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Guidelines for using the Multiplexing Features of RTP
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Abstract

Real-time Transport Protocol (RTP) is a flexible protocol possible to use in a wide range of applications and network and system topologies. This flexibility and the implications of different choices should be understood by any application developer using RTP. To facilitate that understanding, this document contains an in-depth discussion of the usage of RTP's multiplexing points; the RTP session and the Synchronisation Source Identifier (SSRC). The document tries to give guidance and source material for an analysis on the most suitable choices for the application being designed.

Status of this Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on January 17, 2013.

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1. Introduction

Real-time Transport Protocol (RTP) [RFC3550] is a commonly used protocol for real-time media transport. It is a protocol that provides great flexibility and can support a large set of different applications. RTP has several multiplexing points designed for different purposes. These enable support of multiple media streams and switching between different encoding or packetization of the media. By using multiple RTP sessions, sets of media streams can be structured for efficient processing or identification. Thus the question for any RTP application designer is how to best use the RTP session, the SSRC and the payload type to meet the application's needs.

The purpose of this document is to provide clear information about the possibilities of RTP when it comes to multiplexing. The RTP application designer should understand the implications that come from a particular usage of the RTP multiplexing points. The document will recommend against some usages as being unsuitable, in general or for particular purposes.

RTP was from the beginning designed for multiple participants in a communication session. This is not restricted to multicast, as some may believe, but also provides functionality over unicast, using either multiple transport flows below RTP or a network node that re-distributes the RTP packets. The re-distributing node can for example be a transport translator (relay) that forwards the packets unchanged, a translator performing media or protocol translation in addition to forwarding, or an RTP mixer that creates new conceptual sources from the received streams. In addition, multiple streams may occur when a single endpoint have multiple media sources, like multiple cameras or microphones that need to be sent simultaneously.

This document has been written due to increased interest in more advanced usage of RTP, resulting in questions regarding the most appropriate RTP usage. The limitations in some implementations, RTP/RTCP extensions, and signalling has also been exposed. It is expected that some limitations will be addressed by updates or new extensions resolving the shortcomings. The authors also hope that clarification on the usefulness of some functionalities in RTP will result in more complete implementations in the future.

The document starts with some definitions and then goes into the existing RTP functionalities around multiplexing. Both the desired behaviour and the implications of a particular behaviour depend on which topologies are used, which requires some consideration. This is followed by a discussion of some choices in multiplexing behaviour and their impacts. Some arch-types of RTP usage are discussed.

Finally, some recommendations and examples are provided.

This document is currently an individual contribution, but it is the intention of the authors that this should become a WG document that objectively describes and provides suitable recommendations for which there is WG consensus. Currently this document only represents the views of the authors. The authors gladly accept any feedback on the document and will be happy to discuss suitable recommendations.

2. Definitions

2.1. Terminology

The following terms and abbreviations are used in this document:

Endpoint: A single entity sending or receiving RTP packets. It may be decomposed into several functional blocks, but as long as it behaves a single RTP stack entity it is classified as a single endpoint.

Multiparty: A communication situation including multiple end-points. In this document it will be used to refer to situations where more than two end-points communicate.

Media Source: The source of a stream of data of one Media Type, It can either be a single media capturing device such as a video camera, a microphone, or a specific output of a media production function, such as an audio mixer, or some video editing function. Sending data from a Media Source may cause multiple RTP sources to send multiple Media Streams.

Media Stream: A sequence of RTP packets using a single SSRC that together carries part or all of the content of a specific Media Type from a specific sender source within a given RTP session.

RTP Source: The originator or source of a particular Media Stream. Identified using an SSRC in a particular RTP session. An RTP source is the source of a single media stream, and is associated with a single endpoint and a single Media Source. An RTP Source is just called a Source in RFC 3550.

Media Sink: A recipient of a Media Stream. The endpoint sinking media are Identified using one or more SSRCs. There may be more than one Media Sink for one RTP source.

CNAME: "Canonical name" - identifier associated with one or more RTP sources from a single endpoint. Defined in the RTP specification [RFC3550]. A CNAME identifies a synchronisation context. A CNAME is associated with a single endpoint, although some RTP nodes will use an end-points CNAME on that end-points behalf. An endpoint may use multiple CNAMEs. A CNAME is intended to be globally unique and stable for the full duration of a communication session. [RFC6222] gives updated guidelines for choosing CNAMEs.

Media Type: Audio, video, text or data whose form and meaning are defined by a specific real-time application.

Multiplex: The operation of taking multiple entities as input, aggregating them onto some common resource while keeping the individual entities addressable such that they can later be fully and unambiguously separated (de-multiplexed) again.

RTP Session: As defined by [RFC3550], the endpoints belonging to the same RTP Session are those that share a single SSRC space. That is, those endpoints can see an SSRC identifier transmitted by any one of the other endpoints. An endpoint can receive an SSRC either as SSRC or as CSRC in RTP and RTCP packets. Thus, the RTP Session scope is decided by the endpoints' network interconnection topology, in combination with RTP and RTCP forwarding strategies deployed by endpoints and any interconnecting middle nodes.

RTP Session Group: One or more RTP sessions that are used together to perform some function. Examples are multiple RTP sessions used to carry different layers of a layered encoding. In an RTP Session Group, CNAMEs are assumed to be valid across all RTP sessions, and designate synchronisation contexts that can cross RTP sessions.

Source: Term that should not be used alone. An RTP Source, as identified by its SSRC, is the source of a single Media Stream; a Media Source can be the source of mutiple Media Streams.

SSRC: An RTP 32-bit unsigned integer used as identifier for a RTP Source.

CSRC: Contributing Source, A SSRC identifier used in a context, like the RTP headers CSRC list, where it is clear that the Media Source is not the source of the media stream, instead only a contributor to the Media Stream.

Signalling: The process of configuring endpoints to participate in one or more RTP sessions.

(tbd: The terms "SSRC multiplexing" and "session multiplexing" are confusing, with unclear historical meanings; they need to be removed from this document in the interests of clarity)

2.2. Subjects Out of Scope

This document is focused on issues that affect RTP. Thus, issues that involve signalling protocols, such as whether SIP, Jingle or some other protocol is in use for session configuration, the particular syntaxes used to define RTP session properties, or the constraints imposed by particular choices in the signalling protocols, are mentioned only as examples in order to describe the RTP issues more precisely.

This document assumes the applications will use RTCP. While there are such applications that don't send RTCP, they do not conform to the RTP specification, and thus should be regarded as reusing the RTP packet format, not as implementing the RTP protocol.

3. RTP Concepts

This section describes the existing RTP tools that are particularly important when discussing multiplexing of different media streams.

3.1. Session

The RTP Session is the highest semantic level in RTP and contains all of the RTP functionality. RTP itself has no normative statements about the relationship between different RTP sessions.

Identifier: RTP in itself does not contain any Session identifier, but relies either on the underlying transport or on the used signalling protocol, depending on in which context the identifier is used (e.g. transport or signalling). Due to this, a single RTP Session may have multiple associated identifiers belonging to different contexts.

Position: Depending on underlying transport and signalling protocol. For example, when running RTP on top of UDP, an RTP endpoint can identify and delimit an RTP Session from other RTP Sessions through the UDP source and destination transport address, consisting of network address and port number(s).

Commonly, RTP and RTCP use separate ports and the destination transport address is in fact an address pair, but in the case of RTP/RTCP multiplex [RFC5761] there is only a single port. Another example is SDP signalling [RFC4566], where the grouping framework [RFC5888] uses an identifier per "m="-line. If there is a one-to-one mapping between "m="-line and RTP Session, that grouping framework identifier can identify a single RTP Session.

Usage: Identify separate RTP Sessions.

Uniqueness: Globally unique, but identity can only be detected by the general communication context for the specific endpoint.

Inter-relation: Depending on the underlying transport and signalling protocol.

Special Restrictions: None.

A RTP source in an RTP session that changes its source transport address during a session must also choose a new SSRC identifier to avoid being interpreted as a looped source.

The set of participants considered part of the same RTP Session is defined by the RTP specification [RFC3550] as those that share a single SSRC space. That is, those participants that can see an SSRC identifier transmitted by any one of the other participants. A participant can receive an SSRC either as SSRC or CSRC in RTP and RTCP packets. Thus, the RTP Session scope is decided by the participants' network interconnection topology, in combination with RTP and RTCP forwarding strategies deployed by endpoints and any interconnecting middle nodes.

3.2. SSRC

An SSRC identifies a RTP source or a media sink. For end-points that both source and sink media streams its SSRCs are used in both roles. At any given time, a RTP source has one and only one SSRC - although that may change over the lifetime of the RTP source or sink. An RTP Session serves one or more RTP sources, each sending a Media Stream.

Identifier: Synchronisation Source (SSRC), 32-bit unsigned number.

Position: In every RTP and RTCP packet header. May be present in RTCP payload. May be present in SDP signalling.

Usage: Identify individual RTP sources and media sinks within an RTP Session. Refer to individual RTP sources and media sinks in RTCP messages and SDP signalling.

Uniqueness: Randomly chosen, intended to be globally unique within an RTP Session and not dependent on network address. SSRC value collisions may occur and must be handled as specified in RTP [RFC3550].

Inter-relation: SSRC belonging to the same synchronisation context (originating from the same endpoint), within or between RTP Sessions, are indicated through use of identical SDES CNAME items in RTCP compound packets with those SSRC as originating source. SDP signalling can provide explicit SSRC grouping [RFC5576]. When CNAME is inappropriate or insufficient, there exist a few other methods to relate different SSRC. One such case is session-based RTP retransmission [RFC4588]. In some cases, the same SSRC Identifier value is used to relate streams in two different RTP Sessions, such as in Multi-Session Transmission of scalable video [RFC6190].

Special Restrictions: All RTP implementations must be prepared to use procedures for SSRC collision handling, which results in an SSRC number change. A RTP source that changes its RTP Session identifier (e.g. source transport address) during a session must also choose a new SSRC identifier to avoid being interpreted as looped source.

Note that RTP sequence number and RTP timestamp are scoped by SSRC and thus independent between different SSRCs.

A RTP source having an SSRC identifier can be of different types:

Real: Connected to a "physical" media source, for example a camera or microphone.

Conceptual: A source with some attributed property generated by some network node, for example a filtering function in an RTP mixer that provides the most active speaker based on some criteria, or a mix representing a set of other sources.

Media Sink: A source that does not generate any RTP media stream in itself (e.g. an endpoint only receiving in an RTP session), but anyway need a sender SSRC for use as source in RTCP reports.

Note that a endpoint that generates more than one media type, e.g. a conference participant sending both audio and video, need not (and commonly should not) use the same SSRC value across RTP sessions.

RTCP Compound packets containing the CNAME SDES item is the designated method to bind an SSRC to a CNAME, effectively cross-correlating SSRCs within and between RTP Sessions as coming from the same endpoint. The main property attributed to SSRCs associated with the same CNAME is that they are from a particular synchronisation context and may be synchronised at playback.

Note also that RTP sequence number and RTP timestamp are scoped by SSRC and thus independent between different SSRCs.

An RTP receiver receiving a previously unseen SSRC value must interpret it as a new source. It may in fact be a previously existing source that had to change SSRC number due to an SSRC conflict. However, the originator of the previous SSRC should have ended the conflicting source by sending an RTCP BYE for it prior to starting to send with the new SSRC, so the new SSRC is anyway effectively a new source.

3.3. CSRC

The Contributing Source (CSRC) is not a separate identifier, but an usage of the SSRC identifier. It is optionally included in the RTP header as list of up to 15 contributing RTP sources. CSRC shares the SSRC number space and specifies which set of SSRCs that has contributed to the RTP payload. However, even though each RTP packet and SSRC can be tagged with the contained CSRCs, the media representation of an individual CSRC is in general not possible to extract from the RTP payload since it is typically the result of a media mixing (merge) operation (by an RTP mixer) on the individual media streams corresponding to the CSRC identifiers. The exception is the case when only a single CSRC is indicated as this represent forwarding of a media stream, possibly modified. The RTP header extension for Mixer-to-Client Audio Level Indication [RFC6465] expands on the receivers information about a packet with CSRC list. Due to these restrictions, CSRC will not be considered a fully qualified multiplex point and will be disregarded in the rest of this document.

