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Multipath RTP (MPRTP)
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Abstract

The Real-time Transport Protocol (RTP) is used to deliver real-time content and, along with the RTP Control Protocol (RTCP), forms the control channel between the sender and receiver. However, RTP and RTCP assume a single delivery path between the sender and receiver and make decisions based on the measured characteristics of this single path. Increasingly, endpoints are becoming multi-homed, which means that they are connected via multiple Internet paths. Network utilization can be improved when endpoints use multiple parallel paths for communication. The resulting increase in reliability and throughput can also enhance the user experience. This document extends the Real-time Transport Protocol (RTP) so that a single session can take advantage of the availability of multiple paths between two endpoints.

Status of this Memo

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1. Introduction

Multi-homed endpoints are becoming common in today's Internet, e.g., devices that support multiple wireless access technologies such as 3G and Wireless LAN. This means that often there is more than one network path available between two endpoints. Transport protocols, such as RTP, have not been designed to take advantage of the availability of multiple concurrent paths and therefore cannot benefit from the increased capacity and reliability that can be achieved by pooling their respective capacities.

Multipath RTP (MPRTP) is an OPTIONAL extension to RTP [1] that allows splitting a single RTP stream into multiple subflows that transmit over different paths. In effect, this pools the resource capacity of multiple paths. Multipath RTCP (MPRTCP) is an extension to RTCP, it is used along with MPRTP to report per-path sender and receiver characteristics.

Other IETF transport protocols that are capable of using multiple paths include SCTP [9], MPTCP MPTCP [10] and SHIM6 [11]. However, these protocols are not suitable for realtime communications.

1.1. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [2].

1.2. Terminology

- o Endpoint: host either initiating or terminating an RTP connection.
- o Interface: A logical or physical component that is capable of acquiring a unique IP address.
- o Path: sequence of links between a sender and a receiver. Typically, defined by a set of source and destination addresses.
- o Subflow: A flow of RTP packets along a specific path, i.e., a subset of the packets belonging to an RTP stream. The combination of all RTP subflows forms the complete RTP stream.

1.3. Use-cases

The primary use-case for MPRTP is transporting high bit-rate streaming multimedia content between endpoints, where at least one is multi-homed. Such endpoints could be residential IPTV devices that connect to the Internet through two different Internet service

providers (ISPs), or mobile devices that connect to the Internet through 3G and WLAN interfaces. By allowing RTP to use multiple paths for transmission, the following gains can be achieved:

- o Higher quality: Pooling the resource capacity of multiple Internet paths allows higher bit-rate and higher quality codecs to be used. From the application perspective, the available bandwidth between the two endpoints increases.
- o Load balancing: Transmitting one RTP stream over multiple paths can reduce the bandwidth usage, compared to transmitting the same stream along a single path. This reduces the impact on other traffic.
- o Fault tolerance: When multiple paths are used in conjunction with redundancy mechanisms (FEC, re-transmissions, etc.), outages on one path have less impact on the overall perceived quality of the stream.

A secondary use-case for MP RTP is transporting Voice over IP (VoIP) calls to a device with multiple interfaces. Again, such an endpoint could be a mobile device with multiple wireless interfaces. In this case, little is to be gained from resource pooling, i.e., higher capacity or load balancing, because a single path should be easily capable of handling the required load. However, using multiple concurrent subflows can improve fault tolerance, because traffic can shift between the subflows when path outages occur. This results in very fast transport-layer handovers that do not require support from signaling.

2. Goals

This section outlines the basic goals that multipath RTP aims to meet. These are broadly classified as Functional goals and Compatibility goals.

2.1. Functional goals

Allow unicast RTP session to be split into multiple subflows in order to be carried over multiple paths. This may prove beneficial in case of video streaming.

- o Increased Throughput: Cumulative capacity of the two paths may meet the requirements of the multimedia session. Therefore, MP RTP MUST support concurrent use of the multiple paths.

- o Improved Reliability: MPRTTP SHOULD be able to send redundant or re-transmit packets along any available path to increase reliability.

The protocol SHOULD be able to open new subflows for an existing session when new paths appear and MUST be able to close subflows when paths disappear.

2.2. Compatibility goals

MPRTTP MUST be backwards compatible; an MPRTTP stream needs to fall back to be compatible with legacy RTP stacks if MPRTTP support is not successfully negotiated.

- o Application Compatibility: MPRTTP service model MUST be backwards compatible with existing RTP applications, i.e., an MPRTTP stack MUST be able to work with legacy RTP applications and not require changes to them. Therefore, the basic RTP APIs MUST remain unchanged, but an MPRTTP stack MAY provide extended APIs so that the application can configure any additional features provided by the MPRTTP stack.
- o Network Compatibility: individual RTP subflows MUST themselves be well-formed RTP flows, so that they are able to traverse NATs and firewalls. This MUST be the case even when interfaces appear after session initiation. Interactive Connectivity Establishment (ICE) [3] MAY be used for discovering new interfaces or performing connectivity checks.

