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Multiplexing RTP Data and Control Packets on a Single Port
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Abstract

This memo discusses issues that arise when multiplexing RTP data packets and RTP control protocol (RTCP) packets on a single UDP port. It updates RFC 3550 to describe when such multiplexing is, and is not, appropriate, and explains how the Session Description Protocol (SDP) can be used to signal multiplexed sessions.

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1. Introduction

The Real-time Transport Protocol (RTP) [1] comprises two components: a data transfer protocol, and an associated control protocol (RTCP). Historically, RTP and RTCP have been run on separate UDP ports. With increased use of Network Address Translation (NAT) this has become problematic, since opening multiple NAT pinholes can be costly. This memo discusses how the RTP and RTCP flows for a single media type can be run on a single port, to ease NAT traversal, and considers when such multiplexing is appropriate. The multiplexing of several types of media (e.g. audio and video) onto a single port is not considered here (but see Section 5.2 of [1]).

This memo is structured as follows: in Section 2 we discuss the design choices which led to the use of separate ports, and comment on the applicability of those choices to current network environments. We discuss terminology in Section 3, how to distinguish multiplexed packets in Section 4, and then specify when and how RTP and RTCP should be multiplexed in Section 5. Quality of service and bandwidth issues are discussion in Section 6. We conclude with security considerations in Section 7.

This memo updates Section 11 of [1].

2. Background

An RTP session comprises data packets and periodic control (RTCP) packets. RTCP packets are assumed to use "the same distribution mechanism as the data packets" and the "underlying protocol MUST provide multiplexing of the data and control packets, for example using separate port numbers with UDP" [1]. Multiplexing was deferred to the underlying transport protocol, rather than being provided within RTP, for the following reasons:

1. Simplicity: an RTP implementation is simplified by moving the RTP and RTCP demultiplexing to the transport layer, since it need not concern itself with the separation of data and control packets. This allows the implementation to be structured in a very natural fashion, with a clean separation of data and control planes.

2. Efficiency: following the principle of integrated layer processing [13] an implementation will be more efficient when demultiplexing happens in a single place (e.g. according to UDP port) than when split across multiple layers of the stack (e.g. according to UDP port then according to packet type).

3. To enable third party monitors: while unicast voice-over-IP has always been considered, RTP was also designed to support loosely coupled multicast conferences [14] and very large-scale multicast streaming media applications (such as the so-called "triple-play" IPTV service). Accordingly, the design of RTP allows the RTCP packets to be multicast using a separate IP multicast group and UDP port to the data packets. This not only allows participants in a session to get reception quality feedback, but also enables deployment of third party monitors which listen to reception quality without access to the data packets. This was intended to provide manageability of multicast sessions, without compromising privacy.

While these design choices are appropriate for many use of RTP, they are problematic in some cases. There are many RTP deployments which don't use IP multicast, and with the increased use of Network Address Translation (NAT) the simplicity of multiplexing at the transport layer has become a liability, since it requires complex signalling to open multiple NAT pinholes. In environments such as these, it is desirable to provide an alternative to demultiplexing RTP and RTCP using separate UDP ports, instead using only a single UDP port and demultiplexing within the application.

This memo provides such an alternative by multiplexing RTP and RTCP packets on a single UDP port, distinguished by the RTP payload type and RTCP packet type values. This pushes some additional work onto the RTP implementation, in exchange for simplified NAT traversal.

3. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [2].

4. Distinguishable RTP and RTCP Packets

When RTP and RTCP packets are multiplexed onto a single port, they can be distinguished provided: 1) the RTP payload type (PT) values used are distinct from the RTCP packet types used; and 2) for each RTP payload type, PT+128 is distinct from the RTCP packet types used. The first constraint precludes a direct conflict between RTP payload type and RTCP packet type, the second constraint precludes a conflict between an RTP data packet with marker bit set and an RTCP packet. This demultiplexing method works because the RTP payload type and RTCP packet type occupy the same position within the packet.

The following conflicts between RTP and RTCP packet types are known:

- o RTP payload types 64-65 conflict with the RTCP FIR and NACK packets defined in the RTP Payload Format for H.261 [3].
- o RTP payload types 72-76 conflict with the RTCP SR, RR, SDES, BYE and APP packets defined in the RTP specification [1].
- o RTP payload types 77-78 conflict with the RTCP RTPFB and PSFB packets defined in the RTP/AVPF profile [4].
- o RTP payload type 79 conflicts with RTCP Extended Report (XR) [5] packets.
- o RTP payload type 80 conflicts with Receiver Summary Information (RSI) packets defined in the RTCP Extensions for Single-Source Multicast Sessions with Unicast Feedback [6].