3.4. Payload Type

Each Media Stream utilises one or more encoding formats, identified by the Payload Type.

The Payload Type is not a multiplexing point. Appendix A gives some of the many reasons why attempting to use it as a multiplexing point will have bad results.

Identifier: Payload Type number.

Position: In every RTP header and in SDP signalling.

Usage: Identify a specific Media Stream encoding format. The format definition may be taken from [RFC3551] for statically allocated Payload Types, but should be explicitly defined in signalling, such as SDP, both for static and dynamic Payload Types. The term "format" here includes whatever can be described by out-of-band signalling means. In SDP, the term "format" includes media type, RTP timestamp sampling rate, codec, codec configuration, payload format configurations, and various robustness mechanisms such as redundant encodings [RFC2198].

Uniqueness: Scoped by sending endpoint within an RTP Session. To avoid any potential for ambiguity, it is desirable that payload types are unique across all sending endpoints within an RTP session, but this is often not true in practice. All SSRC in an RTP session sent from an single endpoint share the same Payload Types definitions. The RTP Payload Type is designed such that only a single Payload Type is valid at any time instant in the SSRC's RTP timestamp time line, effectively time-multiplexing different Payload Types if any change occurs. Used Payload Type may change on a per-packet basis for an SSRC, for example a speech codec making use of generic Comfort Noise [RFC3389].

Inter-relation: There are some uses where Payload Type numbers need to be unique across RTP Sessions. This is for example the case in Media Decoding Dependency [RFC5583] where Payload Types are used to describe media dependency across RTP Sessions. Another example is session-based RTP retransmission [RFC4588].

Special Restrictions: Using different RTP timestamp clock rates for the RTP Payload Types in use in the same RTP Session have issues such as loss of synchronisation. Payload Type clock rate switching requires some special consideration that is described in the multiple clock rates specification [I-D.ietf-avtext-multiple-clock-rates].

If there is a true need to send multiple Payload Types for the same SSRC that are valid for the same RTP Timestamps, then redundant encodings [RFC2198] can be used. Several additional constraints than the ones mentioned above need to be met to enable this use, one of which is that the combined payload sizes of the different Payload Types must not exceed the transport MTU.

Other aspects of RTP payload format use are described in RTP Payload HowTo [I-D.ietf-payload-rtp-howto].

4. Multiple Streams Alternatives

The reasons why an endpoint may choose to send multiple media streams are widespread. In the below discussion, please keep in mind that the reasons for having multiple media streams vary and include but are not limited to the following:

- Multiple Media Sources
- Multiple Media Streams may be needed to represent one Media Source (for instance when using layered encodings)
- A Retransmission stream may repeat the content of another Media Stream
- An FEC stream may provide material that can be used to repair another Media Stream
- Alternative Encodings, for instance different codecs for the same audio stream
- Alternative formats, for instance multiple resolutions of the same video stream

Thus the choice made due to one reason may not be the choice suitable for another reason. In the above list, the different items have different levels of maturity in the discussion on how to solve them. The clearest understanding is associated with multiple media sources of the same media type. However, all warrant discussion and clarification on how to deal with them.

This section reviews the alternatives to enable multi-stream handling. Let's start with describing mechanisms that could enable multiple media streams, independent of the purpose for having multiple streams.

SSRC Multiplexing: Each additional Media Stream gets its own SSRC within a RTP Session.

Session Multiplexing: Using additional RTP Sessions to handle additional Media Streams

As the below discussion will show, in reality we cannot choose a single one of the two solutions. To utilise RTP well and as

efficiently as possible, both are needed. The real issue is finding the right guidance on when to create RTP sessions and when additional SSRCs in an RTP session is the right choice.

5. RTP Topologies and Issues

The impact of how RTP Multiplex is performed will in general vary with how the RTP Session participants are interconnected; the RTP Topology [RFC5117]. This section describes the topologies and attempts to highlight the important behaviours concerning RTP multiplexing and multi-stream handling. It lists any identified issues regarding RTP and RTCP handling, and introduces additional topologies that are supported by RTP beyond those included in RTP Topologies [RFC5117]. The RTP Topologies that do not follow the RTP specification or do not attempt to utilise the facilities of RTP are ignored in this document.

5.1. Point to Point

This is the most basic use case with two endpoints directly interconnected and no additional entities are involved in the communication.

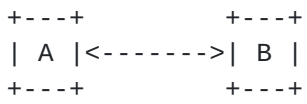


Figure 1: Point to Point

A number of applications are using a single RTP session with one RTP source per endpoint. This is likely the simplest case, as you basically doesn't have to make any choices regarding multiplexing. When you add an additional source to either endpoint you immediately create the question do one send the media stream in the existing RTP session or should I use an additional RTP session.

This raises a number of considerations that are discussed in detail below (Section 6). But the range over such aspects as:

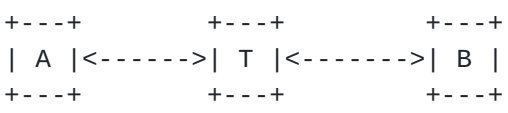
- Does my communication peer support RTP as defined with multiple SSRCs?
- Do I need network differentiation in form of QoS?
- Can the application more easily process and handle the media streams if they are in different RTP sessions?

o etc.

The application designer will have to make choices here. The point to point topology can contain one to many RTP sessions with one to many RTP sources (SSRC) per session.

5.1.1. Translators & Gateways

A point to point communication can end up in a situation when the peer it is communicating with is not compatible with the other peer for various reasons. This is in many situation resolved by the inclusion of a translator in-between the two peers.



Point to Point with Translator

The translator main purpose is to make the peer look to the other peer like something it is compatible with. An RTP translator will commonly not be distinguishable from the actual end-point. It is intentional not identifiable on RTP level. Reasons a translator can be required are:

o No common media codec for a media type thus requiring transcoding.

o Different usages of the RTP multiplexing points

o Usage of different media transport protocols

o Usage of different transport protocols

o Different security solutions

The RTP translator will rewrite RTP and RTCP as required to provide a consistent view to each peer of the traffic the translator forwards and the feedback being provided back to the RTP source.

In some case security policies or the need for monitoring the media streams the direct communication are directed to a pass through a specific middlebox, commonly called a gateway. This is often placed on the border of administrative domain where the security policies are in effect. Many gateways simple relay the RTP and RTCP traffic between the domains, but some may do more by including above mentioned translator functions or even go as far as terminating the RTP session and do application level forwarding of the media traffic. The later places requirements on the gateway to have full

understanding of the application logic and especially be able to cope with any congestion control or media adaptation.

A variant of translator behaviour worth pointing out is when an endpoint A sends a media flow to B. On the path there is a device T that on A's behalf does something with the media streams, for example adds an RTP session with FEC information for A's media streams. T will in this case need to bind the new FEC streams to A's media stream by using the same CNAME as A.

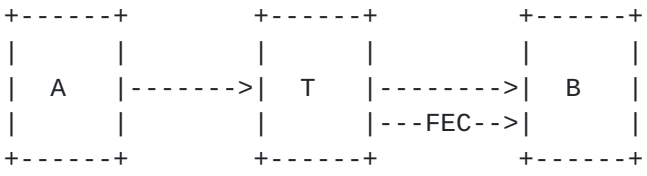


Figure 2: When De-composition is a Translator

This type of functionality where T does something with the media stream on behalf of A is clearly covered under the media translator definition (Section 5.3).

5.2. Point to Multipoint Using Multicast

This section discusses the Point to Multi-point using Multicast to interconnect the session participants. This needs to consider both Any Source Multicast (ASM) and Source-Specific Multicast (SSM). There are large commercial deployments of multicast for applications like IPTV.

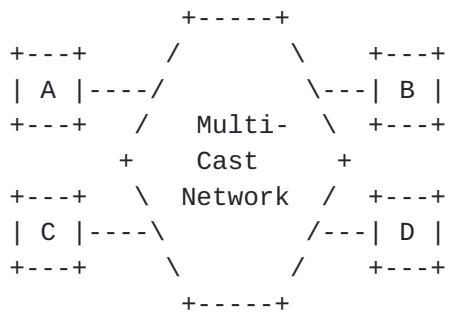
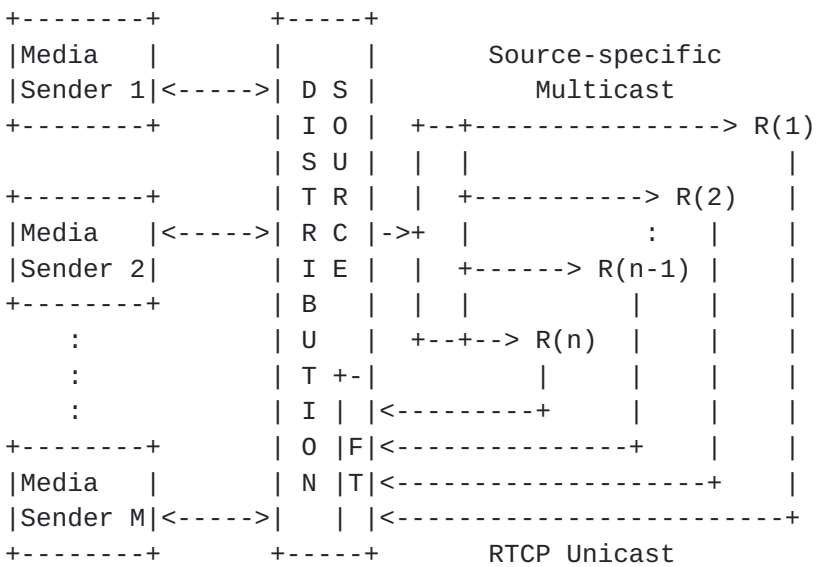


Figure 3: Point to Multipoint Using Any Source Multicast

In Any Source Multicast, any of the participants can send to all the other participants, simply by sending a packet to the multicast group. That is not possible in Source Specific Multicast [RFC4607] where only a single source (Distribution Source) can send to the multicast group, creating a topology that looks like the one below:



FT = Feedback Target
Transport from the Feedback Target to the Distribution Source is via unicast or multicast RTCP if they are not co-located.

Figure 4: Point to Multipoint using Source Specific Multicast

In the SSM topology (Figure 4) a number of RTP sources (1 to M) are allowed to send media to the SSM group. These send media to the distribution source which then forwards the media streams to the multicast group. The media streams reach the Receivers (R(1) to R(n)). The Receivers' RTCP cannot be sent to the multicast group. To support RTCP, an RTP extension for SSM [RFC5760] was defined to use unicast transmission to send RTCP from the receivers to one or more Feedback Targets (FT).

As multicast is a one to many distribution system, this must be taken into consideration. For example, the only practical method for adapting the bit-rate sent towards a given receiver for large groups is to use a set of multicast groups, where each multicast group represents a particular bit-rate. Otherwise the whole group gets media adapted to the participant with the worst conditions. The media encoding is either scalable, where multiple layers can be combined, or simulcast where a single version is selected. By either selecting or combing multicast groups, the receiver can control the bit-rate sent on the path to itself. It is also common that streams that improve transport robustness are sent in their own multicast group to allow for interworking with legacy or to support different levels of protection.

The result of this is some common behaviours for RTP multicast:

1. Multicast applications use a group of RTP sessions, not one. Each endpoint will need to be a member of a number of RTP sessions in order to perform well.

2. Within each RTP session, the number of media sinks is likely to be much larger than the number of RTP sources.

3. Multicast applications need signalling functions to identify the relationships between RTP sessions.

4. Multicast applications need signalling functions to identify the relationships between SSRCs in different RTP sessions.

All multicast configurations share a signalling requirement; all of the participants will need to have the same RTP and payload type configuration. Otherwise, A could for example be using payload type 97 as the video codec H.264 while B thinks it is MPEG-2. It should be noted that SDP offer/answer [RFC3264] has issues with ensuring this property. The signalling aspects of multicast are not explored further in this memo.

Security solutions for this type of group communications are also challenging. First of all the key-management and the security protocol must support group communication. Source authentication becomes more difficult and requires special solutions. For more discussion on this please review Options for Securing RTP Sessions [I-D.ietf-avtcore-rtp-security-options].

