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C. Hopps
LabN Consulting, L.L.C.
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Aggregation and Fragmentation Mode for Encapsulating Security Payload (ESP) and Its Use for IP Traffic Flow Security (IP-TFS)

Abstract

This document describes a mechanism for aggregation and fragmentation of IP packets when they are being encapsulated in Encapsulating Security Payload (ESP). This new payload type can be used for various purposes, such as decreasing encapsulation overhead for small IP packets; however, the focus in this document is to enhance IP Traffic Flow Security (IP-TFS) by adding Traffic Flow Confidentiality (TFC) to encrypted IP-encapsulated traffic. TFC is provided by obscuring the size and frequency of IP traffic using a fixed-size, constant-send-rate IPsec tunnel. The solution allows for congestion control, as well as nonconstant send-rate usage.

Status of This Memo

This is an Internet Standards Track document.

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1. Introduction

Traffic analysis [RFC4301] [AppCrypt] is the act of extracting information about data being sent through a network. While directly obscuring the data with encryption [RFC4303], the patterns in the message traffic may expose information due to variations in its shape

and timing [RFC8546] [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 underutilize the available bandwidth.

This document defines an aggregation and fragmentation (AGGFRAG) mode for ESP, as well as ESP's use for IP Traffic Flow Security (IP-TFS). This solution provides for full TFC without the aforementioned bandwidth limitation. This is accomplished by using a constant-send-rate IPsec [RFC4303] tunnel with fixed-size encapsulating packets; however, these fixed-size packets can contain partial, whole, or multiple IP packets to maximize the bandwidth of the tunnel. A nonconstant 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 TFC solution prescribed in [RFC4303], see Appendix C.

Additionally, IP-TFS provides for operating fairly within congested networks [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.

The mechanisms, such as the AGGFRAG mode, defined in this document are generic with the intent of allowing for non-TFS uses, but such uses are outside the scope of this document.

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 BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

This document assumes familiarity with IP security concepts, including TFC, as described in [RFC4301].

2. The AGGFRAG Tunnel

As mentioned in Section 1, the AGGFRAG mode utilizes an IPsec [RFC4303] tunnel as its transport. For the purpose of IP-TFS, fixed-size encapsulating packets are sent at a constant rate on the AGGFRAG tunnel.

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

The fixed packet size MAY either be specified manually or be determined through other methods, such as the Packetization Layer MTU

Discovery (PLMTUD) [RFC4821] [RFC8899] or Path MTU Discovery (PMTUD) [RFC1191] [RFC8201]. PMTUD is known to have issues, so PLMTUD is considered the more robust option. For PLMTUD, congestion control payloads can be used as in-band probes (see Section 6.1.2 and [RFC8899]).

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

The egress (receiving) side of the AGGFRAG tunnel MUST allow for and expect the ingress (sending) side of the AGGFRAG 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 by introducing an AGGFRAG mode for ESP.

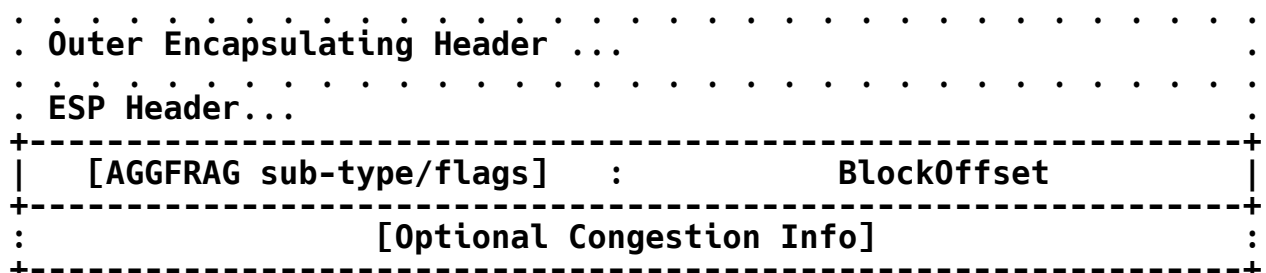
The AGGFRAG mode aggregates and fragments the inner IP traffic flow into encapsulating IPsec tunnel packets. For IP-TFS, the IPsec encapsulating tunnel packets are a fixed size. Padding is only added to the tunnel packets if there is no data available to be sent at the time of tunnel packet transmission or if fragmentation has been disabled by the receiver.

This is accomplished using a new Encapsulating Security Payload (ESP) [RFC4303] Next Header field value AGGFRAG_PAYLOAD (Section 6.1).

Other non-IP-TFS uses of this AGGFRAG mode have been suggested, 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 consists of a 4- or 24-octet header, followed by either a partial data block, a full data block, 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.



