**RTP**

RTP opens two ports for communication. One for the media stream (an even port number) and one for control (QoS feedback and media control) – RTCP. The port numbers are not hard defined, it depends very much upon the application.

* RTP (Real-time Transport Protocol)
* RTCP (Real-time Control Protocol)
  + Adds information for:
  + Packet Loss
  + Jitter
  + Delay
  + Signal Level
  + Call Quality Metrics
  + Echo Return Loss
  + etc.
* RTCP XR (Real-time Control Protocol Extended Reports)
  + All of the RTCP list above plus:
  + [R Factor](https://www.voip-info.org/call-quality-metrics)
  + [MOS](https://www.voip-info.org/call-quality-metrics)
  + and more

Actual voice packets are sent using RTP/RTCP for [SIP](https://www.voip-info.org/sip) VOIP calls. RTP is able to carry media identified by parameters registred by the Internet assigned numbers authority, [IANA](https://www.voip-info.org/iana). These are also used for [SDP](https://www.voip-info.org/sdp) descriptions in [SIP](https://www.voip-info.org/sip) and [MGCP](https://www.voip-info.org/mgcp) messages.

Some of these payloads:

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
| PT | encoding name | audio/video (A/V) | clock rate (Hz) | channels | Ref |
| 0 | G711 PCMU | A | 8000 | 1 | RFC3551 |
| 3 | GSM | A | 8000 | 1 | RFC3551 |
| 4 | G723 | A | 8000 | 1 | Kumar |
| 5 | DVI4 | A | 8000 | 1 | RFC3551 |
| 6 | DVI4 | A | 16000 | 1 | RFC3551 |
| 7 | LPC | A | 8000 | 1 | RFC3551 |
| 8 | PCMA | A | 8000 | 1 | RFC3551 |
| 9 | G722 | A | 8000 | 1 | RFC3551 |
| 10 | L16 | A | 44100 | 2 | RFC3551 |
| 11 | L16 | A | 44100 | 1 | RFC3551 |
| 12 | QCELP | A | 8000 | 1 | – |
| 13 | CN | A | 8000 | 1 | RFC3389 |
| 14 | MPA | A | 90000 |  | RFC3551,RFC2250 |
| 15 | G728 | A | 8000 | 1 | RFC3551 |
| 16 | DVI4 | A | 11025 | 1 | DiPol |
| 17 | DVI4 | A | 22050 | 1 | DiPol |
| 18 | G729 | A | 8000 | 1 |  |
| 19 | reserved | A |  |  |  |
| 20 | unassigned | A |  |  |  |
| 21 | unassigned | A |  |  |  |
| 22 | unassigned | A |  |  |  |
| 23 | unassigned | A |  |  |  |
| 24 | unassigned | V |  |  |  |
| 25 | CelB | V | 90000 |  | RFC2029 |
| 26 | JPEG | V | 90000 |  | RFC2435 |
| 27 | unassigned | V |  |  |  |
| 28 | nv | V | 90000 |  | RFC3551 |
| 29 | unassigned | V |  |  |  |
| 30 | unassigned | V |  |  |  |
| 31 | H261 | V | 90000 |  | RFC2032 |
| 32 | MPV | V | 90000 |  | RFC2250 |
| 33 | MP2T | AV | 90000 |  | RFC2250 |
| 34 | H263 | V | 90000 |  | Zhu |
| 35–71 | unassigned | ? |  |  |  |
| 72–76 | reserved for RTCP conflict avoidance |  |  |  | RFC3550 |
| 77–95 | unassigned | ? |  |  |  |
| 96–127 | dynamic | ? |  |  | RFC3551 |

* [IANA](https://www.voip-info.org/iana): Registred RTP Parameters : [http://www.iana.org/assignments/rtp-parameters](https://www.iana.org/assignments/rtp-parameters)

**RTP and NAT**

In a VOIP session, there are two RTP streams, one in each direction. If one of the parties involved in the session is on a private IP address, that stream from the public client to the NAT box, will not be allowed to reach the client on the inside of the NAT. To handle this, [Symmetric RTP](https://www.voip-info.org/rtp-symmetric) is often used. For more information on NAT and VOIP, see [NAT and VOIP](https://www.voip-info.org/nat-and-voip).

**RFCs**

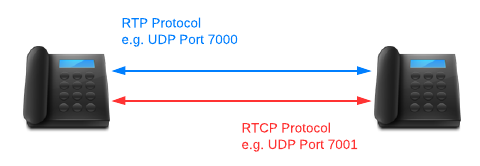
* IETF [RFC 3550](https://www.ietf.org/rfc/rfc3550.txt?number=3550) RTP: A Transport Protocol for Real-Time Applications
* IETF [RFC 3611](https://www.ietf.org/rfc/rfc3611.txt?number=3611) RTP Control Protocol Extended Reports (RTCP XR)
* IETF [RFC 1890](https://www.ietf.org/rfc/rfc1890.txt?number=1890) RTP Profile for Audio and Video Conferences with Minimal Control
* IETF [RFC 2508](https://www.ietf.org/rfc/rfc2508.txt?number=2508) Compressing IP/UDP/RTP Headers for Low-Speed Serial Links
* IETF [RFC 3545](https://www.ietf.org/rfc/rfc3545.txt?number=3545) Enhanced Compressed RTP (CRTP) for Links with High Delay, Packet Loss and Reordering

