

**Draft Recommendation for
Space Data System Standards**

**SPECIFICATION FOR
RTP AS TRANSPORT
FOR AUDIO AND VIDEO
OVER DTN**

DRAFT RECOMMENDED STANDARD

CCSDS 766.3-R-1

RED BOOK
December 2019



CCSDS

The Consultative Committee for Space Data Systems

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FOREWORD

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DRAFT CCSDS RECOMMENDED STANDARD FOR REAL-TIME TRANSPORT PROTOCOL OVER DELAY TOLERANT NETWORKING FOR VIDEO APPLICATIONS

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PREFACE

This document is a draft CCSDS Recommended Standard. Its 'Red Book' status indicates that the CCSDS believes the document to be technically mature and has released it for formal review by appropriate technical organizations. As such, its technical contents are not stable, and several iterations of it may occur in response to comments received during the review process.

Implementers are cautioned **not** to fabricate any final equipment in accordance with this document's technical content.

Recipients of this draft are invited to submit, with their comments, notification of any relevant patent rights of which they are aware and to provide supporting documentation.

DOCUMENT CONTROL

Document	Title	Date	Status
CCSDS 766.3-R-1	Specification for RTP as Transport for Audio and Video over DTN, Draft Recommended Standard, Issue 1	December 2019	Current draft

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1 INTRODUCTION

1.1 OVERVIEW

Motion Imagery/Video Transmission over Delay/Disruption Tolerant Networking (DTN) is not simple because of the nature of video over IP, which generally requires consistent data flow into a video decoder. Up to now, the use of MPEG Transport Stream (MPEG-TS) has been the default standard to encapsulate and format video transmission over IP. However, MPEG-TS does not work well in situations with frequent interruptions or excessive latency and is not very flexible with regard to packet size. Deutsches Zentrum für Luft- und Raumfahrt (DLR) has been testing Real-time Transport Protocol (RTP) (reference [1]) over DTN for video and has found it is much more forgiving of interruptions and long latencies in the network. The fact that RTP packet size is arbitrary makes it much more flexible for video over DTN.

1.2 PURPOSE AND SCOPE

This document provides an overview and proposed methods for transmission of video over DTN using RTP. As more deep space missions will be utilizing DTN for data transmission and the use of video becomes even more prevalent, a standard method of formatting encoded video streams for transmission is required. Standardization of video transmission methodology makes it easier to design DTN nodes and networks. It also helps ensure higher reliability and quality of video transmission.

1.3 APPLICABILITY

This standard is intended for all future missions that produce, consume, or distribute video via the Bundle Protocol. The details (formats, resolutions, bitrates, etc.) of the video to be transmitted are largely omitted in order to prevent immediate obsolescence.

1.4 RATIONALE

This book is written with the assumption that future missions will rely upon a variety of cameras and imaging sensors. The encoding of these videos may be distributed throughout a spacecraft and/or mission, in which the only common thread will be the usage of IP. Following the current trends in the media industry, the various encoders are foreseen to use RTP and/or a format that can be encapsulated into RTP. Because RTP has been shown to be better suited to video over DTN, standardizing a video transmission methodology around current industry trends is an advantage to system designers.

1.5 NOMENCLATURE

1.5.1 NORMATIVE TEXT

The following conventions apply for the normative specifications in this Recommended Standard:

- a) the words ‘shall’ and ‘must’ imply a binding and verifiable specification;
- b) the word ‘should’ implies an optional, but desirable, specification;
- c) the word ‘may’ implies an optional specification;
- d) the words ‘is’, ‘are’, and ‘will’ imply statements of fact.

NOTE – These conventions do not imply constraints on diction in text that is clearly informative in nature.

1.5.2 INFORMATIVE TEXT

In the normative sections of this document, informative text is set off from the normative specifications either in notes or under one of the following subsection headings:

- Overview;
- Background;
- Rationale;
- Discussion.

1.6 REFERENCES

The following publications contain provisions which, through reference in this text, constitute provisions of this document. At the time of publication, the editions indicated were valid. All publications are subject to revision, and users of this document are encouraged to investigate the possibility of applying the most recent editions of the publications indicated below. The CCSDS Secretariat maintains a register of currently valid CCSDS publications.

- [1] H. Schulzrinne, et al. *RTP: A Transport Protocol for Real-Time Applications*. STD 64. Reston, Virginia: ISOC, July 2003.
- [2] *CCSDS Bundle Protocol Specification*. Issue 1. Recommendation for Space Data System Standards (Blue Book), CCSDS 734.2-B-1. Washington, D.C.: CCSDS, September 2015.
- [3] M. Handley, V. Jacobson, and C. Perkins. *SDP: Session Description Protocol*. RFC 4566. Reston, Virginia: ISOC, July 2006.

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OVER DELAY TOLERANT NETWORKING FOR VIDEO APPLICATIONS

- [4] M. Handley, C. Perkins, and E. Whelan. *Session Announcement Protocol*. Experimental. RFC 2974. Reston, Virginia: ISOC, October 2000.
- [5] S. Burleigh. *Compressed Bundle Header Encoding (CBHE)*. RFC 6260. Reston, Virginia: ISOC, May 2011.

2 OVERVIEW

2.1 GENERAL

In the past two decades, crewed spaceflight has undergone many changes. What was once unthinkable is now typical, throughout various components of spacecraft programs. Within spacecraft video systems, the embrace of Commercial Off-The-Shelf (COTS) video equipment, such as action cameras and handheld devices, is widespread. As a result, the once-centralized nature of video transmission, in which all cameras transmitted to a single encoder and transmission system, has been forsaken for a distributed system, in which some cameras are encoded by a centralized system, while others have built-in encoders. Furthermore, the variety of ways video can be transmitted has increased; cameras may use wired networks, uncompressed video mechanisms such as Serial Data Interface (SDI), wireless networks, or point-to-point links. The Real-time Transport Protocol has been used as a single consistent factor for this variety of video transmission opportunities as it is robust to jitter, well-defined within the IETF, easy to implement, and extendable. In the modern broadcast world, everything from uncompressed video (SDI) to highly compressed video (H.265) may be transmitted via RTP, over IP and non-IP networks.