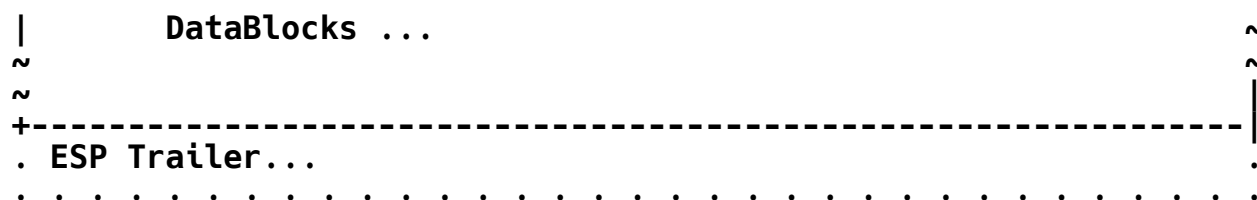


Figure 1: Layout of an AGGFRAG Mode IPsec Packet

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 the data block that is still being reassembled.

If the BlockOffset points past the end of the DataBlocks data, then 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 inner packet in the presence of outer encapsulating packet loss.

An example AGGFRAG mode packet flow can be found in Appendix A.

2.2.1. DataBlocks

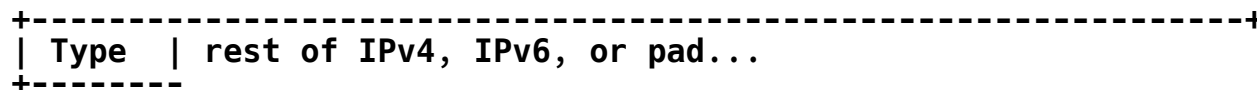


Figure 2: Layout of a Data Block

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's Total Length or IPv6's Payload Length fields.

2.2.2. End Padding

Since a data block's type is identified in its first 4 bits, the only time padding is required is when there is no data to encapsulate. For this end padding, a Pad Data Block is used.

2.2.3. Fragmentation, Sequence Numbers, and All-Pad Payloads

In order for a receiver 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

data block) in between the packets carrying the inner packet fragment payloads. This interleaving of all-pad payloads allows the sender to always 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.

Given the above, the receiver will need to handle out-of-order arrival of outer ESP packets prior to reassembly processing. ESP already provides for optionally detecting replay attacks. Detecting replay attacks normally utilizes a window method. A similar sequence-number-based sliding window can be used to correct reordering of the outer packet stream. Receiving a larger (newer) sequence number packet advances the window, and if any older ESP packets whose sequence numbers the window has passed by are received, then the packets are dropped. A good choice for the size of this window depends on the amount of misordering the user is experiencing; however, a value of 3 has been suggested as a default when no more informed choice exists.

As the amount of misordering that may be present is hard to predict, the window size SHOULD be configurable by the user. Implementations MAY also dynamically adjust the reordering window based on actual misordering seen in arriving packets.

Please note, when IP-TFS sends a continuous stream of packets, there is no requirement for an explicit lost packet timer; however, using a lost packet timer is RECOMMENDED. If an implementation does not use a lost packet timer and only considers an outer packet lost when the reorder window moves by it, the inner traffic can be delayed by up to the reorder window size times the per-packet send rate. This delay could be significant for slower send rates or when larger reorder window sizes are in use. As the lost packet timer affects the delay of inner packet delivery, an implementation or user could choose to set it proportionate to the tunnel rate.

While ESP guarantees an increasing sequence number with subsequently sent packets, it does not actually require the sequence numbers to be generated consecutively (e.g., sending only even-numbered sequence numbers would be allowed, as long as they are always increasing). Gaps in the sequence numbers will not work for this document, so the sequence number stream MUST increase monotonically by 1 for each subsequent packet.

When using the AGGFRAG_PAYLOAD in conjunction with replay detection, the window size for both MAY be reduced to the smaller of the two window sizes. This is because packets outside of the smaller window but inside the larger window would still be dropped by the mechanism with the smaller window size. However, there is also no requirement to make these values the same. Indeed, in some cases, such as slow tunnels where a very small or zero reorder window size is appropriate, the user may still want a large replay detection window to log replayed packets. Additionally, large replay windows can be implemented with very little overhead, compared to large reorder

windows.

Finally, as sequence numbers are reset when switching Security Associations (SAs) (e.g., when rekeying a Child SA), senders **MUST NOT** send initial fragments of an inner packet using one SA and subsequent fragments in a different SA.

A note on BlockOffset values: Senders **MUST** encode the BlockOffset consistently with the immediately preceding non-all-pad payload packet. Specifically, if the immediately preceding non-all-pad payload packet ended with a Pad Data Block, this BlockOffset **MUST** be zero, as Pad Data Blocks are never fragmented. The BlockOffset **MUST** be consistent with the remaining size implied by the length field from the fragmented inner packet.

2.2.3.1. Optional Extra Padding

When the tunnel bandwidth is not being fully utilized, a sender **MAY** pad out the current encapsulating packet in order to deliver an inner packet unfragmented in the following outer packet. The benefit would be to avoid inner packet fragmentation in the presence of a bursty offered load (non-bursty traffic will naturally not fragment). Senders **MAY** also choose to allow for a minimum fragment size to be configured (e.g., as a percentage of the AGGFRAG_PAYLOAD payload size) to avoid fragmentation at the cost of tunnel bandwidth. The costs with these methods are complexity and an added delay of inner traffic. The main advantage to avoiding fragmentation is to minimize inner packet loss in the presence of outer packet loss. When this is worthwhile (e.g., how much loss and what type of loss is required, given different inner traffic shapes and utilization, for this to make sense) and what values to use for the allowable/added delay may be worth researching but is outside the scope of this document.