**See also**

* [Open Source VOIP Software](https://www.voip-info.org/open-source-voip-software) RTP protocol stacks and related software
* [RTP Packet Format with field descriptions](http://www.networksorcery.com/enp/protocol/rtp.htm)
* <http://www.cs.columbia.edu/~hgs/rtp/>: All you want to or need to know about RTP
* [Asterisk RTCP](https://www.voip-info.org/asterisk-rtcp)
* [SRTP](https://www.voip-info.org/srtp): Secure RTP
* [IETF](https://www.voip-info.org/ietf)
* [SIP](https://www.voip-info.org/sip): Session initiation protocol
* [MGCP](https://www.voip-info.org/mgcp)
* [RTP Silence Suppression](https://www.voip-info.org/rtp-silence-suppression) Also known as [VAD](https://www.voip-info.org/vad)

What is RTP – Real-time Transport Protocol?

RTP – short for Real-time Transport Protocol defines a standard packet format for delivering audio and video over the Internet. It is defined in [RFC 1889](http://tools.ietf.org/html/rfc1889/). It was developed by the Audio Video Transport Working group and was first published in 1996. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications, television services and web-based push-to-talk features.



RTP is used in conjunction with the [RTP Control Protocol (RTCP)](https://www.3cx.com/pbx/rtcp/). While RTP carries the media streams (e.g., audio and video), RTCP is used to monitor transmission statistics and quality of service (QoS) and aids synchronization of multiple streams. RTP is originated and received on even port numbers and the associated RTCP communication uses the next higher odd port number. RTP is one of the foundations of VoIP and it is used in conjunction with SIP which assists in setting up the connections across the network.

**RTP Advantages and Usage**

As its name implies, the design goal for RTP is the end-to-end streaming in real-time of media-related data. RTP includes mechanisms for jitter compensation, packet loss detection, as well as out-of-order data packet delivery, issues that are especially common in UDP ([User Datagram Protocol](https://en.wikipedia.org/wiki/User_Datagram_Protocol)) transmissions over IP. As RTP enables data transfer to multiple destination end-points in parallel via [IP multicast](https://en.wikipedia.org/wiki/Multicast), it is the primary standard eployed for audio and video IP network transfers. The mechanisms for the associated profile and payload format, referenced in the design of the [RTP architecture](https://tools.ietf.org/html/rfc3550), are implemented on the level of the application layer, instead of the operating system layer.

Applications such as [VoIP](https://www.3cx.com/voip/) that need to employ real-time streaming of multimedia data, typically require the timely delivery of data, with varying tolerance in packet loss. As an example, audio packet loss in a VoIP application can cause losing some milliseconds of audio data. This loss can be appropriately handled by error compensation algorithms to make it insignificant and imperceptible to the caller(s). TCP ([Transmission Control Protocol](https://en.wikipedia.org/wiki/Transmission_Control_Protocol)) is also standardized for RTP use, even though it is not typically employed in applications due to its error-control mechanisms that can cause delays and affect timely packet delivery. For this reason, most RTP applications most commonly base their implementations on UDP.

**Further Reading**

[What is RTCP?](https://www.3cx.com/pbx/rtcp/)

Real-time transport protocol (RTP) is a way of structuring data packets so that they can be delivered across the internet at lightning speeds and reassembled into a smooth flowing stream suitable for delivering voice or multimedia in a natural way. Without such a protocol, voice over IP would be impossible.

**The evolution of VoIP telephony**

When wired terrestrial phone systems first came into use in the 1880s, each call was carried in the form of a continuous electrical signal travelling along a single wire or series of wires. If you wanted to call Birmingham instead of London, an operator would physically connect you to a different wiring route by rearranging a jack plug. That method continued almost unchanged until the 1960s.

Telephone calls that could find their own way across a congested telephone network came with the introduction of digitally switched exchanges based upon the newly available transistors.

When data is streamed across today’s internet, it is divided into a series of packets and wrapped inside instructions that help it to perform the requisite switching. Each packet can then squeeze through internet traffic bottlenecks as bandwidth permits. Although it is really fast, internet traffic isn’t unlike urban road traffic – with queues, red lights, crossings, one-way streets and toll bridges to navigate. As a result, it is unlikely that packets will arrive in the same order in which they were sent, so each packet is marked with its proper place in the stream so that it can be reassembled on arrival.

**Real-time Transfer Protocol**

There are a variety of ways to wrap data inside additional layers of information to control how it is routed, but delivering a real-time experience has some very specific requirements. For example, if you were sending an executable file across the internet, the speed with which it arrives and is reassembled is unimportant but you cannot lose any data, so there is constant error checking and often requests for packets to be resent. (File Transfer Protocol was developed to do that). In contrast, when sending a conversation, it’s better to drop a missing packet than to delay the delivery of the reassembled audio. If both audio and video are being streamed, there must be additional safeguards to keep them tightly synchronised.