5.3. Point to Multipoint Using an RTP Transport Translator

This mode is described in section 3.3 of RFC 5117.

Transport Translators (Relays) result in an RTP session situation that is very similar to how an ASM group RTP session would behave.

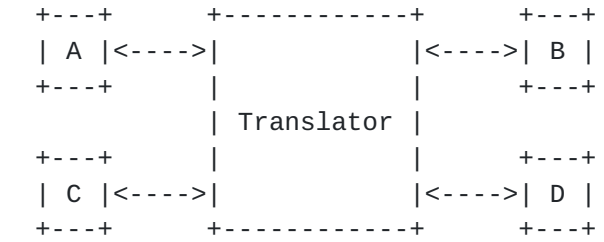


Figure 5: Transport Translator (Relay)

An RTP translator forwards both RTP and RTCP packets from all participants to all other participants.

One of the most important aspects with the simple relay is that it is only rewriting transport headers, no RTP modifications nor media transcoding occur. The most obvious downside of this basic relaying is that the translator has no control over how many streams need to be delivered to a receiver. Nor can it simply select to deliver only certain streams, as this creates session inconsistencies: If the translator temporarily stops a stream, this prevents some receivers from reporting on it. From the sender's perspective it will look like a transport failure. Applications having needs to stop or switch streams in the central node should consider using an RTP mixer to avoid this issue.

The Transport Translator does not need to have an SSRC of itself, nor does it need to send any RTCP reports on the flows that pass it. This as the RTP source will receive feedback for the full path in the RTCP being sent back. However the transport translator may choose to send RTCP reports using its own SSRC, as if it itself contained a media sink, in order to make information about the source-to-translator link available to monitors.

Use of a transport translator results in that all the endpoints will receive multiple SSRCs over a single unicast transport flow from the translator. That is independent of whether the other endpoints have only a single or several SSRCs.

The Transport Translator has the same signalling requirement as multicast: All participants must have the same payload type configuration. Also most of the ASM security issues also arise here. Some alternative when it comes to solution do exist as there after all exist a central node to communicate with. One that also can enforce some security policies depending on the level of trust placed in the node.

5.4. Point to Multipoint Using an RTP Mixer

An mixer (Figure 6) is a centralised node that selects or mixes content in a conference to optimise the RTP session so that each endpoint only needs connect to one entity, the mixer. The mixer can also reduce the bit-rate needed from the mixer down to a conference participants as the media sent from the mixer to the end-point can be optimised in different ways. These optimisations include methods like only choosing media from the currently most active speaker or mixing together audio so that only one audio stream is required instead of three in the depicted scenario (Figure 6).

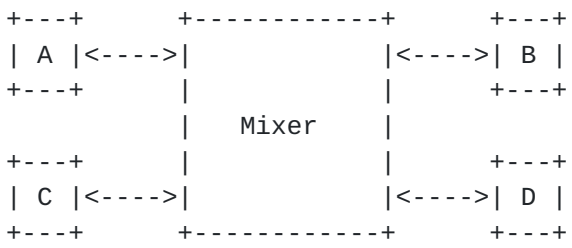


Figure 6: RTP Mixer

Mixers has some downsides, the first is that the mixer must be a trusted node as they either performs media operations or at least repacketize the media. Both type of operations requires when using SRTP that the mixer verifies integrity, decrypts the content, perform its operation and form new RTP packets, encrypts and integrity protect them. This applies to all types of mixers described below. The second downside is that all these operations and optimisation of the session requires processing. How much depends on the implementation as will become evident below.

A mixer, unlike a pure transport translator, is always application specific: the application logic for stream mixing or stream selection has to be embedded within the mixer, and controlled using application specific signalling. The implementation of an mixer can take several different forms and we will discuss the main themes available that doesn't break RTP.

Please note that a Mixer could also contain translator functionalities, like a media transcoder to adjust the media bit-rate or codec used on a particular RTP media stream.

5.4.1. Media Mixing

This type of mixer is one which clearly can be called RTP mixer is likely the one that most thinks of when they hear the term mixer. Its basic pattern of operation is that it will receive the different participants RTP media stream. Select which that are to be included in a media domain mix of the incoming RTP media streams. Then create a single outgoing stream from this mix.

The most commonly deployed media mixer is probably the audio mixer, used in voice conferencing, where the output consists of some mixture of all the input streams; this needs minimal signalling to be successful. Audio mixing is straight forward and commonly possible to do for a number of participants. Lets assume that you want to mix N number of streams from different participants. Then the mixer need to perform N decodings. Then it needs to produce N or N+1 mixes, the reasons that different mixes are needed are so that each contributing

source get a mix which don't contain themselves, as this would result in an echo. When N is lower than the number of all participants one may produce a Mix of all N streams for the group that are currently not included in the mix, thus N+1 mixes. These audio streams are then encoded again, RTP packetised and sent out.

Video can't really be "mixed" and produce something particular useful for the users, however creating an composition out of the contributed video streams can be done. In fact it can be done in a number of ways, tiling the different streams creating a chessboard, selecting someone as more important and showing them large and a number of other sources as smaller overlays is another. Also here one commonly need to produce a number of different compositions so that the contributing part doesn't need to see themselves. Then the mixer re-encodes the created video stream, RTP packetise it and send it out

The problem with media mixing is that it both consume large amount of media processing and encoding resources. The second is the quality degradation created by decoding and re-encoding the RTP media stream. Its advantage is that it is quite simplistic for the clients to handle as they don't need to handle local mixing and composition.

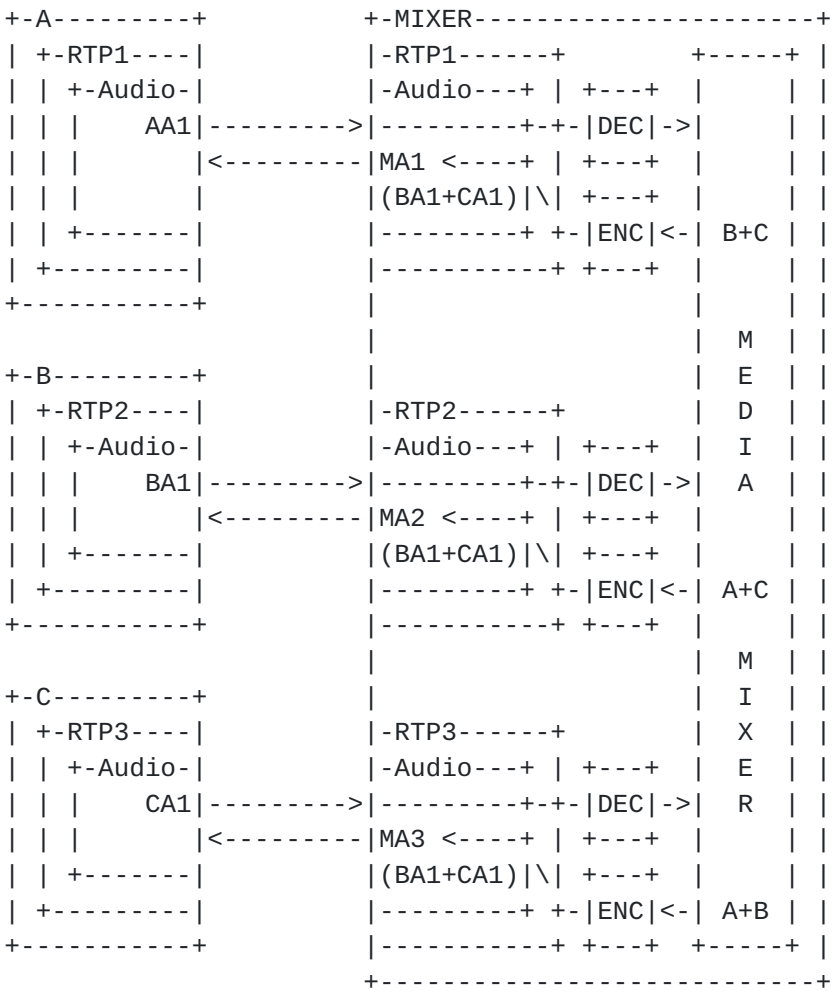


Figure 7: Session and SSRC details for Media Mixer

From an RTP perspective media mixing can be very straight forward as can be seen in Figure 7. The mixer present one SSRC towards the peer client, e.g. MA1 to Peer A, which is the media mix of the other participants. As each peer receives a different version produced by the mixer there are no actual relation between the different RTP sessions in the actual media or the transport level information. There is however one connection between RTP1-RTP3 in this figure. It has to do with the SSRC space and the identity information. When A receives the MA1 stream which is a combination of BA1 and CA1 streams in the other PeerConnections RTP could enable the mixer to include CSRC information in the MA1 stream to identify the contributing source BA1 and CA1.

The CSRC has in its turn utility in RTP extensions, like the in Mixer to Client audio levels RTP header extension [RFC6465]. If the SSRC from endpoint to mixer leg are used as CSRC in another PeerConnection then RTP1, RTP2 and RTP3 becomes one joint session as they have a

common SSRC space. At this stage the mixer also need to consider which RTCP information it need to expose in the different legs. For the above situation commonly nothing more than the Source Description (SDES) information and RTCP BYE for CSRC need to be exposed. The main goal would be to enable the correct binding against the application logic and other information sources. This also enables loop detection in the RTP session.

5.4.1.1. RTP Session Termination

There exist an possible implementation choice to have the RTP sessions being separated between the different legs in the multi-party communication session and only generate RTP media streams in each without carrying on RTP/RTCP level any identity information about the contributing sources. This removes both the functionality that CSRC can provide and the possibility to use any extensions that build on CSRC and the loop detection. It may appear a simplification if SSRC collision would occur between two different end-points as they can be avoided to be resolved and instead remapped between the independent sessions if at all exposed. However, SSRC/CSRC remapping requires that SSRC/CSRC are never used in the application level as reference. This as they only have local importance, if they are used on a multi-party session scope the result would be miss-referencing.

Session termination may appear to resolve some issues, it however creates other issues that needs resolving, like loop detection, identification of contributing sources and the need to handle mapped identities and ensure that the right one is used towards the right identities and never used directly between multiple end-points.

5.4.2. Media Switching

An RTP Mixer based on media switching avoids the media decoding and encoding cycle in the mixer, but not the decryption and re-encryption cycle as one rewrites RTP headers. This both reduces the amount of computational resources needed in the mixer and increases the media quality per transmitted bit. This is achieve by letting the mixer have a number of SSRCs that represents conceptual or functional streams the mixer produces. These streams are created by selecting media from one of the by the mixer received RTP media streams and forward the media using the mixers own SSRCs. The mixer can then switch between available sources if that is required by the concept for the source, like currently active speaker.

To achieve a coherent RTP media stream from the mixer's SSRC the mixer is forced to rewrite the incoming RTP packet's header. First the SSRC field must be set to the value of the Mixer's SSRC. Secondly, the sequence number must be the next in the sequence of

outgoing packets it sent. Thirdly the RTP timestamp value needs to be adjusted using an offset that changes each time one switch media source. Finally depending on the negotiation the RTP payload type value representing this particular RTP payload configuration may have to be changed if the different endpoint mixer legs have not arrived on the same numbering for a given configuration. This also requires that the different end-points do support a common set of codecs, otherwise media transcoding for codec compatibility is still required.

Lets consider the operation of media switching mixer that supports a video conference with six participants (A-F) where the two latest speakers in the conference are shown to each participants. Thus the mixer has two SSRCs sending video to each peer.

