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RTP Header Extension for RTCP Source Description Items
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Abstract

Source Description (SDES) items are normally transported in RTP control protocol (RTCP). In some cases it can be beneficial to speed up the delivery of these items. Mainly when a new source (SSRC) joins an RTP session and the receivers needs this source's identity, relation to other sources, or its synchronization context, all of which may be fully or partially identified using SDES items. To enable this optimization, this document specifies a new RTP header extension that can carry SDES items.

Status of This Memo

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1. Introduction

This specification defines an RTP header extension [RFC3550][RFC5285] that can carry RTCP source description (SDES) items. Normally the SDES items are carried in their own RTCP packet type [RFC3550]. By including selected SDES items in a header extension the determination of relationship and synchronization context for new RTP streams (SSRCs) in an RTP session can be optimized. Which relationship and what information depends on the SDES items carried. This becomes a complement to using only RTCP for SDES Item delivery.

It is important to note that not all SDES items are appropriate to transmit using RTP header extensions. Some SDES items performs binding or identifies synchronization context with strict timeliness requirements, while many other SDES items do not have such requirements. In addition, security and privacy concerns for the SDES item information need to be considered. For example, the Name and Location SDES items are highly sensitive from a privacy perspective and should not be transported over the network without strong security. No use case has identified where this information is required at the same time as the first RTP packets arrive. A few seconds delay before such information is available to the receiver appears acceptable. Therefore only appropriate SDES items will be registered for use with this header extension, such as CNAME.

First, some requirements language and terminology are defined. The following section motivates why this header extension is sometimes required or at least provides a significant improvement compared to waiting for regular RTCP packet transmissions of the information. This is followed by a specification of the header extension and usage recommendations. Next, a sub-space of the header-extension URN is defined to be used for existing and future SDES items, and then the appropriate existing SDES items are registered.

2. Definitions

2.1. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC2119].

2.2. Terminology

This document uses terminology defined in "A Taxonomy of Semantics and Mechanisms for Real-Time Transport Protocol (RTP) Sources" [RFC7656]. In particular the following definitions:

Media Source

RTP Stream

Media Encoder

Participant

3. Motivation

Source Description (SDES) items are associated with a particular SSRC and thus RTP stream. The source description items provide various meta data associated with the SSRC. How important it is to have this data no later than when receiving the first RTP packets depends on the item itself. The CNAME item is one item that is commonly needed either at reception of the first RTP packet for this SSRC, or at least by the time the first media can be played out. If it is not available, the synchronization context cannot be determined and thus any related streams cannot be correctly synchronized. Thus, this is a valuable example for having this information early when a new RTP stream is received.

The main reason for new SSRCs in an RTP session is when media sources are added. This can be either because an end-point is adding a new actual media source, or additional participants in a multi-party session are added to the session. Another reason for a new SSRC can be an SSRC collision that forces both colliding parties to select new SSRCs.

For the case of rapid media synchronization, one may use the RTP header extension for Rapid Synchronization of RTP Flows [RFC6051]. This header extension carries the clock information present in the RTCP sender report (SR) packets. It however assumes that the CNAME binding is known, which can be provided via signaling [RFC5576] in some cases, but not all. Thus an RTP header extension for carrying SDES items like CNAME is a powerful combination to enable rapid synchronization in all cases.

The Rapid Synchronization of RTP Flows specification does provide an analysis of the initial synchronization delay for different sessions depending on number of receivers as well as on session bandwidth (Section 2.1 of [RFC6051]). These results are applicable also for other SDES items that have a similar time dependency until the information can be sent using RTCP. These figures can be used to determine the benefit of reducing the initial delay before information is available for some use cases.

Rapid Synchronization of RTP Flows [RFC6051] also discusses the case of late joiners, and defines an RTCP Feedback format to request synchronization information, which is another potential use case for SDES items in RTP header extension. It would for example be natural to include CNAME SDES item with the header extension containing the NTP formatted reference clock to ensure synchronization.