This paradigm shift is not unique to the video transmission world; spacecraft transmission has suffered the same fate. A single spacecraft may now have a variety of distinct up- and down-link mechanisms, as well as a separate modicum of onboard connectivity options for payloads and visiting vehicles, such as wireless networks based upon 802.11, Proximity-1, etc. The Bundle Protocol has been designed to unify this conglomeration by providing an overlay network for heterogeneous networks, in which the concepts of delay and disruption tolerance were integrated from conception.

This standard addresses the following:

- encapsulation of RTP data on space links;
- RTP performance tuning for BP-based space links;
- addressing of RTP data, in order to facilitate multiple-sender-multiple-receiver scenarios;
- interoperability of standards.

2.2 ASSUMPTIONS

It is assumed that future deep-space networks will rely upon DTN and the Bundle Protocol for their communication, while using RTP for video transmission. Therefore it is pertinent that best-practices for the union of the two be well-defined prior to any future missions.

It is assumed that a Bundle Protocol implementation that is compliant with the CCSDS Bundle Protocol standard (reference [2]) will be used for the underlying transport for RTP data. However, while the utilization of Bundle Protocol multicast is foreseen for many

complex video networks, no standard exists for bundle multicast, other than that outlined in reference [C1]. Therefore this book makes the following assumptions about the underlying Bundle Protocol multicast mechanism, if it is in use:

- a) arbitrary addressing—each multicast-based stream must be uniquely identifiable by a tuple of (address, service number), similar to the (multicast, port) tuple used for IP-based multicast;
- b) arbitrary bundle sizes—the multicast mechanism must not induce arbitrary limits that are less than the maximum bundle size on the size of bundles.

2.3 FLOW DIAGRAM—RTP IN LARGER PICTURE

2.3.1 GENERAL

RTP provides a robust and mature infrastructure for the distribution of video within complex missions, while allowing for a relatively simple interface with onboard and ground hardware. Furthermore, RTP can be interfaced with legacy MPEG-TS-based hardware and software without altering the payload data (e.g., video data). The remainder of this section will provide a theoretical implementation of an onboard RTP-based video system for a partially crewed deep space mission before progressing to its associated ground segment.

The remainder of this section will solely focus on real-time video; as archived video acquired via LOS would also be encapsulated in RTP, the exact details of reordering and replay are outside the scope of this book.

2.3.2 ONBOARD IMPLEMENTATION

Given the operational experiences gained from the International Space Station, it is imperative that the video system of a next-generation spacecraft/mission be designed with extensive flexibility, allowing for new commercial cameras and video sources to be quickly added. Therefore this section envisions a ‘partitioned’ video system, which is, whenever possible, based entirely on IP. The basic concept of a partitioned video system is that while some core functionality (such as cameras and encoders) is inherent to the spacecraft and can be considered to be critical spacecraft data, other sources are transient and/or best-effort (scientific cameras, PR functionality, etc.). Furthermore, each spacecraft and rover within the mission may have its own subset of critical video sources. If it is considered that multiple spacecraft or components thereof may dock or otherwise be attached to each other via high-bandwidth connections, it is logical to design a video system that may use the encoders of one spacecraft to encode or transmit video from another. Besides the core encoding functionality, it must be expected that payloads may contain specialized cameras or video sources. Depending upon the location of the payload, it may be optimal to encode the video in the camera, while in other cases, that video should be transmitted to the core encoders.

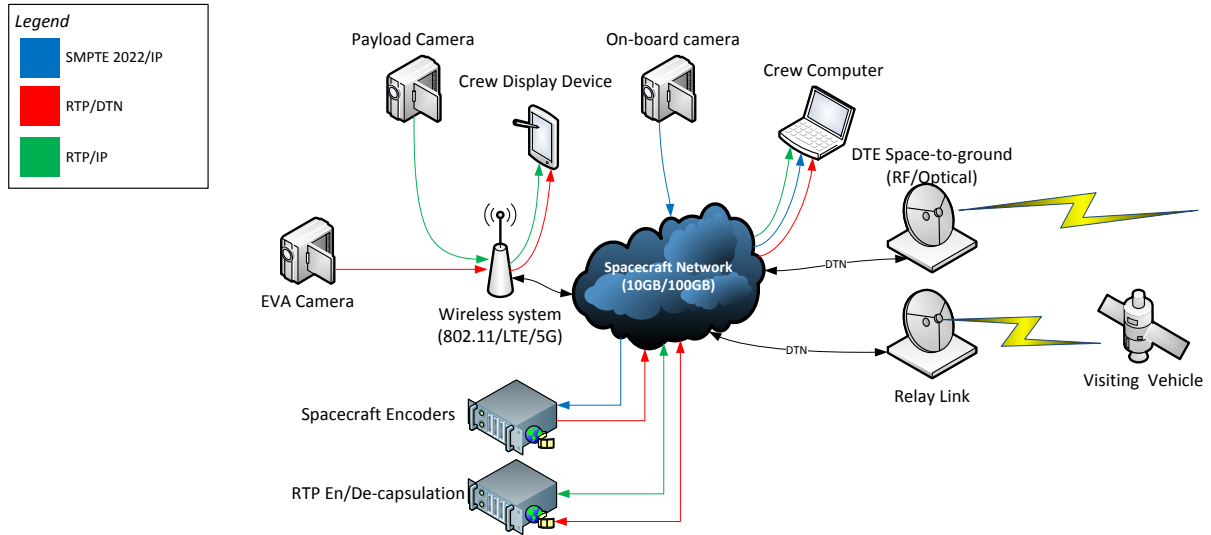


Figure 2-1: Flow Diagram—Onboard Assets

Such a system may be designed entirely around RTP as an encapsulation format; while the best-known uses of RTP are as conveyance for compressed video data, it is also possible to convey uncompressed (or lightly compressed) video data via RTP, using SMPTE-2022.¹ Figure 2-1 shows an example of such a system, although redundant links have been omitted for clarity. This design allows for extreme flexibility and for increased redundancy, in that the failure of individual encoders does not affect the capability to encode video. If additional redundancy is required for individual cameras, multiple network links may be furnished for them. However, it must be noted that uncompressed (SMPTE-2022) video requires extremely high bandwidths, on the order of gigabits/second; therefore the mission designer must carefully design the network to ensure adequate capacity.