New RTCP packet types may be registered in future, and will further reduce the RTP payload types that are available when multiplexing RTP and RTCP onto a single port. To allow this multiplexing, future RTCP packet type assignments SHOULD be made after the current assignments in the range 209-223, then in the range 194-199, so that only the RTP payload types in the range 64-95 are blocked.

Given these constraints, it is RECOMMENDED to follow the guidelines in the RTP/AVP profile [7] for the choice of RTP payload type values, with the additional restriction that payload type values in the range 64-95 MUST NOT be used. Specifically, dynamic RTP payload types SHOULD be chosen in the range 96-127 where possible. Values below 64 MAY be used if that is insufficient, in which case it is RECOMMENDED that payload type numbers that are not statically assigned by [7] be used first.

> Note: since all RTCP packets MUST be sent as compound packets beginning with an SR or an RR packet ([1] Section 6.1), one might wonder why RTP payload types other than 72 and 73 are prohibited when multiplexing RTP and RTCP. This is done to ensure robustness against broken nodes which send non-compliant RTCP packets, which might otherwise be confused with multiplexed RTP packets.

5. Multiplexing RTP and RTCP on a Single Port

The procedures for multiplexing RTP and RTCP on a single port depend on whether the session is a unicast session or a multicast session. For a multicast sessions, also depends whether ASM or SSM multicast is to be used.

5.1. Unicast Sessions

It is acceptable to multiplex RTP and RTCP packets on a single UDP port to ease NAT traversal for unicast sessions, provided the RTP payload types used in the session are chosen according to the rules in Section 4.

When the Session Description Protocol (SDP) [8] is used to negotiate RTP sessions according to the offer/answer model [9], the "a=rtcp:" attribute [10] is used to indicate the port chosen for RTCP traffic, if the default of using an odd/even port pair is not desirable. If RTP and RTCP are to be multiplexed on a single port, this attribute MUST be included in the initial SDP offer, and MUST indicate the the same port as included in the "m=" line. For example:

```
v=0
o=csp 1153134164 1153134164 IN IP6 2001:DB8::211:24ff:fea3:7a2e
s=-
c=IN IP6 2001:DB8::211:24ff:fea3:7a2e
t=1153134164 1153137764
m=audio 49170 RTP/AVP 97
a=rtpmap:97 iLBC/8000
a=rtcp:49170
```

This offer denotes a unicast voice-over-IP session using the RTP/AVP profile with iLBC coding. The answerer is requested to send both RTP and RTCP to port 49170 on IPv6 address 2001:DB8::211:24ff:fea3:7a2e.

If the answerer supports multiplexing of RTP and RTCP onto a single port it MUST include an "a=rtcp:" attribute in the answer. The port specified in that attribute MUST be the same as that used for RTP in the "m=" line of the answer. The RTP payload types used MUST conform to the rules in Section 4, and the answer MUST be rejected if there is a conflict between the chosen RTP payload types and the expected RTCP packet types.

If the answer does not contain an "a=rtcp:" attribute, the offerer MUST NOT multiplex RTP and RTCP packets on a single port. Instead, it must send and receive RTCP on a port allocated according to the usual port pair rules. This will occur when talking to a peer that does not understand the "a=rtcp:" attribute or this specification.

If the answer contains an "a=rtcp:" attribute, but that attribute specifies a different port for RTCP than for RTP, then the answer MUST be rejected, and the session re-negotiated using separate RTP and RTCP ports.

Answerers which support the "a=rtcp:" attribute but do not understand

this memo should reject sessions where the RTP and RTCP ports are the same (Section 11 of [1] prohibits such sessions unless the mechanisms described in this memo are used). It is likely that this is a poorly tested feature of older implementations, however, and implementations should be robust to unexpected behaviour. If the offerer suspects a session was rejected due to the presence of multiplexed RTP and RTCP, it MAY retry the offer using separate ports for RTP and RTCP.

When using SIP with a forking proxy, it is possible that multiple 200 (OK) answers will be received, some supporting multiplexed RTP and RTCP, some not. This is not an issue if a separate RTP session is established with each answerer, since multiplexing occurs on a per session basis, but does prevent a single RTP session being opened between the offerer and all answerers.

TODO: expand this discussion. Does SIP define if this should be a single RTP session or multiple sessions?