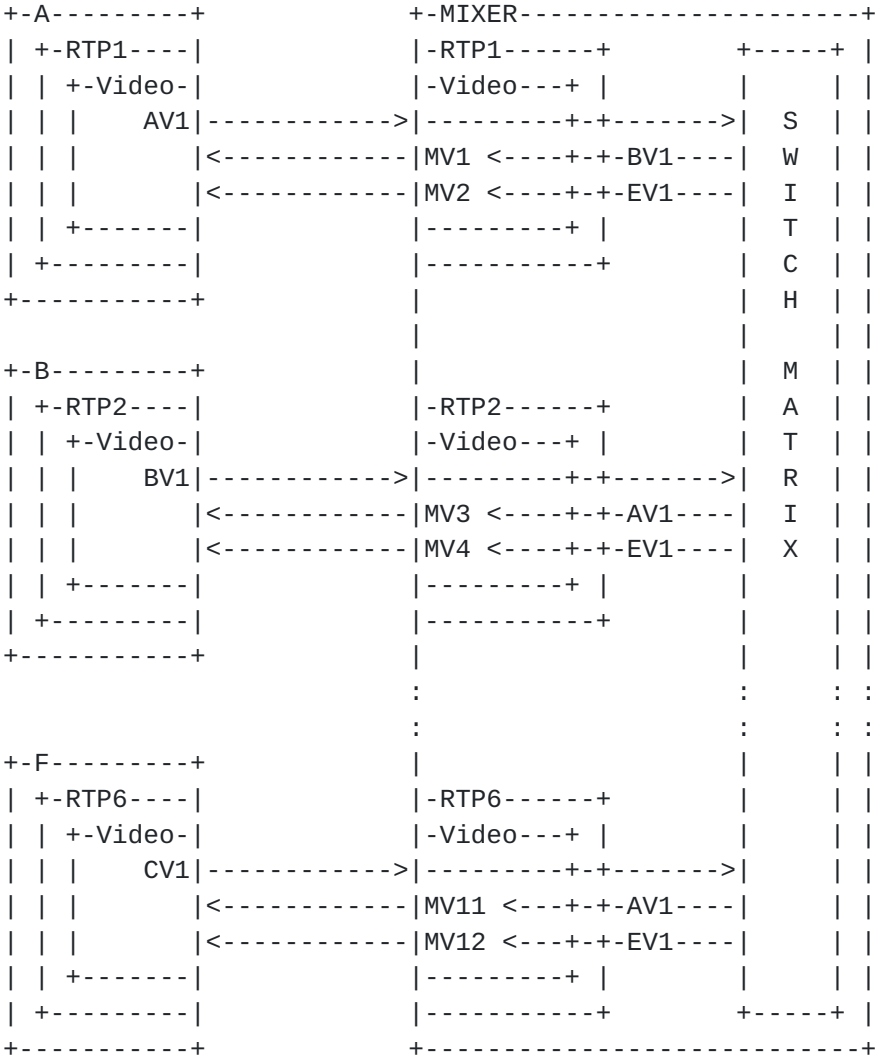


Figure 8: Media Switching RTP Mixer

The Media Switching RTP mixer can similar to the Media Mixing one reduce the bit-rate needed towards the different peers by selecting and switching in a sub-set of RTP media streams out of the ones it receives from the conference participations.

To ensure that a media receiver can correctly decode the RTP media stream after a switch, it becomes necessary to ensure for state saving codecs that they start from default state at the point of switching. Thus one common tool for video is to request that the encoding creates an intra picture, something that isn't dependent on earlier state. This can be done using Full Intra Request [RFC5104] RTCP codec control message.

Also in this type of mixer one could consider to terminate the RTP sessions fully between the different end-point and mixer legs. The same arguments and considerations as discussed in Section 5.4.1.1 applies here.

5.4.3. RTP Source Projecting

Another method for handling media in the RTP mixer is to project all potential RTP sources (SSRCs) into a per end-point independent RTP session. The mixer can then select which of the potential sources that are currently actively transmitting media, despite that the mixer in another RTP session receives media from that end-point. This is similar to the media switching Mixer but have some important differences in RTP details.

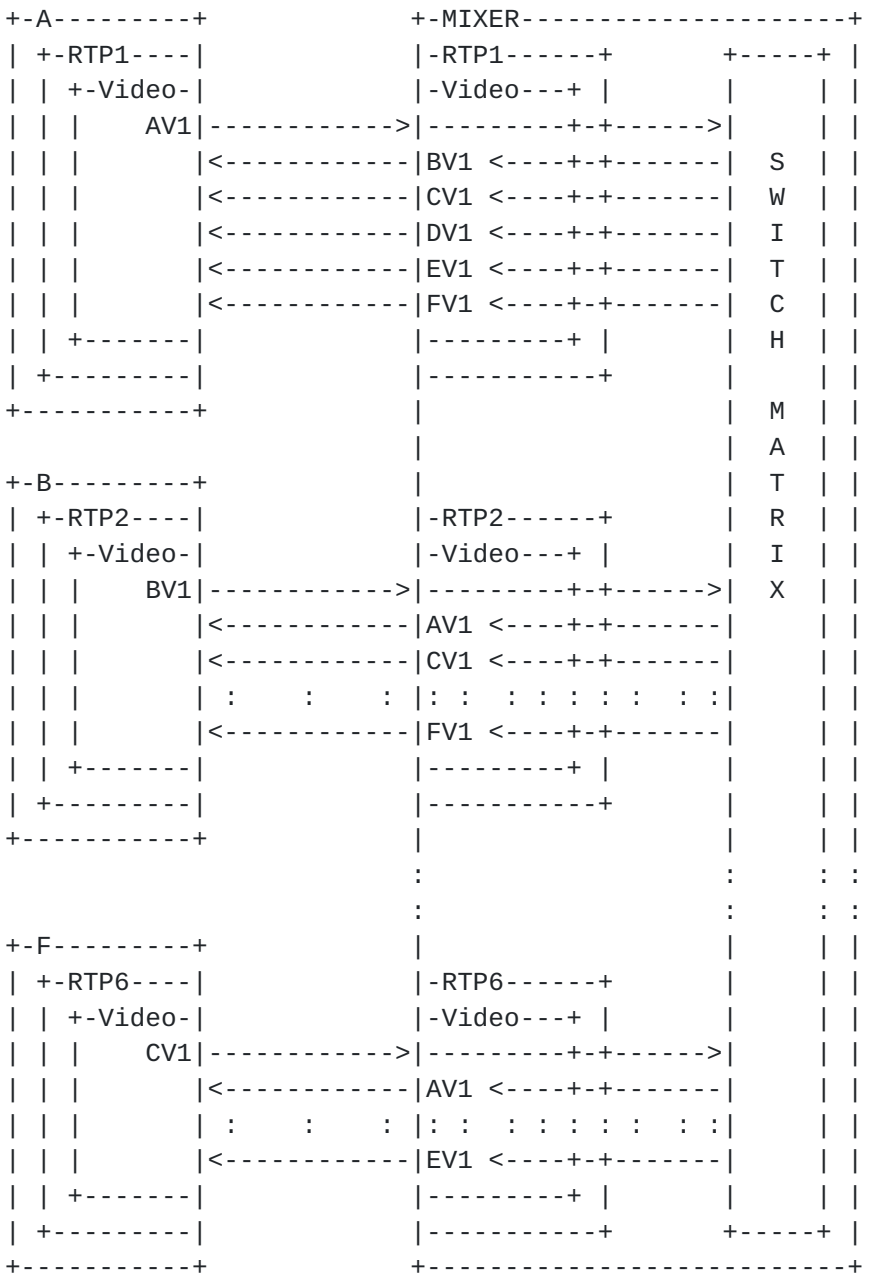


Figure 9: Media Projecting Mixer

So in this six participant conference depicted above in (Figure 9) one can see that end-point A will in this case be aware of 5 incoming SSRCs, BV1-FV1. If this mixer intend to have the same behaviour as in Section 5.4.2 where the mixer provides the end-points with the two latest speaking end-points, then only two out of these five SSRCs will concurrently transmit media to A. As the mixer selects which source in the different RTP sessions that transmit media to the end-points each RTP media stream will require some rewriting when being projected from one session into another. The main thing is that the

sequence number will need to be consecutively incremented based on the packet actually being transmitted in each RTP session. Thus the RTP sequence number offset will change each time a source is turned on in a RTP session.

As the RTP sessions are independent the SSRC numbers used can be handled independently also thus working around any SSRC collisions by having remapping tables between the RTP sessions. This will result that each endpoint may have a different view of the application usage of a particular SSRC. Thus the application must not use SSRC as references to RTP media streams when communicating with other peers directly.

The mixer will also be responsible to act on any RTCP codec control requests coming from an end-point and decide if it can act on it locally or needs to translate the request into the RTP session that contains the media source. Both end-points and the mixer will need to implement conference related codec control functionalities to provide a good experience. Full Intra Request to request from the media source to provide switching points between the sources, Temporary Maximum Media Bit-rate Request (TMMBR) to enable the mixer to aggregate congestion control response towards the media source and have it adjust its bit-rate in case the limitation is not in the source to mixer link.

This version of the mixer also puts different requirements on the end-point when it comes to decoder instances and handling of the RTP media streams providing media. As each projected SSRC can at any time provide media the end-point either needs to handle having thus many allocated decoder instances or have efficient switching of decoder contexts in a more limited set of actual decoder instances to cope with the switches. The WebRTC application also gets more responsibility to update how the media provides is to be presented to the user.

5.5. Point to Multipoint using Multiple Unicast flows

Based on the RTP session definition, it is clearly possible to have a joint RTP session over multiple transport flows like the below three endpoint joint session. In this case, A needs to send its' media streams and RTCP packets to both B and C over their respective transport flows. As long as all participants do the same, everyone will have a joint view of the RTP session.

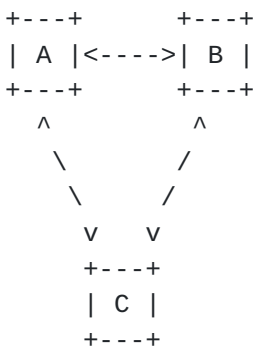


Figure 10: Point to Multi-Point using Multiple Unicast Transports

This doesn't create any additional requirements beyond the need to have multiple transport flows associated with a single RTP session. Note that an endpoint may use a single local port to receive all these transport flows, or it might have separate local reception ports for each of the endpoints.

There exists an alternative structure for establishing the above communication scenario (Figure 10) which uses independent RTP sessions between each pair of peers, i.e. three different RTP sessions. Unless independently adapted the same RTP media stream could be sent in both of the RTP sessions an endpoint has. The difference exists in the behaviours around RTCP, for example common RTCP bandwidth for one joint session, rather than three independent pools, and the awareness based on RTCP reports between the peers of how that third leg is doing.

5.6. De-composite Endpoint

The implementation of an application may desire to send a subset of the application's data to each of multiple devices, each with their own network address. A very basic use case for this would be to separate audio and video processing for a particular endpoint, like a conference room, into one device handling the audio and another handling the video, being interconnected by some control functions allowing them to behave as a single endpoint in all aspects except for transport Figure 11.

Which decomposition that is possible is highly dependent on the RTP session usage. It is not really feasible to decomposed one logical end-point into two different transport node in one RTP session. From a third party monitor of such an attempt the two entities would look like two different end-points with a CNAME collision. This put a requirement on that the only type of de-composited endpoint that RTP really supports is one where the different parts have separate RTP sessions to send and/or receive media streams intended for them.

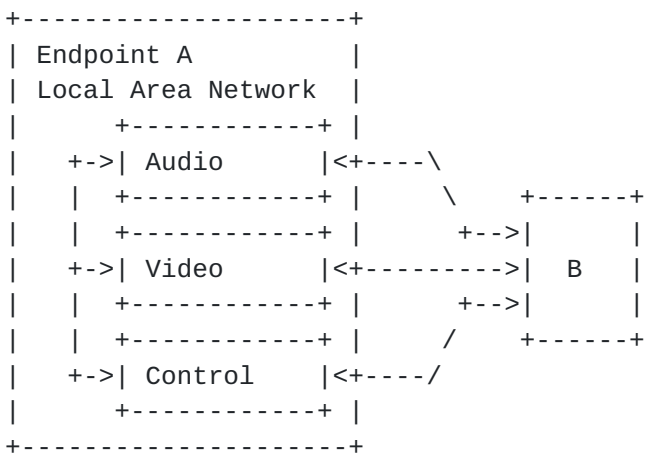


Figure 11: De-composite End-Point

In the above usage, let us assume that the RTP sessions are different for audio and video. The audio and video parts will use a common CNAME and also have a common clock to ensure that synchronisation and clock drift handling works despite the decomposition. Also the RTCP handling works correctly as long as only one part of the de-composite is part of each RTP session. That way any differences in the path between A's audio entity and B and A's video and B are related to different SSRCs in different RTP sessions.