While use of padding to avoid fragmentation does not impact interoperability, if padding is used inappropriately, it can reduce the effective throughput of a tunnel. Senders implementing either of the above approaches will need to take care to not reduce the effective capacity, and overall utility, of the tunnel through the overuse of padding.

2.2.4. Empty Payload

To support reporting of congestion control information (described later) using a non-AGGFRAG_PAYLOAD-enabled SA, it is allowed to send 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.

Currently, this situation is only applicable in use cases without Internet Key Exchange Protocol Version 2 (IKEv2).

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], the Differentiated Services (DS) field [RFC2474], and the Explicit Congestion Notification (ECN) field [RFC3168]. Unlike in [RFC4301], the AGGFRAG mode may, and often will, be encapsulating more than one IP packet per ESP packet. To deal with this, these mappings are restricted further.

2.2.5.1. DF Bit

The AGGFRAG mode never maps the inner DF bit, as it is unrelated to the AGGFRAG tunnel functionality; the AGGFRAG mode never needs to IP fragment the inner packets, and the inner packets will not affect the fragmentation of the outer encapsulation packets.

2.2.5.2. ECN Value

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. The sender MAY still set the ECN value of inner packets based on the normal ECN specification [RFC3168] [RFC4301] [RFC6040].

2.2.5.3. DS Field

By default, the DS field SHOULD NOT be copied, although a sender MAY choose to allow for configuration to override this behavior. A sender SHOULD also allow the DS value to be set by configuration.

2.2.6. IPv4 Time To Live (TTL), IPv6 Hop Limit, and ICMP Messages

How to modify the inner packet IPv4 TTL [RFC0791] or IPv6 Hop Limit [RFC8200] is specified in [RFC4301].

[RFC4301] specifies how to apply policy to authenticated and unauthenticated ICMP error packets (e.g., Destination Unreachable) arriving at or being forwarded through the endpoint, in particular, whether to process, ignore, or forward said packets. With the one exception that this document does not change the handling of these packets, they should be handled as specified in [RFC4301].

The one way in which an AGGFRAG tunnel differs in ICMP error packet mechanics is with PMTU. When fragmentation is enabled on the AGGFRAG tunnel, then no ICMP "Too Big" errors need to be generated for arriving ingress traffic, as the arriving inner packets will be naturally fragmented by the AGGFRAG encapsulation.

Otherwise, when fragmentation has been disabled on the AGGFRAG tunnel, then the treatment of arriving inner traffic exactly maps to that of a non-AGGFRAG ESP tunnel. Explicitly, IPv4 with DF set and IPv6 packets that cannot fit in its own outer packet payload will generate the appropriate ICMP "Too Big" error, as described in [RFC4301], and IPv4 packets without DF set will be IP fragmented, as described in [RFC4301].

Packets egressing the tunnel continue to be handled as specified in [RFC4301].

All other aspects of PMTU and the handling of ICMP "Too Big" messages (i.e., with regards to the outer AGGFRAG/ESP tunnel packet size) also remain unchanged from [RFC4301].

2.2.7. Effective MTU of the Tunnel

Unlike in [RFC4301], there is normally no effective MTU (EMTU) on an AGGFRAG tunnel, as all IP packet sizes are properly transmitted without requiring IP fragmentation prior to tunnel ingress. That said, a sender MAY allow for explicitly configuring an MTU for the tunnel.

If fragmentation has been disabled on the AGGFRAG tunnel, then the tunnel's EMTU and behaviors are the same as normal IPsec tunnels [RFC4301].

2.3. Exclusive SA Use

This document does not specify mixed use of an AGGFRAG_PAYLOAD-enabled SA. A sender MUST only send AGGFRAG_PAYLOAD payloads over an SA configured for AGGFRAG mode.

2.4. Modes of Operation

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

An AGGFRAG tunnel used for IP-TFS 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-size packets over an AGGFRAG tunnel 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 MUST only be used where the user has full administrative control over any path the tunnel will take and MUST NOT be used if this is not the case. 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. The use of circuit breakers is also RECOMMENDED (Section 2.4.2.1).

Users that choose the non-congestion-controlled mode need to understand that this mode will send packets at a constant rate, utilizing a constant, fixed bandwidth, and will not adjust based on congestion. Thus, if they do not guarantee the bandwidth required by the tunnel, the tunnel's operation, as well as the rest of their network, may be negatively impacted.

One expected use case for the non-congestion-controlled mode is to guarantee the full tunnel bandwidth is available and preferred over other non-tunnel traffic. In fact, a typical site-to-site use case might have all of the user traffic utilizing the IP-TFS tunnel.

The non-congestion-controlled mode is also appropriate if ESP over TCP is in use [RFC9329]. However, the use of TCP is considered a fallback-only solution for IPsec; it is highly not preferred. This is also one of the reasons that TCP was not chosen as the encapsulation for IP-TFS instead of AGGFRAG.

