        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 Port Translation (NAPT) [14] this
   has become problematic, since maintaining multiple NAT bindings can
   be costly. It also complicates firewall administration, since
   multiple ports must be opened to allow RTP traffic. 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 simplify firewall
   administration, 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, and how to signal multiplexed sessions, 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 [15] 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 [16] 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 uses 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, the
RTCP packet type field occupies the same position in the packet as
the combination of the RTP marker (M) bit and the RTP payload type
(PT). This field can be used to distinguish RTP and RTCP packets
when two restrictions are observed: 1) the RTP payload type values
used are distinct from the RTCP packet types used; and 2) for each
RTP payload type (PT), 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 the marker bit set and an
RTCP packet.

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

o RTP payload types 64-65 conflict with the (obsolete) RTCP FIR and
   NACK packets defined in the original 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. RTCP packet types in
the ranges 1-191 and 224-254 SHOULD only be used when other values
have been exhausted.

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 multicast sessions, the procedures also depend on whether Any
   Source Multicast (ASM) or Source Specific Multicast (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, and provided that multiplexing is signalled in advance.
   The following sections describe how such multiplexed sessions can be
   signalled using the Session Initiation Protocol (SIP) with the offer/
   answer model.

5.1.1. SDP Signalling

   When the Session Description Protocol (SDP) [8] is used to negotiate
   RTP sessions following the offer/answer model [9], the "a=rtcp-mux"
   attribute (see Section 8) indicates the desire to multiplex RTP and
   RTCP onto a single port. The initial SDP offer MUST include this
   attribute to request multiplexing of RTP and RTCP on a single port.
   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-mux

   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 wishes to multiplex RTP and RTCP onto a single port
   it MUST include an "a=rtcp-mux" attribute in the answer. The RTP
   payload types used in the answer MUST conform to the rules in
   Section 4.

   If the answer does not contain an "a=rtcp-mux" attribute, the offerer
   MUST NOT multiplex RTP and RTCP packets on a single port. Instead,
   it should send and receive RTCP on a port allocated according to the

usual port selection rules (either the port pair, or a signalled port
if the "a=rtcp:" attribute [10] is also included). This will occur
when talking to a peer that does not understand the "a=rtcp-mux"
attribute.

When SDP is used in a declarative manner, the presence of an "a=rtcp-
mux" attribute signals that the sender will multiplex RTP and RTCP on
the same port. The receiver MUST be prepared to receive RTCP packets
on the RTP port, and any resource reservation needs to be made
including the RTCP bandwidth.

5.1.2. Interactions with SIP forking

When using SIP with a forking proxy, it is possible that an INVITE
request may result in multiple 200 (OK) responses. If RTP and RTCP
multiplexing is offered in that INVITE, it is important to be aware
that some answerers may support multiplexed RTP and RTCP, some not.
This will require the offerer to listen for RTCP on both the RTP port
and the usual RTCP port, and to send RTCP on both ports, unless
branches of the call that support multiplexing are re-negotiated to
use separate RTP and RTCP ports.

5.1.3. Interactions with ICE

It is common to use the Interactive Connectivity Establishment (ICE)
[17] methodology to establish RTP sessions in the presence of Network
Address Translation (NAT) devices or other middleboxes. If RTP and
RTCP are sent on separate ports, the RTP media stream comprises two
components in ICE (one for RTP and one for RTCP), with connectivity
checks being performed for each component. If RTP and RTCP are to be
multiplexed on the same port some of these connectivity checks can be
avoided, reducing the overhead of ICE.

If it is desired to use both ICE and multiplexed RTP and RTCP, the
initial offer MUST contain an "a=rtcp-mux" attribute to indicate that
RTP and RTCP multiplexing is desired, and MUST contain "a=candidate:"
lines for both RTP and RTCP along with an "a=rtcp:" line indicating a
fallback port for RTCP in the case that the answerer does not support
RTP and RTCP multiplexing. This MUST be done for each media where
RTP and RTCP multiplexing is desired.

