Rtcp rfc 3550 pdf

RTCP, RTP Control Protocol Description Glossary RFCs Publishing Outdated RFCs Description: MAC headline IP headline UDP headline RTCP headline Data ::: RTCP headline: Version. Two bits. The RTP version, which is the same in RTCP packages as in RTP data packets. Pading. 1 bit. If set, this RTCP package contains some additional bytes of upholstery at the end that are not part of the control information. The latest byte of upholstery is to calculate how many bytes of upholstery should be ignored. Some fixed block encryption algorithms may be required. In the RTCP composite package, upholstery should only be required on the last separate package, as the connection package is encrypted as a whole. Count. 5 bit. 0 to 31. The number of acceptance report blocks contained in this package. Type. 8 bit. Type of RTCP package. TypeDescriptionReferences 0-191 192FIR, full intra-frame request. RFC 2032 193NACK, negative recognition. RFC 2032 194SMPTETC, SMPTE time code mapping. RFC5484 195IJ, Extended Inter-Profit Report. RFC 5450 196-199 200SR, sender report. RFC 3550 201RR, receiver report. RFC 3550 202SDES, source description. RFC 3550 203BYE, bye. RFC 3550 204APP, application determined. RFC 3550 205RTPFB, General RTP Reviews. 206PSFB, payload specific. 207XR, RTCP expansion. RFC 3611 208AVB, AVB RTCP package. IEEE 1733 209RSI, Receiver Consolidated Information. RFC 5760 210-255 Length. 16 bit. The length of this RTCP package is 32 bits of words minus one, including the title and any ups ups and ups. The offset of one makes zero valid length and avoids a possible endless loop in the RTCP compound scanning package, while counting 32 bit words avoids validity checks for several of the 4. Glossary: RFCs: RFC 2032 RTP payload format for H.261 video streams. Category: Standard track. Identifies the types of PACKAGES RTCP 192 (FIR) and 193 (NACK). (RFC 3158) RTP Testing Strategies. (RFC 3550) RTP: Real-time traffic protocol for applications. STDs: 64. Outdated: RFC 1889. (RFC 3551) RTP profile for audio and video conferencing with minimal control. STD: 65. Outdated: RFC 1890. (RFC 3605) Real Time Control Protocol (RTCP) attribute in session description protocol (SDP). Category: Standard track. Defines the SDP rtcp attribute. (RFC 3611) RtP Advanced Reports Monitoring Protocol (RTCP XR). Category: Standard track. Defines the type of RTCP 207 (XR) package. (RFC 3711) Real-time Safe Transport Protocol (SRTP). Category: Standard track defines RTP/SAVP profile. (RFC 5450) Transfer time Shift in RTP streams. Category: Standard track. Defines the type of package RTCP 195 (IJ). Publications: Outdated RFCs: RFC 1889 RTP: Real-time application transport protocol. Category: Standard track. RFC 3550. (RFC 1890) RTP profile for audio and video conferencing with minimal control. Category: Standard track. Outdated: RFC 3551. Description of RFCs RFCs Outdated RFCs ValuesUsage notes the Compatibility of The SpecificationsBrowser The WebRTC API in RTCIceComponent the type listed contains the values of DOMString that each identify a specific component of ICE; it's rtp and rtcp. These lines are displayed at the corresponding numerical values as they appear in the candidate-line in the SDP. The rtp values are determined by ICE transport, which is used for real-time Transport Protocol (RTP), or for an RTP multiplex with an RTP control protocol (RTCP). RTP is defined in RFC 3550. RTCIceComponent of this value corresponds to the component ID field in the a-line candidate with a value of 1. rtcp defines the ICE transport used for RTCP, which is defined in RFC 3550, Section 6. This value of RTCIceComponent corresponds to the ID identifier of ID 2. Use notes that the RTCIceComponent type is used by the component property of RTCIceCandidate and RTCIceTransport. Спецификации Браузер совместимость Обновление данных о совместимости GitHubDesktopMobileChromeEdgeFirenet ExplorerOperaSafariAndroid webviewChrome для AndroidFirefox для AndroidOpera для AndroidSafari на iOSSamsung InternetRTCIceComponentChrome Нет поддержки NoEdge Нет поддержки NoFirefox Нет поддержки NoIE Нет поддержки NoIE Нет поддержки NoE? Safari? WebView Android No support NoChrome Android No support NoFirefox Android No support NoOpera Android? iOS Safari ? Samsung Internet Android No Support No Support No Support Non-Compatibility Unknown Compatibility Unknown Protocol for Delivering Audio and Video Via IP Internet Protocol Application Package Layer BGP DHCP DNS FTP HTTP HTTPS IMAP LDAP MGCP MHT NNTP NTP POP PTP ONC/RPC RTP RTPSP SIP SIP SMTP SNMP Telnet TLS The transport layer