There is an another SDES item that can benefit from timely delivery, and an RTP header extension SDES item was therefore defined for it:

MID: This is a media description identifier that matches the value of the Session Description Protocol (SDP) [RFC4566] a=mid attribute, to associate RTP streams multiplexed on the same transport with their respective SDP media description as described in [I-D.ietf-mmusic-sdp-bundle-negotiation].

4. Specification

This section first specifies the SDES item RTP header extension format, followed by some usage considerations.

4.1. SDES Item Header Extension

An RTP header extension scheme allowing for multiple extensions is defined in "A General Mechanism for RTP Header Extensions" [RFC5285]. That specification defines both short and long item headers. The short headers (One-byte) are restricted to 1 to 16 bytes of data, while the long format (Two-byte) supports a data length of 0 to 255 bytes. Thus the RTP header extension formats are capable of supporting any SDES item from a data length perspective.

The ID field, independent of short or long format, identifies both the type of RTP header extension and, in the case of the SDES item header extension, the type of SDES item. The mapping is done in signaling by identifying the header extension and SDES item type using a URN, which is defined in the IANA consideration (Section 5) for the known SDES items appropriate to use.

4.1.1. One-Byte Format

The one-byte header format for an SDES item extension element consists of the one-byte header (defined in Section 4.2 of [RFC5285]), which consists of a 4-bit ID followed by a 4-bit length field (len) that identifies the number of data bytes (len value +1) following the header. The data part consists of len+1 bytes of UTF-8 text. The type of text and its mapping to the SDES item type is determined by the ID field value.

```
 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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  ID   |  len  | SDES Item text value ...                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 1

4.1.2. Two-Byte Format

The two-byte header format for an SDES item extension element consists of the two-byte header (defined in Section 4.3 of [RFC5285]), which consists of an 8-bit ID followed by an 8-bit length field (len) that identifies the number of data bytes following the header. The data part consists of len bytes of UTF-8 text. The type of text and its mapping to the SDES item type is determined by the ID field value.

```
 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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|      ID       |      len      |  SDES Item text value ...     |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 2

4.2. Usage of the SDES Item Header Extension

This section discusses various usage considerations; which form of header extension to use, the packet expansion, and when to send SDES items in header extension.

4.2.1. One or Two Byte Headers

The RTP header extensions for SDES items MAY use either the one-byte or two-byte header formats, depending on the text value size for the used SDES items and the requirement from any other header extensions used. The one-byte header SHOULD be used when all non SDES item header extensions supports the one-byte format and all SDES item text values contain at most 16 bytes. Note that the RTP header extension specification does not allow mixing one-byte and two-byte headers for the same RTP stream (SSRC), so if the value size of any of the SDES items value requires the two-byte header, then all other header extensions MUST also use the two-byte header format.

For example using CNAMEs that are generated according to "Guidelines for Choosing RTP Control Protocol (RTCP) Canonical Names (CNAMEs)" [RFC7022], using short term persistent values, and if 96-bit random values prior to base64 encoding are sufficient, then they will fit into the one-byte header format.

An RTP middlebox needs to take care choosing between one-byte headers and two-byte headers when creating the first packets for an outgoing stream (SSRC) with header extensions. First of all it needs to consider all the header extensions that may potentially be used. Secondly, it needs to know the size of the SDES items that are going

to be included, and use two bytes headers if any are longer than 16 bytes. An RTP middlebox that forwards a stream, i.e., not mixing it or combing it with other streams, may be able to base its choice on the header size in incoming streams. This is assuming that the middlebox does not modify the stream or add additional header extensions to the stream it sends, in which case it needs to make its own decision.