In practice, communication protocols such as RTP are usually wrapped inside numerous other protocols – each controlling different aspects of the addressing, switching and data protection (including encryption) required en route. Common examples include IP (internet protocol), UDP (user datagram protocol), and RTCP (real time control protocol). All of these are commonly used when voice termination providers, such as IDT, connect your telephone call, video call or fax message across the internet.

In combination, these protocols even monitor the traffic conditions on the network they are crossing, adapting to signal delays and packet errors to prevent any “jitter” or “echo” degrading the quality of your VoIP conversation or video conference. We’ve come a long way since the 1880s!

Real-time Transport Protocol

From Wikipedia, the free encyclopedia

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|  |
| --- |
| [**Internet protocol suite**](https://en.wikipedia.org/wiki/Internet_protocol_suite) |
| [**Application layer**](https://en.wikipedia.org/wiki/Application_layer) |
| * [BGP](https://en.wikipedia.org/wiki/Border_Gateway_Protocol) * [DHCP](https://en.wikipedia.org/wiki/Dynamic_Host_Configuration_Protocol) * [DNS](https://en.wikipedia.org/wiki/Domain_Name_System) * [FTP](https://en.wikipedia.org/wiki/File_Transfer_Protocol) * [HTTP](https://en.wikipedia.org/wiki/Hypertext_Transfer_Protocol) * [HTTPS](https://en.wikipedia.org/wiki/HTTPS) * [IMAP](https://en.wikipedia.org/wiki/Internet_Message_Access_Protocol) * [LDAP](https://en.wikipedia.org/wiki/Lightweight_Directory_Access_Protocol) * [MGCP](https://en.wikipedia.org/wiki/Media_Gateway_Control_Protocol) * [MQTT](https://en.wikipedia.org/wiki/MQTT) * [NNTP](https://en.wikipedia.org/wiki/Network_News_Transfer_Protocol) * [NTP](https://en.wikipedia.org/wiki/Network_Time_Protocol) * [POP](https://en.wikipedia.org/wiki/Post_Office_Protocol) * [PTP](https://en.wikipedia.org/wiki/Precision_Time_Protocol) * [ONC/RPC](https://en.wikipedia.org/wiki/Open_Network_Computing_Remote_Procedure_Call) * RTP * [RTSP](https://en.wikipedia.org/wiki/Real_Time_Streaming_Protocol) * [RIP](https://en.wikipedia.org/wiki/Routing_Information_Protocol) * [SIP](https://en.wikipedia.org/wiki/Session_Initiation_Protocol) * [SMTP](https://en.wikipedia.org/wiki/Simple_Mail_Transfer_Protocol) * [SNMP](https://en.wikipedia.org/wiki/Simple_Network_Management_Protocol) * [SSH](https://en.wikipedia.org/wiki/Secure_Shell) * [Telnet](https://en.wikipedia.org/wiki/Telnet) * [TLS/SSL](https://en.wikipedia.org/wiki/Transport_Layer_Security) * [XMPP](https://en.wikipedia.org/wiki/XMPP) * [*more...*](https://en.wikipedia.org/wiki/Category:Application_layer_protocols) |
| [**Transport layer**](https://en.wikipedia.org/wiki/Transport_layer) |
| * [TCP](https://en.wikipedia.org/wiki/Transmission_Control_Protocol) * [UDP](https://en.wikipedia.org/wiki/User_Datagram_Protocol) * [DCCP](https://en.wikipedia.org/wiki/Datagram_Congestion_Control_Protocol) * [SCTP](https://en.wikipedia.org/wiki/Stream_Control_Transmission_Protocol) * [RSVP](https://en.wikipedia.org/wiki/Resource_Reservation_Protocol) * [*more...*](https://en.wikipedia.org/wiki/Category:Transport_layer_protocols) |
| [**Internet layer**](https://en.wikipedia.org/wiki/Internet_layer) |
| * [IP](https://en.wikipedia.org/wiki/Internet_Protocol)   + [IPv4](https://en.wikipedia.org/wiki/IPv4)   + [IPv6](https://en.wikipedia.org/wiki/IPv6) * [ICMP](https://en.wikipedia.org/wiki/Internet_Control_Message_Protocol) * [ICMPv6](https://en.wikipedia.org/wiki/Internet_Control_Message_Protocol_for_IPv6) * [ECN](https://en.wikipedia.org/wiki/Explicit_Congestion_Notification) * [IGMP](https://en.wikipedia.org/wiki/Internet_Group_Management_Protocol) * [IPsec](https://en.wikipedia.org/wiki/IPsec) * [*more...*](https://en.wikipedia.org/wiki/Category:Internet_layer_protocols) |
| [**Link layer**](https://en.wikipedia.org/wiki/Link_layer) |
| * [ARP](https://en.wikipedia.org/wiki/Address_Resolution_Protocol) * [NDP](https://en.wikipedia.org/wiki/Neighbor_Discovery_Protocol) * [OSPF](https://en.wikipedia.org/wiki/Open_Shortest_Path_First) * [Tunnels](https://en.wikipedia.org/wiki/Tunneling_protocol)   + [L2TP](https://en.wikipedia.org/wiki/Layer_2_Tunneling_Protocol) * [PPP](https://en.wikipedia.org/wiki/Point-to-Point_Protocol) * [MAC](https://en.wikipedia.org/wiki/Medium_access_control)   + [Ethernet](https://en.wikipedia.org/wiki/Ethernet)   + [Wi-Fi](https://en.wikipedia.org/wiki/Wi-Fi)   + [DSL](https://en.wikipedia.org/wiki/Digital_subscriber_line)   + [ISDN](https://en.wikipedia.org/wiki/Integrated_Services_Digital_Network)   + [FDDI](https://en.wikipedia.org/wiki/Fiber_Distributed_Data_Interface) * [*more...*](https://en.wikipedia.org/wiki/Category:Link_protocols) |
| * [v](https://en.wikipedia.org/wiki/Template:IPstack) * [t](https://en.wikipedia.org/wiki/Template_talk:IPstack) * [e](https://en.wikipedia.org/w/index.php?title=Template:IPstack&action=edit) |