Additional assets, such as EVA cameras, which are provided with their own encoders, output compressed video streams via RTP. More interestingly, onboard system cameras may not contain their own encoder; instead, they output SDI via RTP (as permitted by SMPTE-2022), which is encoded by the ‘spacecraft encoders’, which can be positioned in a less extreme environment, such as inside the spacecraft. These devices can also be largely commercial, except for the DTN output functionality. Some payload cameras may also be commercial devices with integrated compressors, which can output RTP over IP, for encapsulation within the avionics system. This diagram also showcases a visiting vehicle link, which may also contain compressed RTP-formatted video, which may be viewed on the crew display. Finally, the space-to-ground link, based upon DTN, is utilized to transport RTP-encapsulated video.

¹ ST 2022-6:2012 Transport of High Bit Rate Media Signals over IP Networks.

2.3.3 GROUND SEGMENT

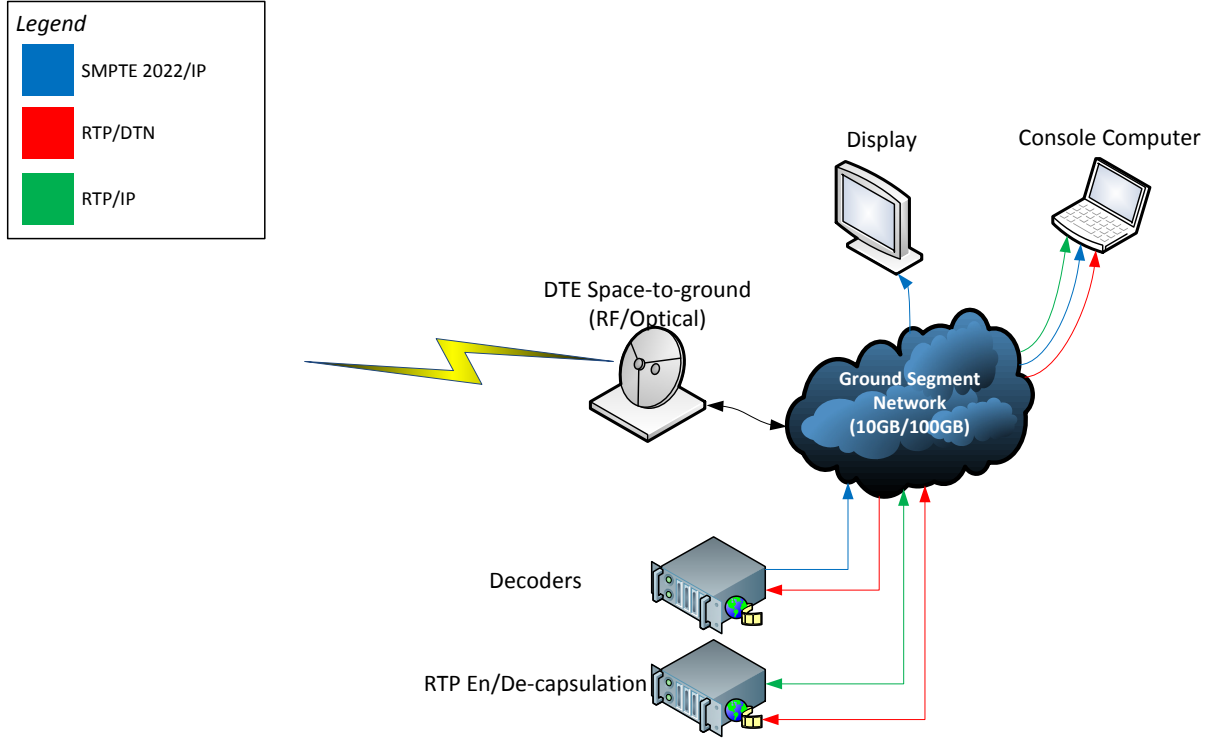


Figure 2-2: Flow Diagram—Ground Segment

Because of the homogeneity provided by RTP, future ground segments may be simplified, thanks to both the usage of commercial software and hardware and the reduced number of interfaces required for reception. An example ground segment may be found in figure 2-2, in which the acute reader may notice many similarities to the onboard implementation. This figure omits some components, such as recording and archiving, although a proposed solution will be discussed in this section.

In the proposed ground segment, DTN-encapsulated RTP data containing compressed video is received from the Earth terminal of the space-to-ground link. This data is sent to the RTP encapsulation/decapsulation unit, which removes the Bundle Protocol headers and transmits the RTP data to the ground segment network. The RTP decapsulation is the only custom component of the system; all further data distribution may be accomplished with COTS hardware and software.

Individual console positions may use a COTS IPTV package (such as VLC Media Player) to select from various channels for display on their individual machines. Decoders may optionally be used, allowing the RTP streams to be decoded to SMPTE-2022. Displays without onboard decoding may be provided with uncompressed data via SMPTE-2022, using COTS IP/SDI converters. External users may make a VPN tunnel to the IPTV package (which, in some cases, also provides a Web site for channel selection), allowing them to select individual channels for display.

Finally, recording and archiving may be provided by the options from the same COTS IPTV packages, which capture RTP data from the network and encapsulate it into files, allowing for rapid archiving and subsequent exchange of video data.