4.2.2. MTU and Packet Expansion

The RTP packet size will clearly increase when a header extension is included. How much depends on the type of header extensions and their data content. The SDES items can vary in size. There are also some use-cases that require transmitting multiple SDES items in the same packet to ensure that all relevant data reaches the receiver. An example of that is when both CNAME, a MID, and the rapid time synchronization extension from RFC 6051 are needed. Such a combination is quite likely to result in at least 16+3+8 bytes of data plus the headers, which will be another 7 bytes for one-byte headers, plus two bytes of header padding to make the complete header extension 32-bit word aligned, thus in total 36 bytes.

If the packet expansion cannot be taken into account when producing the RTP payload, it can cause an issue. An RTP payload that is created to meet a particular IP level Maximum Transmission Unit (MTU), taking the addition of IP/UDP/RTP headers but not RTP header extensions into account, could exceed the MTU when the header extensions are present, thus resulting in IP fragmentation. IP fragmentation is known to negatively impact the loss rate due to middleboxes unwilling or not capable of dealing with IP fragments, as well as increasing the target surface for other types of packet losses.

As this is a real issue, the media encoder and payload packetizer should be flexible and be capable of handling dynamically varying payload size restrictions to counter the packet expansion caused by header extensions. If that is not possible, some reasonable worst case packet expansion should be calculated and used to reduce the RTP payload size of all RTP packets the sender transmits.

4.2.3. Transmission Considerations

The general recommendation is to only send header extensions when needed. This is especially true for SDES items that can be sent in periodic repetitions of RTCP throughout the whole session. Thus, the different usages (Section 4.2.4) have different recommendations. First some general considerations for getting the header extensions delivered to the receiver:

1. The probability for packet loss and burst loss determine how many

repetitions of the header extensions will be required to reach a targeted delivery probability, and if burst loss is likely, what distribution would be needed to avoid getting all repetitions of the header extensions lost in a single burst.

2. If a set of packets are all needed to enable decoding, there is commonly no reason for including the header extension in all of these packets, as they share fate. Instead, at most one instance of the header extension per independently decodable set of media data would be a more efficient use of the bandwidth.

3. How early the SDES item information is needed, from the first received RTP data or only after some set of packets are received, can guide if the header extension(s) should be in all of the first N packets or be included only once per set of packets, for example once per video frame.

4. The use of RTP level robustness mechanisms, such as RTP retransmission [RFC4588], or Forward Error Correction, e.g., [RFC5109] may treat packets differently from a robustness perspective, and SDES header extensions should be added to packets that get a treatment corresponding to the relative importance of receiving the information.

As a summary, the number of header extension transmissions should be tailored to a desired probability of delivery taking the receiver population size into account. For the very basic case, N repetitions of the header extensions should be sufficient, but may not be optimal. N is selected so that the header extension target delivery probability reaches 1-P^N, where P is the probability of packet loss. For point to point or small receiver populations, it might also be possible to use feedback, such as RTCP, to determine when the information in the header extensions has reached all receivers and stop further repetitions. Feedback that can be used includes the RTCP XR Loss RLE report block [RFC3611], which will indicate successful delivery of particular packets. If the RTP/AVPF Transport Layer Feedback Messages for generic NACK [RFC4585] is used, it can indicate the failure to deliver an RTP packet with the header extension, thus indicating the need for further repetitions. The normal RTCP report blocks can also provide an indicator of successful delivery, if no losses are indicated for a reporting interval covering the RTP packets with the header extension. Note that loss of an RTCP packet reporting on an interval where RTP header extension packets were sent, does not necessarily mean that the RTP header extension packets themselves were lost.

4.2.4. Different Usages

4.2.4.1. New SSRC

A new SSRC joins an RTP session. As this SSRC is completely new for everyone, the goal is to ensure, with high probability, that all RTP session participants receives the information in the header extension. Thus, header extension transmission strategies that allow some margins in the delivery probability should be considered.