The **Real-time Transport Protocol** (**RTP**) is a [network protocol](https://en.wikipedia.org/wiki/Network_protocol) for delivering audio and video over [IP networks](https://en.wikipedia.org/wiki/IP_networks). RTP is used in communication and entertainment systems that involve [streaming media](https://en.wikipedia.org/wiki/Streaming_media), such as [telephony](https://en.wikipedia.org/wiki/Telephony), [video teleconference](https://en.wikipedia.org/wiki/Video_teleconference) applications including [WebRTC](https://en.wikipedia.org/wiki/WebRTC), [television services](https://en.wikipedia.org/wiki/IPTV) and web-based [push-to-talk](https://en.wikipedia.org/wiki/Push-to-talk) features.

RTP typically runs over [User Datagram Protocol](https://en.wikipedia.org/wiki/User_Datagram_Protocol) (UDP). RTP is used in conjunction with the [RTP Control Protocol](https://en.wikipedia.org/wiki/RTP_Control_Protocol) (RTCP). While RTP carries the media streams (e.g., audio and video), RTCP is used to monitor transmission statistics and [quality of service](https://en.wikipedia.org/wiki/Quality_of_service) (QoS) and aids [synchronization](https://en.wikipedia.org/wiki/Synchronization) of multiple streams. RTP is one of the technical foundations of [Voice over IP](https://en.wikipedia.org/wiki/Voice_over_IP) and in this context is often used in conjunction with a [signaling protocol](https://en.wikipedia.org/wiki/Signaling_protocol) such as the [Session Initiation Protocol](https://en.wikipedia.org/wiki/Session_Initiation_Protocol) (SIP) which establishes connections across the network.

RTP was developed by the Audio-Video Transport Working Group of the [Internet Engineering Task Force](https://en.wikipedia.org/wiki/Internet_Engineering_Task_Force) (IETF) and first published in 1996 as [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [1889](https://tools.ietf.org/html/rfc1889) which was then superseded by [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3550](https://tools.ietf.org/html/rfc3550) in 2003.



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* [2Profiles and payload formats](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#Profiles_and_payload_formats)
* [3Packet header](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#Packet_header)
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* [9External links](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#External_links)

Overview[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=1)]

RTP is designed for [end-to-end](https://en.wikipedia.org/wiki/End-to-end_principle), [real-time](https://en.wikipedia.org/wiki/Real-time_computing) transfer of [streaming media](https://en.wikipedia.org/wiki/Streaming_media). The protocol provides facilities for [jitter](https://en.wikipedia.org/wiki/Jitter) compensation and detection of [packet loss](https://en.wikipedia.org/wiki/Packet_loss) and [out-of-order delivery](https://en.wikipedia.org/wiki/Out-of-order_delivery), which are common especially during UDP transmissions on an IP network. RTP allows data transfer to multiple destinations through [IP multicast](https://en.wikipedia.org/wiki/IP_multicast).[[1]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Hardy_298-1) RTP is regarded as the primary standard for audio/video transport in IP networks and is used with an associated profile and payload format.[[2]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Perkins_55-2) The design of RTP is based on the architectural principle known as [application-layer framing](https://en.wikipedia.org/wiki/Application-layer_framing) where protocol functions are implemented in the application as opposed to in the operating system's [protocol stack](https://en.wikipedia.org/wiki/Protocol_stack).

