|
+-+-+-+-+-+-+-+-+
```

- P: PT is present
- Q: SEQ is present
- M: Marker bit is present
- B: Value of marker bit
- R: Reserved, **MUST** be set to 0

In particular, an all-zero OHB Config octet (`0x00`) indicates that there have been no modifications from the original header.

If the marker bit is not present (M=0), then `B` **MUST** be set to zero. That is, if `C` represents the value of the Config octet, then the masked value `C & 0x0C` **MUST NOT** have the value `0x80`.

5. RTP Operations

As implied by the use of the word "double" above, this transform applies AES-GCM to the SRTP packet twice. This allows media distributors to be able to modify some header fields while allowing endpoints to verify the end-to-end integrity of a packet.

The first, "inner" application of AES-GCM encrypts the SRTP payload and protects the integrity of a version of the SRTP header with extensions truncated. Omitting extensions from the inner integrity check means that they can be modified by a Media Distributor holding only the outer key.

The second, "outer" application of AES-GCM encrypts the ciphertext produced by the inner encryption (i.e., the encrypted payload and authentication tag), plus an OHB that expresses any changes made between the inner and outer transforms.

A Media Distributor that has the outer key but not the inner key may modify the header fields that can be included in the OHB by decrypting, modifying, and re-encrypting the packet.

5.1. Encrypting a Packet

An endpoint encrypts a packet by using the inner (end-to-end) cryptogrpaphic key and then the outer (hop-by-hop) cryptographic key. The encryption also supports a mode for repair packets that only does the outer (hop-by-hop) encryption. The processes is as follows:

1. Form an RTP packet. If there are any header extensions, they **MUST** use [RFC8285].
2. If the packet is for repair mode data, skip to step 6.
3. Form a synthetic RTP packet with the following contents:
   - Header: The RTP header of the original packet with the following modifications:
     - The X bit is set to zero.
     - The header is truncated to remove any extensions (i.e., keep only the first 12 + 4 * CSRC count (CC) bytes of the header).
   - Payload: The RTP payload of the original packet (including padding when present).
4. Apply the inner cryptographic algorithm to the synthetic RTP packet from the previous step.
5. Replace the header of the protected RTP packet with the header of the original packet (to restore any header extensions and reset the X bit), and append an empty OHB (`0x00`) to the encrypted payload (with the authentication tag) obtained from step 4.

6. Apply the outer cryptographic algorithm to the RTP packet. If encrypting RTP header extensions hop-by-hop, then [RFC6904] **MUST** be used when encrypting the RTP packet using the outer cryptographic key.

When using Encrypted Key Transport (EKT) [EKT-SRTP], the EKTField comes after the SRTP packet, exactly like using EKT with any other SRTP transform.

5.2. Relaying a Packet

The Media Distributor has the part of the key for the outer (hop-by-hop) cryptographic algorithm, but it does not have the part of the key for the inner (end-to-end) cryptographic algorithm. The cryptographic algorithm and key used to decrypt a packet and any encrypted RTP header extensions would be the same as those used in the endpoint's outer algorithm and key.

In order to modify a packet, the Media Distributor decrypts the received packet, modifies the packet, updates the OHB with any modifications not already present in the OHB, and re-encrypts the packet using the outer (hop-by-hop) cryptographic key before transmitting using the following steps:

1. Apply the outer (hop-by-hop) cryptographic algorithm to decrypt the packet. If decrypting RTP header extensions hop-by-hop, then [RFC6904] **MUST** be used. Note that the RTP payload produced by this decryption operation contains the original encrypted payload with the tag from the inner transform and the OHB appended.
2. Make any desired changes to the fields that are allowed to be changed, i.e., PT, SEQ, and M. The Media Distributor **MAY** also make modifications to header extensions, without the need to reflect these changes in the OHB.
3. Reflect any changes to header fields in the OHB:
   - If the Media Distributor changed a field that is not already in the OHB, then it **MUST** add the original value of the field to the OHB. Note that this might result in an increase in the size of the OHB.
