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IEEE Standard for Layer 3 Transport Protocol for Time-Sensitive Applications in Local Area Networks

IEEE Computer Society

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IEEE Standard for Layer 3 Transport Protocol for Time-Sensitive Applications in Local Area Networks

Sponsor

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IEEE Computer Society

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Abstract: Protocol, data encapsulations, connection management, and presentation time procedures used to ensure interoperability between audio and video-based end stations that use standard networking services provided by all IEEE 802 networks meeting QoS requirements for time-sensitive applications by leveraging the Real-time Transport Protocol (RTP) family of protocols and IEEE 802.1 Audio/Video Bridging (AVB) protocols is specified in this standard.
Keywords: bridged LAN, IEEE 802.1AS, IEEE 802.1 AVB protocols, IEEE 802.1Qat, IEEE 802.1Qav, IEEE 1733, LAN, QoS, RTCP, RTP, time-sensitive media streaming, time synchronization

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Introduction

This introduction is not part of IEEE Std 1733-2011, IEEE Standard for Layer 3 Transport Protocol for Time-Sensitive Applications in Local Area Networks.

This standard specifies the protocol, data encapsulations, connection management and presentation time procedures used to ensure interoperability between audio and video based end stations that use standard networking services provided by all IEEE 802 networks meeting quality of service (QoS) requirements for time-sensitive applications by leveraging the Real-time Transport Protocol (RTP) family of protocols and family of IEEE 802.1 Audio/Video Bridging (AVB) protocols.

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IEEE Standard for Layer 3 Transport Protocol for Time-Sensitive Applications in Local Area Networks

1. Overview

The Real-time Transport Protocol (RTP) family of protocols is one of the most commonly used protocols for streaming real-time media across packet networks. While RTP is widely used for media streaming, as originally defined it did not take advantage of recent layer-2 standards created by the IEEE 802.1 Audio Video Bridging (AVB) task group for time-sensitive media streaming. These standards, collectively referred to as the AVB standards, include accurate timing (IEEE Std 802.1AS™-2011),[1] a reservation protocol (added to IEEE Std 802.1Q™-2005 as amended by IEEE Std 802.1Qat™-2010) and forwarding rules for traffic shaping (added to IEEE Std 802.1Q-2005 as amended by amendment 802.1Qav™-2009). This standard builds on and depends on the AVB standards.

This standard is relevant in today's entertainment world as more and more entertainment media is being digitally transported. Streaming audio/video and interactive applications over bridged LANs need to have real-time performance comparable with legacy analog distribution. There is significant end-user and vendor interest in defining a simple yet common IP-based method for handling real-time audio/video suitable for consumer electronics, professional A/V applications, etc.

Although RTP's use of UDP/IP implies that it can be routed between IP subnets, for reasons of simplicity this standard limits its scope to a single IP subnet.

[1] Information on references can be found in Clause 2.

1.1 Scope

This standard specifies the protocol, data encapsulations, connection management and presentation time procedures used to ensure interoperability between audio and video based end stations that use standard networking services provided by all IEEE 802 networks meeting quality of service (QoS) requirements for time-sensitive applications by leveraging the Real-time Transport Protocol (RTP) family of protocols and family of IEEE 802.1 Audio/Video Bridging (AVB) protocols.

1.2 Purpose

This standard will facilitate interoperability between stations that stream time-sensitive audio and/or video across bridged and routed LANs providing time synchronization and latency/bandwidth services by defining the packet format and stream setup, control, synchronization and teardown protocols by leveraging Real-time Transport Protocol (RTP) family of protocols and IEEE 802.1 AVB protocols.

2. Normative references

The following referenced documents are indispensable for the application of this document (i.e., they must be understood and used, so each referenced document is cited in text and its relationship to this document is explained). For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments or corrigenda) applies.

