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RFC 8861
Sending Multiple RTP Streams in a Single RTP Session: Grouping RTP Control Protocol (RTCP) Reception Statistics and Other Feedback

Abstract

RTP allows multiple RTP streams to be sent in a single session but requires each Synchronization Source (SSRC) to send RTP Control Protocol (RTCP) reception quality reports for every other SSRC visible in the session. This causes the number of RTCP reception reports to grow with the number of SSRCs, rather than the number of endpoints. In many cases, most of these RTCP reception reports are unnecessary, since all SSRCs of an endpoint are normally co-located and see the same reception quality. This memo defines a Reporting Group extension to RTCP to reduce the reporting overhead in such scenarios.

Status of This Memo

This is an Internet Standards Track document.

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1. Introduction

The Real-time Transport Protocol (RTP) [RFC3550] is a protocol for group communication, supporting multiparty multimedia sessions. A single RTP session can support multiple participants sending data at once and can also support participants sending multiple simultaneous RTP streams. Examples of the latter might include a participant with multiple cameras who chooses to send multiple views of a scene, or a participant that sends audio and video flows multiplexed in a single RTP session. Rules for handling RTP sessions containing multiple RTP streams are described in [RFC3550], with some clarifications in [RFC8108].

An RTP endpoint will have one or more Synchronization Sources (SSRCs). It will have at least one RTP stream, and thus at least one SSRC, for each media source it sends, and it might use multiple SSRCs per media source when using media scalability features [RFC6190], forward error correction, RTP retransmission [RFC4588], or similar mechanisms. An endpoint that is not sending any RTP streams will have at least one SSRC to use for reporting and any feedback messages. Each SSRC has to send RTP Control Protocol (RTCP) Sender Reports (SRs) corresponding to the RTP packets it sends and Receiver Reports (RRs) for traffic it receives. (SRs and RRs are described in [RFC3550].) That is, every SSRC will send RTCP packets to report on every other SSRC. This rule is simple, but it can be quite inefficient for endpoints that send large numbers of RTP streams in a single RTP session. Consider a session comprising ten participants, each sending three media sources, each media source associated with its own RTP stream. There will be 30 SSRCs in such an RTP session, and each of those 30 SSRCs will send an RTCP SR/RR packet (containing several report blocks) per reporting interval as each SSRC reports on all the others. However, the three SSRCs comprising each participant are commonly co-located such that they see identical reception quality. If there was a way to indicate that several SSRCs are co-located and see the same reception quality, then two-thirds of those RTCP reports could be suppressed. This would allow the remaining RTCP reports to be sent more often, while keeping within the same RTCP bandwidth fraction.

This memo defines such an RTCP extension: RTCP Reporting Groups. This extension is used to indicate the SSRCs that originate from the same endpoint and therefore have identical reception quality, hence allowing the endpoints to suppress unnecessary RTCP reception quality reports.

2. Terminology

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

3. RTCP Reporting Groups

An RTCP Reporting Group is a set of SSRCs that are co-located at a single endpoint (which could be an end host or a middlebox) in an RTP session. Since they are co-located, every SSRC in the RTCP Reporting Group will have an identical view of the network conditions and will see the same lost packets, jitter, etc. This allows a single representative to send RTCP reception quality reports on behalf of the rest of the Reporting Group, reducing the number of RTCP packets that need to be sent without loss of information.

3.1. Semantics and Behavior of RTCP Reporting Groups

A group of co-located SSRCs that see identical network conditions can form an RTCP Reporting Group. If Reporting Groups are in use, an RTP endpoint with multiple SSRCs **MAY** put those SSRCs into a Reporting Group if their view of the network is identical, i.e., if they report on traffic received at the same interface of an RTP endpoint. SSRCs with different views of the network **MUST NOT** be put into the same Reporting Group.

