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IP Traffic Flow Security Using Aggregation and Fragmentation draft-ietf-ipsecme-iptfs-04

Abstract

This document describes a mechanism to enhance IPsec traffic flow security by adding traffic flow confidentiality to encrypted IP encapsulated traffic. Traffic flow confidentiality is provided by obscuring the size and frequency of IP traffic using a fixed-sized, constant-send-rate IPsec tunnel. The solution allows for congestion control as well as non-constant send-rate usage.

Status of This Memo

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1. Introduction

Traffic Analysis ([RFC4301], [AppCrypt]) is the act of extracting information about data being sent through a network. While one may directly obscure the data through the use of encryption [RFC4303], the traffic pattern itself exposes information due to variations in it's shape and timing ([I-D.iab-wire-image], [AppCrypt]). Hiding the size and frequency of traffic is referred to as Traffic Flow Confidentiality (TFC) per [RFC4303].

[RFC4303] provides for TFC by allowing padding to be added to encrypted IP packets and allowing for transmission of all-pad packets (indicated using protocol 59). This method has the major limitation that it can significantly under-utilize the available bandwidth.

The IP-TFS solution provides for full TFC without the aforementioned bandwidth limitation. This is accomplished by using a constant-send-rate IPsec [RFC4303] tunnel with fixed-sized encapsulating packets; however, these fixed-sized packets can contain partial, whole or multiple IP packets to maximize the bandwidth of the tunnel. A non-constant send-rate is allowed, but the confidentiality properties of its use are outside the scope of this document.

For a comparison of the overhead of IP-TFS with the RFC4303 prescribed TFC solution see Appendix C.

Additionally, IP-TFS provides for dealing with network congestion [RFC2914]. This is important for when the IP-TFS user is not in full control of the domain through which the IP-TFS tunnel path flows.

1.1. Terminology & Concepts

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 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

This document assumes familiarity with IP security concepts described in [RFC4301].

2. The IP-TFS Tunnel

As mentioned in Section 1 IP-TFS utilizes an IPsec [RFC4303] tunnel (SA) as it's transport. To provide for full TFC, fixed-sized encapsulating packets are sent at a constant rate on the tunnel.

The primary input to the tunnel algorithm is the requested bandwidth used by the tunnel. Two values are then required to provide for this bandwidth, the fixed size of the encapsulating packets, and rate at which to send them.

The fixed packet size may either be specified manually or can be determined through the use of Path MTU discovery [RFC1191] and [RFC8201].

Given the encapsulating packet size and the requested tunnel used bandwidth, the corresponding packet send rate can be calculated. The packet send rate is the requested bandwidth divided by the size of the encapsulating packet.

The egress of the IP-TFS tunnel MUST allow for and expect the ingress (sending) side of the IP-TFS tunnel to vary the size and rate of sent encapsulating packets, unless constrained by other policy.

2.1. Tunnel Content

As previously mentioned, one issue with the TFC padding solution in [RFC4303] is the large amount of wasted bandwidth as only one IP packet can be sent per encapsulating packet. In order to maximize bandwidth IP-TFS breaks this one-to-one association.

IP-TFS aggregates as well as fragments the inner IP traffic flow into fixed-sized encapsulating IPsec tunnel packets. Padding is only added to the the tunnel packets if there is no data available to be sent at the time of tunnel packet transmission, or if fragmentation has been disabled by the receiver.

This is accomplished using a new Encapsulating Security Payload (ESP, [RFC4303]) type which is identified by the number AGGFRAG_PAYLOAD (Section 6.1).

Other non-IP-TFS uses of this aggregation and fragmentation encapsulation have been identified, such as increased performance through packet aggregation, as well as handling MTU issues using fragmentation. These uses are not defined here, but are also not restricted by this document.

2.2. Payload Content

The AGGFRAG_PAYLOAD payload content defined in this document is comprised of a 4 or 24 octet header followed by either a partial, a full or multiple partial or full data blocks. The following diagram illustrates this payload within the ESP packet. See Section 6.1 for the exact formats of the AGGFRAG_PAYLOAD payload.

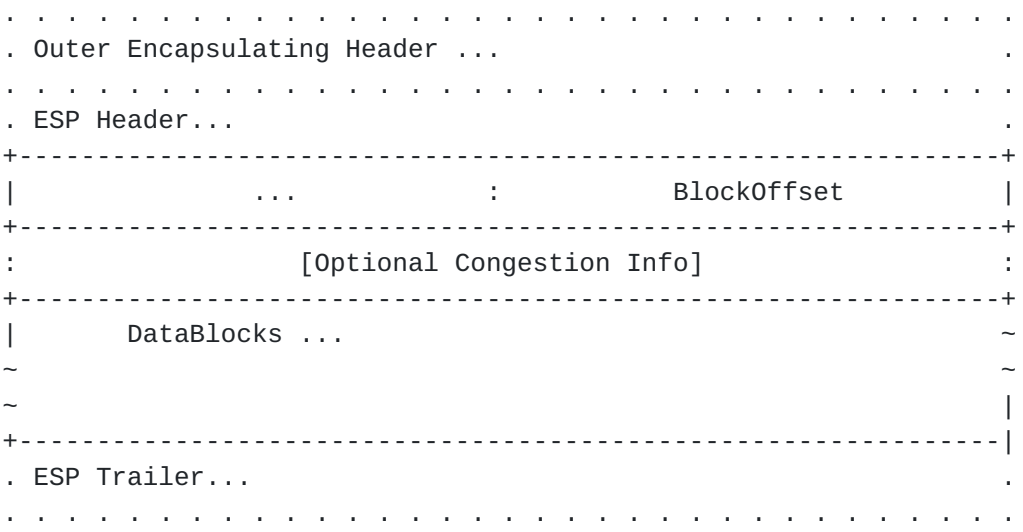


Figure 1: Layout of an IP-TFS IPsec Packet

The "BlockOffset" value is either zero or some offset into or past the end of the "DataBlocks" data.