3. RTP Topologies

RFC 5117 [12] describes a number of scenarios using mixers and translators in single-party (point-to-point), and multi-party (point-to-multipoint) scenarios. RFC 3550 [1] (Section 2.3 and 7.x) discuss in detail the impact of mixers and translators on RTP and RTCP packets. MPRTTP assumes that if a mixer or translator exists in the network, then either all of the multiple paths or none of the multiple paths go via this component.

4. MPRTTP Architecture

In a typical scenario, an RTP session uses a single path. In an MPRTTP scenario, an RTP session uses multiple subflows that each use a different path. Figure 1 shows the difference.

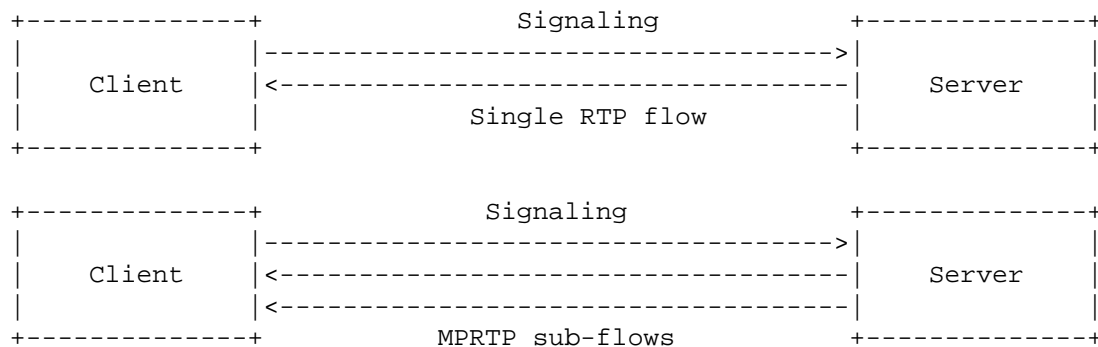


Figure 1: Comparison between traditional RTP streaming and MPRTTP

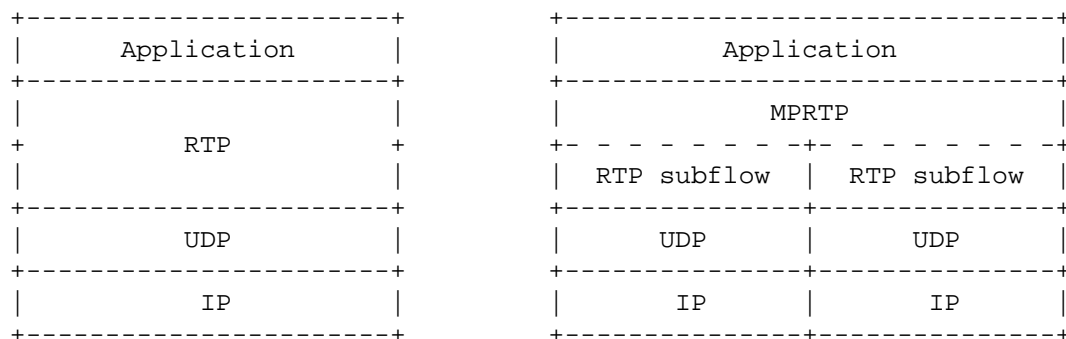


Figure 2: MPRTTP Architecture

Figure 2 illustrates the differences between the standard RTP stack and the MPRTTP stack. MPRTTP receives a normal RTP session from the application and splits it into multiple RTP subflows. Each subflow is then sent along a different path to the receiver. To the network, each subflow appears as an independent, well-formed RTP flow. At the receiver, the subflows are combined to recreate the original RTP session. The MPRTTP layer performs the following functions:

- o Path Management: The layer is aware of alternate paths to the peer, which may, for example, be the peer's multiple interfaces to send differently marked packets along separate paths. MPRTTP also selects interfaces to send and receive data. Furthermore, it manages the port and IP address pair bindings for each subflow.
- o Packet Scheduling: the layer splits a single RTP flow into multiple subflows and sends them across multiple interfaces (paths). The splitting MAY BE done using different path characteristics.

- o Subflow recombination: the layer creates the original stream by recombining the independent subflows. Therefore, the multipath subflows appear as a single RTP stream to applications.

4.1. Relationship of MPRTTP with Session Signaling

Session signaling (e.g., SIP [13], RTSP [14]) SHOULD be done over a failover-capable or multipath-capable transport for e.g., SCTP [9] or MPTCP [10] instead of TCP or UDP.