The requirements that can derived from the above usage is that the transport flows for each RTP session might be under common control but still go to what looks like different endpoints based on addresses and ports. This geometry cannot be accomplished using one RTP session, so in this case, multiple RTP sessions are needed.

6. Multiple Streams Discussion

6.1. Introduction

Using multiple media streams is a well supported feature of RTP. However, it can be unclear for most implementers or people writing RTP/RTCP applications or extensions attempting to apply multiple streams when it is most appropriate to add an additional SSRC in an existing RTP session and when it is better to use multiple RTP sessions. This section tries to discuss the various considerations needed. The next section then concludes with some guidelines.

6.2. RTP/RTCP Aspects

This section discusses RTP and RTCP aspects worth considering when selecting between SSRC multiplexing and Session multiplexing.

6.2.1. The RTP Specification

RFC 3550 contains some recommendations and a bullet list with 5 arguments for different aspects of RTP multiplexing. Let's review Section 5.2 of [RFC3550], reproduced below:

"For efficient protocol processing, the number of multiplexing points should be minimised, as described in the integrated layer processing design principle [ALF]. In RTP, multiplexing is provided by the destination transport address (network address and port number) which is different for each RTP session. For example, in a teleconference composed of audio and video media encoded separately, each medium SHOULD be carried in a separate RTP session with its own destination transport address.

Separate audio and video streams SHOULD NOT be carried in a single RTP session and demultiplexed based on the payload type or SSRC fields. Interleaving packets with different RTP media types but using the same SSRC would introduce several problems:

1. If, say, two audio streams shared the same RTP session and the same SSRC value, and one were to change encodings and thus acquire a different RTP payload type, there would be no general way of identifying which stream had changed encodings.

2. An SSRC is defined to identify a single timing and sequence number space. Interleaving multiple payload types would require different timing spaces if the media clock rates differ and would require different sequence number spaces to tell which payload type suffered packet loss.

3. The RTCP sender and receiver reports (see Section 6.4) can only describe one timing and sequence number space per SSRC and do not carry a payload type field.

4. An RTP mixer would not be able to combine interleaved streams of incompatible media into one stream.

5. Carrying multiple media in one RTP session precludes: the use of different network paths or network resource allocations if appropriate; reception of a subset of the media if desired, for example just audio if video would exceed the available bandwidth; and receiver implementations that use separate processes for the different media, whereas using separate RTP sessions permits either single- or multiple-process implementations.

Using a different SSRC for each medium but sending them in the same RTP session would avoid the first three problems but not the last

two.

On the other hand, multiplexing multiple related sources of the same medium in one RTP session using different SSRC values is the norm for multicast sessions. The problems listed above don't apply: an RTP mixer can combine multiple audio sources, for example, and the same treatment is applicable for all of them. It may also be appropriate to multiplex streams of the same medium using different SSRC values in other scenarios where the last two problems do not apply."

Let's consider one argument at a time. The first is an argument for using different SSRC for each individual media stream, which still is very applicable.

The second argument is advocating against using payload type multiplexing, which still stands as can been seen by the extensive list of issues found in Appendix A.

The third argument is yet another argument against payload type multiplexing.

The fourth is an argument against multiplexing media streams that require different handling into the same session. As we saw in the discussion of RTP mixers, the RTP mixer has to embed application logic in order to handle streams anyway; the separation of streams according to stream type is just another piece of application logic, which may or may not be appropriate for a particular application. A type of application that can mix different media sources "blindly" is the telephone bridge; most other type of application needs application-specific logic to perform the mix correctly.

The fifth argument discusses network aspects that we will discuss more below in Section 6.4. It also goes into aspects of implementation, like decomposed endpoints where different processes or inter-connected devices handle different aspects of the whole multi-media session.

A summary of RFC 3550's view on multiplexing is to use unique SSRCs for anything that is its own media/packet stream, and to use different RTP sessions for media streams that don't share media type. The first this document support as very valid. The later is one thing which is further discussed in this document as something the application developer needs to make a continuous choice for.

6.2.1.1. Different Media Types Recommendations

The above quote from RTP [RFC3550] includes a strong recommendation:

> "For example, in a teleconference composed of audio and video media encoded separately, each medium SHOULD be carried in a separate RTP session with its own destination transport address."

It was identified in "Why RTP Sessions Should Be Content Neutral" [I-D.alvestrand-rtp-sess-neutral] that the above statement is poorly supported by any of the motivations provided in the RTP specification. This has resulted in the creation of a specification Multiple Media Types in an RTP Session specification [I-D.westerlund-avtcore-multi-media-rtp-session] which intend to update this recommendation. That document has a detailed analysis of the potential issues in having multiple media types in the same RTP session. This document tries to provide an more over arching consideration regarding the usage of RTP session and considers multiple media types in one RTP session as possible choice for the RTP application designer.

6.2.2. Multiple SSRCs in a Session

In cases when an endpoint uses multiple SSRCs, we have found two closely related issues. The first is if every SSRC shall report on all other SSRC, even the ones originating from the same endpoint. The reason for this would be to ensure that no monitoring function should suspect a breakage in the RTP session. No monitoring function that gives an alert on non-reporting of an endpoint's own SSRCs has been identified.

The second issue around RTCP reporting arise when an endpoint receives one or more media streams, and when the receiving endpoint itself sends multiple SSRC in the same RTP session. As transport statistics are gathered per endpoint and shared between the nodes, all the endpoint's SSRC will report based on the same received data, the only difference will be which SSRCs sends the report. This could be considered unnecessary overhead, but for consistency it might be simplest to always have all sending SSRCs send RTCP reports on all media streams the endpoint receives.

The current RTP text is silent about sending RTCP Receiver Reports for an endpoint's own sources, but does not preclude either sending or omitting them. The uncertainty in the expected behaviour in those cases has likely caused variations in the implementation strategy. This could cause an interoperability issue where it is not possible to determine if the lack of reports is a true transport issue, or simply a result of implementation.

Although this issue is valid already for the simple point to point case, it needs to be considered in all topologies. From the perspective of an endpoint, any solution needs to take into account

what a particular endpoint can determine without explicit information of the topology. For example, a Transport Translator (Relay) topology will look quite similar to point to point on a transport level but is different on RTP level. Assume a first scenario with two SSRC being sent from an endpoint to a Transport Translator, and a second scenario with two single SSRC remote endpoints sending to the same Transport Translator. The main differences between those two scenarios are that in the second scenario, the RTT may vary between the SSRCs (but it is not guaranteed), and the SSRCs may also have different CNAMEs.

When an endpoint has multiple SSRCs and it needs to send RTCP packets on behalf of these SSRCs, the question arises how RTCP packets with different source SSRCs can be sent in the same compound packet. It appears allowed, however some consideration of the transmission scheduling is needed.

These issues are currently being discussed and a recommendation for how to handle them are developed in "Real-Time Transport Protocol (RTP) Considerations for Endpoints Sending Multiple Media Streams" [I-D.lennox-avtcore-rtp-multi-stream].

6.2.3. Handling Varying sets of Senders

In some applications, the set of simultaneously active sources varies within a larger set of session members. A receiver can then possibly try to use a set of decoding chains that is smaller than the number of senders, switching the decoding chains between different senders. As each media decoding chain may contain state, either the receiver must either be able to save the state of swapped-out senders, or the sender must be able to send data that permits the receiver to reinitialise when it resumes activity.

This behaviour will cause similar issues independent of SSRC or Session multiplexing.

6.2.4. Cross Session RTCP Requests

There currently exists no functionality to make truly synchronised and atomic RTCP messages with some type of request semantics across multiple RTP Sessions. Instead, separate RTCP messages will have to be sent in each session. This gives SSRC multiplexed streams a slight advantage as RTCP messages for different streams in the same session can be sent in a compound RTCP packet. Thus providing an atomic operation if different modifications of different streams are requested at the same time.

In Session multiplexed cases, the RTCP timing rules in the sessions

and the transport aspects, such as packet loss and jitter, prevents a receiver from relying on atomic operations, forcing it to use more robust and forgiving mechanisms.

6.2.5. Binding Related Sources

A common problem in a number of various RTP extensions has been how to bind related sources together. This issue is common to SSRC multiplexing and Session Multiplexing.

Most, if not all, solutions to this problem are implemented in the signalling plane, providing metadata information using SDP.

There exists one solution for grouping RTP sessions together in SDP [RFC5888] to know which RTP session contains for example the FEC data for the source data in another session. However, this mechanism does not work on individual media flows and is thus not directly applicable to the problem. The other solution is also SDP based and can group SSRCs within a single RTP session [RFC5576]. Thus this mechanism can bind media streams in SSRC multiplexed cases. Both solutions have the shortcoming of being restricted to SDP based signalling and also do not work in cases where the session's dynamic properties are such that it is difficult or resource consuming to keep the list of related SSRCs up to date.

One possible solution could be to mandate the same SSRC value being used in all RTP session in case of session multiplexing. We do note that Section 8.3 of the RTP Specification [RFC3550] recommends using a single SSRC space across all RTP sessions for layered coding. However this recommendation has some downsides and is less applicable beyond the field of layered coding. To use the same sender SSRC in all RTP sessions from a particular endpoint can cause issues if an SSRC collision occurs. If the same SSRC is used as the required binding between the streams, then all streams in the related RTP sessions must change their SSRC. This is extra likely to cause problems if the participant populations are different in the different sessions. For example, in case of large number of receivers having selected totally random SSRC values in each RTP session as RFC 3550 specifies, a change due to a SSRC collision in one session can then cause a new collision in another session. This cascading effect is not severe but there is an increased risk that this occurs for well populated sessions (the birthday paradox ensures that if you populate a single session with 9292 SSRCs at random, the chances are approximately 1% that at least one collision will occur). In addition, being forced to change the SSRC affects all the related media streams; instead of having to resynchronise only the originally conflicting stream, all streams will suddenly need to be resynchronised with each other. This will prevent also the media

streams not having an actual collision from being usable during the resynchronisation and also increases the time until synchronisation is finalised. In addition, it requires exception handling in the SSRC generation.

The above collision issue does not occur in case of having only one SSRC space across all sessions and all participants will be part of at least one session, like the base layer in layered encoding. In that case the only downside is the special behaviour that needs to be well defined by anyone using this. But, having an exception behaviour where the SSRC space is common across all session is an issue as this behaviour does not fit all the RTP extensions or payload formats. It is possible to create a situation where the different mechanisms cannot be combined due to the non standard SSRC allocation behaviour.

Existing mechanisms with known issues:

RTP Retransmission: [RFC4588] Has two modes, one for SSRC multiplexing and one for Session multiplexing. The session multiplexing requires the same CNAME and mandates that the same SSRC is used in both sessions. Using the same SSRC does work but will as previously stated potentially have issues in certain cases. In SSRC multiplexed mode the CNAME is used to bind media and retransmission streams together. However, if multiple media streams are sent from the same endpoint in the same session this does not provide non-ambiguous binding. Therefore when the first retransmission request for a media stream is sent, one must not have another retransmission request outstanding for an SSRC which don't have a binding between the original SSRC and the retransmission stream's SSRC. This works but creates some limitations that can be avoided by a explicit mechanism. The SDP based ssrc-group mechanism would be sufficient in this case as long as the application can rely on the signalling based solution.

Scalable Video Coding : As an example of scalable coding, SVC [RFC6190] has various modes. The Multi Session Transmission (MST) uses Session multiplexing to separate scalability layers. However, this specification has failed to be explicit on how these layers are bound together in cases where CNAME is not sufficient. CNAME is no longer sufficient when more than one media source occur within a session that has the same CNAME, for example due to multiple video cameras capturing the same lecture hall. This likely implies that a single SSRC space as recommend by Section 8.3 of RTP [RFC3550] is to be used.

Forward Error Correction: If some type of FEC or redundancy stream is being sent, it needs its own SSRC, with the exception of constructions like redundancy encoding [RFC2198]. Thus in case of transmitting the FEC in the same session as the source data, the inter SSRC relation within a session is needed. In case of sending the redundant data in a separate session from the source, the SSRC in each session needs to be related. This occurs for example in RFC5109 when using session separation of original and FEC data. SSRC multiplexing is not supported, only using redundant encoding is supported.