If the "BlockOffset" value is zero it means that the "DataBlocks" data begins with a new data block.

Conversely, if the "BlockOffset" value is non-zero it points to the start of the new data block, and the initial "DataBlocks" data belongs to a previous data block that is still being re-assembled.

The "BlockOffset" can point past the end of the "DataBlocks" data which indicates that the next data block occurs in a subsequent encapsulating packet.

Having the "BlockOffset" always point at the next available data block allows for recovering the next full inner packet in the presence of outer encapsulating packet loss.

An example IP-TFS packet flow can be found in Appendix A.

2.2.1. Data Blocks

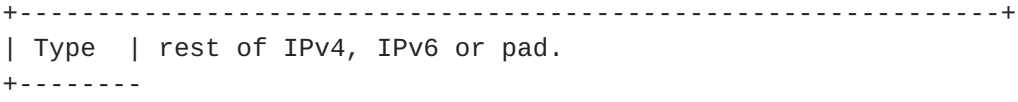


Figure 2: Layout of IP-TFS data block

A data block is defined by a 4-bit type code followed by the data block data. The type values have been carefully chosen to coincide with the IPv4/IPv6 version field values so that no per-data block type overhead is required to encapsulate an IP packet. Likewise, the length of the data block is extracted from the encapsulated IPv4 or IPv6 packet's length field.

2.2.2. No Implicit End Padding Required

It's worth noting that since a data block type is identified by its first octet there is never a need for an implicit pad at the end of an encapsulating packet. Even when the start of a data block occurs near the end of a encapsulating packet such that there is no room for the length field of the encapsulated header to be included in the current encapsulating packet, the fact that the length comes at a known location and is guaranteed to be present is enough to fetch the length field from the subsequent encapsulating packet payload. Only when there is no data to encapsulated is end padding required, and then an explicit "Pad Data Block" would be used to identify the padding.

2.2.3. Fragmentation, Sequence Numbers and All-Pad Payloads

In order for a receiver to be able to reassemble fragmented inner-packets, the sender MUST send the inner-packet fragments back-to-back in the logical outer packet stream (i.e., using consecutive ESP sequence numbers). However, the sender is allowed to insert "all-pad" payloads (i.e., payloads with a "BlockOffset" of zero and a single pad "DataBlock") in between the packets carrying the inner-packet fragment payloads. This possible interleaving of all-pad payloads allows the sender to always be able to send a tunnel packet, regardless of the encapsulation computational requirements.

When a receiver is reassembling an inner-packet, and it receives an "all-pad" payload, it increments the expected sequence number that the next inner-packet fragment is expected to arrive in.

2.2.4. Empty Payload

In order to support reporting of congestion control information (described later) on a non-AGGFRAG_PAYLOAD enabled SA, IP-TFS allows for the sending of an AGGFRAG_PAYLOAD payload with no data blocks (i.e., the ESP payload length is equal to the AGGFRAG_PAYLOAD header length). This special payload is called an empty payload.

2.2.5. IP Header Value Mapping

[RFC4301] provides some direction on when and how to map various values from an inner IP header to the outer encapsulating header, namely the Don't-Fragment (DF) bit ([RFC0791] and [RFC8200]), the Differentiated Services (DS) field [RFC2474] and the Explicit Congestion Notification (ECN) field [RFC3168]. Unlike [RFC4301], IP-TFS may and often will be encapsulating more than one IP packet per ESP packet. To deal with this, these mappings are restricted further. In particular IP-TFS never maps the inner DF bit as it is unrelated to the IP-TFS tunnel functionality; IP-TFS never IP fragments the inner packets and the inner packets will not affect the fragmentation of the outer encapsulation packets. Likewise, the ECN value need not be mapped as any congestion related to the constant-send-rate IP-TFS tunnel is unrelated (by design!) to the inner traffic flow. Finally, by default the DS field SHOULD NOT be copied although an implementation MAY choose to allow for configuration to override this behavior. An implementation SHOULD also allow the DS value to be set by configuration.

2.3. Exclusive SA Use

It is not the intention of this specification to allow for mixed use of an AGGFRAG_PAYLOAD enabled SA. In other words, an SA that has AGGFRAG_PAYLOAD enabled MUST NOT have non-AGGFRAG_PAYLOAD payloads such as IP (IP protocol 4), TCP transport (IP protocol 6), or ESP pad packets (protocol 59) intermixed with non-empty AGGFRAG_PAYLOAD payloads. While it's possible to envision making the algorithm work in the presence of sequence number skips in the AGGFRAG_PAYLOAD payload stream, the added complexity is not deemed worthwhile. Other IPsec uses can configure and use their own SAs.

2.4. Modes of Operation

Just as with normal IPsec/ESP tunnels, IP-TFS tunnels are unidirectional. Bidirectional IP-TFS functionality is achieved by setting up 2 IP-TFS tunnels, one in either direction.

An IP-TFS tunnel can operate in 2 modes, a non-congestion controlled mode and congestion controlled mode.

2.4.1. Non-Congestion Controlled Mode

In the non-congestion controlled mode IP-TFS sends fixed-sized packets at a constant rate. The packet send rate is constant and is not automatically adjusted regardless of any network congestion (e.g., packet loss).