2.4 TECHNICAL OVERVIEW

2.4.1 GENERAL

RTP (reference [1]) provides a lightweight packet format for the transmission of media-related payloads over IP-based networks and is a fundamental component of many higher-level protocols, such as the Session Initiation Protocol (SIP) (reference [C3]), used in IP telephony. The remainder of this section serves as a brief overview of RTP; it is not to be misconstrued as an attempt at a complete description of the protocol, which may be found in (reference [1]).

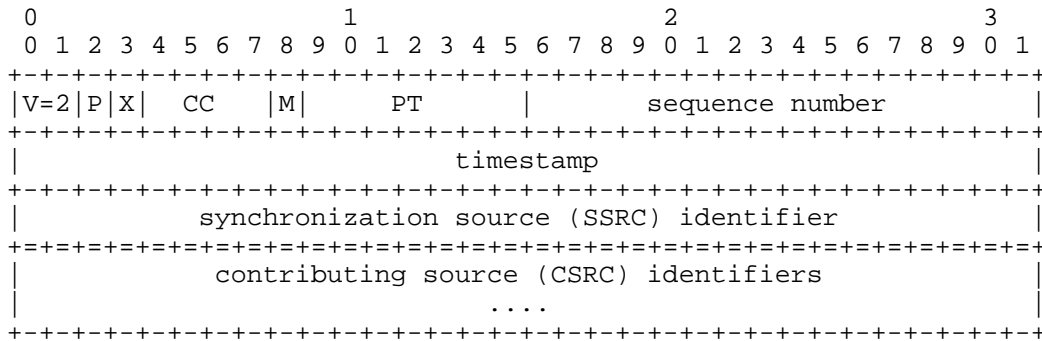


Figure 2-3: RTP Packet Header

RTP is designed around a fixed 12-byte header, along with a variable-length header extension field, as can be seen in figure 2-3. Among other things, this header contains the Payload Type (PT) component, which specifies the type of data conveyed in this packet. Additionally, the header contains a timestamp and sequence counters, both of which will be discussed further. The last element to be aware of is the marker bit, which specifies that this packet contains ‘important data’, although the definition of importance is left to the payload type. Figure 2-4 provides an overview of the various IETF RFCs that describe RTP.

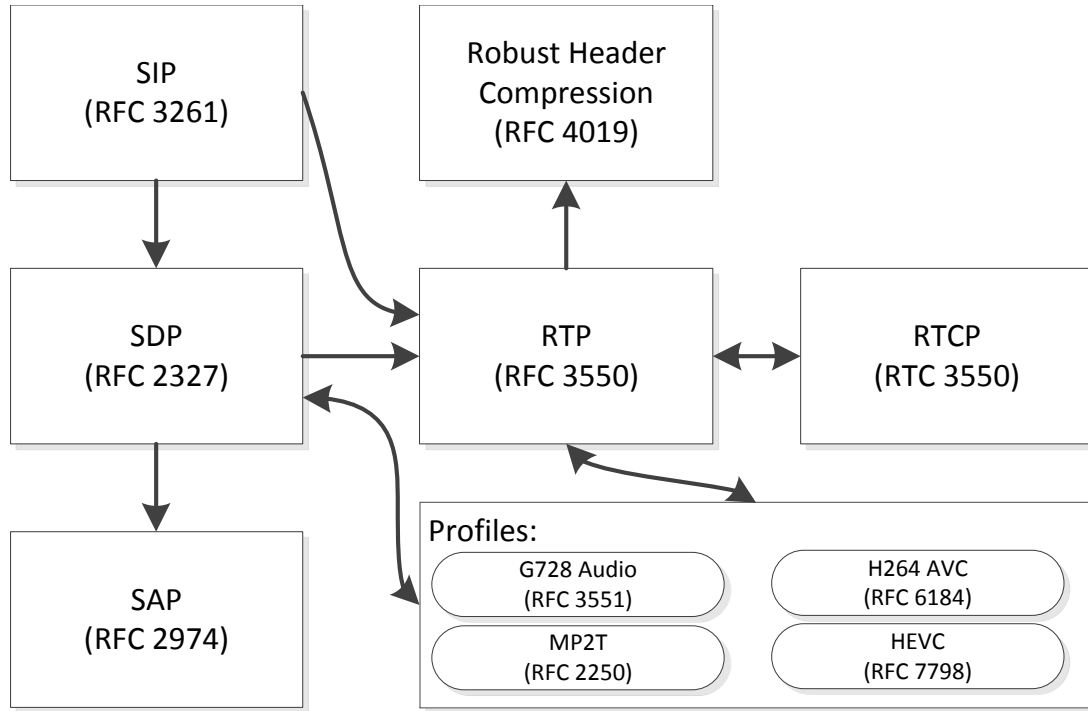


Figure 2-4: RTP Relationships

It is important to note that the RTP packet does not encode the size of the payload data, instead leaving that task to the underlying transport protocols. Furthermore, in order to prevent potential confusion, unless specifically mentioned, the term ‘payload’ refers to the payload of the RTP packet, as opposed to the bundle payload.

2.4.2 RTP PROGRAM STRUCTURE

2.4.2.1 General

RTP profiles, with the exception of the RTP MPEG-TS profile, stipulate that each RTP stream contains a single type of data, such as audio or video. Therefore a single RTP program may comprise multiple RTP streams, each of which exists on their own channel. In an IP/multicast-based network, it is the responsibility of the receiver to subscribe to all relevant multicasts. This is a flexible mechanism that allows for robust programming; for example, a single RTP-based video encoder could transmit two unique video streams at different bitrates along with two audio streams in different languages. Based on the out-of-band signaling information, which is discussed further throughout section 3, the receiver may select the video stream that is best suited to the available bandwidth as well as the preferred audio language.