5. Example Media Flow diagrams

There may be many complex technical scenarios for MPRTTP, however, this memo only considers the following two scenarios: 1) an unidirectional media flow that represents the streaming use-case, and 2) a bidirectional media flow that represents a conversational use-case.

5.1. Streaming use-case

In the unidirectional scenario, the receiver (client) initiates a multimedia session with the sender (server). The receiver or the sender may have multiple interfaces and both endpoints are MPRTTP-capable. Figure 3 shows this scenario. In this case, host A has multiple interfaces. Host B performs connectivity checks on host A's multiple interfaces. If the interfaces are reachable, then host B streams multimedia data along multiple paths to host A. Moreover, host B also sends RTCP Sender Reports (SR) for each subflow and host A responds with a standard RTCP Receiver Report (RR) for the overall session and receiver statistics for each subflow. Host B distributes the packets into the subflows based on the individually measured path characteristics.

Alternatively, to reduce media startup time, host B may start streaming multimedia data to host A's initiating interface and then perform connectivity checks for the other interfaces. This method of updating a single path session to a multipath session is called "multipath session upgrade".

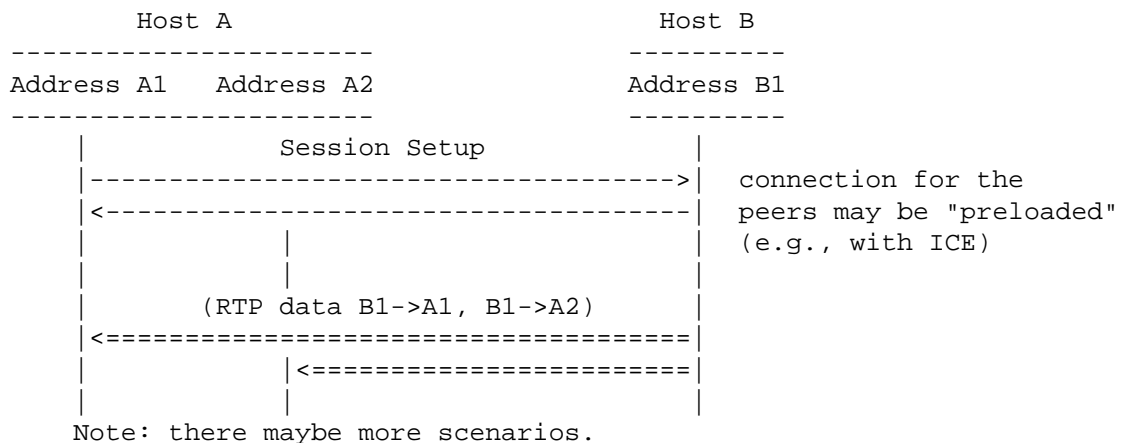


Figure 3: Unidirectional media flow

5.2. Conversational use-case

In the bidirectional scenario, multimedia data flows in both directions. The two hosts exchange their lists of interfaces with each other and perform connectivity checks. Communication begins after each host finds suitable address, port pairs. All interfaces that receive data send back RTCP receiver statistics for each path. The peers balance their multimedia stream over multiple links based on the reception statistics from its peer and its own volume of traffic. Figure 4 describes an example of a bidirectional flow.