TCP UDP DCCP SCTP RSVP more details... Internet layer IP IPv4 IPv6 ICMP ICMPv6 ECN IGMP IPsec more ... Link layer ARP NDP OSPF Tunnels L2 PPPTP MAC Ethernet Wi-Fi DSL ISDN FDDI more... vte Real-time Transport Protocol (RTP) is a network protocol for delivering audio and video via IP networks. RTP is used in communication and entertainment systems that include streaming media such as telephony, video teleconference applications including WebRTC, television services, and push-to-talk web features. RTP usually works on the user's datagram (UDP) protocol. RTP is used in conjunction with the RTCP Protocol. While RTP carries media streams (such as audio and video), RTCP is used to monitor transmission and service quality statistics (zoS) and helps synchronize multiple streams. RTP is one of the technical foundations of Voices over IP and in this context is often used in conjunction with an alarm protocol, such as the Session Initiation Protocol (SIP), which establishes connections throughout the network. RTP was Working Group on Audio Video Transport task force on Internet Engineering (IETF) and first published in 1996 as RFC 1889 1889 was then supersed by the RFC 3550 in 2003. The RTP review is designed for the ultimate, real-time streaming media transmission. The protocol provides opportunities to compensate for fright and detect loss of packages and delivery outside of order that are common, especially during the transmission of UDP to the IP network. RTP allows data to be transmitted in several directions through IP multicast. RTP is considered the main standard for audio/video transport in IP networks and is used with an accompanying profile and payload format. The RTP design is based on an architectural principle known as application level framing, where protocol functions are implemented in an application, as opposed to the operating system protocol. Real-time multimedia streaming applications require timely delivery of information and can often tolerate some package losses to achieve this goal. For example, losing a package in an audio application can result in a fraction of a second of audio data being lost, which can be discreetly with appropriate error-hiding algorithms. The Transmission Control Protocol (TCP), although standardized for the use of RTP, is not usually used in RTP applications because TCP advocates for reliability over time. Instead, most RTP implementations are based on the User Datagram (UDP) Protocol. Other transport protocols specifically designed for multimedia sessions are SCTP and DCCP, although they are not widely used as of 2012. RTP was developed by the IETF Audio/Video Transportation Working Group. RTP is used in conjunction with other protocols such as H.323 and RTSP. The RTP specification describes two protocols: RTP and RTCP. RTP is used to transmit multimedia data, and RTCP is used for periodic transmission of control information and AIA parameters. The data protocol, RTP, contains real-time data. The information provided by this protocol includes time stamps (for synchronization), sequence numbers (to detect package loss and re-order) and a payload format that indicates a coded data format. The control protocol, RTCP, is used for quality feedback and synchronization between media streams. RTCP traffic bandwidth compared to RTP is generally small, usually around 5%. RTP sessions are usually initiated between communication nodes using an alarm protocol such as H.323, Session Initiation Protocol (SIP), RTSP, or Jingle (XMPP). These protocols can use the Session Description Protocol to describe session settings. An RTP session is set for each media stream. Audio and video streams can use individual RTP sessions, allowing the recipient to selectively receive components of a particular thread. The design of RTP and RTCP does not depend on Protocol. Apps most most use UDP with port numbers in unprivileged ranges (1024 to 65535). The Flow Transfer Protocol (SCTP) and the Datagram Congestion Control Protocol (DCCP) can be used if a reliable transport protocol is desired. The RTP specification recommends even port numbers for RTP, as well as the use of the next odd port number for a related RTCP session. One port is used for RTP and RTCP in applications that multiplex protocols. RTP is used by real-time multimedia applications such as VOICE voice, IP audio, WebRTC and Internet Protocol TV profiles and payload formats Main article: RTP payload formats are designed to run multiple multimedia formats, allowing the development of new formats without revising the RTP standard. To this end, the