IEEE 802.1AS™-2011, IEEE Standard for Local and Metropolitan Area Networks—Timing and Synchronization for Time-Sensitive Applications in Bridged Local Area Networks.[2, 3]

IEEE Std 802.1Q™-2005, IEEE Standard for Local and Metropolitan Area Networks—Virtual Bridged Local Area Networks.

IEEE 802.1Qat™-2010, IEEE Standard for Local and Metropolitan Area Networks—Virtual Bridged Local Area Networks Amendment 14: Stream Reservation Protocol (SRP).

IEEE 802.1Qav™-2009, IEEE Standard for Local and Metropolitan Area Networks—Virtual Bridged Local Area Networks Amendment 12: Forwarding and Queuing Enhancements for Time-Sensitive Streams.

Real-Time Transport Protocol (RTP) Parameters: http://www.iana.org/assignments/rtp-parameters.

3. Definitions

For the purposes of this document, the following terms and definitions apply. The *IEEE Standards Dictionary: Glossary of Terms & Definitions*[4] [B3][5] should be referenced for terms not defined in this clause.

AVB RTCP packet: A Real-time Transport Protocol (RTCP) packet that carries the Audio/Video Bridge (AVB) control information from talker to listener.

[2] The IEEE standards or products referred to in this clause are trademarks of the Institute of Electrical and Electronics Engineers, Inc.
[3] IEEE publications are available from the Institute of Electrical and Electronics Engineers, 445 Hoes Lane, Piscataway, NJ 08854, USA (http://standards.ieee.org/).
[4] The *IEEE Standards Dictionary: Glossary of Terms and Definitions* is available at http://shop.ieee.org.
[5] The numbers in brackets correspond to those of the bibliography in Annex A.

grandmaster clock: Within a AS clock domain, a clock that is the ultimate source of time for clock synchronization using the IEEE 802.1AS protocol.

IEEE 802.1AVB: IEEE family of protocols that includes IEEE Std 802.1AS and IEEE Std 802.1Q-2005 as amended by IEEE Std 802.1Qat-2010 and IEEE Std 802.1Qav-2009.

listener: The destination of stream within the Local Area Network (LAN). If stream is terminating outside the LAN, listener is defined as exit point of stream in the LAN.

RTP profile: The Real-time Transport Protocol (RTP) profile is defined as a set of payload type codes and their mapping to payload formats (e.g., media encodings), extensions or modifications to RTP that are specific to a particular class of applications.

talker: The source of stream within the Local Area Network (LAN). If stream is originating outside the LAN, talker is defined as entry point of stream in the LAN.

4. Abbreviations and acronyms

AV	Audio/Video
AVB	Audio/Video Bridge
CSRC	contributing source
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
LAN	Local Area Network
LLC	Logical Link Control
MAC	Media Access Control
P	padding
QoS	quality of service
RTP	Real-time Transport Protocol
RTCP	RTP Control Protocol
RTSP	Real-time Streaming Protocol
SRP	Stream Reservation Protocol
SSRC	synchronization source
TAI	International Atomic Time
UTC	Universal Coordinated Time
V	version

5. Real-time Transport Protocol (RTP) and IEEE 802.1 AVB

This clause provides an overview of the RTP and RTCP protocols used for streaming media (audio and video). It also provides an overview of the IEEE 802.1 AVB protocols, which specify Time Synchronization and Layer-2 QoS extensions for streaming media.

5.1 RTP

The RTP provides end-to-end delivery services for data with real-time characteristics such as interactive audio and video. The RTP family of protocols consists of the two closely linked parts:

a) RTP: Carry data that has real-time properties.

b) RTCP: Monitor the quality of service and to convey information between participants of an on-going session.

RTP is usually integrated into the application to provide information required by the application rather than being implemented as a separate layer. RTP runs on the top of transport user datagram protocol (UDP) and network internet protocols (IP) to send RTP data and RTCP control streams. Some of the example applications where RTP is used are simple multicast audio conference, audio and video conference, etc. (for a detailed description of the RTP packet, refer IETF RFC 3550 [B6]).