4.2.4.2. Late Joiner

In a multi-party RTP session where one or a small number of receivers join a session where the majority of receivers already have all necessary information, the use of header extensions to deliver relevant information should be tailored to reach the new receivers. The trigger to send header extensions can for example either be RTCP from new receiver(s) or an explicit request like the Rapid Resynchronization Request defined in [RFC6051]. In centralized topologies where an RTP middlebox is present, it can be responsible for transmitting the known information, possibly stored, to the new session participant only, and not repeat it to all the session participants.

4.2.4.3. Information Change

If the SDES information is tightly coupled with the RTP data, and the SDES information needs to be updated, then the use of the RTP header extension is superior to RTCP. Using the RTP header extension ensures that the information is updated on reception of the related RTP media, ensuring synchronization between the two. Continued use of the old SDES information can lead to undesired effects in the application. Thus, header extension transmission strategies with high probability of delivery should be chosen.

4.2.5. SDES Items in RTCP

The RTP header extension information, i.e., SDES items, can and will be sent also in RTCP. Therefore, it is worth making some reflections on this interaction. As an alternative to the header extension, it is possible to schedule a non-regular RTCP packet transmission containing important SDES items, if one uses an RTP/AVPF-based RTP profile. Depending on which mode one's RTCP feedback transmitter is working on, extra RTCP packets may be sent as immediate or early packets, enabling more timely SDES information delivery.

There are however two aspects that differ between using RTP header extensions and any non-regular transmission of RTCP packets. First,

as the RTCP packet is a separate packet, there is no direct relation and also no fate sharing between the relevant media data and the SDES information. The order of arrival for the packets will matter. With a header-extension, the SDES items can be ensured to arrive if the media data to play out arrives. Secondly, it is difficult to determine if an RTCP packet is actually delivered. This, as the RTCP packets lack both sequence number and a mechanism providing feedback on the RTCP packets themselves.

4.2.6. Update Flaps

The SDES item may arrive both in RTCP and in RTP header extensions, potentially causing the value to flap back and forth at the time of updating. There at least two reasons for these flaps. The first one is packet reordering, where a pre-update RTP or RTCP packet with an SDES item is delivered to the receiver after the first RTP/RTCP packet with the updated value. The second reason is the different code-paths for RTP and RTCP in implementations. An update to the sender's SDES item parameter can take a different time to propagate to the receiver than the corresponding media data. For example, an RTCP packet with the SDES item included that may have been generated prior to the update can still reside in a buffer and be sent unmodified. The update of the item's value can at the same time cause RTP packets to be sent including the header extension, prior to the RTCP packet being sent.

However, most of these issues can be avoided by the receiver performing some checks before updating the receiver's stored value. To handle flaps caused by reordering, SDES items received in RTP packets with the same or a lower extended sequence number than the last change MUST NOT be applied, i.e., discard items that can be determined to be older than the current one. For compound RTCP packets, which will contain a Sender Report (SR) packet (assuming an active RTP sender), the receiver can use the RTCP SR Timestamp field to determine at what approximate time it was transmitted. If the timestamp is earlier than the last received RTP packet with a header extension carrying an SDES item, and especially if carrying a previously used value, the SDES item in the RTCP SDES packet can be ignored. Note that media processing and transmission pacing can easily cause the RTP header timestamp field as well as the RTCP SR timestamp field to not match with the actual transmission time.

5. IANA Considerations

This section makes the following requests to IANA:

- o Create a new sub-registry reserved for RTCP SDES items with the URN sub-space "urn:ietf:params:rtp-hdrext:sdes:" in the RTP Compact Header Extensions registry.

- o Register the SDES items appropriate for use with the RTP header extension defined in this document.

RFC-editor note: Please replace all occurrences of RFCXXXX with the RFC number this specification receives when published.