information required by the specific application of the protocol is not included in the overall RTP headline. For each app class (such as audio, video), RTP identifies the profile and associated payload formats. Each instant RTP work in a particular application requires profile and payload specifications. The profile identifies the codeks used to encode payload data and display payload codes in the RTP payload type (PT) protocol field. Each profile is accompanied by several payload specifications, each describing the transportation of specific coded data. Examples of sound payload formats are G.711, G.723, G.726, G.729, GSM, zELP, MP3 and DTMF, and examples of video-by-loads are H.261, H.263, H.264, H.265 and MPEG-1/MPEG-2. The MPEG-4 audio/video streams are shown in RTP packages in RFC 3016, and the H.263 payload video is described in RFC 2429. Examples of RTP profiles include: RTP profile for audio and video conferencing with minimal control (RFC 3551) identifies a set of static payload-type assignments and a dynamic display mechanism between the payload format and the PT value using the Session Description Protocol (SDP). The Real Time Safe Transport Protocol (RFC 3711) defines the RTP profile that provides cryptographic services for payload data. Experimental control data profile for RTP (RTP/CDP) for machine-machine communication. The RTP package headers are created at the app level and passed to the transport layer for delivery. Each RTP media data unit created by the app starts with the rtP package header. RTP Package Headline Offsets Octet 0 1 2 3 Octet Bit 0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 20 21 22 23 24 25 26 27 28 29 30 31 0 Version P X CC M PT Sequence Number 4 32 Timestamp 8 64 SSRC ID 12 96 CSRC identifiers... 12'4×CC 96×CC Profile-specific Expansion ID Expansion headline length 16×4×CC 128×CC Expansion headline ... Headline RTP has a minimum size size Bytes. After the title, there may be additional title extensions. This is followed by an RTP payload, the format of which is determined by a specific class of application. The fields in the title are as follows: Version: (2 bits) points to a version of the protocol. Current version 2. P (Padding): (1 bit) Is used to indicate whether there are additional upholstery bytes at the end of the RTP package. The padding can be used to fill a block of a certain size, for example, as required by the encryption algorithm. The last byte of the upholstery contains the number of upholstery bytes that have been added (including themselves). X (expansion): (1 bit) indicates that there is an extension header between the title and the payload data. The extension title is a specific app or profile. CC (CSRC count): (4 bits) contains the number of CSRC identifiers (defined below) that follow the SSRC (also defined below). (Marker): (1 bit) Signaling used at the application level in a profile manner. If it is installed, it means that current data is of particular importance to the application. PT (payload type): (7 bits) indicates the payload format and thus determines its interpretation by the application. The values are profile-specific and can be dynamically assigned. Sequence number: (16 bits) The sequence number is increments for each RTP data package sent and must be used by the recipient to detect the loss of the package and to place the delivery outside the order. The initial value of the sequence number must be randomized to make real-time attacks on the Secure Transport Protocol more difficult. Timestamp: (32 bits) Is used by the receiver to play the samples at the appropriate time and interval. With multiple media streams, tags can be independent in each thread. The time detail is specific for use. For example, an audio application that samples data once every 125 x (8 kHz, a common sample speed in digital telephony) will use this value as its resolution of the watch. Video streams typically use 90 kHz watches. The watch detail is one of the details that is listed in the RTP profile for the app. SSRC: (32 bits) The synchronization source ID clearly identifies the source of the flow. The synchronization sources in the same RTP session will be unique. 