TODO: discuss interactions between multiplexed RTP and RTCP, and Interactive Connectivity Establishment (ICE) [15].

5.2. Any Source Multicast Sessions

The problem of NAT traversal is less severe for any source multicast (ASM) RTP sessions than for unicast RTP sessions, and the benefit of using separate ports for RTP and RTCP is greater, due to the ability to support third party RTCP only monitors. Accordingly, RTP and RTCP packets SHOULD NOT be multiplexed onto a single port when using ASM multicast RTP sessions, and SHOULD instead use separate ports and multicast groups.

5.3. Source Specific Multicast Sessions

RTP sessions running over Source Specific Multicast (SSM) send RTCP packets from the source to receivers via the multicast channel, but use a separate unicast feedback mechanism [6] to send RTCP from the receivers back to the source, with the source either reflecting the RTCP packets to the group, or sending aggregate summary reports.

Following the terminology of [6], we identify three RTP/RTCP flows in an SSM session:

1. RTP and RTCP flows between media sender and distribution source. In many scenarios, the media sender and distribution source are collocated, so multiplexing is not a concern. If the media sender and distribution source are connected by a unicast connection, the rules in Section 5.1 of this memo apply to that connection. If the media sender and distribution source are

connected by an Any Source Multicast connection, the rules in Section 5.2 apply to that connection. If the media sender and distribution source are connected by a Source Specific Multicast connection, the RTP and RTCP packets MAY be multiplexed on a single port, provided this is signalled (for example, using "a=rtcp:" with the same port number as specified for RTP on the "m=" line, if using SDP).

2. RTP and RTCP sent from the distribution source to the receivers. The distribution source MAY multiplex RTP and RTCP onto a single port to ease NAT traversal issues on the forward SSM path, since this does not hinder third party monitoring. When using SDP, the multiplexing MUST be signalled using the "a=rtcp:" attribute [10] with the same port number as specified for RTP on the "m=" line.

3. RTCP sent from receivers to distribution source. This is an RTCP only path, so multiplexing is not a concern.

Multiplexing RTP and RTCP onto a single port is more acceptable for an SSM session than for an ASM session, since SSM sessions cannot readily make use of third party reception quality monitoring devices that listen to the multicast RTCP traffic but not the data traffic (since the RTCP traffic is unicast to the distribution source, rather than multicast, and since one cannot subscribe to only the RTCP packets on the SSM channel, even if sent on a separate port).

6. Multiplexing, Bandwidth, and Quality of Service

Multiplexing RTP and RTCP has implications on the use of Quality of Service (QoS) mechanism that handles flow that are determined by a three or five tuple (protocol, port and address for source and/or destination). In these cases the RTCP flow will be merged with the RTP flow when multiplexing them together. Thus the RTCP bandwidth requirement needs to be considered when doing QoS reservations for the combinded RTP and RTCP flow. However from an RTCP perspective it is beneficial to receive the same treatment of RTCP packets as for RTP as it provides more accurate statistics for the measurements performed by RTCP.

The bandwidth required for a multiplexed stream comprises the session bandwidth of the RTP stream, plus the bandwidth used by RTCP. In the usual case, the RTP session bandwidth is signalled in the SDP "b=AS:" line, and the RTCP traffic is limited to 5% of this value. Any QoS reservation SHOULD therefore be made for 105% of the "b=AS:" value. If a non-standard RTCP bandwidth fraction is used, signalled by the SDP "b=RR:" and/or "b=RS:" lines [11], then any QoS reservation SHOULD be made for bandwidth equal to (AS + RS + RR), taking the RS

and RR values from the SDP answer.

7. Security Considerations

The security considerations in the RTP specification [1] and any applicable RTP profile (e.g. [7]) and payload format(s) apply.

If the Secure Real-time Transport Protocol (SRTP) [12] is to be used in conjunction with multiplexed RTP and RTCP, then multiplexing MUST be done below the SRTP layer. The sender generates SRTP and SRTCP packets in the usual manner, based on their separate cryptographic contexts, and multiplexes them onto a single port immediately before transmission. At the receiver, the cryptographic context is derived from the SSRC, destination network address and destination transport port number in the usual manner, augmented using the RTP payload type and RTCP packet type to demultiplex SRTP and SRTCP according to the rules in Section 4 of this memo. After the SRTP and SRTCP packets have been demultiplexed, cryptographic processing happens in the usual manner.

8. IANA Considerations

No IANA actions are required.

9. Acknowledgements

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