Though the existing RTP data packet header definition is complete with respect to the application classes that RTP can support, it can be modified according to profile specifications as shown in the following examples:

— A profile may tailor the octets containing the marker field and payload type fields to suit different requirements.

— Applications may add additional functionality by defining additional fixed fields in the header.

— New implementations may extend the RTP data packet header to support new payload-independent functions.

5.1.1 RTP control protocol (RTCP)

The RTCP provides the control information for RTP. It describes the periodic transmission of control packets to all participants of a RTP session. RTCP has the following three functions:

1) Provides feedback on the quality of data distribution.

2) Carries the transport-level identifier CNAME (canonical name) for a RTP source.

3) Conveys necessary session control information.

There are several RTCP packet types to carry a variety of control information. The different types are as follows:

— SR: Sender Report

— RR: Receiver Report

— SDES: Source Description Items

— BYE: Indicates end of participation

— APP: Application-specific functions

— AVB RTCP: AVB specific RTCP packet defined in this standard

This standard uses the AVB RTCP packet for sending the necessary AVB information.

Each RTCP packet begins with a fixed part that is followed by elements of variable length. However, the variable length elements must end on a 32-bit boundary. RTCP packets are always sent in a compound packet containing at least two individual packets where the first packet is always a report (SR or RR). The RTCP compound packet also always includes a SDES packet containing the CNAME. The AVB RTCP packet (described in Clause 8) will follow the above RTCP packet types in a compound RTCP packet.

5.2 IEEE 802.1 Audio video bridging (AVB)

Ethernet and heterogeneous LANs of today do not guarantee bandwidth or latency as needed by many media streaming applications, including home and studio applications. IEEE 802.1 AVB fills this QoS gap for audio/video streaming over a bridged LAN.

Three main QoS issues are addressed by AVB as follows:

1) Ensure sufficient bandwidth on the network link to which the talker is attached to
2) Ensure sufficient bandwidth on all of the other links on the path to the listener(s)
3) Ensure sufficiently low latency for some of the applications like gaming, live video, etc.

For some applications, QoS is not enough and accurate clock synchronization is also needed. For example multi-speakers/displays require time synchronization both for simultaneous start and to counteract long-term drift. In addition, media streaming using push mode requires frequency synchronization.

The following AVB standards enable streaming of time sensitive AV media:

— IEEE Std 802.1AS-2011: Standard for accurate time synchronization between two end stations

— IEEE Std 802.1Qat-2010: Standard for Layer-2 Stream Reservation Protocol, which reserves network bandwidth for particular streams

— IEEE Std 802.1Qav-2009: Standard for traffic shaping, which guarantees the bandwidth and latency parameters of QoS

6. Operation of RTP over AVB

This clause describes how the RTP streaming protocol is expected to work over an IEEE 802.1 AVB network to provide QoS and time clock synchronization support for media streaming. Overall this clause talks about operation of two AVB standards: IEEE Std 802.1AS-2011 and IEEE Std 802.1Q-2005 Stream reservation Protocol (Clause 35) as amended by IEEE Std 802.1Qat-2009.

6.1 IEEE Std 802.1AS-2011

This standard specifies the protocol and procedures used to ensure that synchronization requirements are met for time sensitive applications, such as audio and video, across Bridged and Virtual Bridged LANs.

This standard enables stations attached to bridged LANs to meet the respective jitter, wander and time synchronization requirements for time-sensitive applications.

If the source of a media stream is not a real-time source (e.g., is being read from a mass storage device) and if there is a single listener, then there is no need to use this standard, nor is there a need to synchronize media clocks. However, if either condition is true (the media source is arriving at a rate that is outside the control of the listener(s), or if there are multiple listeners that must render the stream synchronously), some method must be employed to synchronize both the start time of a stream and also maintain the talker's media rate at the listener(s).