For similar reasons as given in [RFC7510] the non-congestion controlled mode should only be used where the user has full administrative control over the path the tunnel will take. This is required so the user can guarantee the bandwidth and also be sure as to not be negatively affecting network congestion [RFC2914]. In this case packet loss should be reported to the administrator (e.g., via syslog, YANG notification, SNMP traps, etc) so that any failures due to a lack of bandwidth can be corrected.

2.4.2. Congestion Controlled Mode

With the congestion controlled mode, IP-TFS adapts to network congestion by lowering the packet send rate to accommodate the congestion, as well as raising the rate when congestion subsides. Since overhead is per packet, by allowing for maximal fixed-size packets and varying the send rate transport overhead is minimized.

The output of the congestion control algorithm will adjust the rate at which the ingress sends packets. While this document does not require a specific congestion control algorithm, best current practice RECOMMENDS that the algorithm conform to [RFC5348]. Congestion control principles are documented in [RFC2914] as well. An example of an implementation of the [RFC5348] algorithm which matches the requirements of IP-TFS (i.e., designed for fixed-size packet and send rate varied based on congestion) is documented in [RFC4342].

The required inputs for the TCP friendly rate control algorithm described in [RFC5348] are the receiver's loss event rate and the sender's estimated round-trip time (RTT). These values are provided by IP-TFS using the congestion information header fields described in Section 3. In particular these values are sufficient to implement the algorithm described in [RFC5348].

At a minimum, the congestion information must be sent, from the receiver and from the sender, at least once per RTT. Prior to establishing an RTT the information SHOULD be sent constantly from the sender and the receiver so that an RTT estimate can be established. The lack of receiving this information over multiple consecutive RTT intervals should be considered a congestion event that causes the sender to adjust it's sending rate lower. For

example, [RFC4342] calls this the "no feedback timeout" and it is equal to 4 RTT intervals. When a "no feedback timeout" has occurred [RFC4342] halves the sending rate.

An implementation MAY choose to always include the congestion information in it's IP-TFS payload header if sending on an IP-TFS enabled SA. Since IP-TFS normally will operate with a large packet size, the congestion information should represent a small portion of the available tunnel bandwidth. An implementation choosing to always send the data MAY also choose to only update the "LossEventRate" and "RTT" header field values it sends every "RTT" though.

When an implementation is choosing a congestion control algorithm (or a selection of algorithms) one should remember that IP-TFS is not providing for reliable delivery of IP traffic, and so per packet ACKs are not required and are not provided.

It's worth noting that the variable send-rate of a congestion controlled IP-TFS tunnel, is not private; however, this send-rate is being driven by network congestion, and as long as the encapsulated (inner) traffic flow shape and timing are not directly affecting the (outer) network congestion, the variations in the tunnel rate will not weaken the provided inner traffic flow confidentiality.

2.4.2.1. Circuit Breakers

In additional to congestion control, implementations MAY choose to define and implement circuit breakers [RFC8084] as a recovery method of last resort. Enabling circuit breakers is also a reason a user may wish to enable congestion information reports even when using the non-congestion controlled mode of operation. The definition of circuit breakers are outside the scope of this document.

3. Congestion Information

In order to support the congestion control mode, the sender needs to know the loss event rate and also be able to approximate the RTT ([RFC5348]). In order to obtain these values the receiver sends congestion control information on it's SA back to the sender. Thus, in order to support congestion control the receiver must have a paired SA back to the sender (this is always the case when the tunnel was created using IKEv2). If the SA back to the sender is a non-AGGFRAG_PAYLOAD enabled SA then an AGGFRAG_PAYLOAD empty payload (i.e., header only) is used to convey the information.

In order to calculate a loss event rate compatible with [RFC5348], the receiver needs to have a round-trip time estimate. Thus the

sender communicates this estimate in the "RTT" header field. On startup this value will be zero as no RTT estimate is yet known.

In order for the sender to estimate it's "RTT" value, the sender places a timestamp value in the "TVal" header field. On first receipt of this "TVal", the receiver records the new "TVal" value along with the time it arrived locally, subsequent receipt of the same "TVal" MUST not update the recorded time. When the receiver sends it's CC header it places this latest recorded value in the "TEcho" header field, along with 2 delay values, "Echo Delay" and "Transmit Delay". The "Echo Delay" value is the time delta from the recorded arrival time of "TVal" and the current clock in microseconds. The second value, "Transmit Delay", is the receiver's current transmission delay on the tunnel (i.e., the average time between sending packets on it's half of the IP-TFS tunnel). When the sender receives back it's "TVal" in the "TEcho" header field it calculates 2 RTT estimates. The first is the actual delay found by subtracting the "TEcho" value from it's current clock and then subtracting "Echo Delay" as well. The second RTT estimate is found by adding the received "Transmit Delay" header value to the senders own transmission delay (i.e., the average time between sending packets on it's half of the IP-TFS tunnel). The larger of these 2 RTT estimates SHOULD be used as the "RTT" value. The two estimates are required to handle different combinations of faster or slower tunnel packet paths with faster or slower fixed tunnel rates. Choosing the larger of the two values guarantees that the "RTT" is never considered faster than the aggregate transmission delay based on the IP-TFS tunnel rate (the second estimate), as well as never being considered faster than the actual RTT along the tunnel packet path (the first estimate).

The receiver also calculates, and communicates in the "LossEventRate" header field, the loss event rate for use by the sender. This is slightly different from [RFC4342] which periodically sends all the loss interval data back to the sender so that it can do the calculation. See Appendix B for a suggested way to calculate the loss event rate value. Initially this value will be zero (indicating no loss) until enough data has been collected by the receiver to update it.