This issue appears to need action to harmonise and avoid future shortcomings in extension specifications. A proposed solution for handling this issue is [I-D.westerlund-avtext-rtcp-sdes-srcname].

6.2.6. Forward Error Correction

There exist a number of Forward Error Correction (FEC) based schemes for how to reduce the packet loss of the original streams. Most of the FEC schemes will protect a single source flow. The protection is achieved by transmitting a certain amount of redundant information that is encoded such that it can repair one or more packet loss over the set of packets they protect. This sequence of redundant information also needs to be transmitted as its own media stream, or in some cases instead of the original media stream. Thus many of these schemes create a need for binding the related flows as discussed above. They also create additional flows that need to be transported. Looking at the history of these schemes, there is both SSRC multiplexed and Session multiplexed solutions and some schemes that support both.

Using a Session multiplexed solution supports the case where some set of receivers may not be able to utilise the FEC information. By placing it in a separate RTP session, it can easily be ignored.

In usages involving multicast, having the FEC information on its own multicast group, and therefore in its own RTP session, allows for flexibility, for example when using Rapid Acquisition of Multicast Groups (RAMS) [RFC6285]. During the RAMS burst where data is received over unicast and where it is possible to combine with unicast based retransmission [RFC4588], there is no need to burst the FEC data related to the burst of the source media streams needed to catch up with the multicast group. This saves bandwidth to the receiver during the burst, enabling quicker catch up. When the receiver has caught up and joins the multicast group(s) for the source, it can at the same time join the multicast group with the FEC information. Having the source stream and the FEC in separate groups allows for easy separation in the Burst/Retransmission Source (BRS)

without having to individually classify packets.

6.2.7. Transport Translator Sessions

A basic Transport Translator relays any incoming RTP and RTCP packets to the other participants. The main difference between SSRC multiplexing and Session multiplexing resulting from this use case is that for SSRC multiplexing it is not possible for a particular session participant to decide to receive a subset of media streams. When using separate RTP sessions for the different sets of media streams, a single participant can choose to leave one of the sessions but not the other.

6.3. Interworking

There are several different kinds of interworking, and this section discusses two related ones. The interworking between different applications and the implications of potentially different choices of usage of RTP's multiplexing points. The second topic relates to what limitations may have to be considered working with some legacy applications.

6.3.1. Types of Interworking

It is not uncommon that applications or services of similar usage, especially the ones intended for interactive communication, ends up in a situation where one want to interconnect two or more of these applications.

In these cases one ends up in a situation where one might use a gateway to interconnect applications. This gateway then needs to change the multiplexing structure or adhere to limitations in each application.

There are two fundamental approaches to gatewaying: RTP bridging, where the gateway acts as an RTP Translator, and the two applications are members of the same RTP session, and RTP termination, where there are independent RTP sessions running from each interconnected application to the gateway.

6.3.2. RTP Translator Interworking

From an RTP perspective the RTP Translator approach could work if all the applications are using the same codecs with the same payload types, have made the same multiplexing choices, have the same capabilities in number of simultaneous media streams combined with the same set of RTP/RTCP extensions being supported. Unfortunately this may not always be true.

When one is gatewaying via an RTP Translator, a natural requirement is that the two applications being interconnected must use the same approach to multiplexing. Furthermore, if one of the applications is capable of working in several modes (such as being able to use SSRC multiplexing or RTP session multiplexing at will), and the other one is not, successful interconnection depends on locking the more flexible application into the operating mode where interconnection can be successful, even if no participants using the less flexible application are present when the RTP sessions are being created.

6.3.3. Gateway Interworking

When one terminates RTP sessions at the gateway, there are certain tasks that the gateway must carry out:

- Generating appropriate RTCP reports for all media streams (possibly based on incoming RTCP reports), originating from SSRCs controlled by the gateway.

- Handling SSRC collision resolution in each application's RTP sessions.

- Signalling, choosing and policing appropriate bit-rates for each session.

If either of the applications has any security applied, e.g. in the form of SRTP, the gateway must be able to decrypt incoming packets and re-encrypt them in the other application's security context. This is necessary even if all that's required is a simple remapping of SSRC numbers. If this is done, the gateway also needs to be a member of the security contexts of both sides, of course.

Other tasks a gateway may need to apply include transcoding (for incompatible codec types), rescaling (for incompatible video size requirements), suppression of content that is known not to be handled in the destination application, or the addition or removal of redundancy coding or scalability layers to fit the need of the destination domain.

From the above, we can see that the gateway needs to have an intimate knowledge of the application requirements; a gateway is by its nature application specific, not a commodity product.

This fact reveals the potential for these gateways to block evolution of the applications by blocking unknown RTP and RTCP extensions that the regular application has been extended with.

If one uses security functions, like SRTP, they can as seen above

incur both additional risk due to the gateway needing to be in security association between the endpoints, unless the gateway is on the transport level, and additional complexities in form of the decrypt-encrypt cycles needed for each forwarded packet. SRTP, due to its keying structure, also requires that each RTP session must have different master keys, as use of the same key in two RTP sessions can result in two-time pads that completely breaks the confidentiality of the packets.

6.3.4. Multiple SSRC Legacy Considerations

Historically, the most common RTP use cases have been point to point Voice over IP (VoIP) or streaming applications, commonly with no more than one media source per endpoint and media type (typically audio and video). Even in conferencing applications, especially voice only, the conference focus or bridge has provided a single stream with a mix of the other participants to each participant. It is also common to have individual RTP sessions between each endpoint and the RTP mixer, meaning that the mixer functions as an RTP-terminating gateway.

When establishing RTP sessions that may contain endpoints that aren't updated to handle multiple streams following these recommendations, a particular application can have issues with multiple SSRCs within a single session. These issues include:

1. Need to handle more than one stream simultaneously rather than replacing an already existing stream with a new one.

2. Be capable of decoding multiple streams simultaneously.

3. Be capable of rendering multiple streams simultaneously.

This indicates that gateways attempting to interconnect to this class of devices must make sure that only one media stream of each type gets delivered to the endpoint if it's expecting only one, and that the multiplexing format is what the device expects. It is highly unlikely that RTP translator-based interworking can be made to function successfully in such a context.

6.4. Network Aspects

The multiplexing choice has impact on network level mechanisms that need to be considered by the implementor.

6.4.1. Quality of Service

When it comes to Quality of Service mechanisms, they are either flow based or marking based. RSVP [RFC2205] is an example of a flow based mechanism, while Diff-Serv [RFC2474] is an example of a Marking based one. For a marking based scheme, the method of multiplexing will not affect the possibility to use QoS.

However, for a flow based scheme there is a clear difference between the methods. SSRC multiplexing will result in all media streams being part of the same 5-tuple (protocol, source address, destination address, source port, destination port) which is the most common selector for flow based QoS. Thus, separation of the level of QoS between media streams is not possible. That is however possible for session based multiplexing, where each media stream for which a separate QoS handling is desired can be in a different RTP session that can be sent over different 5-tuples.

6.4.2. NAT and Firewall Traversal

In today's network there exist a large number of middleboxes. The ones that normally have most impact on RTP are Network Address Translators (NAT) and Firewalls (FW).

Below we analyze and comment on the impact of requiring more underlying transport flows in the presence of NATs and Firewalls:

End-Point Port Consumption: A given IP address only has 65536 available local ports per transport protocol for all consumers of ports that exist on the machine. This is normally never an issue for an end-user machine. It can become an issue for servers that handle large number of simultaneous streams. However, if the application uses ICE to authenticate STUN requests, a server can serve multiple endpoints from the same local port, and use the whole 5-tuple (source and destination address, source and destination port, protocol) as identifier of flows after having securely bound them to the remote endpoint address using the STUN request. In theory the minimum number of media server ports needed are the maximum number of simultaneous RTP Sessions a single endpoint may use. In practice, implementation will probably benefit from using more server ports to simplify implementation or avoid performance bottlenecks.

NAT State: If an endpoint sits behind a NAT, each flow it generates to an external address will result in a state that has to be kept in the NAT. That state is a limited resource. In home or Small Office/Home Office (SOHO) NATs, memory or processing are usually the most limited resources. For large scale NATs serving many

internal endpoints, available external ports are typically the scarce resource. Port limitations is primarily a problem for larger centralised NATs where endpoint independent mapping requires each flow to use one port for the external IP address. This affects the maximum number of internal users per external IP address. However, it is worth pointing out that a real-time video conference session with audio and video is likely using less than 10 UDP flows, compared to certain web applications that can use 100+ TCP flows to various servers from a single browser instance.

NAT Traversal Excess Time: Making the NAT/FW traversal takes a certain amount of time for each flow. It also takes time in a phase of communication between accepting to communicate and the media path being established which is fairly critical. The best case scenario for how much extra time it takes after finding the first valid candidate pair following the specified ICE procedures are: 1.5*RTT + Ta*(Additional_Flows-1), where Ta is the pacing timer, which ICE specifies to be no smaller than 20 ms. That assumes a message in one direction, and then an immediate triggered check back. The reason it isn't more is that ICE first finds one candidate pair that works prior to attempting to establish multiple flows. Thus, there is no extra time until one has found a working candidate pair. Based on that working pair the needed extra time is to in parallel establish the, in most cases 2-3, additional flows. However, packet loss causes extra delays, at least 100 ms which is the minimal retransmission timer for ICE.

NAT Traversal Failure Rate: Due to the need to establish more than a single flow through the NAT, there is some risk that establishing the first flow succeeds but that one or more of the additional flows fail. The risk that this happens is hard to quantify, but it should be fairly low as one flow from the same interfaces has just been successfully established. Thus only rare events such as NAT resource overload, or selecting particular port numbers that are filtered etc, should be reasons for failure.

Deep Packet Inspection and Multiple Streams: Firewalls differ in how deeply they inspect packets. There exist some potential that deeply inspecting firewalls will have similar legacy issues with multiple SSRCs as some stack implementations.

SSRC multiplexing keeps the additional media streams within one RTP Session and does not introduce any additional NAT traversal complexities per media stream. This can be compared with normally one or two additional transport flows per RTP session when using session multiplexing. Additional lower layer transport flows will be required, unless an explicit de-multiplexing layer is added between

RTP and the transport protocol. A proposal for how to multiplex multiple RTP sessions over the same single lower layer transport exist in [I-D.westerlund-avtcore-transport-multiplexing].

6.4.3. Multicast

Multicast groups provides a powerful semantics for a number of real-time applications, especially the ones that desire broadcast-like behaviours with one endpoint transmitting to a large number of receivers, like in IPTV. But that same semantics do result in a certain number of limitations.

One limitation is that for any group, sender side adaptation to the actual receiver properties causes degradation for all participants to what is supported by the receiver with the worst conditions among the group participants. In most cases this is not acceptable. Instead various receiver based solutions are employed to ensure that the receivers achieve best possible performance. By using scalable encoding and placing each scalability layer in a different multicast group, the receiver can control the amount of traffic it receives. To have each scalability layer on a different multicast group, one RTP session per multicast group is used.

RTP can't function correctly if media streams sent over different multicast groups where considered part of the same RTP session. First of all the different layers needs different SSRCs or the sequence number space seen for a receiver of any sub set of the layers would have sender side holes. Thus triggering packet loss reactions. Also any RTCP reporting of such a session would be non consistent and making it difficult for the sender to determine the sessions actual state.

Thus it appears easiest and most straightforward to use multiple RTP sessions. In addition, the transport flow considerations in multicast are a bit different from unicast. First of all there is no shortage of port space, as each multicast group has its own port space.

6.4.4. Multiplexing multiple RTP Session on a Single Transport

For applications that doesn't need flow based QoS and like to save ports and NAT/FW traversal costs and where usage of multiple media types in one RTP session is not suitable, there is a proposal for how to achieve multiplexing of multiple RTP sessions over the same lower layer transport [I-D.westerlund-avtcore-transport-multiplexing]. Using such a solution would allow session multiplexing without most of the perceived downsides of additional RTP sessions creating a need for additional transport flows.

6.5. Security Aspects

When dealing with point-to-point, 2-member RTP sessions only, there are few security issues that are relevant to the choice of having one RTP session or multiple RTP sessions. However, there are a few aspects of multiparty sessions that might warrant consideration.