   - If the Media Distributor took a field that had previously been modified and reset to its original value, then it **SHOULD** drop the corresponding information from the OHB. Note that this might result in a decrease in the size of the OHB.
   - Otherwise, the Media Distributor **MUST NOT** modify the OHB.
4. Apply the outer (hop-by-hop) cryptographic algorithm to the packet. If the RTP sequence number has been modified, SRTP processing happens as defined in SRTP and will end up using the new sequence number. If encrypting RTP header extensions hop-by-hop, then [RFC6904] **MUST** be used.

In order to avoid nonce reuse, the cryptographic contexts used in steps 1 and 4 **MUST** use different, independent master keys. Note that this means that the key used for decryption by the MD **MUST** be different from the key used for re-encryption to the end recipient.

Note that if multiple MDs modify the same packet, then the first MD to alter a given header field is the one that adds it to the OHB. If a subsequent MD changes the value of a header field that has already been changed, then the original value will already be in the OHB, so no update to the OHB is required.

A Media Distributor that decrypts, modifies, and re-encrypts packets in this way **MUST** use an independent key for each recipient, and **MUST NOT** re-encrypt the packet using the sender's keys. If the Media Distributor decrypts and re-encrypts with the same key and salt, it will result in the reuse of a (key, nonce) pair, undermining the security of AES-GCM.

5.3. Decrypting a Packet

To decrypt a packet, the endpoint first decrypts and verifies using the outer (hop-by-hop) cryptographic key, then uses the OHB to reconstruct the original packet, which it decrypts and verifies with the inner (end-to-end) cryptographic key using the following steps:

1. Apply the outer cryptographic algorithm to the packet. If the integrity check does not pass, discard the packet. The result of this is referred to as the outer SRTP packet. If decrypting RTP header extensions hop-by-hop, then [RFC6904] **MUST** be used when decrypting the RTP packet using the outer cryptographic key.
2. If the packet is for repair mode data, skip the rest of the steps. Note that the packet that results from the repair algorithm will still have encrypted data that needs to be decrypted as specified by the repair algorithm sections.
3. Remove the inner authentication tag and the OHB from the end of the payload of the outer SRTP packet.
4. Form a new synthetic SRTP packet with:
   - Header = Received header, with the following modifications:
     - Header fields replaced with values from OHB (if any).
     - The X bit is set to zero.
     - The header is truncated to remove any extensions (i.e., keep only the first 12 + 4 * CC bytes of the header).
   - Payload is the encrypted payload from the outer SRTP packet (after the inner tag and OHB have been stripped).
   - Authentication tag is the inner authentication tag from the outer SRTP packet.
5. Apply the inner cryptographic algorithm to this synthetic SRTP packet. Note if the RTP sequence number was changed by the Media Distributor, the synthetic packet has the original sequence number. If the integrity check does not pass, discard the packet.

Once the packet has been successfully decrypted, the application needs to be careful about which information it uses to get the correct behavior. The application **MUST** use only the information found in the synthetic SRTP packet and **MUST NOT** use the other data that was in the outer SRTP packet with the following exceptions:

- The PT from the outer SRTP packet is used for normal matching to Session Description Protocol (SDP) and codec selection.
- The sequence number from the outer SRTP packet is used for normal RTP ordering.

The PT and sequence number from the inner SRTP packet can be used for collection of various statistics.

If the RTP header of the outer packet contains extensions, they **MAY** be used. However, because extensions are not protected end-to-end, implementations **SHOULD** reject an RTP packet containing headers that would require end-to-end protection.

6. RTCP Operations

Unlike RTP, which is encrypted both hop-by-hop and end-to-end using two separate cryptographic keys, RTCP is encrypted using only the outer (hop-by-hop) cryptographic key. The procedures for RTCP encryption are specified in [RFC3711], and this document introduces no additional steps.