This model is also applicable to space-based RTP networks, albeit with some caveats. It is foreseen that a spacecraft network may be heterogeneous, with portions built around standard IP networking, while other segments may be built around DTN, as shown in figure 2-5. As with standard IP-based networks, data that is destined for a DTN network may be transmitted

in one of two ways: unicast or multicast. If more than two DTN endpoints have been factored into the mission design, BP Multicast (reference [C1]) may be used for transmission over space networks.

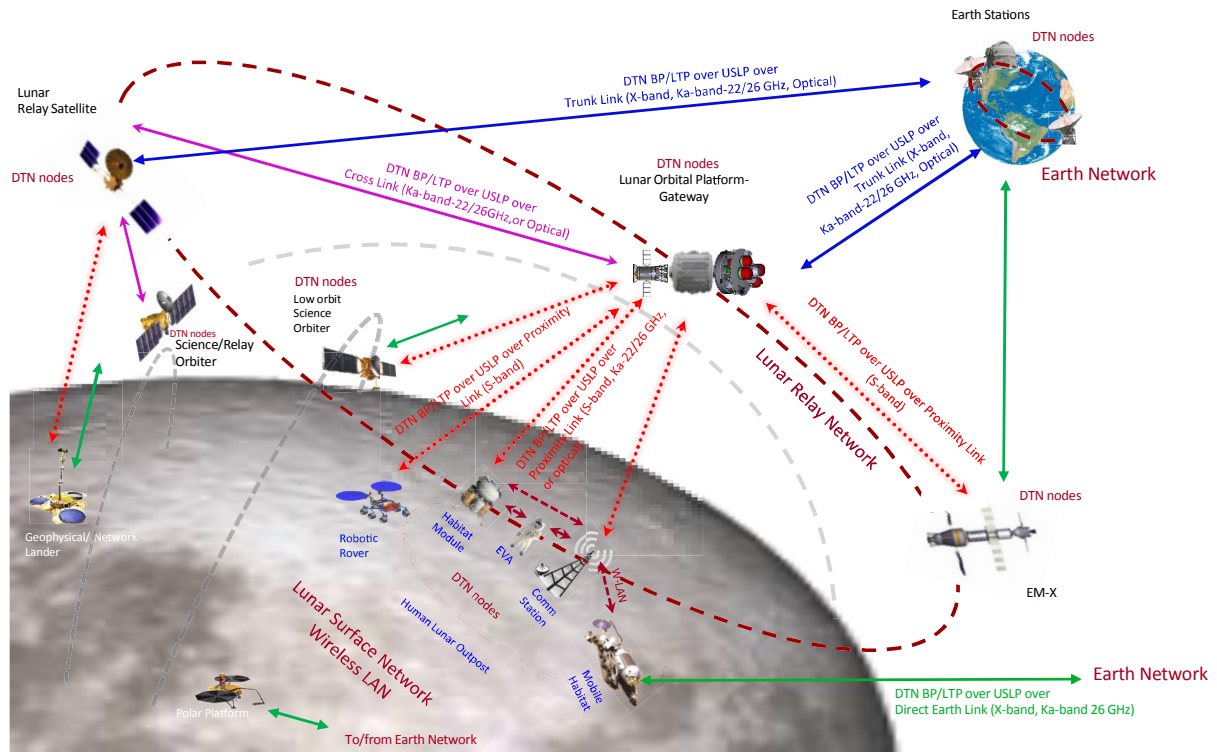


Figure 2-5: Spacecraft Network Design

2.4.2.2 RTP Concatenation Mechanisms

In order to facilitate variable Maximum Transmission Units (MTUs) in a single transmission path, it is possible to concatenate RTP packets. This is not to be confused with the concatenation function provided by the H.264/265 Network Abstraction Layer (NAL). This concatenation function is applicable to data that is encapsulated within NAL transports and is relatively complex. It is not recommended to do NAL concatenation within the encoded video data stream with additional concatenation being performed at the Network Layer.

The timestamp contained within the RTP header is critical for the decoding of data, but may be constant across multiple packets. The presence of a non-incrementing timestamp across multiple packets indicates that the data from all packets must be decoded at once. Therefore the timestamp must be constant across all packets to be concatenated. Furthermore, when concatenating RTP data, care must be taken to avoid the loss of any data contained within the RTP header, as well as to prevent the interleaving of important and non-important data.

When concatenating RTP packets for DTN transport, a single RTP header is used followed by a variable-length blob of data. The transmitter is responsible for the concatenation, while the receiver must recreate the individual RTP packets, prudent to the maximum applicable

MTU for the receiving IP network. This is a similar approach to that which is taken by Robust Header Compression (RFC 4019), but is not identical. RFC 4019 relies upon packet identifiers and/or sequence numbers, while DTN-based RTP concatenation relies upon the sequence numbers and timestamps from within the Bundle Protocol.

2.4.3 RTP SYNCHRONIZATION PRIMITIVES

When compared to terrestrial RTP-based video systems, space-based networks are subject to unique difficulties with regards to the synchronization of video streams. Mission teams may require synchronization between received video and other sources of data, such as audio (either on camera or from a voice system) or telemetry. Therefore some care must be taken in order to ensure reliable synchronization of audio and video.

RTP timestamps are based upon the sampling frequency of the encoded data and are intended to be independent of each other. For example, the sampling frequency of most video codecs is 90 kHz, while the sampling frequency of telephone-quality audio is 8 kHz. Therefore even if these two programs are intended to be viewed simultaneously, their respective timestamps will increment by different quantities and at different frequencies. Furthermore, the timestamp may remain constant across multiple packets. RTP provides synchronization primitives that are applicable to streams sourced by a single encoder and/or multiple encoders and provides sufficient fidelity to achieve ‘wall-clock’ synchronization.