An endpoint that has combined its SSRCs into an RTCP Reporting Group will choose one (or a subset) of those SSRCs to act as "reporting source(s)" for that RTCP Reporting Group. A reporting source will send RTCP SR/RR reception quality reports on behalf of the other members of the RTCP Reporting Group. A reporting source **MUST** suppress the RTCP SR/RR reports that relate to other members of the Reporting Group and only report on remote SSRCs. The other members (non-reporting sources) of the RTCP Reporting Group will suppress their RTCP reception quality reports and will instead send an RTCP Reporting Group Reporting Sources (RGRS) packet (see Section 3.2.2) to indicate that they are part of an RTCP Reporting Group and give the SSRCs of the reporting sources.

If there are large numbers of remote SSRCs in the RTP session, then the reception quality reports generated by the reporting source might grow too large to fit into a single compound RTCP packet, forcing the reporting source to use a round-robin policy to determine what remote SSRCs it includes in each compound RTCP packet, and so reducing the frequency of reports on each SSRC. To avoid this, in sessions with large numbers of remote SSRCs, an RTCP Reporting Group **MAY** use more than one reporting source. If several SSRCs are acting as reporting sources for an RTCP Reporting Group, then each reporting source **MUST** have non-overlapping sets of remote SSRCs it reports on.

An endpoint **MUST NOT** create an RTCP Reporting Group that comprises only a single local SSRC (i.e., an RTCP Reporting Group where the reporting source is the only member of the group), unless it is anticipated that the group might have additional SSRCs added to it in the future.

If a reporting source leaves the RTP session (i.e., if it sends an RTCP BYE packet or it leaves the session without sending a BYE according to the rules of [RFC3550], Section 6.3.7), the remaining members of the RTCP Reporting Group **MUST** (a) have another reporting source -- if one exists --

report on the remote SSRCs that the leaving SSRC had reported on, (b) choose a new reporting source, or (c) disband the RTCP Reporting Group and begin sending reception quality reports per [RFC3550] and [RFC8108].

The RTCP timing rules assign different bandwidth fractions to senders and receivers. This lets senders transmit RTCP reception quality reports more often than receivers. If a reporting source in an RTCP Reporting Group is a receiver but one or more non-reporting SSRCs in the RTCP Reporting Group are senders, then the endpoint **MAY** treat the reporting source as a sender for the purpose of RTCP bandwidth allocation, increasing its RTCP bandwidth allocation, provided it also treats one of the senders as if it were a receiver and makes the corresponding reduction in RTCP bandwidth for that SSRC. However, the application needs to consider the impact on the frequency of transmitting of the synchronization information included in RTCP SRs.

3.2. Identifying Members of an RTCP Reporting Group

When RTCP Reporting Groups are in use, the other SSRCs in the RTP session need to be able to identify which SSRCs are members of an RTCP Reporting Group. Two RTCP extensions are defined to support this: the RTCP Reporting Group (RGRP) Source Description (SDES) item is used by the reporting source(s) to identify an RTCP Reporting Group, and the RTCP RGRS packet is used by other members of an RTCP Reporting Group to identify the reporting source(s).

3.2.1. Definition and Use of the RTCP RGRP SDES Item

This document defines a new RTCP RGRP SDES item to identify an RTCP Reporting Group. The motivation for giving a Reporting Group an identifier is to ensure that (1) the RTCP Reporting Group and its member SSRCs can be correctly associated when there are multiple reporting sources and (2) a reporting SSRC can be associated with the correct Reporting Group if an SSRC collision occurs.

This document defines the RTCP RGRP SDES item. The RTCP RGRP SDES item **MUST** be sent by the reporting sources in a Reporting Group and **MUST NOT** be sent by other members of the Reporting Group or by SSRCs that are not members of any RTCP Reporting Group. Specifically, every reporting source in an RTCP Reporting Group **MUST** include an RTCP SDES packet containing an RGRP item in every compound RTCP packet in which it sends an RR or SR packet (i.e., in every RTCP packet it sends, unless Reduced-Size RTCP [RFC5506] is in use).

Syntactically, the format of the RTCP RGRP SDES item is identical to that of the RTCP SDES CNAME item [RFC7022], except that the SDES item type field **MUST** have value RGRP=11 instead of CNAME=1. The value of the RTCP RGRP SDES item **MUST** be chosen with the same concerns about global uniqueness and the same privacy considerations as the RTCP SDES CNAME. The value of the RTCP RGRP SDES item **MUST** be stable throughout the lifetime of the Reporting Group, even if some or all of the reporting sources change their SSRC due to collisions or if the set of reporting sources changes.