Real-time [multimedia](https://en.wikipedia.org/wiki/Multimedia) streaming applications require timely delivery of information and often can tolerate some packet loss to achieve this goal. For example, loss of a packet in audio application may result in loss of a fraction of a second of audio data, which can be made unnoticeable with suitable [error concealment](https://en.wikipedia.org/wiki/Error_concealment) algorithms.[[3]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Perkins_46-3) The [Transmission Control Protocol](https://en.wikipedia.org/wiki/Transmission_Control_Protocol) (TCP), although standardized for RTP use,[[4]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-4) is not normally used in RTP applications because TCP favors reliability over timeliness. Instead the majority of the RTP implementations are built on the [User Datagram Protocol](https://en.wikipedia.org/wiki/User_Datagram_Protocol) (UDP).[[3]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Perkins_46-3) Other transport protocols specifically designed for multimedia sessions are [SCTP](https://en.wikipedia.org/wiki/SCTP)[[5]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-5) and [DCCP](https://en.wikipedia.org/wiki/DCCP),[[6]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-6) although, as of 2012, they are not in widespread use.[[7]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-7)

RTP was developed by the Audio/Video Transport working group of the IETF standards organization. RTP is used in conjunction with other protocols such as [H.323](https://en.wikipedia.org/wiki/H.323) and [RTSP](https://en.wikipedia.org/wiki/RTSP).[[2]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Perkins_55-2) The RTP specification describes two protocols: RTP and RTCP. RTP is used for the transfer of multimedia data, and the RTCP is used to periodically send control information and QoS parameters.[[8]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Peterson_430-8)

The data transfer protocol, RTP, carries real-time data. Information provided by this protocol includes timestamps (for synchronization), sequence numbers (for packet loss and reordering detection) and the payload format which indicates the encoded format of the data.[[9]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Colins_56-9) The control protocol, RTCP, is used for quality of service (QoS) feedback and synchronization between the media streams. The bandwidth of RTCP traffic compared to RTP is small, typically around 5%.[[9]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Colins_56-9)[[10]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-10)

RTP sessions are typically initiated between communicating peers using a signaling protocol, such as H.323, the [Session Initiation Protocol](https://en.wikipedia.org/wiki/Session_Initiation_Protocol) (SIP), RTSP, or [Jingle](https://en.wikipedia.org/wiki/Jingle_(protocol)) ([XMPP](https://en.wikipedia.org/wiki/XMPP)). These protocols may use the [Session Description Protocol](https://en.wikipedia.org/wiki/Session_Description_Protocol) to specify the parameters for the sessions.[[11]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-11)

An RTP session is established for each multimedia stream. Audio and video streams may use separate RTP sessions, enabling a receiver to selectively receive components of a particular stream.[[12]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Zurawski_28-12) The RTP and RTCP design is independent of the transport protocol. Applications most typically use UDP with port numbers in the unprivileged range (1024 to 65535).[[13]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Collins_47-13) The [Stream Control Transmission Protocol](https://en.wikipedia.org/wiki/Stream_Control_Transmission_Protocol) (SCTP) and the [Datagram Congestion Control Protocol](https://en.wikipedia.org/wiki/Datagram_Congestion_Control_Protocol) (DCCP) may be used when a reliable transport protocol is desired. The RTP specification recommends even port numbers for RTP, and the use of the next odd port number for the associated RTCP session.[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):68 A single port be used for RTP and RTCP in applications that multiplex the protocols.[[15]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-15)

RTP is used by real-time multimedia applications such as [voice over IP](https://en.wikipedia.org/wiki/Voice_over_IP), [audio over IP](https://en.wikipedia.org/wiki/Audio_over_IP), [WebRTC](https://en.wikipedia.org/wiki/WebRTC) and [Internet Protocol television](https://en.wikipedia.org/wiki/Internet_Protocol_television)

Profiles and payload formats[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=2)]

*Main article:*[*RTP payload formats*](https://en.wikipedia.org/wiki/RTP_payload_formats)

RTP is designed to carry a multitude of multimedia formats, which permits the development of new formats without revising the RTP standard. To this end, the information required by a specific application of the protocol is not included in the generic RTP header. For each class of application (e.g., audio, video), RTP defines a *profile* and associated *payload formats*.[[8]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Peterson_430-8) Every instantiation of RTP in a particular application requires a profile and payload format specifications.[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):71

The profile defines the codecs used to encode the payload data and their mapping to payload format codes in the protocol field *Payload Type* (PT) of the RTP header. Each profile is accompanied by several payload format specifications, each of which describes the transport of particular encoded data.[[2]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Perkins_55-2) Examples of audio payload formats are [G.711](https://en.wikipedia.org/wiki/G.711), [G.723](https://en.wikipedia.org/wiki/G.723), [G.726](https://en.wikipedia.org/wiki/G.726), [G.729](https://en.wikipedia.org/wiki/G.729), [GSM](https://en.wikipedia.org/wiki/GSM), [QCELP](https://en.wikipedia.org/wiki/QCELP), [MP3](https://en.wikipedia.org/wiki/MP3), and [DTMF](https://en.wikipedia.org/wiki/DTMF), and examples of video payloads are [H.261](https://en.wikipedia.org/wiki/H.261), [H.263](https://en.wikipedia.org/wiki/H.263), [H.264](https://en.wikipedia.org/wiki/H.264), [H.265](https://en.wikipedia.org/wiki/H.265) and [MPEG-1](https://en.wikipedia.org/wiki/MPEG-1)/[MPEG-2](https://en.wikipedia.org/wiki/MPEG-2).[[16]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-perkins_60-16) The mapping of [MPEG-4](https://en.wikipedia.org/wiki/MPEG-4) audio/video streams to RTP packets is specified in [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3016](https://tools.ietf.org/html/rfc3016), and H.263 video payloads are described in [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [2429](https://tools.ietf.org/html/rfc2429).[[17]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Chou2007-17)