Inter-program synchronization may be accomplished via several mechanisms, each of which is covered in one or more IETF RFCs. The primary mechanism specified by the RTP standard is via the usage of Real-Time Control Protocol (RTCP) messages. RTCP, described in section 6 of reference [1] provides monitoring and control functionality for RTP streams. In an IP network, RTCP data for a given stream is provided on the next odd-numbered port. For example, an RTP stream that exists at 224.0.0.1:4220 would have an accompanying RTCP stream on port 4221. RTP and RTCP share similar packet structures, as shown in figure 2-6. However, an RTCP packet must be aligned on a 32-bit boundary.

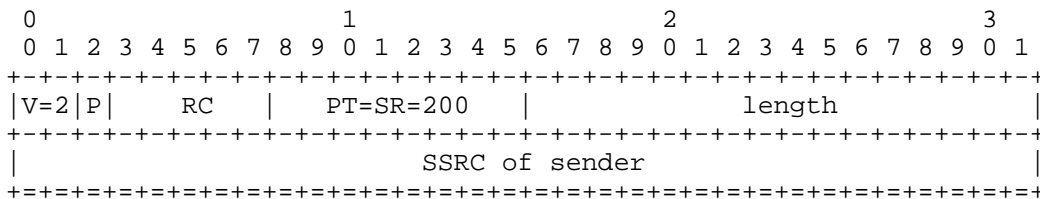


Figure 2-6: RTCP Overview

RTCP has several defined message types, each of which is intended to accomplish different tasks, and will not be described here. For synchronization, the RTCP Sender Report (SR) must be used. The Sender Report, described in section 6.4.1 of reference [1] and shown in figure 2-7 is intended to be transmitted by all RTP senders or multiplexers, and contains all information required to synchronize RTP streams.

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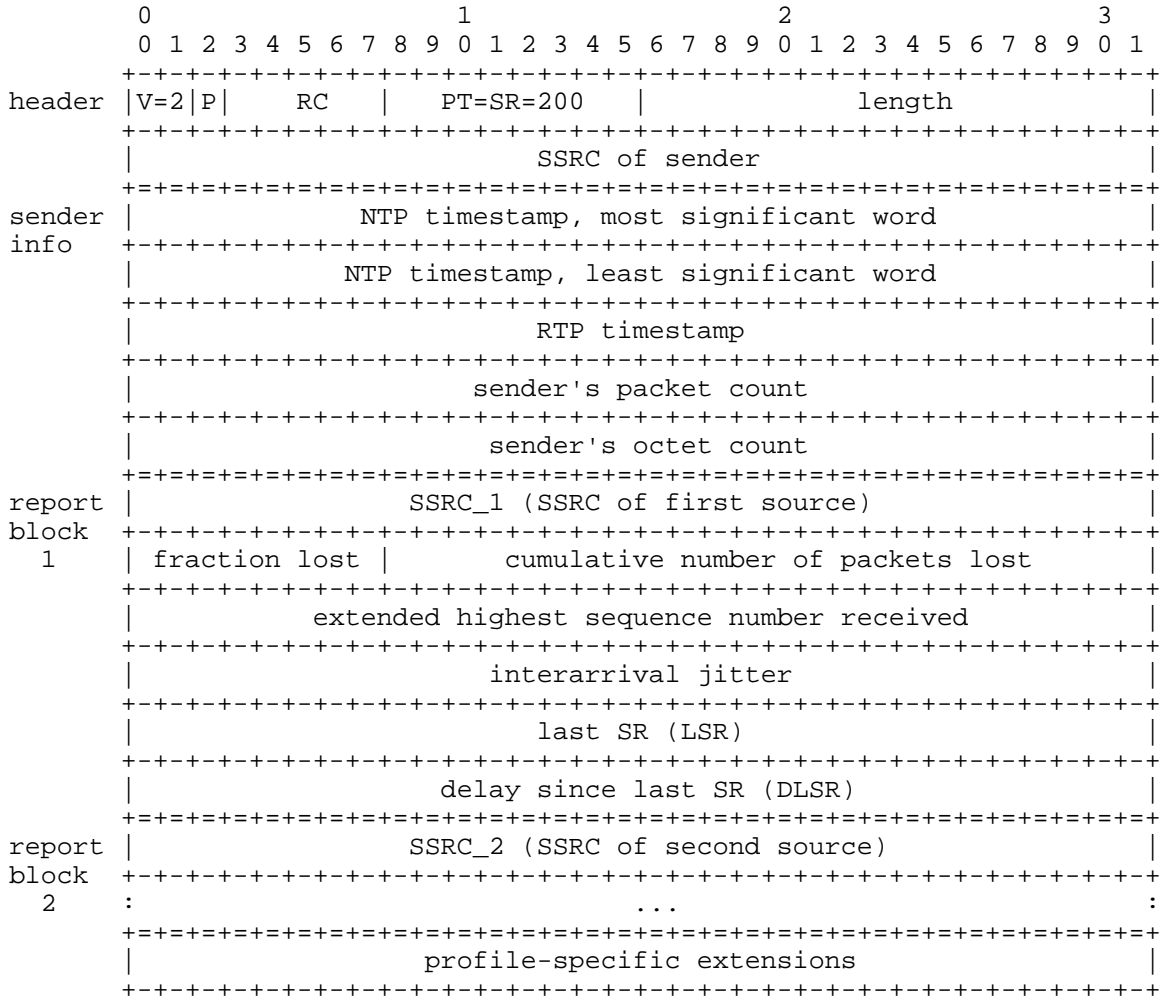


Figure 2-7: RTCP Sender Report Message

The Sender Report message is divided into one sender block and one or more reporting blocks. Encoders and processors that generate this block may omit the reporting blocks, and they may be stripped by re-multiplexers. The sender block contains the NTP timestamp at the time of generation as well as the RTP timestamp for that media source, valid at the time of SR transmission. Receivers can use that information, along with the sampling frequency of the RTP stream, to determine the offset from real time. By receiving multiple sender reports for all components of the RTP stream, the streams may be coordinated.