An RTP mixer or translator that forwards RTCP SR or RR packets from members of a Reporting Group **MUST** forward the corresponding RTCP RGRP SDES items as well, even if it otherwise strips SDES items other than the CNAME item.

3.2.2. Definition and Use of the RTCP RGRS Packet

A new RTCP packet type is defined to allow the members of an RTCP Reporting Group to identify the reporting sources for that group. This allows participants in an RTP session to distinguish an SSRC that is sending empty RTCP reception reports because it is a member of an RTCP Reporting Group from an SSRC that is sending empty RTCP reception reports because it is not receiving any traffic. It also explicitly identifies the reporting sources, allowing other members of the RTP session to (1) know which SSRCs are acting as the reporting sources for an RTCP Reporting Group and (2) detect if RTCP packets from any of the reporting sources are being lost.

The format of the RTCP RGRS packet is defined below. It comprises the fixed RTCP header that indicates the packet type and length, the SSRC of the packet sender, and a list of reporting sources for the RTCP Reporting Group of which the packet sender is a member.

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|    SC   | PT=RGRS(212)  |             length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                     SSRC of packet sender                     |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
:          List of SSRC(s) for the Reporting Source(s)          :
:                              ...                              :
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

The fields in the RTCP RGRS packet have the following definitions:

version (V): 2-bit unsigned integer. This field identifies the RTP version. The current RTP version is 2.

padding (P): 1 bit. If set, the padding bit indicates that the RTCP packet contains additional padding octets at the end that are not part of the control information but are included in the length field. See [RFC3550].

Source Count (SC): 5-bit unsigned integer. Indicates the number of reporting source SSRCs that are included in this RTCP packet. As the RTCP RGRS packet **MUST NOT** be sent by reporting sources, all the SSRCs in the list of reporting sources will be different from the SSRC of the packet sender. Every RTCP RGRS packet **MUST** contain at least one reporting source SSRC.

Payload type (PT): 8-bit unsigned integer. The RTCP packet type number that identifies the packet as being an RTCP RGRS packet. The RGRS RTCP packet has the value 212.

Length: 16-bit unsigned integer. The length of this packet in 32-bit words minus one, including the header and any padding. This is in line with the definition of the length field used in RTCP SRs and RRs [RFC3550]. Since all RTCP RGRS packets include at least one reporting source SSRC, the length will always be 2 or greater.

SSRC of packet sender: 32 bits. The SSRC of the sender of this packet.

List of SSRCs for the Reporting Source(s): A variable number (as indicated by the SC header field) of 32-bit SSRC values of the reporting sources for the RTCP Reporting Group of which the packet sender is a member.

Every source that belongs to an RTCP Reporting Group but is not a reporting source **MUST** include an RTCP RGRS packet in every compound RTCP packet in which it sends an RR or SR packet (i.e., in every RTCP packet it sends, unless Reduced-Size RTCP [RFC5506] is in use). Each RTCP RGRS packet **MUST** contain the SSRC identifier of at least one reporting source. If there are more reporting sources in an RTCP Reporting Group than can fit into an RTCP RGRS packet, the members of that Reporting Group **MUST** send the SSRCs of the reporting sources in a round-robin fashion in consecutive RTCP RGRS packets, such that all the SSRCs of the reporting sources are included over the course of several RTCP reporting intervals.

An RTP mixer or translator that forwards RTCP SR or RR packets from members of a Reporting Group **MUST** also forward the corresponding RGRS RTCP packets. If the RTP mixer or translator rewrites SSRC values of the packets it forwards, it **MUST** make the corresponding changes to the RTCP RGRS packets.

3.3. Interactions with the RTP/AVPF Feedback Profile

The use of the RTP/AVPF Feedback Profile [RFC4585] allows SSRCs to send rapid RTCP feedback requests and codec control messages. If the use of the RTP/AVPF profile has been negotiated in an RTP session, members of an RTCP Reporting Group can send rapid RTCP feedback and codec control messages per [RFC5104], per [RFC4585] as updated by Section 5.4 of [RFC8108], and by the following considerations.