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. There is an example in [RFC4342] of the algorithm in [RFC5348], which matches the requirements of IP-TFS (i.e., designed for fixed-size packets and send rate varied based on congestion).

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. Not receiving this information over multiple consecutive RTT intervals should be considered a congestion event that causes the sender to adjust its sending rate lower. For example, this is called the "no feedback timeout" in [RFC4342], and it is equal to 4 RTT intervals. When a "no feedback timeout" has occurred, the sending rate is halved, as per [RFC4342].

An implementation MAY choose to always include the congestion information in its AGGFRAG payload header if it is sending it 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 through.

When choosing a congestion control algorithm (or a selection of

algorithms), note that IP-TFS is not providing for reliable delivery of IP traffic, and so per-packet acknowledgements (ACKs) are not required and are not provided.

It is worth noting that the variable send rate of a congestion-controlled AGGFRAG 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 addition to congestion control, implementations that support the non-congestion-control mode SHOULD implement circuit breakers [RFC8084] as a recovery method of last resort. When circuit breakers are enabled, an implementation SHOULD also enable congestion control reports so that circuit breakers have information to act on.

The pseudowire congestion considerations [RFC7893] are equally applicable to the mechanisms defined in this document, notably the text on inelastic traffic.

One example of a simple, slow-trip circuit breaker that an implementation may provide would utilize 2 values: the amount of persistent loss rate required to trip the circuit breaker and the required length of time this persistent loss rate must be seen to trip the circuit breaker. These 2 value are required configurations from the user. When the circuit breaker is tripped, the tunnel traffic is disabled and an appropriate log message or other management type alarm is triggered, indicating operation intervention is required.

2.5. Summary of Receiver Processing

An AGGFRAG-enabled SA receiver has a few tasks to perform.

The receiver MAY process incoming AGGFRAG_PAYLOAD payloads as soon as they arrive, as much as it can, i.e., if the incoming AGGFRAG_PAYLOAD packet contains complete inner packet(s), the receiver should extract and transmit them immediately. For partial packets, the receiver needs to keep the partial packets in the memory until they fall out from the reordering window or until the missing parts of the packets are received, in which case, it will reassemble and transmit them. If the AGGFRAG_PAYLOAD payload contains multiple packets, they SHOULD be sent out in the order they are in the AGGFRAG_PAYLOAD (i.e., keep the original order they were received on the other end). The cost of using this method is that an amplification of out-of-order delivery of inner packets can occur due to inner packet aggregation.

Instead of the method described in the previous paragraph, the receiver MAY reorder out-of-order AGGFRAG_PAYLOAD payloads received into in-sequence-order AGGFRAG_PAYLOAD payloads (Section 2.2.3), and only after it has an in-order AGGFRAG_PAYLOAD payload stream would the receiver transmit the inner packets. Using this method will ensure the inner packets are sent in order. The cost of this method

is that a lost packet will cause a delay of up to the lost packet timer interval (or the full reorder window if no lost packet timer is used). Additionally, there can be extra burstiness in the output stream. This burstiness can happen when a lost packet is dropped from the reorder window, and the remaining outer packets in the reorder window are immediately processed and sent out back to back.

Additionally, if congestion control is enabled, the receiver sends congestion control data (Section 6.1.2) back to the sender, as described in Sections 2.4.2 and 3.

Finally, a note on receiving incorrect BlockOffset values: To account for misbehaving senders, a receiver SHOULD gracefully handle the case where the BlockOffset of consecutive packets, and/or the inner packet they share, do not agree. It MAY drop the inner packet or one or both of the outer packets.

3. Congestion Information

In order to support the congestion-controlled mode, the sender needs to know the loss event rate and to approximate the RTT [RFC5348]. In order to obtain these values, the receiver sends congestion control information on its SA back to the sender. Thus, 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 an RTT 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 its 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 its congestion control header, it places this latest recorded TVal 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 its half of the AGGFRAG tunnel).

When the sender receives back its TVal in the TEcho header field, it calculates 2 RTT estimates. The first is the actual delay found by subtracting the TEcho value from its current clock and then subtracting the Echo Delay as well. The second RTT estimate is found by adding the received Transmit Delay header value to the sender's own transmission delay (i.e., the average time between sending packets on its half of the AGGFRAG tunnel). The larger of these 2

RTT estimates SHOULD be used as the RTT value.

The two RTT 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 send 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 addition to normal packet loss information, the AGGFRAG mode 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 (receiving) side 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.

In [RFC6040], which updates [RFC3168] and [RFC4301], behaviors for marking the outer ECN field value based on the ECN field of the inner packet are defined. As the AGGFRAG mode may have multiple inner packets present in a single outer packet, and there is no obvious correct way to map these multiple values to the single outer packet ECN field value, the tunnel ingress endpoint SHOULD operate in the "compatibility" mode, rather than the "default" mode from [RFC6040]. In particular, this means that the ingress (sending) endpoint of the tunnel always sets the newly constructed outer encapsulating packet header ECN field to Not-ECT [RFC6040].