Examples of RTP profiles include:

* The *RTP profile for Audio and video conferences with minimal control* ([RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3551](https://tools.ietf.org/html/rfc3551)) defines a set of static payload type assignments, and a dynamic mechanism for mapping between a payload format, and a PT value using [Session Description Protocol](https://en.wikipedia.org/wiki/Session_Description_Protocol) (SDP).
* The [Secure Real-time Transport Protocol](https://en.wikipedia.org/wiki/Secure_Real-time_Transport_Protocol) (SRTP) ([RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3711](https://tools.ietf.org/html/rfc3711)) defines an RTP profile that provides [cryptographic](https://en.wikipedia.org/wiki/Cryptography) services for the transfer of payload data.[[18]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-18)
* The experimental *Control Data Profile for RTP* (RTP/CDP) for [machine-to-machine](https://en.wikipedia.org/wiki/Machine-to-machine) communications.[[19]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Breese2010-19)

Packet header[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=3)]

RTP packets are created at the application layer and handed to the transport layer for delivery. Each unit of RTP media data created by an application begins with the RTP packet header.

|  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
| **RTP packet header** | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| ***Offsets*** | **Octet** | **0** | | | | | | | | **1** | | | | | | | | **2** | | | | | | | | **3** | | | | | | | |
| **Octet** | **Bit**[[a]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-20) | **0** | **1** | **2** | **3** | **4** | **5** | **6** | **7** | **8** | **9** | **10** | **11** | **12** | **13** | **14** | **15** | **16** | **17** | **18** | **19** | **20** | **21** | **22** | **23** | **24** | **25** | **26** | **27** | **28** | **29** | **30** | **31** |
| **0** | **0** | Version | | P | X | CC | | | | M | PT | | | | | | | Sequence number | | | | | | | | | | | | | | | |
| **4** | **32** | Timestamp | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| **8** | **64** | SSRC identifier | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| **12** | **96** | CSRC identifiers ... | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| **12+4×CC** | **96+32×CC** | Profile-specific extension header ID | | | | | | | | | | | | | | | | Extension header length | | | | | | | | | | | | | | | |  |  |
| **16+4×CC** | **128+32×CC** | Extension header ... | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |

The RTP header has a minimum size of 12 bytes. After the header, optional header extensions may be present. This is followed by the RTP payload, the format of which is determined by the particular class of application.[[20]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-21) The fields in the header are as follows:

* **Version**: (2 bits) Indicates the version of the protocol. Current version is 2.[[21]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-peterson_431-22)
* **P (Padding)**: (1 bit) Used to indicate if there are extra padding bytes at the end of the RTP packet. Padding may be used to fill up a block of certain size, for example as required by an encryption algorithm. The last byte of the padding contains the number of padding bytes that were added (including itself).[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):12[[21]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-peterson_431-22)
* **X (Extension)**: (1 bit) Indicates presence of an *extension header* between the header and payload data. The extension header is application or profile specific.[[21]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-peterson_431-22)
* **CC (CSRC count)**: (4 bits) Contains the number of CSRC identifiers (defined below) that follow the SSRC (also defined below).[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):12
* **M (Marker)**: (1 bit) Signaling used at the application level in a profile-specific manner. If it is set, it means that the current data has some special relevance for the application.[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):13
* **PT (Payload type)**: (7 bits) Indicates the format of the payload and thus determines its interpretation by the application. Values are profile specific and may be dynamically assigned.[[22]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-23)
* **Sequence number**: (16 bits) The sequence number is incremented for each RTP data packet sent and is to be used by the receiver to detect packet loss[[1]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-Hardy_298-1) and to accommodate [out-of-order delivery](https://en.wikipedia.org/wiki/Out-of-order_delivery). The initial value of the sequence number should be randomized to make [known-plaintext attacks](https://en.wikipedia.org/wiki/Known-plaintext_attack) on [Secure Real-time Transport Protocol](https://en.wikipedia.org/wiki/Secure_Real-time_Transport_Protocol) more difficult.[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):13
* **Timestamp**: (32 bits) Used by the receiver to play back the received samples at appropriate time and interval. When several media streams are present, the timestamps may be independent in each stream.[[b]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-24) The granularity of the timing is application specific. For example, an audio application that samples data once every 125 µs (8 kHz, a common sample rate in digital telephony) would use that value as its clock resolution. Video streams typically use a 90 kHz clock. The clock granularity is one of the details that is specified in the RTP profile for an application.[[23]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-peterson_432-25)
* **SSRC**: (32 bits) *Synchronization source identifier* uniquely identifies the source of a stream. The synchronization sources within the same RTP session will be unique.[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):15
* **CSRC**: (32 bits each, the number of entries is indicated by the *CSRC count* field) *Contributing source IDs* enumerate contributing sources to a stream which has been generated from multiple sources.[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):15
* **Header extension**: (optional, presence indicated by *Extension* field) The first 32-bit word contains a profile-specific identifier (16 bits) and a length specifier (16 bits) that indicates the length of the extension in 32-bit units, excluding the 32 bits of the extension header. The extension header data follows.[[14]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-RFC3550-14):18