The members of an RTCP Reporting Group will all see identical network conditions. Accordingly, one might therefore think that it doesn't matter which SSRC in the Reporting Group sends the RTP/AVPF feedback or codec control messages. There might be, however, cases where the sender of the feedback/codec control message has semantic importance, or when only a subset of the members of an RTCP Reporting Group might want to send RTP/AVPF feedback or a codec control message in response to a particular event. For example, an RTP video sender might choose to treat packet loss feedback received from SSRCs known to be audio receivers with less urgency than feedback that it receives from video receivers when deciding what packets to retransmit, and a multimedia receiver using Reporting Groups might want to choose the outgoing SSRC for feedback packets to reflect this.

Each member of an RTCP Reporting Group **SHOULD** therefore send RTP/AVPF feedback/codec control messages independently of the other members of the Reporting Group, to respect the semantic meaning of the message sender. The suppression rules of [RFC4585] will ensure that only a single copy of each feedback packet is (typically) generated, even if several members of a Reporting Group send the same feedback. When an endpoint knows that several members of its RTCP Reporting Group will be sending identical feedback and that the sender of the feedback is not semantically important, that endpoint **MAY** choose to send all its feedback from the reporting source and deterministically suppress feedback packets generated by the other sources in the Reporting Group.

It is important to note that the RTP/AVPF timing rules operate on a per-SSRC basis. Using a single reporting source to send all feedback for a Reporting Group will hence limit the amount of feedback that can be sent to that which can be sent by one SSRC. If this limit is a problem, then the Reporting Group can allow each of its members to send its own feedback, using its own SSRC.

If the RTP/AVPF feedback messages or codec control requests are sent as compound RTCP packets, then those compound RTCP packets **MUST** include either an RTCP RGRS packet or an RTCP RGRP SDES item, depending on whether they are sent by the reporting source or a non-reporting source in the RTCP Reporting Group, respectively. The contents of noncompound RTCP feedback or codec control messages are not affected by the use of RTCP Reporting Groups.

3.4. Interactions with RTCP Extended Report (XR) Packets

When using RTCP Extended Report (XR) packets [RFC3611] with RTCP Reporting Groups, it is **RECOMMENDED** that the reporting source be used to send the RTCP XR packets. If multiple reporting sources are in use, the reporting source that sends the SR/RR packets that relate to a particular remote SSRC **SHOULD** send the RTCP XR reports about that SSRC. This is motivated as one commonly combine the RTCP XR metrics with the regular report block to more fully understand the situation. Receiving these blocks in different compound packets reduces their value, as the measuring intervals are not synchronized in those cases.

Some RTCP XR report blocks are specific to particular types of media and might be relevant to only some members of a Reporting Group. For example, it would make no sense for an SSRC that is receiving video to send a Voice over IP (VoIP) metric RTCP XR report block. Such media-specific RTCP XR report blocks **MUST** be sent by the SSRC to which they are relevant and **MUST NOT** be included in the common report sent by the reporting source. This might mean that some SSRCs send RTCP XR packets in compound RTCP packets that contain an empty RTCP SR/RR packet and that the time period covered by the RTCP XR packet is different from that covered by the RTCP SR/RR packet. If it is important that the RTCP XR packet and RTCP SR/RR packet cover the same time period, then that source **SHOULD** be removed from the RTCP Reporting Group, and standard RTCP packets be sent instead.

3.5. Middlebox Considerations

Many different types of middleboxes are used with RTP. RTCP Reporting Groups are potentially relevant to those types of RTP middleboxes that have their own SSRCs and generate RTCP reports for the traffic they receive. RTP middleboxes that do not have their own SSRC and that do not send RTCP reports on the traffic they receive cannot use the RTCP Reporting Group extension, since they generate no RTCP reports to that group.