6.5.1. Security Context Scope

When using SRTP [RFC3711] the security context scope is important and can be a necessary differentiation in some applications. As SRTP's crypto suites (so far) is built around symmetric keys, the receiver will need to have the same key as the sender. This results in that no one in a multi-party session can be certain that a received packet really was sent by the claimed sender or by another party having access to the key. In most cases this is a sufficient security property, but there are a few cases where this does create situations.

The first case is when someone leaves a multi-party session and one wants to ensure that the party that left can no longer access the media streams. This requires that everyone re-keys without disclosing the keys to the excluded party.

A second case is when using security as an enforcing mechanism for differentiation. Take for example a scalable layer or a high quality simulcast version which only premium users are allowed to access. The mechanism preventing a receiver from getting the high quality stream can be based on the stream being encrypted with a key that user can't access without paying premium, having the key-management limit access to the key.

SRTP [RFC3711] has not special functions for dealing with different sets of master keys for different SSRCs. The key-management functions has different capabilities to establish different set of keys, normally on a per end-point basis. DTLS-SRTP [RFC5764] and Security Descriptions [RFC4568] for example establish different keys for outgoing and incoming traffic from an end-point. This key usage must be written into the cryptographic context, possibly associated with different SSRCs.

6.5.2. Key Management for Multi-party session

Performing key-management for multi-party session can be a challenge. This section considers some of the issues.

Multi-party sessions, such as transport translator based sessions and multicast sessions, cannot use Security Description [RFC4568] nor

DTLS-SRTP [RFC5764] without an extension as each endpoint provides its set of keys. In centralised conference, the signalling counterpart is a conference server and the media plane unicast counterpart (to which DTLS messages would be sent) is the transport translator. Thus an extension like Encrypted Key Transport [I-D.ietf-avt-srtp-ekt] is needed or a MIKEY [RFC3830] based solution that allows for keying all session participants with the same master key.

6.5.3. Complexity Implications

The usage of security functions can surface complexity implications of the choice of multiplexing and topology. This becomes especially evident in RTP topologies having any type of middlebox that processes or modifies RTP/RTCP packets. Where there is very small overhead for an RTP translator or mixer to rewrite an SSRC value in the RTP packet of an unencrypted session, the cost of doing it when using cryptographic security functions is higher. For example if using SRTP [RFC3711], the actual security context and exact crypto key are determined by the SSRC field value. If one changes it, the encryption and authentication tag must be performed using another key. Thus changing the SSRC value implies a decryption using the old SSRC and its security context followed by an encryption using the new one.

7. Arch-Types

This section discusses some arch-types of how RTP multiplexing can be used in applications to achieve certain goals and a summary of their implications. For each arch-type there is discussion of benefits and downsides.

7.1. Single SSRC per Session

In this arch-type each endpoint in a point-to-point session has only a single SSRC, thus the RTP session contains only two SSRCs, one local and one remote. This session can be used both unidirectional, i.e. only a single media stream or bi-directional, i.e. both endpoints have one media stream each. If the application needs additional media flows between the endpoints, they will have to establish additional RTP sessions.

The Pros:

1. This arch-type has great legacy interoperability potential as it will not tax any RTP stack implementations.

2. The signalling has good possibilities to negotiate and describe the exact formats and bit-rates for each media stream, especially using today's tools in SDP.

3. It does not matter if usage or purpose of the media stream is signalled on media stream level or session level as there is no difference.

4. It is possible to control security association per RTP session with current key-management.

The Cons:

a. The number of required RTP sessions cannot really be higher, which has the implications:

   * Linear growth of the amount of NAT/FW state with number of media streams.

   * Increased delay and resource consumption from NAT/FW traversal.

   * Likely larger signalling message and signalling processing requirement due to the amount of session related information.

   * Higher potential for a single media stream to fail during transport between the endpoints.

b. When the number of RTP sessions grows, the amount of explicit state for relating media stream also grows, linearly or possibly exponentially, depending on how the application needs to relate media streams.

c. The port consumption may become a problem for centralised services, where the central node's port consumption grows rapidly with the number of sessions.

d. For applications where the media streams are highly dynamic in their usage, i.e. entering and leaving, the amount of signalling can grow high. Issues arising from the timely establishment of additional RTP sessions can also arise.

e. Cross session RTCP requests needs is likely to exist and may cause issues.

f. If the same SSRC value is reused in multiple RTP sessions rather than being randomly chosen, interworking with applications that uses another multiplexing structure than this application will

have issues and require SSRC translation.

g. Cannot be used with Any Source Multicast (ASM) as one cannot guarantee that only two endpoints participate as packet senders. Using SSM, it is possible to restrict to these requirements if no RTCP feedback is used.

h. For most security mechanisms, each RTP session or transport flow requires individual key-management and security association establishment thus increasing the overhead.

i. Does not support multiparty session within a session. Instead each multi-party participant will require an individual RTP session to a given endpoint, even if a central node is used.

RTP applications that need to inter-work with legacy RTP applications, like VoIP and video conferencing, can potentially benefit from this structure. However, a large number of media descriptions in SDP can also run into issues with existing implementations. For any application needing a larger number of media flows, the overhead can become very significant. This structure is also not suitable for multi-party sessions, as any given media stream from each participant, although having same usage in the application, must have its own RTP session. In addition, the dynamic behaviour that can arise in multi-party applications can tax the signalling system and make timely media establishment more difficult.

7.2. Multiple SSRCs of the Same Media Type

In this arch-type, each RTP session serves only a single media type. The RTP session can contain multiple media streams, either from a single endpoint or due to multiple endpoints. This commonly creates a low number of RTP sessions, typically only two one for audio and one for video with a corresponding need for two listening ports when using RTP and RTCP multiplexing.

The Pros:

1. Low number of RTP sessions needed compared to single SSRC case. This implies:

   * Reduced NAT/FW state

   * Lower NAT/FW Traversal Cost in both processing and delay.

2. Allows for early de-multiplexing in the processing chain in RTP applications where all media streams of the same type have the same usage in the application.

3. Works well with media type de-composite endpoints.

4. Enables Flow-based QoS with different prioritisation between media types.

5. For applications with dynamic usage of media streams, i.e. they come and go frequently, having much of the state associated with the RTP session rather than an individual SSRC can avoid the need for in-session signalling of meta-information about each SSRC.

6. Low overhead for security association establishment.

The Cons:

a. May have some need for cross session RTCP requests for things that affect both media types in an asynchronous way.

b. Some potential for concern with legacy implementations that does not support the RTP specification fully when it comes to handling multiple SSRC per endpoint.

c. Will not be able to control security association for sets of media streams within the same media type with today's key-management mechanisms, only between SDP media descriptions.

For RTP applications where all media streams of the same media type share same usage, this structure provides efficiency gains in amount of network state used and provides more faith sharing with other media flows of the same type. At the same time, it is still maintaining almost all functionalities when it comes to negotiation in the signalling of the properties for the individual media type and also enabling flow based QoS prioritisation between media types. It handles multi-party session well, independently of multicast or centralised transport distribution, as additional sources can dynamically enter and leave the session.

7.3. Multiple Sessions for one Media type

In this arch-type one goes one step further than in the above (Section 7.2) by using multiple RTP sessions also for a single media type. The main reason for going in this direction is that the RTP application needs separation of the media streams due to their usage. Some typical reasons for going to this arch-type are scalability over multicast, simulcast, need for extended QoS prioritisation of media streams due to their usage in the application, or the need for fine granular signalling using today's tools.

The Pros:

1. More suitable for Multicast usage where receivers can individually select which RTP sessions they want to participate in, assuming each RTP session has its own multicast group.

2. Detailed indication of the application's usage of the media stream, where multiple different usages exist.

3. Less need for SSRC specific explicit signalling for each media stream and thus reduced need for explicit and timely signalling.

4. Enables detailed QoS prioritisation for flow based mechanisms.

5. Works well with de-composite endpoints.

6. Handles dynamic usage of media streams well.

7. For transport translator based multi-party sessions, this structure allows for improved control of which type of media streams an endpoint receives.

8. The scope for who is included in a security association can be structured around the different RTP sessions, thus enabling such functionality with existing key-management.

The Cons:

a. Increases the amount of RTP sessions compared to Multiple SSRCs of the Same Media Type.

b. Increased amount of session configuration state.

c. May need synchronised cross-session RTCP requests and require some consideration due to this.

d. For media streams that are part of scalability, simulcast or transport robustness it will be needed to bind sources, which must support multiple RTP sessions.

e. Some potential for concern with legacy implementations that does not support the RTP specification fully when it comes to handling multiple SSRC per endpoint.

f. Higher overhead for security association establishment.

g. If the applications need finer control than on media type level over which session participants that are included in different sets of security associations, most of today's key-management will have difficulties establishing such a session.

For more complex RTP applications that have several different usages for media streams of the same media type and / or uses scalability or simulcast, this solution can enable those functions at the cost of increased overhead associated with the additional sessions. This type of structure is suitable for more advanced applications as well as multicast based applications requiring differentiation to different participants.

7.4. Multiple Media Types in one Session

This arch-type is to use a single RTP session for multiple different media types, like audio and video, and possibly also transport robustness mechanisms like FEC or Retransmission. Each media stream will use its own SSRC and a given SSRC value from a particular endpoint will never use the SSRC for more than a single media type.

The Pros:

1. Single RTP session which implies:

   * Minimal NAT/FW state.

   * Minimal NAT/FW Traversal Cost.

   * Fate-sharing for all media flows.

2. Enables separation of the different media types based on the payload types so media type specific endpoint or central processing can still be supported despite single session.

3. Can handle dynamic allocations of media streams well on an RTP level. Depends on the application's needs for explicit indication of the stream usage and how timely that can be signalled.

4. Minimal overhead for security association establishment.

The Cons:

a. Less suitable for interworking with other applications that uses individual RTP sessions per media type or multiple sessions for a single media type, due to need of SSRC translation.

b. Negotiation of bandwidth for the different media types is currently not possible in SDP. This requires SDP extensions to enable payload or source specific bandwidth. Likely to be a problem due to media type asymmetry in required bandwidth.

c. Not suitable for de-composite end-points as it requires higher bandwidth and processing.

d. Flow based QoS cannot provide separate treatment to some media streams compared to other in the single RTP session.

e. If there is significant asymmetry between the media streams RTCP reporting needs, there are some challenges in configuration and usage to avoid wasting RTCP reporting on the media stream that does not need that frequent reporting.

f. Not suitable for applications where some receivers like to receive only a subset of the media streams, especially if multicast or transport translator is being used.

g. Additional concern with legacy implementations that does not support the RTP specification fully when it comes to handling multiple SSRC per endpoint, as also multiple simultaneous media types needs to be handled.

h. If the applications need finer control over which session participants that are included in different sets of security associations, most key-management will have difficulties establishing such a session.

The analysis in this document and considerations in ??? implies that this is suitable only in a set of restricted use cases. The aspect in the above list that can be most difficult to judge long term is likely the potential need for interworking with other applications and services.

7.5. Summary

There are some clear relations between these arch-types. Both the "single SSRC per RTP session" and the "multiple media types in one session" are cases which require full explicit signalling of the media stream relations. However, they operate on two different levels where the first primarily enables session level binding, and the second needs to do it all on SSRC level. From another perspective, the two solutions are the two extreme points when it comes to number of RTP sessions required.

The two other arch-types "Multiple SSRCs of the Same Media Type" and "Multiple Sessions for one Media Type" are examples of two other cases that first of all allows for some implicit mapping of the role or usage of the media streams based on which RTP session they appear in. It thus potentially allows for less signalling and in particular reduced need for real-time signalling in dynamic sessions. They also

represent points in between the first two when it comes to amount of RTP sessions established, i.e. representing an attempt to reduce the amount of sessions as much as possible without compromising the functionality the session provides both on network level and on signalling level.

8. Summary considerations and guidelines

8.1. Guidelines

This section contains a number of recommendations for implementors or specification writers when it comes to handling multi-stream.

Do not Require the same SSRC across Sessions: As discussed in Section 6.2.5 there exist drawbacks in using the same SSRC in multiple RTP sessions as a mechanism to bind related media streams together. It is instead recommended that a mechanism to explicitly signal the relation is used, either in RTP/RTCP or in the used signalling mechanism that establishes the RTP session(s).