Application design[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=4)]

A functional multimedia application requires other protocols and standards used in conjunction with RTP. Protocols such as SIP, [Jingle](https://en.wikipedia.org/wiki/Jingle_(protocol)), RTSP, [H.225](https://en.wikipedia.org/wiki/H.225) and [H.245](https://en.wikipedia.org/wiki/H.245) are used for session initiation, control and termination. Other standards, such as H.264, MPEG and H.263, are used for encoding the payload data as specified by the applicable RTP profile.[[24]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-perkins_11-26)

An RTP sender captures the multimedia data, then encodes, frames and transmits it as RTP packets with appropriate timestamps and increasing timestamps and sequence numbers. The sender sets the *payload type* field in accordance with connection negotiation and the RTP profile in use. The RTP receiver detects missing packets and may reorder packets. It decodes the media data in the packets according to the payload type and presents the stream to its user.[[24]](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_note-perkins_11-26)

Standards documents[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=5)]

* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3550](https://tools.ietf.org/html/rfc3550), Standard 64, *RTP: A Transport Protocol for Real-Time Applications*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3551](https://tools.ietf.org/html/rfc3551), Standard 65, *RTP Profile for Audio and Video Conferences with Minimal Control*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [4855](https://tools.ietf.org/html/rfc4855), *Media Type Registration of RTP Payload Formats*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [4856](https://tools.ietf.org/html/rfc4856), *Media Type Registration of Payload Formats in the RTP Profile for Audio and Video Conferences*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [7656](https://tools.ietf.org/html/rfc7656), *A Taxonomy of Semantics and Mechanisms for Real-Time Transport Protocol (RTP) Sources*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3190](https://tools.ietf.org/html/rfc3190), *RTP Payload Format for 12-bit*[*DAT Audio*](https://en.wikipedia.org/wiki/Digital_Audio_Tape)*and 20- and 24-bit Linear Sampled Audio*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [6184](https://tools.ietf.org/html/rfc6184), *RTP Payload Format for*[*H.264*](https://en.wikipedia.org/wiki/H.264)*Video*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3640](https://tools.ietf.org/html/rfc3640), *RTP Payload Format for Transport of MPEG-4 Elementary Streams*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [6416](https://tools.ietf.org/html/rfc6416), *RTP Payload Format for*[*MPEG-4*](https://en.wikipedia.org/wiki/MPEG-4)*Audio/Visual Streams*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [2250](https://tools.ietf.org/html/rfc2250), *RTP Payload Format for*[*MPEG1*](https://en.wikipedia.org/wiki/MPEG1)*/*[*MPEG2 Video*](https://en.wikipedia.org/wiki/MPEG-2_Video)
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [4175](https://tools.ietf.org/html/rfc4175), *RTP Payload Format for Uncompressed Video*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [6295](https://tools.ietf.org/html/rfc6295), [*RTP Payload Format for MIDI*](https://en.wikipedia.org/wiki/RTP-MIDI)
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [4696](https://tools.ietf.org/html/rfc4696), *An Implementation Guide for RTP MIDI*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [7587](https://tools.ietf.org/html/rfc7587), *RTP Payload Format for the*[*Opus*](https://en.wikipedia.org/wiki/Opus_(audio_format))*Speech and Audio Codec*
* [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [7798](https://tools.ietf.org/html/rfc7798), *RTP Payload Format for*[*High Efficiency Video Coding*](https://en.wikipedia.org/wiki/High_Efficiency_Video_Coding)*(HEVC)*

See also[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=6)]

* [Real Time Streaming Protocol](https://en.wikipedia.org/wiki/Real_Time_Streaming_Protocol)
* [Real Data Transport](https://en.wikipedia.org/wiki/Real_Data_Transport)
* [ZRTP](https://en.wikipedia.org/wiki/ZRTP)

Notes[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=7)]

* 1. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-20) Bits are ordered most significant to least significant; bit offset 0 is the most significant bit of the first octet. Octets are transmitted in [network order](https://en.wikipedia.org/wiki/Network_order). Bit transmission order is medium dependent.
  2. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-24) [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [7273](https://tools.ietf.org/html/rfc7273) provides a means for signalling the relationship between media clocks of different streams.

References[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=8)]