7. Use with Other RTP Mechanisms

Media Distributors sometimes interact with RTP media packets sent by endpoints, e.g., to provide recovery or receive commands via dual-tone multi-frequency (DTMF) signaling. When media packets are encrypted end-to-end, these procedures require modification. (End-to-end interactions, including end-to-end recovery, are not affected by end-to-end encryption.)

Repair mechanisms, in general, will need to perform recovery on encrypted packets (double-encrypted when using this transform), since the Media Distributor does not have access to the plaintext of the packet, only an intermediate, E2E-encrypted form.

When the recovery mechanism calls for the recovery packet itself to be encrypted, it is encrypted with only the outer, hop-by-hop key. This allows a Media Distributor to generate recovery packets without having access to the inner, end-to-end keys. However, it also results in recovery packets being triple-encrypted, twice for the base transform, and once for the recovery protection.

7.1. RTP Retransmission (RTX)

When using RTX [RFC4588] with the double transform, the cached payloads **MUST** be the double-encrypted packets, i.e., the bits that are sent over the wire to the other side. When encrypting a retransmission packet, it **MUST** be encrypted like a packet in repair mode (i.e., with only the hop-by-hop key).

If the Media Distributor were to cache the inner, E2E-encrypted payload and retransmit it with an RTX original sequence number field prepended, then the modifications to the payload would cause the inner integrity check to fail at the receiver.

A typical RTX receiver would decrypt the packet, undo the RTX transformation, then process the resulting packet normally by using the steps in Section 5.3.

7.2. Redundant Audio Data (RED)

When using RED [RFC2198] with the double transform, the processing at the sender and receiver is the same as when using RED with any other SRTP transform.

The main difference between the double transform and any other transform is that in an intermediated environment, usage of RED must be end-to-end. A Media Distributor cannot synthesize RED packets, because it lacks access to the plaintext media payloads that are combined to form a RED payload.

Note that Flexible Forward Error Correction (Flex FEC) may often provide similar or better repair capabilities compared to RED. For most applications, Flex FEC is a better choice than RED; in particular, Flex FEC has modes in which the Media Distributor can synthesize recovery packets.

7.3. Forward Error Correction (FEC)

When using Flex FEC [RFC8627] with the double transform, repair packets **MUST** be constructed by first double-encrypting the packet, then performing FEC. Processing of repair packets proceeds in the opposite order, performing FEC recovery and then decrypting. This ensures that the original media is not revealed to the Media Distributor but, at the same time, allows the Media Distributor to repair media. When encrypting a packet that contains the Flex FEC data, which is already encrypted, it **MUST** be encrypted with only the outer, hop-by-hop transform.

The algorithm recommended in [WEBRTC-FEC] for repair of video is Flex FEC [RFC8627]. Note that for interoperability with WebRTC, [WEBRTC-FEC] recommends not using additional FEC-only "m=" lines in SDP for the repair packets.

7.4. DTMF

When DTMF is sent using the mechanism in [RFC4733], it is end-to-end encrypted; the relay cannot read it, so it cannot be used to control the relay. Other out-of-band methods to control the relay need to be used instead.

8. Recommended Inner and Outer Cryptographic Algorithms

This specification recommends and defines AES-GCM as both the inner and outer cryptographic algorithms, identified as DOUBLE_AEAD_AES_128_GCM_AEAD_AES_128_GCM and DOUBLE_AEAD_AES_256_GCM_AEAD_AES_256_GCM. These algorithms provide for authenticated

encryption and will consume additional processing time double-encrypting for hop-by-hop and end-to-end. However, the approach is secure and simple; thus, it is viewed as an acceptable trade-off in processing efficiency.

Note that names for the cryptographic transforms are of the form DOUBLE_(inner algorithm)_(outer algorithm).