If the talker and listener(s) shared a common time reference [such as global positioning system (GPS)], it would be possible for the talker to specify the time at which the media should be presented. For example; present audio sample zero at GPS-traceable time 08:23:17.0000000 PST, present audio sample 96 at GPS-traceable time 8:23:17.002182 PST. In this example, the talker's media clock's relationship to the shared GPS time is communicated to the listener(s) that can then ensure that they are rendering the audio samples at the same rate as the talker. If they did not, their sample buffers could run out of data or overflow. In addition, the talkers would be capable of rendering the first sample at the correct time (e.g., simultaneously).

Rather than transport the IEEE 802.1AS timestamp in every RTP packet, the correlation between IEEE 802.1AS time and the media timestamp is communicated in an RTCP packet extension.

Along with the timestamp information the Grand Master information and any time discontinuities are also sent in the RTCP packet. These new RTCP packets containing the necessary clock information are sent periodically over the LAN to ensure RTP media clocks remain synchronized.

In order to derive the relation between the RTP sampling clock and the IEEE 802.1AS clock, at least two RTCP packets containing the above clock synchronization information are sent to the listener(s). Two RTCP packets are required to find the relation between the RTP timestamps and the IEEE 802.1AS timestamps.

In order to keep the clocks synchronized between the RTP sampling clock and the IEEE 802.1AS clock, RTCP packet containing the above clock synchronization needs to be sent periodically throughout the life of a stream.

In order to avoid the discontinuities associated with UTC, IEEE 802.1AS clock uses the International Atomic Time (TAI) time scale.

6.2 IEEE Std 802.1Qat-2009

This standard defines the Stream Reservation Protocol (SRP) which allows reservation of network resources (bandwidth, with latency bounds) for streams traversing a bridged LAN.

SRP specifies a Stream ID (64 bits) attribute to identify a specific media stream for a particular RTP session. The SRP reserves network bandwidth between end stations for each stream of a RTP session and once the reservation is successful; the Stream ID corresponding to that RTP session is passed to the RTP layer. This Stream ID is passed with every RTP packet down the network stack to the MAC layer. Thus, RTP packet containing a specific Stream ID is streamed to the destination station using the corresponding network resources identified by Stream ID.

Figure 1 illustrates how the necessary information is passed along with the RTP and RTCP packets to the lower layers for the RTP (implemented in the session layer) to function properly over the AVB protocol (implemented in the MAC layer) at the sender station, and how this same information passes to the RTP stack from the lower layers at the receiver station.

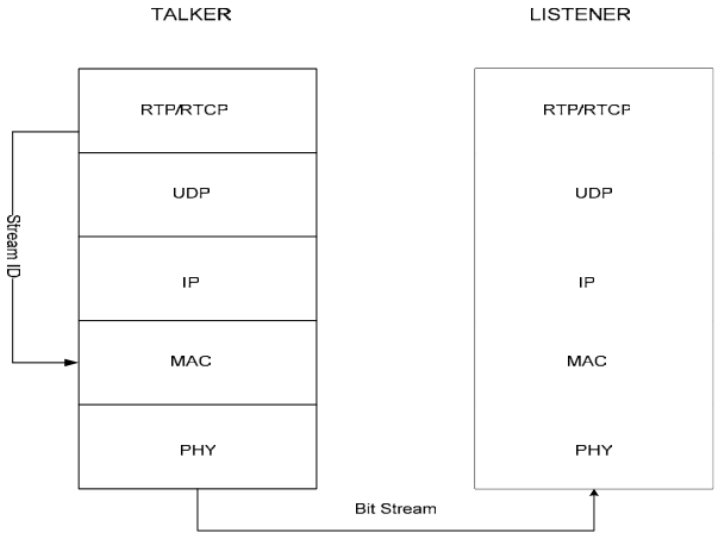

Figure 1—Flow of data and information in the network stack

7. RTP and network stack

There are no changes to the RTP header or packet. In particular, the meanings of the timestamp and SSRC fields are unchanged by this standard.