3.1. ECN Support

In additional to normal packet loss information IP-TFS supports use of the ECN bits in the encapsulating IP header [RFC3168] for identifying congestion. If ECN use is enabled and a packet arrives at the egress endpoint with the Congestion Experienced (CE) value set, then the receiver considers that packet as being dropped, although it does not drop it. The receiver MUST set the E bit in any

AGGFRAG_PAYLOAD payload header containing a "LossEventRate" value derived from a CE value being considered.

As noted in [RFC3168] the ECN bits are not protected by IPsec and thus may constitute a covert channel. For this reason ECN use SHOULD NOT be enabled by default.

4. Configuration

IP-TFS is meant to be deployable with a minimal amount of configuration. All IP-TFS specific configuration should be able to be specified at the unidirectional tunnel ingress (sending) side. It is intended that non-IKEv2 operation is supported, at least, with local static configuration.

4.1. Bandwidth

Bandwidth is a local configuration option. For non-congestion controlled mode the bandwidth SHOULD be configured. For congestion controlled mode one can configure the bandwidth or have no configuration and let congestion control discover the maximum bandwidth available. No standardized configuration method is required.

4.2. Fixed Packet Size

The fixed packet size to be used for the tunnel encapsulation packets can be configured manually or can be automatically determined using Path MTU discovery (see [RFC1191] and [RFC8201]). No standardized configuration method is required.

4.3. Congestion Control

Congestion control is a local configuration option. No standardized configuration method is required.

5. IKEv2

5.1. USE_AGGFRAG Notification Message

As mentioned previously IP-TFS tunnels utilize ESP payloads of type AGGFRAG_PAYLOAD.

When using IKEv2, a new "USE_AGGFRAG" Notification Message is used to enable use of the AGGFRAG_PAYLOAD payload on a child SA pair. The method used is similar to how USE_TRANSPORT_MODE is negotiated, as described in [RFC7296].

To request using the AGGFRAG_PAYLOAD payload on the Child SA pair, the initiator includes the USE_AGGFRAG notification in an SA payload requesting a new Child SA (either during the initial IKE_AUTH or during non-rekeying CREATE_CHILD_SA exchanges). If the request is accepted then response MUST also include a notification of type USE_AGGFRAG. If the responder declines the request the child SA will be established without AGGFRAG_PAYLOAD payload use enabled. If this is unacceptable to the initiator, the initiator MUST delete the child SA.

The USE_AGGFRAG notification MUST NOT be sent, and MUST be ignored, during a CREATE_CHILD_SA rekeying exchange as it is not allowed to change use of the AGGFRAG_PAYLOAD payload type during rekeying.

The USE_AGGFRAG notification contains a 1 octet payload of flags that specify any requirements from the sender of the message. If any requirement flags are not understood or cannot be supported by the receiver then the receiver should not enable use of AGGFRAG_PAYLOAD payload type (either by not responding with the USE_AGGFRAG notification, or in the case of the initiator, by deleting the child SA if the now established non-AGGFRAG_PAYLOAD using SA is unacceptable).

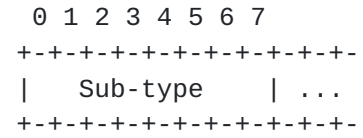
The notification type and payload flag values are defined in Section 6.1.4.

6. Packet and Data Formats

6.1. AGGFRAG_PAYLOAD Payload

ESP Payload Type: 0x5

An IP-TFS payload is identified by the ESP payload type AGGFRAG_PAYLOAD which has the value 0x5. The first octet of this payload indicates the format of the remaining payload data.

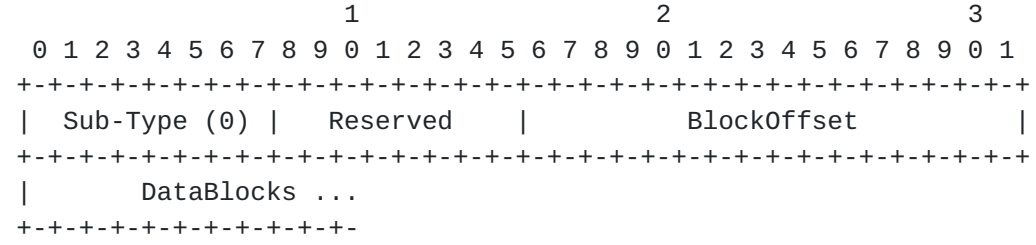


Sub-type:
An 8 bit value indicating the payload format.

This specification defines 2 payload sub-types. These payload formats are defined in the following sections.

6.1.1. Non-Congestion Control AGGFRAG_PAYLOAD Payload Format

The non-congestion control AGGFRAG_PAYLOAD payload is comprised of a 4 octet header followed by a variable amount of "DataBlocks" data as shown below.



Sub-type:
An octet indicating the payload format. For this non-congestion control format, the value is 0.