4. Configuration of AGGFRAG Tunnels for IP-TFS

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

YANG and MIB documents have been defined for IP-TFS in [RFC9348] and [RFC9349].

4.1. Bandwidth

Bandwidth is a local configuration option. For the non-congestion-controlled mode, the bandwidth SHOULD be configured. For the

congestion-controlled mode, the bandwidth can be configured or the congestion control algorithm discovers and uses 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 MAY be configured manually or can be automatically determined using other methods, such as PLMTUD [RFC4821] [RFC8899] or PMTUD [RFC1191] [RFC8201]. As PMTUD is known to have issues, PLMTUD is considered the more robust option. 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, AGGFRAG tunnels utilize ESP payloads of type AGGFRAG_PAYLOAD.

When using IKEv2, a new "USE_AGGFRAG" notification message enables 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 use of 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 CREATE_CHILD_SA exchanges). If the request is accepted, then the 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.

As the use of the AGGFRAG_PAYLOAD payload is currently only defined for non-transport-mode tunnels, the USE_AGGFRAG notification MUST NOT be combined with the USE_TRANSPORT notification.

The USE_AGGFRAG notification contains a 1-octet payload of flags that specify requirements from the sender of the notification. If any requirement flags are not understood or cannot be supported by the receiver, then the receiver SHOULD NOT enable use of AGGFRAG_PAYLOAD (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

The packet and data formats defined below are generic with the intent of allowing for non-IP-TFS uses, but such uses are outside the scope of this document.

6.1. AGGFRAG_PAYLOAD Payload

ESP Next Header value: 144

An AGGFRAG payload is identified by the ESP Next Header value AGGFRAG_PAYLOAD, which has the value 144, which has been reserved in the IP protocol numbers space. The first octet of the payload indicates the format of the remaining payload data.

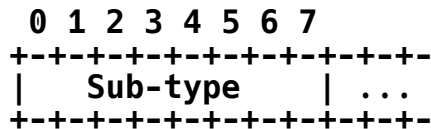


Figure 3: AGGFRAG_PAYLOAD Payload Format

Sub-type:

An 8-bit value indicating the payload format.

This document 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 consists of a 4-octet header, followed by a variable amount of DataBlocks data, as shown below.

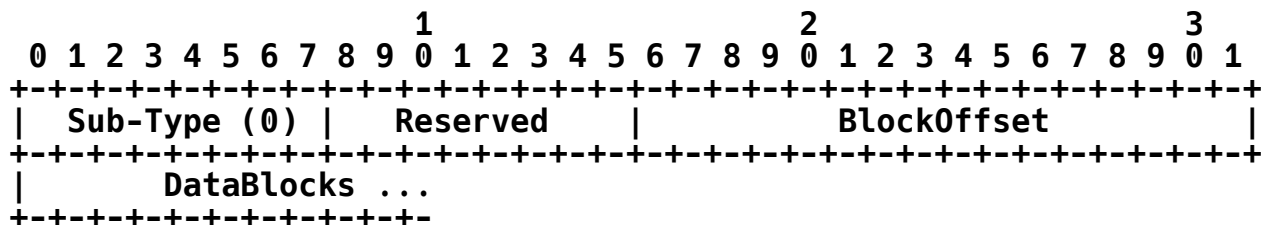


Figure 4: Non-Congestion-Control Payload Format

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. If the start of a new data block occurs in a subsequent payload, the BlockOffset will point past the end of the DataBlocks data. In this case, all the DataBlocks data belongs to the current data

block being assembled. When the BlockOffset extends into subsequent payloads, it continues to only count DataBlocks data (i.e., it does not count subsequent packets of the non-DataBlocks data, such as header 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 consists of a 24-octet header, followed by a variable amount of DataBlocks data, as shown below.