An RTP middlebox that has several SSRCs of its own can use the RTCP Reporting Group extension to group the RTCP reports it generates. This can occur, for example, if a middlebox is acting as an RTP mixer for both audio and video flows that are multiplexed onto a single RTP session, where the middlebox has one SSRC for the audio mixer and one for the video mixer part, and when the middlebox wants to avoid cross-reporting between audio and video.

A middlebox cannot use the RTCP Reporting Group extension to group RTCP packets from the SSRCs that it is forwarding. It can, however, group the RTCP packets from the SSRCs it is forwarding into compound RTCP packets, following the rules in Section 6.1 of [RFC3550] and Section 5.3 of [RFC8108]. If the middlebox is using RTCP Reporting Groups for its own SSRCs, it **MAY** include RTCP packets from the SSRCs that it is forwarding as part of the compound RTCP packets its reporting source generates.

A middlebox that forwards RTCP SR or RR packets sent by members of a Reporting Group **MUST** forward the corresponding RTCP RGRP SDES items, as described in Section 3.2.1. A middlebox that forwards RTCP SR or RR packets sent by members of a Reporting Group **MUST** also forward the corresponding RTCP RGRS packets, as described in Section 3.2.2. Failure to forward these packets can cause compatibility problems, as described in Section 4.2.

If a middlebox rewrites SSRC values in the RTP and RTCP packets that it is forwarding, then it **MUST** make the corresponding changes in RTCP SDES packets containing RGRP items and in RTCP RGRS packets, to allow them to be associated with the rewritten SSRCs.

3.6. SDP Signaling for Reporting Groups

This document defines the "a=rtcp-rgrp" Session Description Protocol (SDP) [RFC4566] attribute to indicate if the session participant is capable of supporting RTCP Reporting Groups for applications that use SDP for configuration of RTP sessions. It is a property attribute and hence takes no value. The multiplexing category [RFC8859] is IDENTICAL, as the functionality applies at the RTP session level. A participant that proposes the use of RTCP Reporting Groups **SHALL** itself support the reception of RTCP Reporting Groups. The formal definition of this attribute is as follows:

Name: rtcp-rgrp
Value: None
Usage Level: session, media
Charset Dependent: no
Example: a=rtcp-rgrp

When using SDP Offer/Answer [RFC3264], the following procedures are to be used:

Generating the initial SDP offer:
: If the offerer supports the RTCP Reporting Group extensions and is willing to accept RTCP packets containing those extensions, then it **MUST** include an "a=rtcp-rgrp" attribute in the initial offer. If the offerer does not support RTCP Reporting Group extensions or is not willing to accept RTCP packets containing those extensions, then it **MUST NOT** include the "a=rtcp-rgrp" attribute in the offer.

Generating the SDP answer:
: If the SDP offer contains an "a=rtcp-rgrp" attribute, and if the answerer supports RTCP Reporting Groups and is willing to receive RTCP packets using the RTCP Reporting Group extensions, then the answerer **MAY** include an "a=rtcp-rgrp" attribute in the answer and **MAY** send RTCP packets containing the RTCP Reporting Group extensions. If the offer does not

contain an "a=rtcp-rgrp" attribute, or if the offer does contain such an attribute but the answerer does not wish to accept RTCP packets using the RTCP Reporting Group extensions, then the answer **MUST NOT** include an "a=rtcp-rgrp" attribute.

Offerer processing of the SDP answer:
If the SDP answer contains an "a=rtcp-rgrp" attribute and the corresponding offer also contained an "a=rtcp-rgrp" attribute, then the offerer **MUST** be prepared to accept and process RTCP packets that contain the Reporting Group extensions and **MAY** send RTCP packets that contain the Reporting Group extensions. If the SDP answer contains an "a=rtcp-rgrp" attribute but the corresponding offer did not contain the "a=rtcp-rgrp" attribute, then the offerer **MUST** reject the call. If the SDP answer does not contain an "a=rtcp-rgrp" attribute, then the offerer **MUST NOT** send packets containing the RTCP Reporting Group extensions and does not need to process packets containing the RTCP Reporting Group extensions.