14:15 CSRC: (32 bits each, number of entries is indicated by the CSRC count box) By making the original ID images list the sources of the thread that was created from multiple sources. Title extension: (optional presence specified by the extension field) The first 32-bit word contains a profile ID (16 bits) and a length specifier (16 bits), which indicates the length of the extension in 32-bit units, with the exception of 32 bits of the extension header. Extension data below. The Design of the Functional Multimedia App requires other protocols and standards used in conjunction with RTP. Protocols such as SIP, Jingle, RTSP, H.225 and H.245 are used to initiate session, control, and discontinue. Other standards, such as H.264, MPEG and H.263, are used to encode payload data listed in the applicable RTP profile. The RTP sender captures the multimedia data, then encodes, frames and transmits it as RTP packages with appropriate time stamps and increases in time and sequence. The sender installs a payload-type field in accordance with the connection negotiations and the RTP profile used. The RTP receiver detects missing packages and can reorder the packages. It deciphers media data in packages according to payload type and presents the flow to the user. Standards document RFC 3550, Standard 64, RTP: Real-time transport protocol for RFC 3551 applications, Standard 65, RTP PROFILE for audio and video conferencing with minimal RFC 4855 control, Media Type Registration of RFC RFC 4856 payload formats, Media Type Registration of RTP load formats in THE RTP profile for RTP audio and video conferencing RFC 7666, Media Type Registration of load formats in the RTP profile for audio and video conferencing RFC 7666, Media Type Registration of payload formats in the RTP profile. Taxonomity of semantics and mechanisms for real-time transportation protocol (RTP) Sources RFC 3190, RTP payload format for 12-bit DAT Audio and 20- and 24-bit linear audio sample RFC 6184, RTP payload format for H.26 26 26 26 Video RFC 3640, RTP payload format for transporting MPEG-4 Elementary streams RFC 6416, RTP Useful image for MPEG-4 Audio/Visual Streams RFC 2250, RTP payload format for MPEG1/MPEG2 Video RFC 4175 RTP payload format for RFC 6295, RTP payload format for MIDI RFC 4696, RTP's implementation guide for RTP MIDI RFC 7587, RTP payload format for Opus Speech and Audio Codec RFC 7798, RTP payload format for high-fuffefefefing video (HEVC) Bit bias 0 is the most significant bit of the first Octet. Octets are transmitted online. The order of transmission of a bit of medium dependency. RFC 7273 provides the tools to signal the relationship between media hours of different streams. Links to b Daniel Hardy (2002). Network. De Bok University. page 298. a b c Perkins 2003, p. 55 - b Perkins 2003, p. 46 - RFC 4571 - Farrel, Adrian (2004). The Internet and its protocols. Morgan Kaufmann. page 363. ISBN 978-1-55860-913-6. Hzaktas, Khaldun M.; Levent Onural (2007). 3D TELEVISION. Springer. page 356. ISBN 978-3-540-72531-2. Hogg, Scott. What about the Control Flow Transfer Protocol (SCTP)?. The network world. Received 2017-10-04. a b Larry L. Peterson (2007). Computer networks. Morgan Kaufmann. page 430. Isbn a b Перкинс 2003, стр. 56 56 p. 435 harvnb error: no purpose: CITEREFPeterson2007 (help) - RFC 4566: SDP: Session Description Protocol, M. Hingley, W. Jacobson, K. Perkins, IETF (July 2006) - Surawski, Richard (2004). Protocols of RTP, RTCP and RTSP. A guide to industrial information technology. CRC Press. 28-7. ISBN 978-0-8493-1985-3. Daniel Collins (2002). Carrying voice using IP. Voice of the carrier class on IP. McGraw-Hill Professional. page 47. ISBN 978-0-07-136326-6. - b c d e f h i RFC 3550 - Multiplexing of data packets and RTP management in one port. Ietf. April 2010. doi:10.17487/RFC5761. RFC 5761. Received on November 21, 2015. Perkins 2003, page 60, Chow, Philip A.; Michaela van der Shaar (2007). Multimedia via IP and wireless networks. Academic press. page 514. ISBN 978-0-12-088480-3. Perkins 2003, page 367 - Braz, Finley (2010). RTP/CDP serial communication. BoD - Books on demand. ISBN 978-3-8391-8460-8. Peterson 2007, page 430 harvnb error: no goal: CITEREFPeterson2007 (help) p. 431 harvnb error: no goal: CITEREFPeterson2007 (help) - Perkins 2003, p. 59 - Peterson, p.432 - b Perkins 2003, p. 11-13 Rtp. Addison-Wesley. ISBN 978-0-672-32249-5. Larry L. Peterson; Davis, Bruce S. (2007). Computer networks (4th. Morgan Kaufmann. ISBN 978-0-12-374013-7. RTP. A guide to network protocols. Jawwin Technologies. 2005. ISBN 978-0-9740945-2-6. RTP, Broadband Networks, Ministry of Human Resources, India, 2008 External oRTP Links, Linphone RTP Library, written on The RtP C Henning Schultzrinna page (including frequently asked questions) GNU ccRTP JRTPLIB, C'RTP Managed Media Aggregation: .NET C' RFC compatible implementation RTP/RTCP written in fully managed code. Extracted from the rtcp rfc 3550 pdf

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