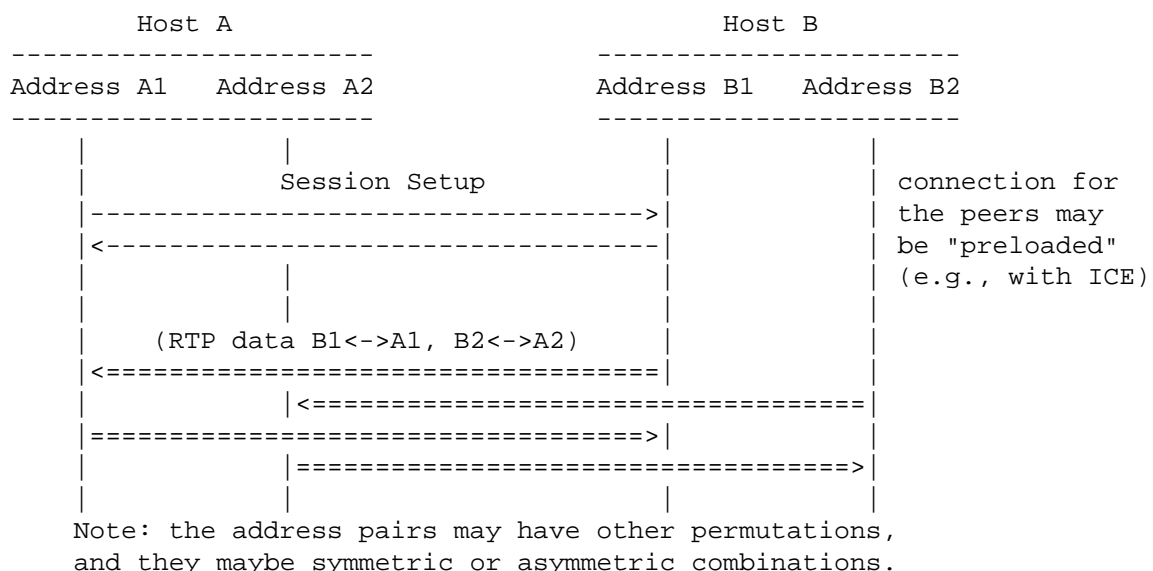


Figure 4: Bidirectional flow

5.3. Challenges with Multipath Interface Discovery

For some applications, where the user expects immediate playback, e.g., High Definition Media Streaming or IPTV, it may not be possible to perform connectivity checks within the given time bound. In these cases, connectivity checks MAY need to be done ahead of time.

[Open Issue: ICE or any other system would have to be aware of the endpoint's interfaces ahead of time].

6. MP RTP Functional blocks

This section describes some of the functional blocks needed for MP RTP. We then investigate each block and consider available mechanisms in the next section.

1. **Session Setup:** Multipath session setup is an upgrade or add-on to a typical RTP session. Interfaces may appear or disappear at anytime during the session. To preserve backward compatibility with legacy applications, a multipath session MUST look like a bundle of individual RTP sessions.
2. **Expanding RTP:** For a multipath session, each subflow MUST look like an independent RTP flow, so that individual RTCP messages can be generated per subflow. Furthermore, MP RTP splits the single multimedia stream into multiple subflows based on path characteristics (e.g. RTT, loss-rate, receiver rate, bandwidth-delay product etc.) and dynamically adjusts the load on each link.
3. **Adding Interfaces:** Interfaces on the host need to be regularly discovered and signaled. This can be done at session setup and/or during the session. When discovering and receiving new interfaces, the MP RTP layer needs to select address and port pairs.
4. **Expanding RTCP:** MP RTP MUST recombine RTCP reports from each path to re-create a single RTCP message to maintain backward compatibility with legacy applications.
5. **Maintenance and Failure Handling:** In a multi-homed endpoint interfaces may appear and disappear. If this happens at the sender, it has to re-adjust the load on the available links. On the other hand, if this occurs on the receiver, then the multimedia data transmitted by the sender to those interfaces is lost. This data may be re-transmitted along a different path i.e., to a different interface on the receiver. Furthermore, the

receiver has to explicitly signal the disappearance of an interface, or the sender has to detect it. What happens if the interface that setup the session disappears? does the control channel also failover? re-start the session?

6. Teardown: The MP RTP layer releases the occupied ports on the interfaces.

7. Available mechanisms within the functional blocks

This section discusses some of the possible alternatives for each functional block mentioned in the previous section.

7.1. Session Setup

MP RTP session can be set up in many possible ways e.g., during handshake, or upgraded mid-session. The capability exchange may be done using out-of-band signaling (e.g., SDP [15] in SIP [13], RTSP [14]) or in-band signaling (e.g., RTP/RTCP header extension). Furthermore, ICE [3] may be used for discovering and performing connectivity checks during session setup.

7.2. Expanding RTP

RTCP [1] is generated per media session. However, with MP RTP, the media sender spreads the RTP load across several interfaces. The media sender SHOULD make the path selection, load balancing and fault tolerance decisions based on the characteristics of each path. Therefore, apart from normal RTP sequence numbers defined in [1], the MP RTP sender MUST add subflow-specific sequence numbers to help calculate fractional losses, jitter, RTT, playout time, etc., for each path and a flow identifier to associate the characteristics to a path. The RTP header extension for MP RTP is shown in Section 9).

7.3. Adding New Interfaces

When interfaces appear and disappear mid-session, ICE [3] may be used for discovering interfaces and performing connectivity checks. However, MP RTP may require a capability re-negotiation (using SDP) to include all these new interfaces. This method is referred to as out-of-band multipath advertisement.

Alternatively, when new interfaces appear the interface advertisements may be done in-band using RTP/RTCP extensions. The peers perform connectivity checks (see Figure 5 for more details). If the connectivity packets are received by the peers, then multimedia data can flow between the new address, port pairs.

7.4. Expanding RTCP

To provide accurate per path information an MPRTTP host MUST send (SR/RR) report for each unique subflow along with the overall overall session RTCP report. Therefore, the additional sub flow reporting affects the RTCP bandwidth and the RTCP reporting interval for each subflow. RTCP report scheduling for each subflow may cause a problem for RTCP recombination and reconstruction in cases when 1) RTCP for a subflow is lost, and 2) RTCP for a subflow arrives later than the other subflows. (There maybe other cases as well.)