In declarative usage of SDP, such as the Real-Time Streaming Protocol (RTSP) [RFC7826] and the Session Announcement Protocol (SAP) [RFC2974], the presence of the attribute indicates that the session participant **MAY** use RTCP Reporting Groups in its RTCP transmissions. An implementation that doesn't explicitly support RTCP Reporting Groups **MAY** join an RTP session as long as it has been verified that the implementation doesn't suffer from the problems discussed in Section 4.2.

4. Properties of RTCP Reporting Groups

This section provides additional information on what the resulting properties are (i.e., resulting effects or impacts) as related to the design specified in Section 3. The content of this section is non-normative.

4.1. Bandwidth Benefits of RTCP Reporting Groups

To understand the benefits of RTCP Reporting Groups, consider a scenario in which the two endpoints in a session each have a hundred sources, of which eight each are sending within any given reporting interval.

For ease of analysis, we can make the simplifying approximation that the duration of the RTCP reporting interval is equal to the total size of the RTCP packets sent during an RTCP interval, divided by the RTCP bandwidth. (This will be approximately true in scenarios where the bandwidth is not so high that the minimum RTCP interval is reached.) To further simplify, we can assume that RTCP senders are following the recommendations regarding compound RTCP packets in [RFC8108]; thus, the per-packet transport-layer overhead will be small relative to the RTCP data. Thus, only the actual RTCP data itself need be considered.

In a report interval in this scenario, there will, as a baseline, be 200 SDES packets, 184 RR packets, and 16 SR packets. This amounts to approximately 6.5 KB of RTCP packets per report interval, assuming 16-byte CNAMEs and no other SDES information.

Using the original "everyone reports on every sender" feedback rules [RFC3550], each of the 184 receivers will send 16 report blocks, and each of the 16 senders will send 15. This amounts to approximately 76 KB of report block traffic per interval; 92% of RTCP traffic consists of report blocks.

If Reporting Groups are used, however, there is only 0.4 KB of reports per interval, with no loss of useful information. Additionally, there will be (assuming 16-byte RGRPs and a single reporting source per Reporting Group) an additional 2.4 KB per cycle of RTCP RGRP SDES items and RGRS packets. Put another way, the unmodified reporting interval per [RFC3550] is approximately 9 times longer than if Reporting Groups are in use.

4.2. Compatibility of RTCP Reporting Groups

The RTCP traffic generated by receivers using RTCP Reporting Groups might appear, to observers unaware of these semantics, to be generated by receivers who are experiencing a network disconnection, as the non-reporting sources appear not to be receiving a given sender at all.

This could be a potentially critical problem for such a sender using RTCP for congestion control, as such a sender might think that it is sending so much traffic that it is causing complete congestion collapse.

However, such an interpretation of the session statistics would require a fairly sophisticated RTCP analysis. Any receiver of RTCP statistics that is just interested in information about itself needs to be prepared for the possibility that any given reception report might not contain information about a specific media source, because reception reports in large conferences can be round-robined.

Thus, the extent to which such backward-compatibility issues would actually cause trouble in practice is unclear.

5. Security Considerations

The security considerations of [RFC3550] and [RFC8108] apply. If the RTP/AVPF profile is in use, then the security considerations of [RFC4585] (and [RFC5104], if used) also apply. If RTCP XR is used, the security considerations of [RFC3611], including security considerations regarding any XR report blocks used, also apply.

The RTCP RGRP SDES item is vulnerable to malicious modifications unless integrity protection is used. A modification of this item's length field causes the parsing of the RTCP packet in which it is contained to fail. Depending on the implementation, parsing of the full compound RTCP packet can also fail, causing the whole packet to be discarded. A modification of the value of this SDES item would make the receiver of the report think that the sender of the report was a member of a different RTCP Reporting Group. This will potentially create an inconsistency, when the RGRS reports the source as being in the same Reporting Group as another source with another Reporting Group identifier. The impacts on a receiver implementation that such inconsistencies could cause are difficult to fully predict. One case is that when congestion control or other adaptation mechanisms are used, an inconsistent report can result in a media sender reducing its bitrate. However, a direct modification of the RR or a feedback message itself would be a more efficient attack and would be equally costly to perform.