Use SSRC multiplexing for additional Media Sources: In the cases an RTP endpoint needs to transmit additional media streams of the same media type in the application, with the same processing requirements at the network and RTP layers, it is recommended to send them as additional SSRCs in the same RTP session. For example a telepresence room where there are three cameras, and each camera captures 2 persons sitting at the table, sending each camera as its own SSRC within a single RTP session is recommended.

Use additional RTP sessions for streams with different requirements: When media streams have different processing requirements from the network or the RTP layer at the endpoints, it is recommended that the different types of streams are put in different RTP sessions. This includes the case where different participants want different subsets of the set of RTP streams.

When using Session Multiplexing use grouping: When using Session Multiplexing solutions, it is recommended to be explicitly group the involved RTP sessions using the signalling mechanism, for example The Session Description Protocol (SDP) Grouping Framework. [RFC5888], using some appropriate grouping semantics.

RTP/RTCP Extensions May Support SSRC and Session Multiplexing: When defining an RTP or RTCP extension, the creator needs to consider if this extension is applicable in both SSRC multiplexed and Session multiplexed usages. Any extension intended to be generic is recommended to support both. Applications that are not as

generally applicable will have to consider if interoperability is better served by defining a single solution or providing both options.

Transport Support Extensions: When defining new RTP/RTCP extensions intended for transport support, like the retransmission or FEC mechanisms, they are recommended to include support for both SSRC and Session multiplexing so that application developers can choose freely from the set of mechanisms without concerning themselves with which of the multiplexing choices a particular solution supports.

9. IANA Considerations

This document makes no request of IANA.

Note to RFC Editor: this section may be removed on publication as an RFC.

10. Security Considerations

There is discussion of the security implications of choosing SSRC vs Session multiplexing in Section 6.5.

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Appendix A. Dismissing Payload Type Multiplexing

This section documents a number of reasons why using the payload type as a multiplexing point for most things related to multiple streams is unsuitable. If one attempts to use Payload type multiplexing beyond it's defined usage, that has well known negative effects on RTP. To use Payload type as the single discriminator for multiple streams implies that all the different media streams are being sent with the same SSRC, thus using the same timestamp and sequence number space. This has many effects:

1. Putting restraint on RTP timestamp rate for the multiplexed media. For example, media streams that use different RTP timestamp rates cannot be combined, as the timestamp values need to be consistent across all multiplexed media frames. Thus streams are forced to use the same rate. When this is not possible, Payload Type multiplexing cannot be used.

2. Many RTP payload formats may fragment a media object over multiple packets, like parts of a video frame. These payload formats need to determine the order of the fragments to

correctly decode them. Thus it is important to ensure that all fragments related to a frame or a similar media object are transmitted in sequence and without interruptions within the object. This can relatively simple be solved on the sender side by ensuring that the fragments of each media stream are sent in sequence.

3. Some media formats require uninterrupted sequence number space between media parts. These are media formats where any missing RTP sequence number will result in decoding failure or invoking of a repair mechanism within a single media context. The text/T140 payload format [RFC4103] is an example of such a format. These formats will need a sequence numbering abstraction function between RTP and the individual media stream before being used with Payload Type multiplexing.

4. Sending multiple streams in the same sequence number space makes it impossible to determine which Payload Type and thus which stream a packet loss relates to.

5. If RTP Retransmission [RFC4588] is used and there is a loss, it is possible to ask for the missing packet(s) by SSRC and sequence number, not by Payload Type. If only some of the Payload Type multiplexed streams are of interest, there is no way of telling which missing packet(s) belong to the interesting stream(s) and all lost packets must be requested, wasting bandwidth.

6. The current RTCP feedback mechanisms are built around providing feedback on media streams based on stream ID (SSRC), packet (sequence numbers) and time interval (RTP Timestamps). There is almost never a field to indicate which Payload Type is reported, so sending feedback for a specific media stream is difficult without extending existing RTCP reporting.

7. The current RTCP media control messages [RFC5104] specification is oriented around controlling particular media flows, i.e. requests are done addressing a particular SSRC. Such mechanisms would need to be redefined to support Payload Type multiplexing.

8. The number of payload types are inherently limited. Accordingly, using Payload Type multiplexing limits the number of streams that can be multiplexed and does not scale. This limitation is exacerbated if one uses solutions like RTP and RTCP multiplexing [RFC5761] where a number of payload types are blocked due to the overlap between RTP and RTCP.

9. At times, there is a need to group multiplexed streams and this is currently possible for RTP Sessions and for SSRC, but there is no defined way to group Payload Types.

10. It is currently not possible to signal bandwidth requirements per media stream when using Payload Type Multiplexing.

11. Most existing SDP media level attributes cannot be applied on a per Payload Type level and would require re-definition in that context.

12. A legacy endpoint that doesn't understand the indication that different RTP payload types are different media streams may be slightly confused by the large amount of possibly overlapping or identically defined RTP Payload Types.

Appendix B. Proposals for Future Work

The above discussion and guidelines indicates that a small set of extension mechanisms could greatly improve the situation when it comes to using multiple streams independently of Session multiplexing or SSRC multiplexing. These extensions are:

Media Source Identification: A Media source identification that can be used to bind together media streams that are related to the same media source. A proposal [I-D.westerlund-avtext-rtcp-sdes-srcname] exist for a new SDES item SRCNAME that also can be used with the a=ssrc SDP attribute to provide signalling layer binding information.

SSRC limitations within RTP sessions: By providing a signalling solution that allows the signalling peers to explicitly express both support and limitations on how many simultaneous media streams an endpoint can handle within a given RTP Session. That ensures that usage of SSRC multiplexing occurs when supported and without overloading an endpoint. This extension is proposed in [I-D.westerlund-avtcore-max-ssrc].

Appendix C. RTP Specification Clarifications

This section describes a number of clarifications to the RTP specifications that are likely necessary for aligned behaviour when RTP sessions contain more SSRCs than one local and one remote.

All of the below proposals are under consideration in [I-D.lennox-avtcore-rtp-multi-stream].

C.1. RTCP Reporting from all SSRCs

When one has multiple SSRC in an RTP node, all these SSRC must send some RTP or RTCP packet as long as the SSRC exist. It is not sufficient that only one SSRC in the node sends report blocks on the incoming RTP streams; any SSRC that intends to remain in the session must send some packets to avoid timing out according to the rules in RFC 3550 section 6.3.5.

It has been hypothesised that a third party monitor may be confused by not necessarily being able to determine that all these SSRC are in fact co-located and originate from the same stack instance; if this hypothesis is true, this may argue for having all the sources send full reception reports, even though they are reporting the same packet delivery.

The contrary argument is that such double reporting may confuse the third party monitor even more by making it seem that utilisation of the last-hop link to the recipient is (number of SSRCs) times higher than what it actually is.

C.2. RTCP Self-reporting

For any RTP node that sends more than one SSRC, there is the question if SSRC1 needs to report its reception of SSRC2 and vice versa. The reason that they in fact need to report on all other local streams as being received is report consistency. The hypothetical third party monitor that considers the full matrix of media streams and all known SSRC reports on these media streams would detect a gap in the reports which could be a transport issue unless identified as in fact being sources from the same node.

C.3. Combined RTCP Packets

When a node contains multiple SSRCs, it is questionable if an RTCP compound packet can only contain RTCP packets from a single SSRC or if multiple SSRCs can include their packets in a joint compound packet. The high level question is a matter for any receiver processing on what to expect. In addition to that question there is the issue of how to use the RTCP timer rules in these cases, as the existing rules are focused on determining when a single SSRC can send.

Appendix D. Signalling considerations

Signalling is not an architectural consideration for RTP itself, so this discussion has been moved to an appendix. However, it is hugely

important for anyone building complete applications, so it is deserving of discussion.

The issues raised here need to be addressed in the WGs that deal with signalling; they cannot be addressed by tweaking, extending or profiling RTP.

D.1. Signalling Aspects

There exist various signalling solutions for establishing RTP sessions. Many are SDP [RFC4566] based, however SDP functionality is also dependent on the signalling protocols carrying the SDP. Where RTSP [RFC2326] and SAP [RFC2974] both use SDP in a declarative fashion, while SIP [RFC3261] uses SDP with the additional definition of Offer/Answer [RFC3264]. The impact on signalling and especially SDP needs to be considered as it can greatly affect how to deploy a certain multiplexing point choice.

D.1.1. Session Oriented Properties

One aspect of the existing signalling is that it is focused around sessions, or at least in the case of SDP the media description. There are a number of things that are signalled on a session level/ media description but those are not necessarily strictly bound to an RTP session and could be of interest to signal specifically for a particular media stream (SSRC) within the session. The following properties have been identified as being potentially useful to signal not only on RTP session level:

- Bitrate/Bandwidth exist today only at aggregate or a common any media stream limit, unless either codec-specific bandwidth limiting or RTCP signalling using TMMBR is used.

- Which SSRC that will use which RTP Payload Types (this will be visible from the first media packet, but is sometimes useful to know before packet arrival).

Some of these issues are clearly SDP's problem rather than RTP limitations. However, if the aim is to deploy an SSRC multiplexed solution that contains several sets of media streams with different properties (encoding/packetization parameter, bit-rate, etc), putting each set in a different RTP session would directly enable negotiation of the parameters for each set. If insisting on SSRC multiplexing only, a number of signalling extensions are needed to clarify that there are multiple sets of media streams with different properties and that they shall in fact be kept different, since a single set will not satisfy the application's requirements.

For some parameters, such as resolution and framerate, a SSRC-linked mechanism has been proposed: [I-D.lennox-mmusic-sdp-source-selection].

D.1.2. SDP Prevents Multiple Media Types

SDP chose to use the m= line both to delineate an RTP session and to specify the top level of the MIME media type; audio, video, text, image, application. This media type is used as the top-level media type for identifying the actual payload format bound to a particular payload type using the rtpmap attribute. This binding has to be loosened in order to use SDP to describe RTP sessions containing multiple MIME top level types.

There is an accepted WG item in the MMUSIC WG to define how multiple media lines describe a single underlying transport [I-D.ietf-mmusic-sdp-bundle-negotiation] and thus it becomes possible in SDP to define one RTP session with media types having different MIME top level types.

D.1.3. Signalling Media Stream Usage

Media streams being transported in RTP has some particular usage in an RTP application. This usage of the media stream is in many applications so far implicitly signalled. For example, an application may choose to take all incoming audio RTP streams, mix them and play them out. However, in more advanced applications that use multiple media streams there will be more than a single usage or purpose among the set of media streams being sent or received. RTP applications will need to signal this usage somehow. The signalling used will have to identify the media streams affected by their RTP-level identifiers, which means that they have to be identified either by their session or by their SSRC + session.

In some applications, the receiver cannot utilise the media stream at all before it has received the signalling message describing the media stream and its usage. In other applications, there exists a default handling that is appropriate.

If all media streams in an RTP session are to be treated in the same way, identifying the session is enough. If SSRCs in a session are to be treated differently, signalling must identify both the session and the SSRC.

If this signalling affects how any RTP central node, like an RTP mixer or translator that selects, mixes or processes streams, treats the streams, the node will also need to receive the same signalling to know how to treat media streams with different usage in the right

fashion.

Appendix E. Changes from -01 to -02

- Added Harald Alvestrand as co-author.
- Removed unused term "Media aggregate".
- Added term "RTP session group", noted that CNAMEs are assumed to bind across the sessions of an RTP session group, and used it when appropriate (TODO)
- Moved discussion of signalling aspects to appendix
- Removed all suggestion that PT can be a multiplexing point
- Normalised spelling of "endpoint" to follow RFC 3550 and not use a hyphen.
- Added CNAME to definition list.
- Added term "Media Sink" for the thing that is identified by a listen-only SSRC.
- Added term "RTP source" for the thing that transmits one media stream, separating it from "Media Source". [[OUTSTANDING: Whether to use "RTP Source" or "Media Sender" here]]
- Rewrote section on distributed endpoint, noting that this, like any endpoint that wants a subset of a set of RTP streams, needs multiple RTP sessions.
- Removed all substantive references to the undefined term "purpose" from the main body of the document when it referred to the purpose of an RTP stream.
- Moved the summary section of section 6 to the guidelines section that it most closely supports.
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