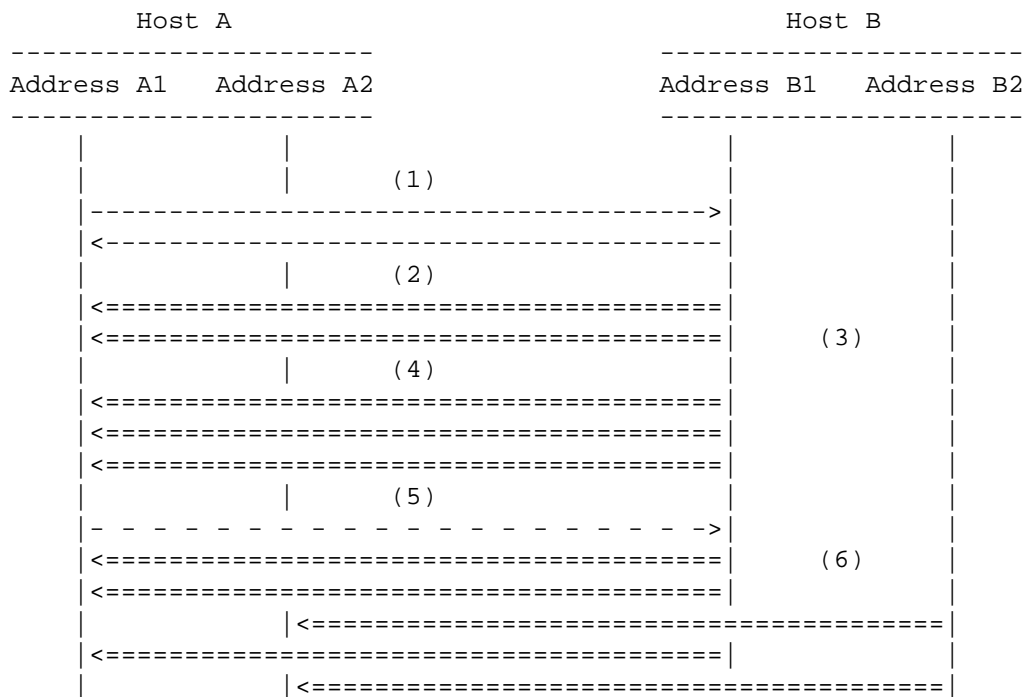
The sender distributes the media across different paths using the per path RTCP reports. However, this document doesn't cover algorithms for congestion control or load balancing.

7.5. Checking and Failure Handling

[Note: If the original interface that setup the session disappears then does the session signaling failover to another interface? Can we recommend that SIP/RTSP be run over MPTCP, SCTP].

8. MPRTTP Protocol

To enable a quick start of a multimedia session, a multipath session MUST be upgraded from a single path session. Therefore, no explicit changes are needed in multimedia session setup and the session can be setup as before.



Key:

| Interface
 ---> Signaling Protocol
 <=== RTP Packets
 - -> RTCP Packet

Figure 5: MPRTTP New Interface

8.1. Overview

The bullet points explain the different steps shown in Figure 5 for upgrading a standard single path multimedia session to multipath session.

- (1) The first two interactions between the hosts describes the standard session setup. This may be SIP or RTSP.
- (2) Following the setup, like in a conventional RTP scenario, host B using interface B1 starts to stream data to host A at interface A1.
- (3) Host B is an MPRTTP-capable media sender and becomes aware of another interface B2.

(4) Host B advertises the multiple interface addresses using an RTP header extensions.

(5) Host A is an MPRTTP-capable media receiver and becomes aware of another interface A2. It advertises the multiple interface addresses using an RTCP RR extension.

Side note, if an MPRTTP-capable host has only one interface even then it SHOULD advertise its single interface.

(6) Each host receives information about the additional interfaces and performs the connectivity tests (not shown in figure). If the paths are reachable then the host starts to stream the multimedia content using the additional paths.

8.1.1. Subflow or Interface Advertisement

To advertise the multiple interfaces, an MPRTTP-capable endpoint MUST add the MPRTTP Interface Advertisement defined in Figure 8 with the RTCP Sender Report (SR). Each unique address is encapsulated in an Interface Advertisement block and contains the IP address, RTP and RTCP port addresses. The Interface Advertisement blocks are ordered based on a decreasing priority level. On receiving the MPRTTP Interface Advertisement, an MPRTTP-capable receiver MUST respond with its own set of interfaces.

If the sender and receiver have only one interface, then the endpoints MUST respond with the default IP, RTP port and RTCP port addresses. If an endpoint receives an RTCP report without the MPRTTP Interface Advertisement, then the endpoint MUST assume that the other endpoint is not MPRTTP capable.