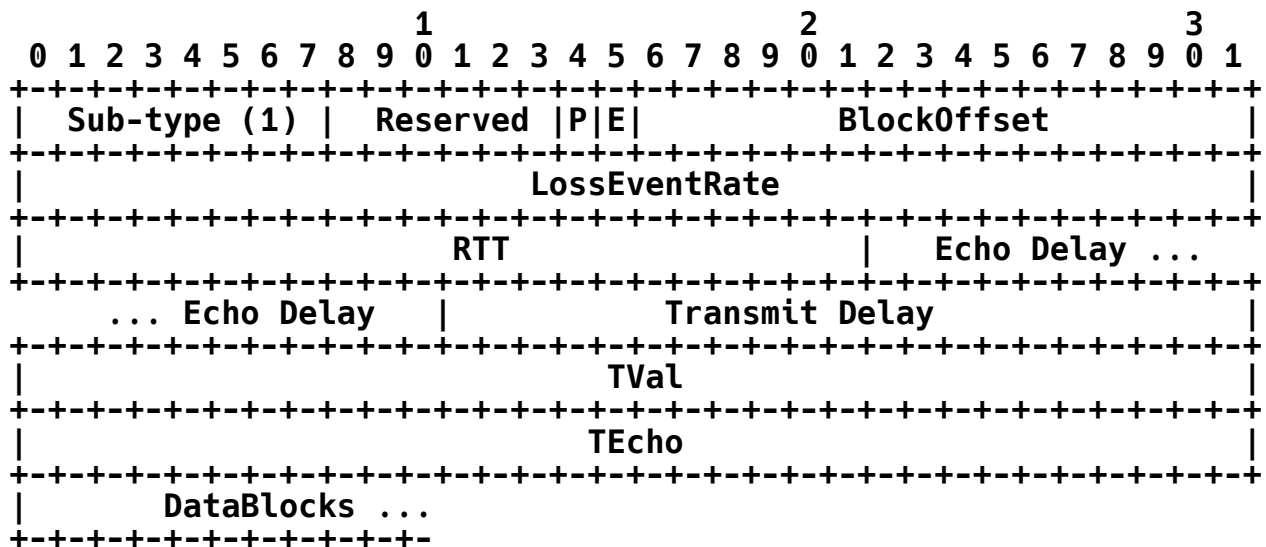


Figure 5: Congestion Control Payload Format

Sub-type:

An octet indicating the payload format. For this congestion control format, the value is 1.

Reserved:

A 6-bit field set to 0 on generation and ignored on receipt.

P:

A 1-bit value that, if set, indicates that PLMTUD probing is in progress. This information can be used to avoid treating missing packets as loss events by the congestion control algorithm when running the PLMTUD probe algorithm.

E:

A 1-bit value that, 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 $1/\text{LossEventRate}$.

RTT:

A 22-bit value specifying the sender's current RTT estimate in microseconds. The value MAY be zero prior to the sender having calculated an RTT estimate. The value SHOULD be set to zero on non-AGGFRAG PAYLOAD-enabled SAs. If the RTT is equal to or larger than $0x3FFFFFF$, the value MUST be set to $0x3FFFFFF$.

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 delay is equal to or larger than $0x1FFFFFF$, the value MUST be set to $0x1FFFFFF$.

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 IP-TFS tunnel. If the delay is equal to or larger than $0x1FFFFFF$, the value MUST be set to $0x1FFFFFF$.

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 a non-IP-TFS-enabled SA, this field MUST be empty (i.e., be zero octets long).

6.1.3. Data Blocks

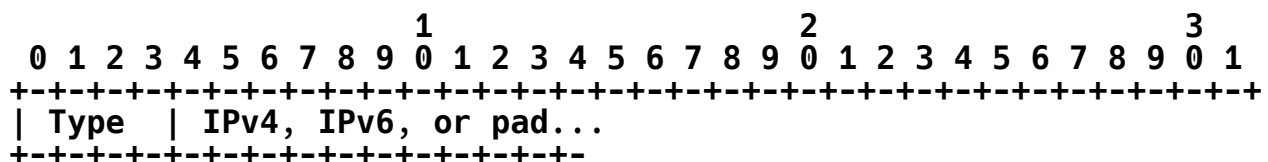


Figure 6: Data Block Format

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

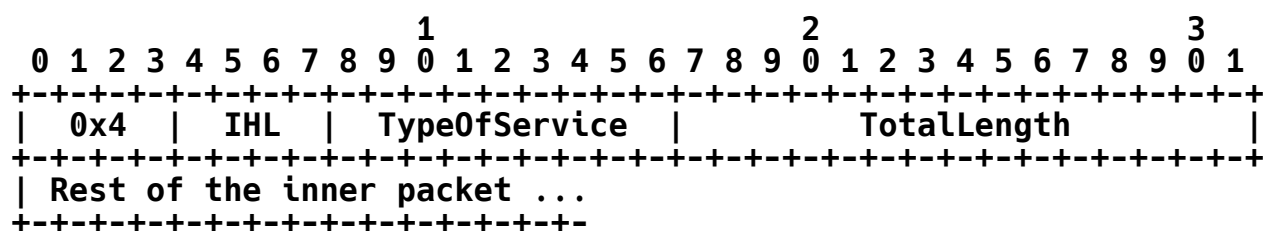


Figure 7: IPv4 Data Block Format

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

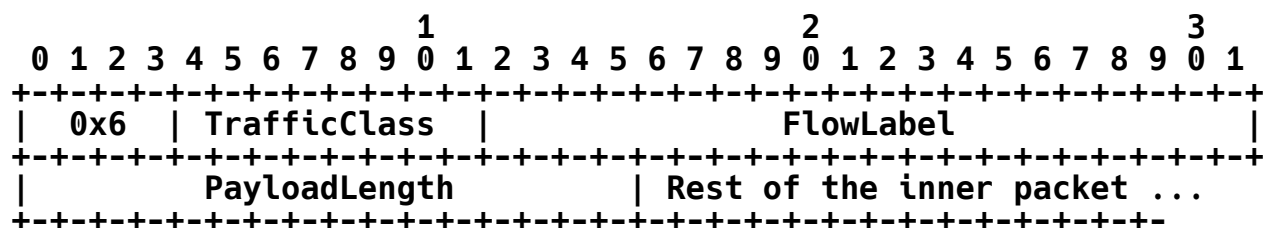


Figure 8: IPv6 Data Block Format

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

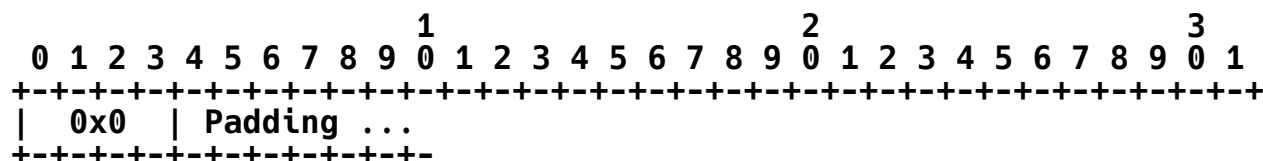


Figure 9: Pad Data Block Format

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 Next Header value.