While this document only defines a profile based on AES-GCM, it is possible for future documents to define further profiles with different inner and outer algorithms in this same framework. For example, if a new SRTP transform were defined that encrypts some or all of the RTP header, it would be reasonable for systems to have the option of using that for the outer algorithm. Similarly, if a new transform were defined that provided only integrity, that would also be reasonable to use for the outer transform as the payload data is already encrypted by the inner transform.

The AES-GCM cryptographic algorithm introduces an additional 16 octets to the length of the packet. When using AES-GCM for both the inner and outer cryptographic algorithms, the total additional length is 32 octets. The OHB will consume an additional 1-4 octets. Packets in repair mode will carry additional repair data, further increasing their size.

9. Security Considerations

This SRTP transform provides protection against two classes of attacker: a network attacker that knows neither the inner nor outer keys and a malicious MD that knows the outer key. Obviously, it provides no protections against an attacker that holds both the inner and outer keys.

The protections with regard to the network are the same as with the normal SRTP AES-GCM transforms. The major difference is that the double transforms are designed to work better in a group context. In such contexts, it is important to note that because these transforms are symmetric, they do not protect against attacks within the group. Any member of the group can generate valid SRTP packets for any SSRC in use by the group.

With regard to a malicious MD, the recipient can verify the integrity of the base header fields and confidentiality and integrity of the payload. The recipient has no assurance, however, of the integrity of the header extensions in the packet.

The main innovation of this transform relative to other SRTP transforms is that it allows a partly trusted MD to decrypt, modify, and re-encrypt a packet. When this is done, the cryptographic contexts used for decryption and re-encryption **MUST** use different, independent master keys. If the same context is used, the nonce formation rules for SRTP will cause the same key and nonce to be used with two different plaintexts, which substantially degrades the security of AES-GCM.

In other words, from the perspective of the MD, re-encrypting packets using this protocol will involve the same cryptographic operations as if it had established independent AES-GCM crypto contexts with the sender and the receiver. If the MD doesn't modify any header fields, then an MD that supports AES-GCM could be unused unmodified.

10. IANA Considerations

10.1. DTLS-SRTP

IANA has added the following protection profiles to the "DTLS-SRTP Protection Profiles" registry defined in [RFC5764].

Value	Profile	Reference
{0x00, 0x09}	DOUBLE_AEAD_AES_128_GCM_AEAD_AES_128_GCM	RFC 8723
{0x00, 0x0A}	DOUBLE_AEAD_AES_256_GCM_AEAD_AES_256_GCM	RFC 8723

Table 1: Updates to the DTLS-SRTP Protection Profiles Registry

The SRTP transform parameters for each of these protection profiles are:

DOUBLE_AEAD_AES_128_GCM_AEAD_AES_128_GCM	
cipher:	AES_128_GCM then AES_128_GCM
cipher_key_length:	256 bits
cipher_salt_length:	192 bits
aead_auth_tag_length:	256 bits
auth_function:	NULL
auth_key_length:	N/A
auth_tag_length:	N/A
maximum lifetime:	at most 2^31 SRTCP packets and at most 2^48 SRTP packets
DOUBLE_AEAD_AES_256_GCM_AEAD_AES_256_GCM	
cipher:	AES_256_GCM then AES_256_GCM
cipher_key_length:	512 bits
cipher_salt_length:	192 bits
aead_auth_tag_length:	256 bits
auth_function:	NULL
auth_key_length:	N/A

auth_tag_length:	N/A
maximum lifetime:	at most 2^31 SRTCP packets and at most 2^48 SRTP packets

Table 2: SRTP Transform Parameters

The first half of the key and salt is used for the inner (end-to-end) algorithm and the second half is used for the outer (hop-by-hop) algorithm.