If the answerer wishes to multiplex RTP and RTCP on a single port, it
MUST generate an answer containing an "a=rtcp-mux" attribute, and a
single "a=candidate:" line corresponding to the RTP port (i.e. there
is no candidate for RTCP), for each media where it is desired to use
RTP and RTCP multiplexing. The answerer then performs connectivity
checks for that media as if the offer had contained only a single
candidate for RTP. If the answerer does not want to multiplex RTP

and RTCP on a single port, it MUST NOT include the "a=rtcp-mux" attribute in its answer, and MUST perform connectivity checks for all offered candidates in the usual manner.

On receipt of the answer, the offerer looks for the presence of the "a=rtcp-mux" line for each media where multiplexing was offered. If this is present, then connectivity checks proceed as if only a single candidate (for RTP) were offered, and multiplexing is used once the session is established. If the "a=rtcp-mux" line is not present, the session proceeds with connectivity checks using both RTP and RTCP candidates, eventually leading to a session being established with RTP and RTCP on separate ports (as signalled by the "a=rtcp:" attribute).

5.1.4. Interactions with Header Compression

Multiplexing RTP and RTCP packets onto a single port may negatively impact header compression schemes, for example Compressed RTP (CRTP) [18] and RObust Header Compression (ROHC) [19]. Header compression exploits patterns of change in the RTP headers of consecutive packets to send an indication that the packet changed in the expected way, rather than sending the complete header each time. This can lead to significant bandwidth savings if flows have uniform behaviour.

The presence of RTCP packets multiplexed with RTP data packets can disrupt the patterns of change between headers, and has the potential to significantly reduce header compression efficiency. The extent of this disruption depends on the header compression algorithm used, and on the way flows are classified. A well designed classifier should be able to separate RTP and RTCP multiplexed on the same port into different compression contexts, using the payload type field, such that the effect on the compression ratio is small. A classifier that assigns compression contexts based only on the IP addresses and UDP ports will not perform well. It is expected that implementations of header compression will need to be updated to efficiently support RTP and RTCP multiplexed on the same port.

This effect of multiplexing RTP and RTCP on header compression may be especially significant in those environments, such as some wireless telephony systems, which rely on the efficiency of header compression to match the media to a limited capacity channel. The implications of multiplexing RTP and RTCP should be carefully considered before use in such environments.

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
    co-located, 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 (using "a=rtcp-mux" 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,
    although doing so may hinder third party monitoring devices if
    the session uses the simple feedback model. When using SDP, the
    multiplexing SHOULD be signalled using the "a=rtcp-mux"
    attribute.

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

Multiplexing RTP and RTCP packets on a single port in an SSM session
has the potential for interactions with header compression described
in Section 5.1.4.

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 combined 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:"
   (or "b=TIAS:" [11]) 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 [12], 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 usage of multiplexing RTP and RTCP is not believed to introduce
   any new security considerations. Known major issues are, integrity
   and authentication of the signalling used to setup the multiplexing,
   the integrity, authentication and confidentiality of the actual RTP
   and RTCP traffic. 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) [13] 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

   Following the guidelines in [8], the IANA is requested to register
   one new SDP attribute:

   o Contact name/email: authors of RFC XXXX

   o Attribute name: rtcp-mux

   o Long-form attribute name: RTP and RTCP multiplexed on one port

   o Type of attribute: media level

   o Subject to charset: no

   This attribute is used to signal that RTP and RTCP traffic should be
   multiplexed on a single port (see Section 5 of this memo). It is a
   property attribute, which does not take a value.

   The rules for assignment of RTP RTCP Control Packet Types in the RTP
   Parameters registry are updated as follows. When assigning RTP RTCP
   Control Packet types, IANA is requested to assign unused values from
   the range 200-223 where possible. If that range is fully occupied,
   values from the range 194-199 may be used, and then values from the
   ranges 1-191 and 224-254. This improves header validity checking of
   RTCP packets compared to RTP packets or other unrelated packets. The
   values 0 and 255 are avoided for improved validity checking relative
   to random packets since all-zeros and all-ones are common values.

   Note to RFC Editor: please replace "RFC XXXX" above with the RFC
   number of this memo, and remove this note.

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