The USE_AGGFRAG Notification Message State Type is 16442.

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

```
+--+--+--+--+--+--+--+--+
|0|0|0|0|0|0|C|D|
+--+--+--+--+--+--+--+--+
```

Figure 10: USE_AGGFRAG Requirement Flags

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, it 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 its USE_AGGFRAG notification.

7. IANA Considerations

7.1. ESP Next Header Value

IANA has allocated an IP protocol number from the "Protocol Numbers - Assigned Internet Protocol Numbers" registry as follows.

Decimal: 144

Keyword: AGGFRAG

Protocol: AGGFRAG encapsulation payload for ESP

Reference: RFC 9347

7.2. AGGFRAG_PAYLOAD Sub-Types

IANA has created a registry called "AGGFRAG_PAYLOAD Sub-Types" under a new category named "ESP AGGFRAG_PAYLOAD". The registration policy for this registry is "Expert Review" [RFC8126] [RFC7120].

Name: AGGFRAG_PAYLOAD Sub-Types
Description: AGGFRAG_PAYLOAD Payload Formats
Reference: RFC 9347

This initial content for this registry is as follows:

Sub-Type	Name	Reference
0	Non-Congestion-Control Format	RFC 9347
1	Congestion Control Format	RFC 9347
3-255	Reserved	

Table 1: AGGFRAG_PAYLOAD Sub-Types

7.3. USE_AGGFRAG Notify Message Status Type

IANA has allocated a status type USE_AGGFRAG from the "IKEv2 Notify Message Types - Status Types" registry.

Decimal: 16442
Name: USE_AGGFRAG
Reference: RFC 9347

8. Security Considerations

This document describes an aggregation and fragmentation mechanism to efficiently implement TFC for IP traffic. This approach is expected to reduce the efficacy of traffic analysis on IPsec communication. 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 in Section 3.1, 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.

As noted previously in Section 2.4.2, for TFC to be 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.

9. References

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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 one may occur in a single encapsulating packet.

```
Offset: 0           Offset: 100       Offset: 2000      Offset: 600
[ ESP1 (1404) ][ ESP2 (1404) ][ ESP3 (1404) ][ ESP4 (1404) ]
[--750--][--750--][60][-240-][--3000-----][pad]
```

Figure 11: Inner and Outer Packet Flow

Each outer encapsulating ESP space is a fixed size of 1404 octets, the first 4 octets of which contain the AGGFRAG header. The encapsulated IP packet flow (lengths include the IP header and payload) is as follows: a 750-octet packet, a 750-octet packet, a 60-octet packet, a 240-octet packet, and a 3000-octet packet.

The BlockOffset values in the 4 AGGFRAG payload headers for this packet flow would thus be: 0, 100, 2000, and 600, respectively. The first encapsulating packet (ESP1) has a zero BlockOffset, which points at the IP data block immediately following the AGGFRAG header. The following packet's (ESP2) 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 3000-octet data block, so the offset points past its end and into the fourth encapsulating packet. The fourth packet's (ESP4) offset is 600, pointing at the padding that follows the completion of the continued 3000-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:

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

X is the send rate in packets per second, R is the RTT 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_rcv 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

For comparing overhead, the overhead of ESP for both normal and AGGFRAG tunnel packets must be calculated, and so an algorithm for encryption and authentication must be chosen. For the data below, AES-GCM-256 was selected. This leads to an IP+ESP overhead of 54.

$$54 = 20 \text{ (IP)} + 8 \text{ (ESPH)} + 2 \text{ (ESPF)} + 8 \text{ (IV)} + 16 \text{ (ICV)}$$

Additionally, for IP-TFS, non-congestion-control AGGFRAG_PAYLOAD headers were chosen, which adds 4 octets, for a total overhead of 58.