However, as a result of successful admission of a stream using Clause 35 of IEEE Std 802.1Q-2005, Stream Reservation Protocol (SRP), a Stream ID (64 bits) is generated which is used to identify each packet of the RTP session. The Stream ID of each RTP packet shall be known by the lower layers of the network stack so that proper IEEE Std 802.1Q, Clause 36 traffic shaping can be performed independently on each RTP stream (and then per class). The Stream ID has a one-to-one correlation with the RTP session, which is identified by the RTP SSRC field and 5-tuple. How Stream ID is passed to the lower layers with an RTP packet within the talker is out of scope of this standard.

8. AVB RTCP usage

RTCP control packets are used to synchronize the listeners' media clocks to the talker's media clock using the IEEE 802.1AS clock as a common reference between the talker and listener, and to relate synchronization source (SSRC) to Stream ID. To enable sending this information, this specification uses the AVB RTCP packet. This RTCP packet is sent only from talker to listener. The detailed frame format is shown in Figure 2.

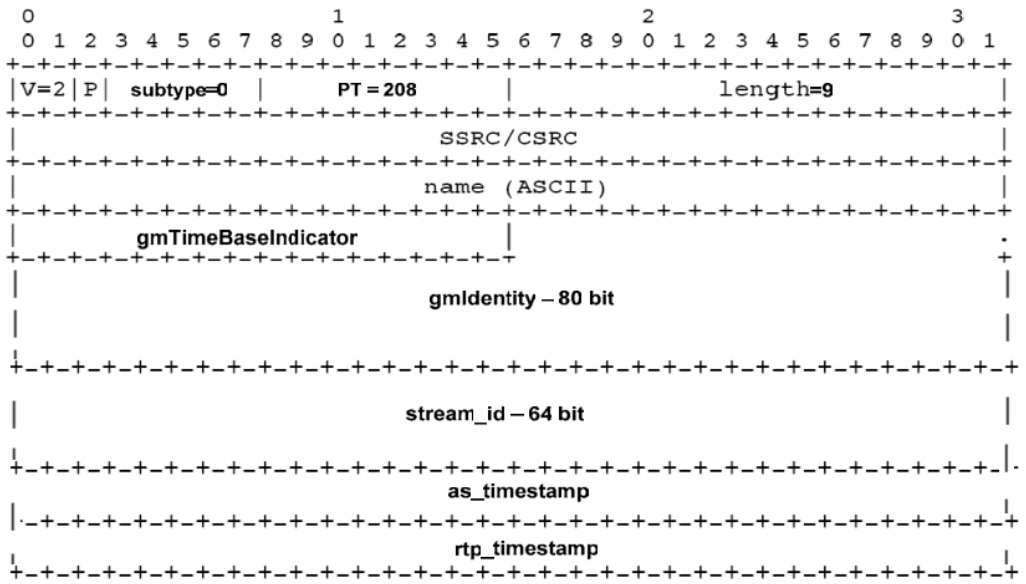

Figure 2—AVB RTCP packet format

8.1 V, P, SSRC/CSRC

Refer to section 6.4.1 of IETF RFC 3550 [B6] for description.

8.2 PT (8 bits)

This field value shall be 208 to identify the packet as an AVB RTCP packet.

8.3 Subtype (5 bits)

This field may be used as a subtype to allow a set of RTCP packets to be defined under one unique name, or for any application-dependent data. For this specification, **subtype** value shall be 0.

8.4 Length (16 bits)

This field contains the length of this RTCP packet in 32-bit words minus one, including the header and any padding. For this specification packet length value shall be 9.

8.5 Name (32 bits)

This field contains a name chosen to uniquely identify this set of RTCP packets with respect to other RTCP packets that the application might receive. For this specification, the use of this field is left to implementation.