5.1. Registration of an SDES Base URN

IANA is requested to register the below entry in the RTP Compact Header Extensions registry:

```
Extension URI: urn:ietf:params:rtp-hdrext:sdes
Description:   Reserved as base URN for RTCP SDES items that are also
               defined as RTP Compact header extensions.
Contact:       Authors of [RFCXXXX]
Reference:     [RFCXXXX]
```

The reason to register a base URN for an SDES sub-space is that the name represents an RTCP Source Description item, where a specification is strongly recommended [RFC3550].

5.2. Creation of an SDES Sub-Registry

IANA is requested to create a sub-registry to the RTP Compact Header Extensions registry, with the same basic requirements, structure and layout as the RTP Compact Header Extensions registry.

- o Registry name: RTP SDES Compact Header Extensions o Specification: RFCXXXX and RFCs updating RFCXXXX

o Information required: Same as for RTP Header Extensions [RFC5285] registry

o Review process: Same as for RTP Header Extensions [RFC5285] registry, with the following requirements added to the expert review:

1. Any registration using an Extension URI that starts with "urn:ietf:params:rtp-hdrext:sdes:" (Section 5.1) MUST also have a registered Source Description item in the "RTP SDES item types" registry.

2. A security and privacy consideration for the SDES item MUST be provided with the registration.

3. Information MUST be provided on why this SDES item requires timely delivery, motivating it to be transported in a header extension rather than as RTCP only.

o Size and format of entries: Same as for RTP Header Extensions [RFC5285] registry.

o Initial assignments: See Section 5.3 below.

5.3. Registration of SDES Items

It is requested that the following SDES item is registered in the newly formed RTP SDES Compact Header Extensions registry:

```
Extension URI: urn:ietf:params:rtp-hdrext:sdes:cname
Description:   Source Description: Canonical End-Point Identifier
               (SDES CNAME)
Contact:       Authors of [RFCXXXX]
Reference:     [RFCXXXX]
```

6. Security Considerations

Source Description items may contain data that are sensitive from a security perspective. There are SDES items that are or may be sensitive from a user privacy perspective, like CNAME, NAME, EMAIL, PHONE, LOC and H323-CADDR. Some may contain sensitive information, like NOTE and PRIV, while others may be sensitive from profiling implementations for vulnerability or other reasons, like TOOL. The CNAME sensitivity can vary depending on how it is generated and what persistence it has. A short term CNAME identifier generated using a random number generator [RFC7022] may have minimal security implications, while a CNAME of the form user@host has privacy concerns, and a CNAME generated from a MAC address has long term tracking potentials.

In RTP sessions where any type of confidentiality protection is enabled for RTCP, the SDES item header extensions MUST also be protected. This implies that to provide confidentiality, users of SRTP need to implement and use encrypted header extensions per [RFC6904]. The security level that is applied to RTCP packets carrying SDES items SHOULD also be applied to SDES items carried as RTP header extensions. If the security level is chosen to be different for an SDES item in RTCP and RTP header extension, it is important to motivate the exception, and to consider the security

properties as the worst in each aspect for the different configurations.

The general RTP header extension mechanism [RFC5285] does not itself contain any functionality that is a significant risk for a denial of service attack, neither from processing nor storage requirements. The extension for SDES items defined, can potentially be a risk. The risk depends on the received SDES item and its content. If the SDES item causes the receiver to perform a large amount of processing, create significant storage structures, or emit network traffic, such a risk does exist. The CNAME SDES item in the RTP header extension is only a minor risk, as reception of a CNAME item will create an association between the stream carrying the SDES item and other RTP streams with the same SDES item. This usually results in time synchronizing the media streams, thus some additional processing is performed. However, the application's media quality is likely more affected by an erroneous or changing association and media synchronization, than the application quality impact caused by the additional processing.

As the SDES items are used by the RTP based application to establish relationships between RTP streams or between an RTP stream and information about the originating participant, there SHOULD be strong integrity protection and source authentication of the header extensions. If not, an attacker can modify the SDES item value to create erroneous relationship bindings in the receiving application. For information regarding options for securing RTP, see [RFC7201].

7. Acknowledgments

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