C.1.1. IP-TFS Overhead

For comparison, the overhead of an AGGFRAG payload is 58 octets per outer packet. Therefore, the octet overhead per inner packet is 58 divided by the number of outer packets required (fractions allowed). The overhead as a percentage of inner packet size is a constant based on the Outer MTU size.

$$\begin{aligned} \text{OH} &= 58 / \text{Outer Payload Size} / \text{Inner Packet Size} \\ \text{OH \% of Inner Packet Size} &= 100 * \text{OH} / \text{Inner Packet Size} \\ \text{OH \% of Inner Packet Size} &= 5800 / \text{Outer Payload Size} \end{aligned}$$

Type	IP-TFS	IP-TFS	IP-TFS
MTU	576	1500	9000
PSize	518	1442	8942
40	11.20%	4.02%	0.65%
576	11.20%	4.02%	0.65%

1500	11.20%	4.02%	0.65%	
+-----+	+-----+	+-----+	+-----+	+-----+
9000	11.20%	4.02%	0.65%	
+-----+	+-----+	+-----+	+-----+	+-----+

Table 2: 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., original 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 (54) 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 Payload Size = IP Packet Size - IP Overhead

Outer Payload Size = MTU - IPsec Overhead

$$NF0 = \frac{\text{Inner Payload Size}}{\text{Outer Payload Size} - \text{IP Overhead}}$$

NF = CEILING(NF0)

$$\begin{aligned} OH &= NF * (\text{IP Overhead} + \text{IPsec Overhead}) \\ &\quad - \text{IP Overhead} \\ &\quad + NF * (\text{Outer Payload Size} - \text{IP Overhead}) \\ &\quad - \text{Inner Payload Size} \end{aligned}$$

$$OH = NF * (\text{IPsec Overhead} + \text{Outer Payload Size}) - (\text{IP Overhead} + \text{Inner Payload Size})$$

$$OH = NF * (\text{IPsec Overhead} + \text{Outer Payload Size}) - \text{Inner Packet Size}$$

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	522	1446	8946	518	1442	8942	

40	482	1406	8906	4.5	1.6	0.3
128	394	1318	8818	14.3	5.1	0.8
256	266	1190	8690	28.7	10.3	1.7
518	4	928	8428	58.0	20.8	3.4
576	576	870	8370	64.5	23.2	3.7
1442	286	4	7504	161.5	58.0	9.4
1500	228	1500	7446	168.0	60.3	9.7
8942	1426	1558	4	1001.2	359.7	58.0
9000	1368	1500	9000	1007.7	362.0	58.4

Table 3: 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	522	1446	8946	518	1442	8942
40	1205.0%	3515.0%	22265.0%	11.20%	4.02%	0.65%
128	307.8%	1029.7%	6889.1%	11.20%	4.02%	0.65%
256	103.9%	464.8%	3394.5%	11.20%	4.02%	0.65%
518	0.8%	179.2%	1627.0%	11.20%	4.02%	0.65%
576	100.0%	151.0%	1453.1%	11.20%	4.02%	0.65%
1442	19.8%	0.3%	520.4%	11.20%	4.02%	0.65%
1500	15.2%	100.0%	496.4%	11.20%	4.02%	0.65%
8942	15.9%	17.4%	0.0%	11.20%	4.02%	0.65%
9000	15.2%	16.7%	100.0%	11.20%	4.02%	0.65%

Table 4: 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 and 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 Cyclic Redundancy Check (CRC) plus an 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
0H	92	92	92	96	96	96	38	74
40	614	1538	9038	47	42	40	84	114
128	614	1538	9038	151	136	129	166	202
256	614	1538	9038	303	273	258	294	330
518	614	1538	9038	614	552	523	574	610
576	1228	1538	9038	682	614	582	614	650
1442	1842	1538	9038	1709	1538	1457	1498	1534
1500	1842	3076	9038	1777	1599	1516	1538	1574
8942	11052	10766	9038	10599	9537	9038	8998	9034
9000	11052	10766	18076	10667	9599	9096	9038	9074

Table 5: 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
0H	92	92	92	96	96	96	38	74
40	2.0M	0.8M	0.1M	26.4M	29.3M	30.9M	14.9M	11.0M
128	2.0M	0.8M	0.1M	8.2M	9.2M	9.7M	7.5M	6.2M
256	2.0M	0.8M	0.1M	4.1M	4.6M	4.8M	4.3M	3.8M
518	2.0M	0.8M	0.1M	2.0M	2.3M	2.4M	2.2M	2.1M
576	1.0M	0.8M	0.1M	1.8M	2.0M	2.1M	2.0M	1.9M

128	1.05 us	7.05 us	1.10 us	7.10 us	
+-----+	+-----+	+-----+	+-----+	+-----+	+
256	0.95 us	6.95 us	1.00 us	7.00 us	
+-----+	+-----+	+-----+	+-----+	+-----+	+
518	0.74 us	6.74 us	0.79 us	6.79 us	
+-----+	+-----+	+-----+	+-----+	+-----+	+
576	0.70 us	6.70 us	0.74 us	6.74 us	
+-----+	+-----+	+-----+	+-----+	+-----+	+
1442	0.00 us	6.00 us	0.05 us	6.05 us	
+-----+	+-----+	+-----+	+-----+	+-----+	+
1500	1.20 us	5.96 us	0.00 us	6.00 us	
+-----+	+-----+	+-----+	+-----+	+-----+	+

Table 8: 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 plain Ethernet, ESP with padding often greatly reduces available bandwidth.

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Contributors

The following person made significant contributions to this document.

Lou Berger
LabN Consulting, L.L.C.
Email: lberger@labn.net

Author's Address

Christian Hopps
LabN Consulting, L.L.C.
Email: chopps@chopps.org