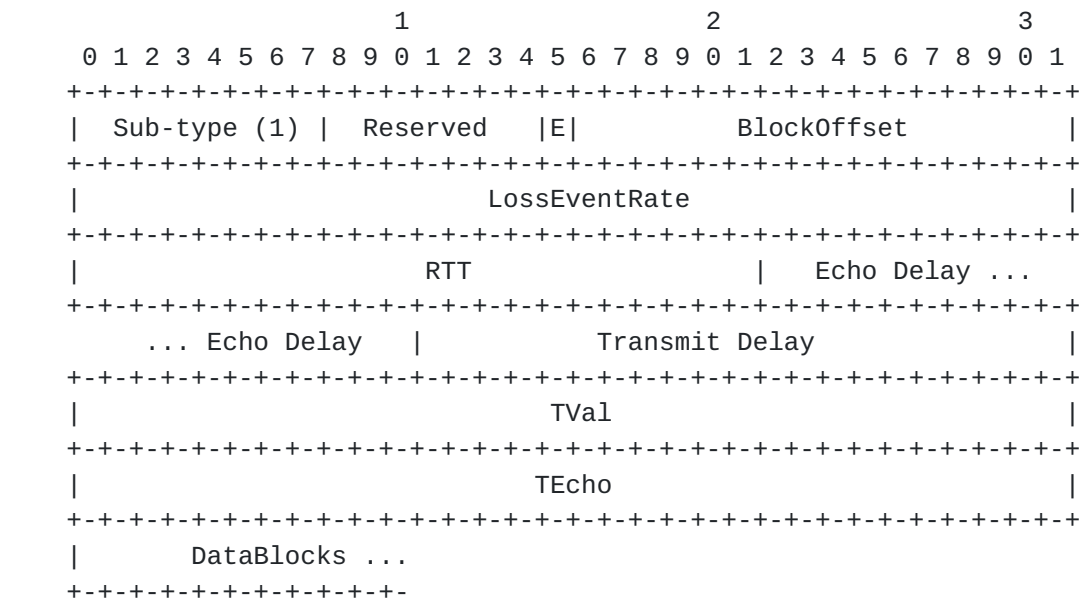
Reserved:
An octet set to 0 on generation, and ignored on receipt.

BlockOffset:
A 16 bit unsigned integer counting the number of octets of "DataBlocks" data before the start of a new data block. "BlockOffset" can count past the end of the "DataBlocks" data in which case all the "DataBlocks" data belongs to the previous data block being re-assembled. If the "BlockOffset" extends into subsequent packets it continues to only count subsequent "DataBlocks" data (i.e., it does not count subsequent packets non-"DataBlocks" octets).

DataBlocks:
Variable number of octets that begins with the start of a data block, or the continuation of a previous data block, followed by zero or more additional data blocks.

6.1.2. Congestion Control AGGFRAG_PAYLOAD Payload Format

The congestion control AGGFRAG_PAYLOAD payload is comprised of a 24 octet header followed by a variable amount of "DataBlocks" data as shown below.



Sub-type:
An octet indicating the payload format. For this congestion control format, the value is 1.

Reserved:
A 7 bit field set to 0 on generation, and ignored on receipt.

E:
A 1 bit value if set indicates that Congestion Experienced (CE) ECN bits were received and used in deriving the reported "LossEventRate".

BlockOffset:
The same value as the non-congestion controlled payload format value.

LossEventRate:
A 32 bit value specifying the inverse of the current loss event rate as calculated by the receiver. A value of zero indicates no loss. Otherwise the loss event rate is "1/LossEventRate".

RTT:
A 22 bit value specifying the sender's current round-trip time estimate in microseconds. The value MAY be zero prior to the sender having calculated a round-trip time estimate. The value SHOULD be set to zero on non-AGGFRAG_PAYLOAD enabled SAs. If the value is equal to or larger than "0x3FFFFF" it MUST be set to "0x3FFFFF".

Echo Delay:

A 21 bit value specifying the delay in microseconds incurred between the receiver first receiving the "TVal" value which it is sending back in "TEcho". If the value is equal to or larger than "0x1FFFFF" it MUST be set to "0x1FFFFF".

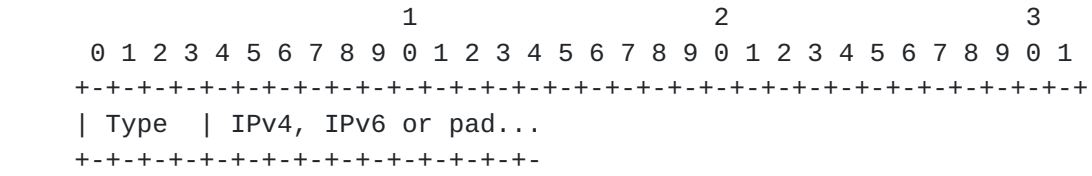
Transmit Delay:
A 21 bit value specifying the transmission delay in microseconds. This is the fixed (or average) delay on the receiver between it sending packets on the IPTFS tunnel. If the value is equal to or larger than "0x1FFFFF" it MUST be set to "0x1FFFFF".

TVal:
An opaque 32 bit value that will be echoed back by the receiver in later packets in the "TEcho" field, along with an "Echo Delay" value of how long that echo took.

TEcho:
The opaque 32 bit value from a received packet's "TVal" field. The received "TVal" is placed in "TEcho" along with an "Echo Delay" value indicating how long it has been since receiving the "TVal" value.

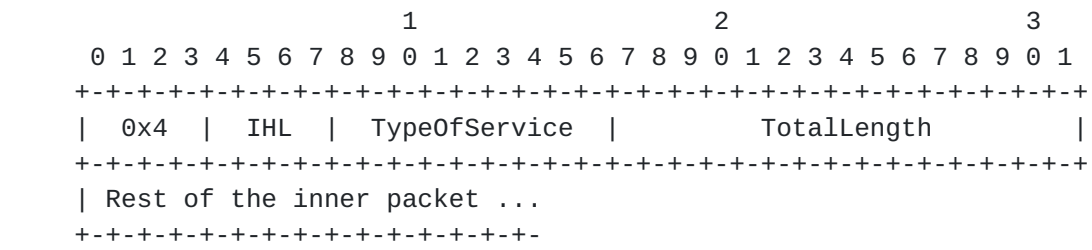
DataBlocks:
Variable number of octets that begins with the start of a data block, or the continuation of a previous data block, followed by zero or more additional data blocks. For the special case of sending congestion control information on an non-IP-TFS enabled SA this value MUST be empty (i.e., be zero octets long).

6.1.3. Data Blocks



Type:
A 4 bit field where 0x0 identifies a pad data block, 0x4 indicates an IPv4 data block, and 0x6 indicates an IPv6 data block.