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  2. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Perkins_55_2-0) [***b***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Perkins_55_2-1) [***c***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Perkins_55_2-2) [Perkins 2003](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#CITEREFPerkins2003), p. 55
  3. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Perkins_46_3-0) [***b***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Perkins_46_3-1) [Perkins 2003](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#CITEREFPerkins2003), p. 46
  4. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-4) [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [4571](https://tools.ietf.org/html/rfc4571)
  5. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-5) *Farrel, Adrian (2004).*[*The Internet and its protocols*](https://books.google.com/?id=LtBegQowqFsC&pg=PA363&dq=rtp+sctp)*. Morgan Kaufmann. p. 363.*[*ISBN*](https://en.wikipedia.org/wiki/ISBN_(identifier))[*978-1-55860-913-6*](https://en.wikipedia.org/wiki/Special:BookSources/978-1-55860-913-6)*.*
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  7. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-7) *Hogg, Scott.*[*"What About Stream Control Transmission Protocol (SCTP)?"*](http://www.networkworld.com/article/2222277/cisco-subnet/what-about-stream-control-transmission-protocol--sctp--.html)*. Network World. Retrieved 2017-10-04.*
  8. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Peterson_430_8-0) [***b***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Peterson_430_8-1) *Larry L. Peterson (2007). Computer Networks. Morgan Kaufmann. p.*[*430*](https://books.google.com/books?id=zGVVuO-6w3IC&pg=PA430)*.*[*ISBN*](https://en.wikipedia.org/wiki/ISBN_(identifier))[*978-1-55860-832-0*](https://en.wikipedia.org/wiki/Special:BookSources/978-1-55860-832-0)*.*
  9. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Colins_56_9-0) [***b***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-Colins_56_9-1) [Perkins 2003](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#CITEREFPerkins2003), p. 56
  10. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-10) [Peterson 2007](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#CITEREFPeterson2007), p. 435
  11. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-11) [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [4566](https://tools.ietf.org/html/rfc4566): *SDP: Session Description Protocol*, M. Handley, V. Jacobson, C. Perkins, IETF (July 2006)
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  14. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-0) [***b***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-1) [***c***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-2) [***d***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-3) [***e***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-4) [***f***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-5) [***g***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-6) [***h***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-7) [***i***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-RFC3550_14-8) [RFC](https://en.wikipedia.org/wiki/RFC_(identifier)) [3550](https://tools.ietf.org/html/rfc3550)
  15. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-15) [*Multiplexing RTP Data and Control Packets on a Single Port*](https://tools.ietf.org/html/rfc5761)*. IETF. April 2010.*[*doi*](https://en.wikipedia.org/wiki/Doi_(identifier))*:*[*10.17487/RFC5761*](https://doi.org/10.17487%2FRFC5761)*.*[*RFC*](https://en.wikipedia.org/wiki/RFC_(identifier))[*5761*](https://tools.ietf.org/html/rfc5761)*. Retrieved November 21, 2015.*
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  21. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-peterson_431_22-0) [***b***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-peterson_431_22-1) [***c***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-peterson_431_22-2) [Peterson 2007](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#CITEREFPeterson2007), p. 431
  22. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-23) [Perkins 2003](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#CITEREFPerkins2003), p. 59
  23. [**^**](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-peterson_432_25-0) Peterson, p.[432](https://books.google.com/books?id=zGVVuO-6w3IC&pg=PA432)
  24. ^ [Jump up to:***a***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-perkins_11_26-0) [***b***](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#cite_ref-perkins_11_26-1) [Perkins 2003](https://en.wikipedia.org/wiki/Real-time_Transport_Protocol#CITEREFPerkins2003), pp. 11–13
* *Perkins, Colin (2003).*[*RTP*](https://books.google.com/?id=OM7YJAy9_m8C)*. Addison-Wesley.*[*ISBN*](https://en.wikipedia.org/wiki/ISBN_(identifier))[*978-0-672-32249-5*](https://en.wikipedia.org/wiki/Special:BookSources/978-0-672-32249-5)*.*
* *Peterson, Larry L.; Davie, Bruce S. (2007). Computer Networks (4 ed.). Morgan Kaufmann.*[*ISBN*](https://en.wikipedia.org/wiki/ISBN_(identifier))[*978-0-12-374013-7*](https://en.wikipedia.org/wiki/Special:BookSources/978-0-12-374013-7)*.*
* [*"RTP"*](https://books.google.com/books?id=D_GrQa2ZcLwC&pg=PA144)*. Network Protocols Handbook. Javvin Technologies. 2005.*[*ISBN*](https://en.wikipedia.org/wiki/ISBN_(identifier))[*978-0-9740945-2-6*](https://en.wikipedia.org/wiki/Special:BookSources/978-0-9740945-2-6)*.*
* [*"RTP"*](https://www.youtube.com/watch?v=OaL2vVFbCG4&feature=channel_page)*, Broadband Networks, Ministry of Human resources, India, 2008*

External links[[edit](https://en.wikipedia.org/w/index.php?title=Real-time_Transport_Protocol&action=edit&section=9)]

* [oRTP, RTP library from Linphone written in C](http://www.linphone.org/eng/documentation/dev/ortp.html)
* [Henning Schulzrinne's RTP page](https://www.cs.columbia.edu/~hgs/rtp) (including [FAQ](https://www.cs.columbia.edu/~hgs/rtp/faq.html))
* [GNU ccRTP](https://www.gnu.org/software/ccrtp/)
* [JRTPLIB, a C++ RTP library](http://research.edm.uhasselt.be/~jori/page/index.php?n=CS.Jrtplib)
* [Managed Media Aggregation](http://net7mma.codeplex.com/): [.NET](https://en.wikipedia.org/wiki/.NET_Framework) [C#](https://en.wikipedia.org/wiki/C_Sharp_(programming_language)) RFC compliant implementation of RTP / RTCP written in completely managed code.