The new RGRS RTCP packet type is very simple. The common RTCP packet type header shares the same security risks as those that affect previous RTCP packet types. Errors or modification of the length field can cause the full compound packet to fail header validation (see Appendix A.2 of [RFC3550]), resulting in the whole compound RTCP packet being discarded. Modification of the SC field or the P field would cause an inconsistency when processing the RTCP packet, likely resulting in the packet being classified as invalid. A modification of the PT field would cause the packet to be interpreted according to some other packet type's rules. In such a case, the result might be more or less predictable but would be specific to the packet type. Modification of the "SSRC of packet sender" field would attribute this packet to another sender, resulting in a receiver believing that the Reporting Group also applies for this SSRC, if it exists. If it doesn't exist, unless corresponding modifications are also done on an SR/RR packet and an SDES packet, the RTCP packet **SHOULD** be discarded. If consistent changes are done, such a scenario could be part of a resource exhaustion attack on a receiver implementation. Modification of the "List of SSRCs for the Reporting Source(s)" field would change the SSRC the receiver expects to report on behalf of this SSRC. If that SSRC exists, this situation could potentially change the Reporting Group used for this SSRC. A change to another Reporting Group belonging to another endpoint is likely detectable, as there would be a mismatch between the SSRC of the packet sender's endpoint information, transport addresses, SDES CNAME, etc., and the corresponding information from the Reporting Group indicated.

In general, the Reporting Group is providing limited-impact attacks on the endpoints. The most significant result from a deliberate attack would be to cause the information to be discarded or be inconsistent, including the discarding of all RTCP packets that are modified. This causes a lack of information at any receiver entity, possibly disregarding the endpoint's participation in the session.

To protect against such attacks from external non-trusted entities, integrity and source authentication **SHOULD** be applied. This can be done, for example, by using the Secure Real-time Transport Protocol (SRTP) [RFC3711] with appropriate key management; other options exist, as discussed in "Options for Securing RTP Sessions" [RFC7201].

The Reporting Group Identifier has properties that could potentially impact privacy. If this identifier were to be generated by an implementation in a way that makes it long-term stable or predictable, it could be used for tracking a particular endpoint. Therefore, it is **RECOMMENDED** that it be generated as a short-term persistent RGRP, following the rules for short-term persistent CNAMEs in [RFC7022]. The rest of the information revealed, i.e., the SSRCs, the size of the Reporting Group, and the number of reporting sources in a Reporting Group, is of a less sensitive nature, considering that the SSRCs and the communication would be revealed without this extension anyway. By encrypting the Reporting Group extensions, the confidentiality of the SSRC values would be preserved, but the values can still be revealed if SRTP [RFC3711] is used. The size of the Reporting Groups and the number of reporting sources are likely determinable from analysis of the packet pattern and sizes. However, this information appears to have limited value.

6. IANA Considerations

IANA has registered a new RTCP RGRP SDES item in the "RTP SDES Item Types" registry, as follows:

Value	Abbrev	Name	Reference
11	RGRP	Reporting Group Identifier	RFC 8861

Table 1: New RTCP RGRP SDES Item: Reporting Group Identifier

The definition of the RTCP RGRP SDES item is given in Section 3.2.1 of this memo.

IANA has registered a new RTCP packet type in the "RTCP Control Packet Types (PT)" registry, as follows:

Value	Abbrev	Name	Reference
212	RGRS	Reporting Group Reporting Sources	RFC 8861

Table 2: New RTCP Packet Type: Reporting Group Reporting Sources

The definition of the RTCP RGRS packet type is given in Section 3.2.2 of this memo.

IANA has also registered a new SDP attribute.

SDP Attribute ("att-field"):

Contact Name: IESG

Contact Email: iesg@ietf.org

Attribute name: rtcp-rgrp

Long form: RTCP Reporting Groups

Type of name: att-field

Type of attribute: Media or session level

Subject to charset: No

Purpose: To negotiate or configure the use of the RTCP Reporting Group extension

Reference: RFC 8861

Value: None

Mux Category: IDENTICAL

The definition of the "a=rtcp-rgrp" SDES attribute is given in Section 3.6 of this memo.

7. References

7.1. Normative References

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