6.1.3.1. IPv4 Data Block

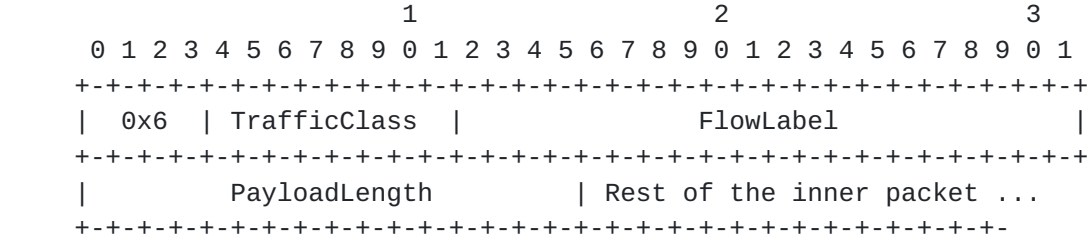


These values are the actual values within the encapsulated IPv4 header. In other words, the start of this data block is the start of the encapsulated IP packet.

Type:
A 4 bit value of 0x4 indicating IPv4 (i.e., first nibble of the IPv4 packet).

TotalLength:
The 16 bit unsigned integer "Total Length" field of the IPv4 inner packet.

6.1.3.2. IPv6 Data Block

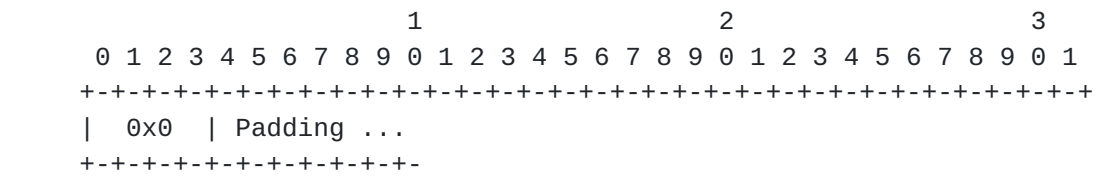


These values are the actual values within the encapsulated IPv6 header. In other words, the start of this data block is the start of the encapsulated IP packet.

Type:
A 4 bit value of 0x6 indicating IPv6 (i.e., first nibble of the IPv6 packet).

PayloadLength:
The 16 bit unsigned integer "Payload Length" field of the inner IPv6 inner packet.

6.1.3.3. Pad Data Block



Type:
A 4 bit value of 0x0 indicating a padding data block.

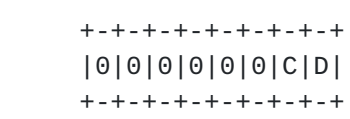
Padding:
extends to end of the encapsulating packet.

6.1.4. IKEv2 USE_AGGFRAG Notification Message

As discussed in Section 5.1 a notification message USE_AGGFRAG is used to negotiate use of the ESP AGGFRAG_PAYLOAD payload type.

The USE_AGGFRAG Notification Message State Type is (TBD2).

The notification payload contains 1 octet of requirement flags. There are currently 2 requirement flags defined. This may be revised by later specifications.



0:
6 bits - reserved, MUST be zero on send, unless defined by later specifications.

C:
Congestion Control bit. If set, then the sender is requiring that congestion control information MUST be returned to it periodically as defined in Section 3.

D:
Don't Fragment bit, if set indicates the sender of the notify message does not support receiving packet fragments (i.e., inner packets MUST be sent using a single "Data Block"). This value only applies to what the sender is capable of receiving; the sender MAY still send packet fragments unless similarly restricted by the receiver in it's USE_AGGFRAG notification.

7. IANA Considerations

7.1. AGGFRAG_PAYLOAD Sub-Type Registry

This document requests IANA create a registry called "AGGFRAG_PAYLOAD Sub-Type Registry" under a new category named "ESP AGGFRAG_PAYLOAD Parameters". The registration policy for this registry is "Standards Action" ([RFC8126] and [RFC7120]).

Name:
AGGFRAG_PAYLOAD Sub-Type Registry

Description:
AGGFRAG_PAYLOAD Payload Formats.

Reference:
This document

This initial content for this registry is as follows:

Sub-Type	Name	Reference
0	Non-Congestion Control Format	This document
1	Congestion Control Format	This document
3-255	Reserved	

7.2. USE_AGGFRAG Notify Message Status Type

This document requests a status type USE_AGGFRAG be allocated from the "IKEv2 Notify Message Types - Status Types" registry.

Value:
TBD2

Name:
USE_AGGFRAG

Reference:
This document

8. Security Considerations

This document describes a mechanism to add Traffic Flow Confidentiality to IP traffic. Use of this mechanism is expected to increase the security of the traffic being transported. Other than the additional security afforded by using this mechanism, IP-TFS utilizes the security protocols [RFC4303] and [RFC7296] and so their security considerations apply to IP-TFS as well.

As noted previously in Section 2.4.2, for TFC to be fully maintained the encapsulated traffic flow should not be affecting network congestion in a predictable way, and if it would be then non-congestion controlled mode use should be considered instead.

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Appendix A. Example Of An Encapsulated IP Packet Flow

Below an example inner IP packet flow within the encapsulating tunnel packet stream is shown. Notice how encapsulated IP packets can start and end anywhere, and more than one or less than 1 may occur in a single encapsulating packet.