8.1.2. Path selection

After MPRTTP support has been discovered and interface advertisements have been exchanged, the sender MUST initiate connectivity checks to determine which interface pairs offer valid paths between the sender and the receiver. Each combination of IP addresses and port pairs (5-tuple) is a unique subflow. An endpoint MUST associate a Flow ID to each unique subflow. To initiate a connectivity check, the endpoints send an RTP packet using the appropriate MPRTTP extension header (See Figure 7), associated Flow ID and no RTP payload. The receiving endpoint replies to each connectivity check with an RTCP packet with the appropriate packet type (See Figure 9) and Flow ID. After the endpoint receives the reply, the path is considered a valid candidate for sending data. An endpoint MAY choose to do any number of connectivity checks for any interface pairs at any point in a session.

[Open Issue: How should the endpoint adjust the RTCP Reporting interval/schedule the RTCP packet on receiving a connectivity check containing a new FlowID? Editor: One option is Immediately as defined in [4]]

8.1.3. Opening subflows

The sender MAY open any number of subflows from the set of candidate subflows after performing connectivity checks. To use the subflow, the sender simply starts sending the RTP packets with an MPRTTP extension shown in Figure 6. The MPRTTP extension carries a mapping of a subflow packet to the aggregate flow. Namely, sequence numbers and timestamps associated with the subflow.

[Open Issue: How to differentiate between Passive and Active connections?]

[Open Issue: How to keep a passive connection alive, if not actively used?]

8.2. RTP Transmission

The MPRTTP layer SHOULD associate an RTP packet to a subflow based on a scheduling strategy. The scheduling strategy may either choose to augment the paths to create higher throughput or use the alternate paths for enhancing resilience or error-repair. Due to the changes in path characteristics, an MPRTTP sender can change its scheduling strategy during an ongoing session. The MPRTTP sender MUST also populate the flow specific fields described in the MPRTTP extension header (see [Section 9.1](#)).

8.3. Playout Considerations at the Receiver

A media receiver, irrespective of MPRTTP support or not, should be able to playback the media stream because the received RTP packets are compliant to [1], i.e., a non-MPRTTP receiver will ignore the MPRTTP header and still be able to playback the RTP packets. However, the variation of jitter and loss per path may affect proper playout. By calculating optimum skew across all paths, the receiver can compensate for the jitter by modifying the playout delay (adaptive playout) of the received RTP packets.

8.4. Flow specific RTCP Statistics and RTCP Aggregation

Aggregate RTCP provides the overall media statistics and follows the standard RTCP defined in [RFC3550](#) [1]. However, flow specific RTCP provides the per path media statistics because the aggregate RTCP report may not provide sufficient per path information to an MPRTTP

scheduler. Specifically, the scheduler should be aware of each path's RTT and loss-rate, which an aggregate RTCP cannot provide. The sender/receiver MUST use non-compound RTCP reports defined in [RFC5506](#) [5] to transmit the aggregate and flow-specific RTCP reports. Also, each subflow and the aggregate RTCP report MUST follow the timing rules defined in [4].

The RTCP reporting interval is locally implemented and the scheduling of the RTCP reports may depend on the the behavior of each path. For instance, the RTCP interval may be different for a passive path than an active path to keep port bindings alive. Additionally, a peer may decide to share the RTCP reporting bit rate equally across all its paths or schedule based on the receiver rate on each path.

8.5. RTCP Transmission

The sender sends an RTCP SR on each active path. For each SR the receiver gets, it echoes one back to the same IP address-port pair that sent the SR. The receiver tries to choose the symmetric path and if the routing is symmetric then the per-path RTT calculations will work out correctly. However, even if the paths are not symmetric, the sender would at maximum, under-estimate the RTT of the path by a factor of half of the actual path RTT.

9. Packet Formats

In this sub-section we define the protocol structures described in the previous sections.

9.1. MP RTP Header Extension

The MP RTP header extension is used to 1) distribute a single RTP stream over multiple subflows, 2) advertise the endpoint's multiple interface addresses, and 3) perform connectivity checks on the advertised interfaces.