2.4.4 STREAM SIGNALING

2.4.4.1 General

RTP focuses upon the transmission of media and does not contain media description functionality as is present in the MPEG-TS standard. Therefore external protocols must be used to ensure that all parties are aware of the available video streams. The remainder of this section serves to describe the primary media description protocols that are in use in the RTP ecosystem.

In a full-featured implementation of RTP-over-DTN, it is assumed that there may be several signaling channels used for synchronization and discovery of various RTP programs. In order to simplify the transmission of these streams, all signaling may follow the basic rules outlined in 3.6, regardless of the content of the signaling channel, and outlined in the following subsections.

2.4.4.2 SDP

2.4.4.2.1 General

The Session Description Protocol (SDP) (reference [3]) is a versatile text-based format for the description of media streams. It provides a mechanism for the exchange of information that is required by a user or decoder, such as the multicast port and addresses for various stream types, as well as the start and end times, names, etc. SDP messages are designed to be relatively small, allowing for their conveyance in a multicast packet, file, or other mechanism (such as DLNA or another service-discovery mechanism).

An SDP description is formatted as a set of key=value tokens delimited by newline characters. Each key must be one-letter, while each value may be an indeterminate length. Per the specification, the key and value must be separated with an equal sign ('=') without spaces. The full list of keys is available in reference [3] and is not explained here. However, in order to describe the components of a single source uniquely, each component must be described by the connection information (c=) and the media name and transport address (m=). If the SDP file describes an IP-based stream, the connection parameter is typically a multicast address. The parameter is formatted as a 3-tuple of <nettype> <addrtype> <connection-address>.

If the connection parameter describes a multicast program, then *nettype* must be 'IN', *addrtype* must be 'IP4' or 'IP6' (depending upon the usage of IPv6), and the connection address must describe the message. If describing a IPv4 multicast stream, the connection address must be formatted as <base multicast>/<ttl>. Optionally, the address may be formatted as <base multicast>/[ttl]/<number of addresses>. If the number of addresses is provided, it is assumed that the data is encoded using a layered or hierarchical mechanism, and the network will prune unused addresses.

Within a DTN network, SDP occupies a similar position as in an IP-based network, acting as a mechanism to signal the presence, configuration, and location of program components. However, as the Bundle Protocol provides all necessary transport and validation mechanisms, SDP is transmitted ‘as is’, with no additional protocol overhead (e.g., SAP). Figure 2-8 shows an example of properly formatted SDP data, intended for transmission over the Bundle Protocol.

```

c=DTN BP IPN:1
m=video 2 RTP/AVP 96
a=rtpmap:96 H264/90000
a=fmtp:96 profile-level-id=640014;sprop-
parameter-
sets=Z2QAFKzZQ0R+f/zBfMMAQAAAA
wBAAAAKI8UKZYA=,aOvssiw=
    
```

Figure 2-8: Example of SDP Data

As the video and SDP data transits the DTN network, the SDP data must be created or altered by the transmitting node. Therefore the following requirements apply to DTN-transmitted SDP data.

The details of SDP addressing as used within a DTN network is further expanded in 3.6.2.

2.4.4.2.2 Transport of SDP Messages

SDP may be transported over an IP-based network within (among others) Session Announcement Protocol (SAP) (reference [4]) and the Real Time Streaming protocol (RTSP) (reference [C2]). It is up to the implementer to determine which transport mechanisms should be supported within the DTN transmitter and receiver.

2.4.4.2.3 SDP Translation Between Network Domains

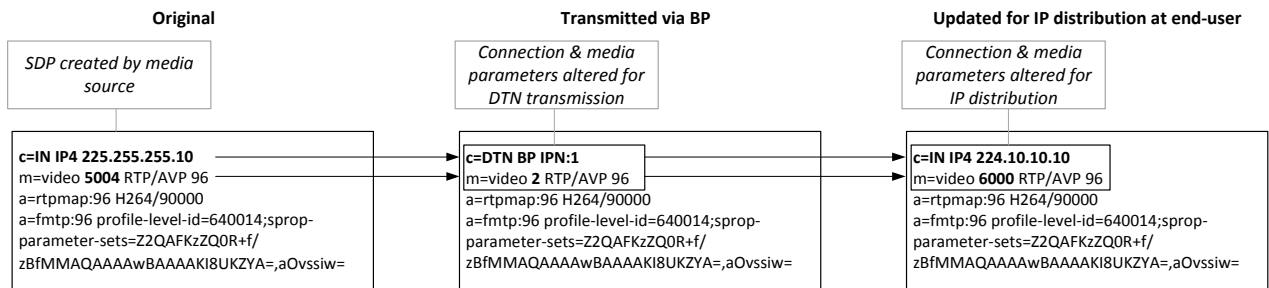


Figure 2-9: SDP Translation Between DTN and IP Domains

The SDP connection and media descriptor parameters are intended to identify the media source across networks. The addition of the DTN and BP parameter types extend SDP to identify media sources across RTP networks. Implementers may freely translate between IP and DTN environments by modifying the relevant portions of the SDP data, as shown in figure 2-9. An IP-to-DTN RTP converter may map the DTN EIDs containing media sources to multicast addresses and ports, allowing for a one-to-one modification of all connection information parameters to multicast addresses and media descriptors to ports within that particular address.

2.5 RTP OVER DTN

2.5.1 GENERAL

RTP may be encapsulated into DTN bundles with minimal modification, instead treating the entirety of the RTP packet as a single bundle, as outlined in 2.5.2. However, astute observation of the RTP packet format may reveal redundancy between the RTP packet and the BP primary header; namely, both provide a timestamp and sequence number. Implementers of RTP over DTN must accept this redundancy and not attempt to remove the RTP timestamp and sequence, as they are designed to accomplish differing goals; while the RTP timestamp is based upon the sample rate of the payload, the BP timestamp is a wall-clock timestamp, expressed in seconds. Furthermore, the timestamp for the RTP packet can be repeated for a media stream in order to allow for fragmentation of RTP data. RTP states that all the payloads of all packets for a single stream with a single timestamp must be presented to the decoder at the same moment.