8.6 gm TimeBaseIndicator (16 bits)

This field, together with the gmIdentity allows listeners and talkers to know that they are referencing the same time base on the same IEEE 802.1AS grand master. At each step change in time or frequency (at each discontinuity in phase or rate), the GM increments the gmTimeBaseIndicator and sends it to all IEEE 802.1AS clocks and also passes along the magnitude of the phase and/or frequency change associated with the discontinuity that it introduced into the IEEE 802.1AS domain. For example, it would not help to know the relation between the IEEE 802.1AS clock and the media clock at the talker and the relation between the listener's media clock and a different IEEE 802.1AS time if the relationship between the IEEE 802.1AS time bases were not known.

If the value of the gmTimeBaseIndicator and gmIdentity in the RTCP packet (i.e., of the talker) matches the gmTimeBaseIndicator and gmIdentity of the IEEE 802.1AS clock of the listener, the listener can be assured that the IEEE 802.1AS reference on both talker and listener are coherent. If they do not match, actions performed by the listener are application dependent, but may include entering a holdover mode until a positive match is again achieved.

8.7 gmIdentity (80 bits)

The gmIdentity field is used to communicate the Grand Master currently being 'listened to' by the talker for purposes of relating media and IEEE 802.1AS time.

8.8 stream_id (IEEE 802.1Q stream identifier) (64 bits)

The stream ID field is the same field as specified in Clause 7. This field shall always contain a valid Stream ID as per Clause 7.

8.9 as_timestamp (32 bits)

The 32 bit as_timestamp field shall contain the IEEE 802.1AS time (relative to the gmIdentity and gmTimebaseIndicator fields in this packet) which corresponds to the rtp_timestamp field of this AVB RTCP packet. The as_timestamp shall have units of nanoseconds and a maximum value of 2^{32}–1 nanoseconds, or just over 4 s. A renderer shall present the content (e.g. audio and/or video) associated with the rtp_timestamp field to the application media interface at the IEEE 802.1AS time indicated by this field. To facilitate rapid initial adjustment of the renderer's media clock it is recommended that the server send two AVB RTCP packets immediately following the first two RTP packets of the stream.

8.10 rtp_timestamp field (32 bits)

The 32 bit rtp_timestamp is used for intra- and inter-media synchronization for sources and renderers whose IEEE 802.1AS timebases are synchronized, and is used by receivers, together with the as_timestamp

field(s), to compute the actual media clock frequency. The value of this field shall be equal to the timestamp field in the header of a previously transmitted RTP packet.

9. Translator rules for AVB RTCP packet

A translator is an intermediate system that forwards RTP packets with their synchronization source identifier (SSRC) intact. The translator has different rules for handling different types of RTCP packets. A translator would be required to route AVB packets between subnets.

For the RTCP AVB packet type the translator shall forward the packet unchanged.

10. Different payload format support

All existing and future media formats conforming to the RTP framework are expected to work without modification.

Annex A

(informative)

Bibliography

[B1] IEEE Std 1588™-2008, IEEE Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control Systems.

[B2] IEEE Std 802.1D™-2004, IEEE Standard for Local and metropolitan area networks — Media Access Control (MAC) Bridges.

[B3] IEEE Standard Dictionary: Glossary of Terms & Definitions.

[B4] IETF RFC 2205 (September 1997)—Resource ReSerVation Protocol (RSVP)—Version 1 Functional Specification, Braden, R., Zhang, L., Berson, S., Herzog, S. and Jamin, S.

[B5] IETF RFC 2326 (April 1998), Real Time Streaming Protocol, Schulzrinne, H., Rao, A. and Lanphier, R.

[B6] IETF RFC 3550 (July 2003), RTP: A Transport Protocol for Real-Time Applications, Schulzrinne H., Casner S., Frederick R. and Jacobson V.

[B7] IETF RFC 3551 (July 2003), RTP Profile for Audio and Video Conference with Minimal Control, Schulzrinne, H. and Casner, S.