9.1.1. MP RTP header extension for a subflow

The RTP header for each subflow is defined below:

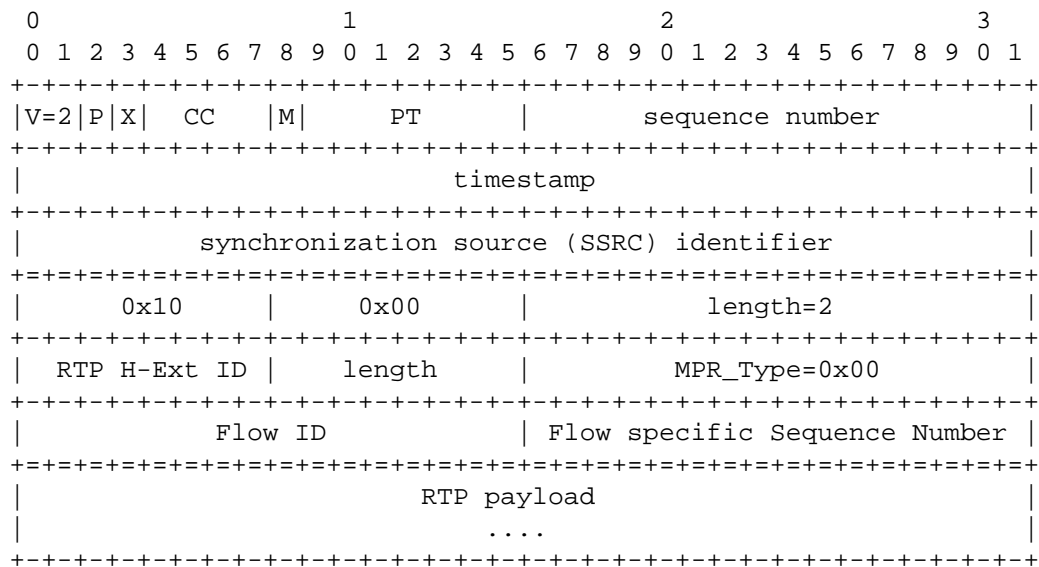


Figure 6: MPRTTP header for subflow

RTP H-Ext ID and length: 8-bits each

It conforms to the 2-byte RTP header extension defined in [6].

RTP H-Ext=TBD

The 8-bit length field is the length of extension data in bytes not including the RTP H-Ext ID and length fields.

MPR_Type: 16-bits

The MPR_Type field corresponds to the type of RTP packet. Namely:

MPR_Type Value	Use
0x00	Subflow RTP Header. For this case the Length is set to 7
0x01	Connectivity Check. For this case the length is set to 0
TBD	Keep Alive Packet.

Figure 7: RTP header extension values for MPRTTP (MPR_Type)

Flow ID: Identifier of the subflow. Every RTP packet belonging to the same subflow carries the same unique flow identifier.

Flow specific Sequence Number: Sequence of the packet in the subflow. Each subflow has its own strictly monotonically increasing sequence number space.

9.2. MPRTP RTCP Header Extension

The MPRTP RTCP header extension is used 1) to provide RTCP feedback per subflow to gauge the characteristics of each path, 2) to advertise the multiple interface addresses for a media receiver, and 3) perform connectivity check on the new interfaces.

9.2.1. MPRTP RTCP header extension for flow specific SR/RR

TBD

9.2.2. MPRTP RTCP header extension for Interface advertisement

This sub-section defines the RTCP header extension for in-band interface advertisement by the receiver, instead of relying on ICE or in situations when the interface appears after SDP session establishment.

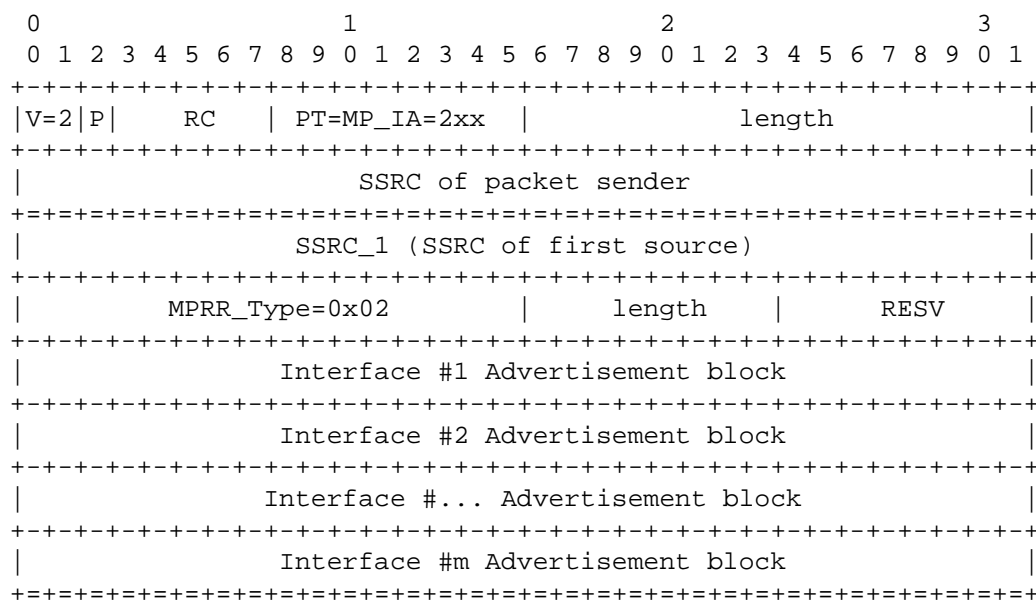


Figure 8: MPRTP Interface Advertisement. (RTCP SR/RR header extension)

MP_IA: 8 bits

Indicates that it is a RTCP extension for interface advertisement.