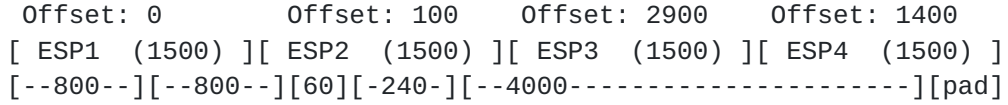

Figure 3: Inner and Outer Packet Flow

The encapsulated IP packet flow (lengths include IP header and payload) is as follows: an 800 octet packet, an 800 octet packet, a 60 octet packet, a 240 octet packet, a 4000 octet packet.

The "BlockOffset" values in the 4 IP-TFS payload headers for this packet flow would thus be: 0, 100, 2900, 1400 respectively. The first encapsulating packet ESP1 has a zero "BlockOffset" which points at the IP data block immediately following the IP-TFS header. The following packet ESP2s "BlockOffset" points inward 100 octets to the start of the 60 octet data block. The third encapsulating packet ESP3 contains the middle portion of the 4000 octet data block so the offset points past its end and into the forth encapsulating packet. The fourth packet ESP4s offset is 1400 pointing at the padding which follows the completion of the continued 4000 octet packet.

Appendix B. A Send and Loss Event Rate Calculation

The current best practice indicates that congestion control SHOULD be done in a TCP friendly way. A TCP friendly congestion control algorithm is described in [RFC5348]. For this IP-TFS use case (as with [RFC4342]) the (fixed) packet size is used as the segment size for the algorithm. The main formula in the algorithm for the send rate is then as follows:

$$X = \frac{1}{R * (\sqrt{2*p/3} + 12*\sqrt{3*p/8}*p*(1+32*p^2))}$$

Where "X" is the send rate in packets per second, "R" is the round trip time estimate and "p" is the loss event rate (the inverse of which is provided by the receiver).

In addition the algorithm in [RFC5348] also uses an "X_recv" value (the receiver's receive rate). For IP-TFS one MAY set this value according to the sender's current tunnel send-rate ("X").

The IP-TFS receiver, having the RTT estimate from the sender can use the same method as described in [RFC5348] and [RFC4342] to collect the loss intervals and calculate the loss event rate value using the weighted average as indicated. The receiver communicates the inverse of this value back to the sender in the AGGFRAG_PAYLOAD payload header field "LossEventRate".

The IP-TFS sender now has both the "R" and "p" values and can calculate the correct sending rate. If following [RFC5348] the sender SHOULD also use the slow start mechanism described therein when the IP-TFS SA is first established.

Appendix C. Comparisons of IP-TFS

C.1. Comparing Overhead

C.1.1. IP-TFS Overhead

The overhead of IP-TFS is 40 bytes per outer packet. Therefore the octet overhead per inner packet is 40 divided by the number of outer packets required (fractional allowed). The overhead as a percentage of inner packet size is a constant based on the Outer MTU size.

```
OH = 40 / Outer Payload Size / Inner Packet Size
OH % of Inner Packet Size = 100 * OH / Inner Packet Size
OH % of Inner Packet Size = 4000 / Outer Payload Size
```

Type	IP-TFS	IP-TFS	IP-TFS
MTU	576	1500	9000
PSize	536	1460	8960
40	7.46%	2.74%	0.45%
576	7.46%	2.74%	0.45%
1500	7.46%	2.74%	0.45%
9000	7.46%	2.74%	0.45%

Figure 4: IP-TFS Overhead as Percentage of Inner Packet Size

C.1.2. ESP with Padding Overhead

The overhead per inner packet for constant-send-rate padded ESP (i.e., traditional IPsec TFC) is 36 octets plus any padding, unless fragmentation is required.

When fragmentation of the inner packet is required to fit in the outer IPsec packet, overhead is the number of outer packets required to carry the fragmented inner packet times both the inner IP overhead (20) and the outer packet overhead (36) minus the initial inner IP overhead plus any required tail padding in the last encapsulation packet. The required tail padding is the number of required packets times the difference of the Outer Payload Size and the IP Overhead minus the Inner Payload Size. So:

```
Inner Paylaod Size = IP Packet Size - IP Overhead
Outer Payload Size = MTU - IPsec Overhead

              Inner Payload Size
NF0 = ---------------------------------
       Outer Payload Size - IP Overhead

NF = CEILING(NF0)

OH = NF * (IP Overhead + IPsec Overhead)
     - IP Overhead
     + NF * (Outer Payload Size - IP Overhead)
     - Inner Payload Size

OH = NF * (IPsec Overhead + Outer Payload Size)
     - (IP Overhead + Inner Payload Size)

OH = NF * (IPsec Overhead + Outer Payload Size)
     - Inner Packet Size
```

C.2. Overhead Comparison

The following tables collect the overhead values for some common L3 MTU sizes in order to compare them. The first table is the number of octets of overhead for a given L3 MTU sized packet. The second table is the percentage of overhead in the same MTU sized packet.

Type	ESP+Pad	ESP+Pad	ESP+Pad	IP-TFS	IP-TFS	IP-TFS
L3 MTU	576	1500	9000	576	1500	9000
PSize	540	1464	8964	536	1460	8960
40	500	1424	8924	3.0	1.1	0.2
128	412	1336	8836	9.6	3.5	0.6
256	284	1208	8708	19.1	7.0	1.1
536	4	928	8428	40.0	14.7	2.4
576	576	888	8388	43.0	15.8	2.6
1460	268	4	7504	109.0	40.0	6.5
1500	228	1500	7464	111.9	41.1	6.7
8960	1408	1540	4	668.7	245.5	40.0
9000	1368	1500	9000	671.6	246.6	40.2