11. References

11.1. Normative References

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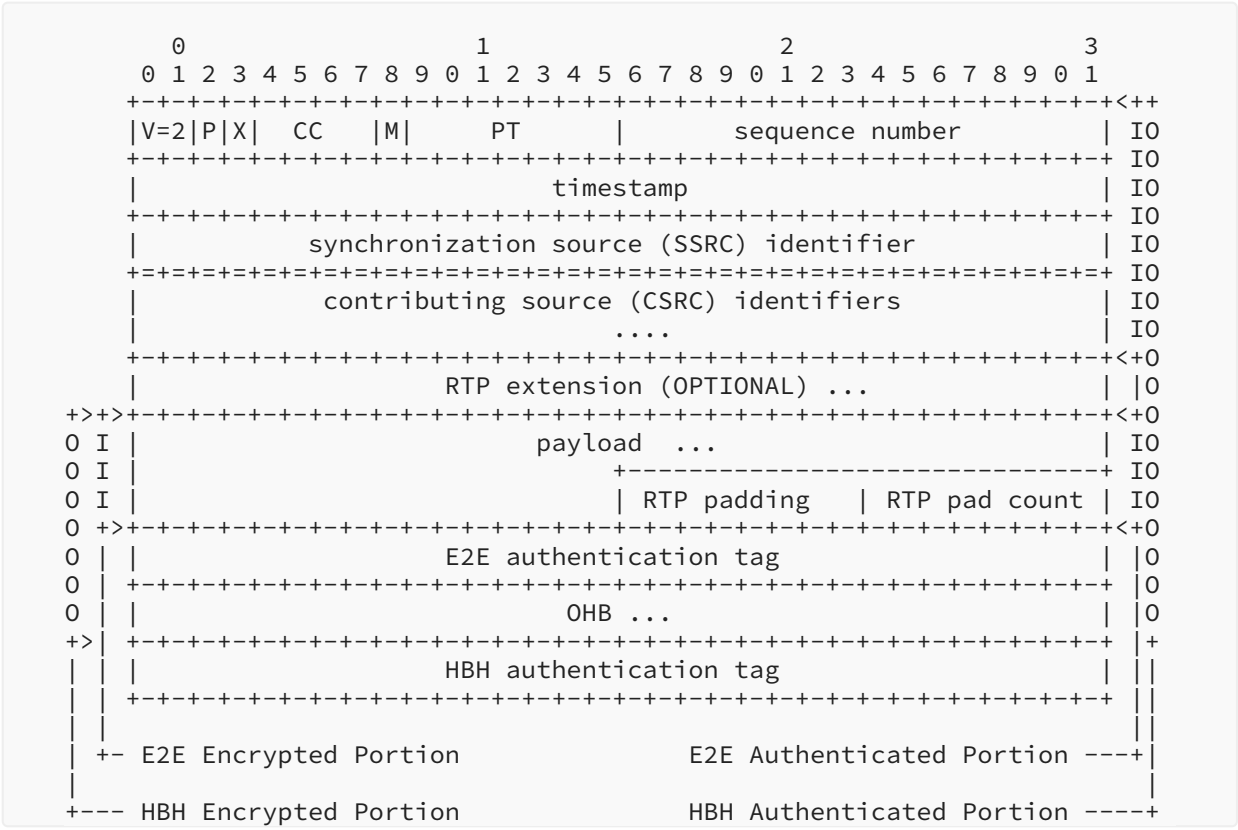
Appendix A. Encryption Overview

The following figure shows a double-encrypted SRTP packet. The sides indicate the parts of the packet that are encrypted and authenticated by the hop-by-hop and end-to-end operations.



Acknowledgments

Thank you to Alex Gouaillard, David Benham, Magnus Westerlund, Nils Ohlmeier, Paul Jones, Roni Even, and Suhas Nandakumar for reviews and improvements to this specification. In addition, thank you to Sergio Garcia Murillo, who proposed the change of transporting the OHB information in the RTP payload instead of the RTP header.

Authors' Addresses

Cullen Jennings
Cisco Systems
Email: fluffy@iii.ca

Paul E. Jones
Cisco Systems
Email: paulej@packetizer.com

Richard Barnes
Cisco Systems
Email: rlb@ipv.sx

Adam Roach
Mozilla
Email: adam@nostrum.com