MPRR_Type: 16-bits

The MPRR_Type field corresponds to the type of MPRTCP RTCP packet. Namely:

MPRR_Type Value	Use
0x00	Interface Advertisement
0x01	Connectivity Check. For this case the length is set to 0
TBD	Keep Alive Packet.

Figure 9: RTP header extension values for MPRTCP (MPR_Type)

length: 8-bits

The 8-bit length field is the length of extension data in bytes not including the MPRR_Type and length fields. The value zero indicates there is no data following.

Interface Advertisement block: variable size

Defined later in the section.

9.2.3. Interface Address Advertisement block

This block describes a method to represent IPv4, IPv6 and generic DNS-type addresses in a block format. It is based on the sub-reporting block in [RFC 5760](#) [7].

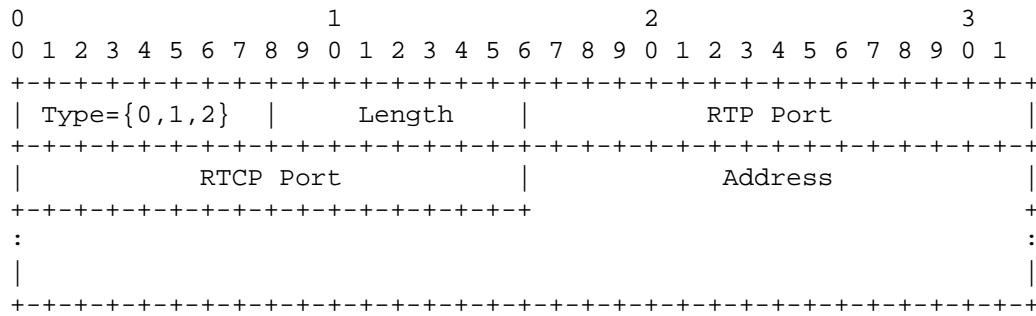


Figure 10: Interface Address Advertisement block during path discovery

Type: 8 bits

The Type corresponds to the type of address. Namely:

0: IPv4 address

1: IPv6 address

2: DNS name

Length: 8 bits

The length of the Interface Advertisement block in bytes.

For an IPv4 address, this should be 9 (i.e., 5 octets for the header and 4 octets for IPv4 address).

For an IPv6 address, this should be 21.

For a DNS name, the length field indicates the number of octets making up the string plus the 5 byte header.

RTP Port: 2 octets

The port number to which the sender sends RTP data. A port number of 0 is invalid and MUST NOT be used.

RTCP Port: 2 octets

The port number to which receivers send feedback reports. A port number of 0 is invalid and MUST NOT be used.

Address: 4 octets (IPv4), 16 octets (IPv6), or n octets (DNS name)

The address to which receivers send feedback reports. For IPv4 and IPv6, fixed-length address fields are used. A DNS name is an arbitrary-length string. The string MAY contain Internationalizing Domain Names in Applications (IDNA) domain names and MUST be UTF-8 encoded [8].

10. SDP Considerations

The packet formats specified in this document define extensions for RTP and RTCP. The use of MP RTP is left to the discretion of the sender and receiver.

A participant of a media session MAY use SDP to signal that it supports MP RTP. Not providing this information may/will make the sender or receiver ignore the header extensions. However, MP RTP MAY be used by either sender or receiver without prior signaling.

```
mp rtp-attr ib = "a=" "mp rtp" [ ":"  
    mp rtp-optional-parameter]  
    CRLF    ; flag to enable MP RTP
```

The literal 'mp rtp' MUST be used to indicate support for MP RTP. Generally, senders and receivers SHOULD indicate this capability if they support MP RTP and would like to use it in the specific media session being signaled. However, it is possible for an MP RTP sender to stream data using multiple paths to a non-MP RTP client.

Currently, there are no extensions defined for the literal 'mp rtp' but we provide the opportunity to extend it using the mp rtp-optional-parameter.

10.1. Increased Throughput

The MP RTP layer MAY choose to augment paths to increase throughput. If the desired media rate exceeds the current media rate, the peers MUST renegotiate the application specific ("b=AS:") [16] bandwidth.

10.2. MP RTP using preloaded ICE

TBD

11. Acknowledgements

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12. IANA Considerations

This document defines a new SDP attribute, "mprtp", within the existing IANA registry of SDP Parameters.

TBD.

13. Security Considerations

All drafts are required to have a security considerations section. See [RFC 3552](#) [17] for a guide.

14. References

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