Figure 5: Overhead comparison in octets

Type	ESP+Pad	ESP+Pad	ESP+Pad	IP-TFS	IP-TFS	IP-TFS
MTU	576	1500	9000	576	1500	9000
PSize	540	1464	8964	536	1460	8960
40	1250.0%	3560.0%	22310.0%	7.46%	2.74%	0.45%
128	321.9%	1043.8%	6903.1%	7.46%	2.74%	0.45%
256	110.9%	471.9%	3401.6%	7.46%	2.74%	0.45%
536	0.7%	173.1%	1572.4%	7.46%	2.74%	0.45%
576	100.0%	154.2%	1456.2%	7.46%	2.74%	0.45%
1460	18.4%	0.3%	514.0%	7.46%	2.74%	0.45%
1500	15.2%	100.0%	497.6%	7.46%	2.74%	0.45%
8960	15.7%	17.2%	0.0%	7.46%	2.74%	0.45%
9000	15.2%	16.7%	100.0%	7.46%	2.74%	0.45%

Figure 6: Overhead as Percentage of Inner Packet Size

C.3. Comparing Available Bandwidth

Another way to compare the two solutions is to look at the amount of available bandwidth each solution provides. The following sections consider and compare the percentage of available bandwidth. For the sake of providing a well understood baseline normal (unencrypted) Ethernet as well as normal ESP values are included.

C.3.1. Ethernet

In order to calculate the available bandwidth the per packet overhead is calculated first. The total overhead of Ethernet is 14+4 octets of header and CRC plus and additional 20 octets of framing (preamble, start, and inter-packet gap) for a total of 38 octets. Additionally the minimum payload is 46 octets.

Size	E + P	E + P	E + P	IPTFS	IPTFS	IPTFS	Enet	ESP
MTU	590	1514	9014	590	1514	9014	any	any
OH	74	74	74	78	78	78	38	74
40	614	1538	9038	45	42	40	84	114
128	614	1538	9038	146	134	129	166	202
256	614	1538	9038	293	269	258	294	330
536	614	1538	9038	614	564	540	574	610
576	1228	1538	9038	659	606	581	614	650
1460	1842	1538	9038	1672	1538	1472	1498	1534
1500	1842	3076	9038	1718	1580	1513	1538	1574
8960	11052	10766	9038	10263	9438	9038	8998	9034
9000	11052	10766	18076	10309	9480	9078	9038	9074

Figure 7: L2 Octets Per Packet

Size	E + P	E + P	E + P	IPTFS	IPTFS	IPTFS	Enet	ESP
MTU	590	1514	9014	590	1514	9014	any	any
OH	74	74	74	78	78	78	38	74
40	2.0M	0.8M	0.1M	27.3M	29.7M	31.0M	14.9M	11.0M
128	2.0M	0.8M	0.1M	8.5M	9.3M	9.7M	7.5M	6.2M
256	2.0M	0.8M	0.1M	4.3M	4.6M	4.8M	4.3M	3.8M
536	2.0M	0.8M	0.1M	2.0M	2.2M	2.3M	2.2M	2.0M
576	1.0M	0.8M	0.1M	1.9M	2.1M	2.2M	2.0M	1.9M
1460	678K	812K	138K	747K	812K	848K	834K	814K
1500	678K	406K	138K	727K	791K	826K	812K	794K
8960	113K	116K	138K	121K	132K	138K	138K	138K
9000	113K	116K	69K	121K	131K	137K	138K	137K

Figure 8: Packets Per Second on 10G Ethernet

Size	E + P 590 74	E + P 1514 74	E + P 9014 74	IPTFS 590 78	IPTFS 1514 78	IPTFS 9014 78	Enet any 38	ESP any 74
40	6.51%	2.60%	0.44%	87.30%	94.93%	99.14%	47.62%	35.09%
128	20.85%	8.32%	1.42%	87.30%	94.93%	99.14%	77.11%	63.37%
256	41.69%	16.64%	2.83%	87.30%	94.93%	99.14%	87.07%	77.58%
536	87.30%	34.85%	5.93%	87.30%	94.93%	99.14%	93.38%	87.87%
576	46.91%	37.45%	6.37%	87.30%	94.93%	99.14%	93.81%	88.62%
1460	79.26%	94.93%	16.15%	87.30%	94.93%	99.14%	97.46%	95.18%
1500	81.43%	48.76%	16.60%	87.30%	94.93%	99.14%	97.53%	95.30%
8960	81.07%	83.22%	99.14%	87.30%	94.93%	99.14%	99.58%	99.18%
9000	81.43%	83.60%	49.79%	87.30%	94.93%	99.14%	99.58%	99.18%

Figure 9: Percentage of Bandwidth on 10G Ethernet

A sometimes unexpected result of using IP-TFS (or any packet aggregating tunnel) is that, for small to medium sized packets, the available bandwidth is actually greater than native Ethernet. This is due to the reduction in Ethernet framing overhead. This increased bandwidth is paid for with an increase in latency. This latency is the time to send the unrelated octets in the outer tunnel frame. The following table illustrates the latency for some common values on a 10G Ethernet link. The table also includes latency introduced by padding if using ESP with padding.

	ESP+Pad 1500	ESP+Pad 9000	IP-TFS 1500	IP-TFS 9000
40	1.14 us	7.14 us	1.17 us	7.17 us
128	1.07 us	7.07 us	1.10 us	7.10 us
256	0.97 us	6.97 us	1.00 us	7.00 us
536	0.74 us	6.74 us	0.77 us	6.77 us
576	0.71 us	6.71 us	0.74 us	6.74 us
1460	0.00 us	6.00 us	0.04 us	6.04 us
1500	1.20 us	5.97 us	0.00 us	6.00 us

Figure 10: Added Latency

Notice that the latency values are very similar between the two solutions; however, whereas IP-TFS provides for constant high bandwidth, in some cases even exceeding native Ethernet, ESP with padding often greatly reduces available bandwidth.

Appendix D. Acknowledgements

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