2.5.2 ONE-TO-ONE ENCAPSULATION OF RTP IN DTN

The one-to-one encapsulation of RTP within the payload of a bundle may be accomplished without any modification of the RTP packet structure. This is the most basic possible implementation, but may not be efficient because of the fact that the average IP-based encoder uses an MTU of 1400 bytes, which is acceptable for IP networks. However, given that the Bundle Protocol and convergence layer adapter will add their own headers, this may create unnecessary amounts of overhead. If possible, the MTU should be set to a value that represents the largest reasonable MTU for a user network before fragmentation may occur. However, it is preferable that the RTP concatenation mechanism from 3.3 be used.

2.5.3 RTP-OVER-DTN TO IP

2.5.3.1 General

The normative section of this book specifies a robust set of guidelines that, taken in conjunction, allow for creation of an interoperable DTN-based video system. However, in many cases, the utility of such a system is limited by the lack of interfaces between standard IP and the DTN network. The remainder of this section provides an outline for an IP-to-DTN-to-IP-based video system.

Figure 2-10 shows a real-world example of an IP-to-DTN-to-IP as configured by DLR. ION 3.6.0B, as developed by JPL, was used as the DTN engine for his test. This configuration is in full compliance with section 3 of this book, as well as with references [1] and [3].

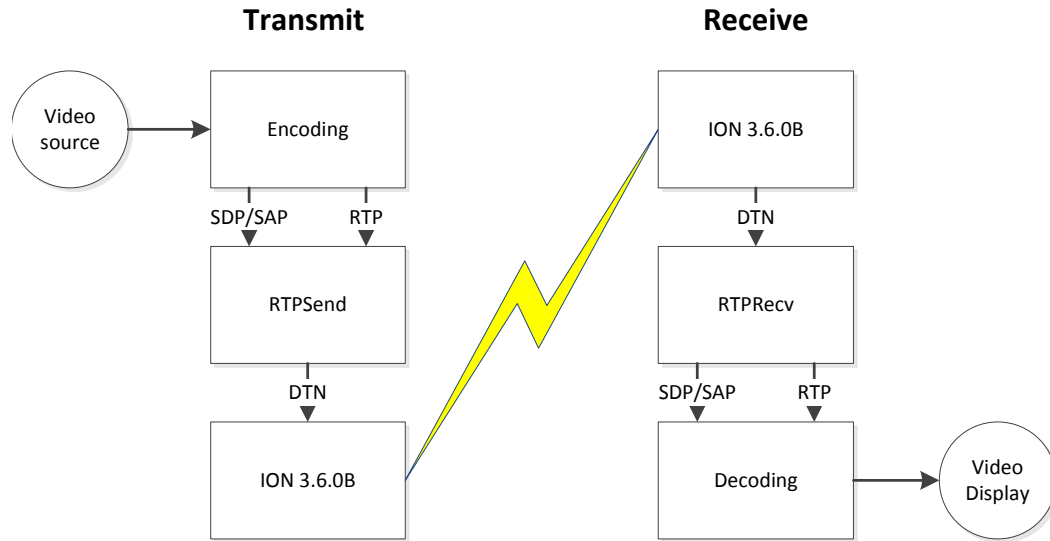


Figure 2-10: Theoretical IP-to-RTP System

2.5.3.2 IP-to-RTP

In order to provide an easily usable interface between IP and DTN, SAP (reference [4]) is used to enumerate video sources within a multicast network. The media streams announced from SAP are described by multicast IP:port tuples and are subscribed to by the application, which assigns each media source a unique DTN EID. The SDP data received in the SAP message is permuted in order to remap the EIDs, as described in 3.6.2.

The RTP concatenation process is run on every RTP stream and sent to the DTN network, as well as the SDP messages.

2.5.3.3 RTP-to-IP

On the DTN receiver, a process listens to the specific EID containing the SDP data. The process from 3.6.2 is followed in reverse; the media sources described in the SDP data are mapped to a user-defined multicast IP, with incrementing ports, starting from a user-defined port.

The application listens to all the EIDs found within the SDP data and waits for the reception of RTP data on those endpoint IDs. Once RTP data is received, the fragmentation process from 3.4 is performed, creating a range of RTP streams. Finally, the SDP data (with the new multicast IP addresses) is retransmitted as SAP messages.

2.5.3.4 Interoperability

As discussed in previous sections, the reception of a DTN-encapsulated RTP program may be envisioned as a set of concurrent operations, each of which acts upon different DTN EIDs:

- a) division and re-transmission of RTP packets;
- b) division and re-transmission of RTCP packets, if required;
- c) reception and retransmission of signaling data, if required.

In order to facilitate a flexible and interoperable structure for the receiving applications, it is recommended that data structures be made available for exchange between the various applications.

Name	From	Description
EID Map	Signaling	A list of all EIDs for this program, used for subscription to BP multicast groups.
SSRC->port Map	RTP Transmitter	A list of all SSRCs, along with the multicast ports that are being used for transmission.

2.6 LEGACY COMPATIBILITY

The progression from MPEG-TS to RTP has been a multi-decade process, supported by the vast resources of the broadcast industry. Therefore well-developed methods exist for legacy MPEG-TS support via RTP-based transports.

If the MPEG-TS RTP profile is used, it is the responsibility of the mission developer to announce the stream via SDP over the DTN network.

3 RTP IN SPACE-BASED LINKS

3.1 OVERVIEW

This section provides an implementer with requirements for the implementation of RTP over DTN. It is assumed that a Bundle Protocol implementation compliant with reference [2] is used.

3.2 RTP PDU FORMATTING

3.2.1 All bundles must use an RTP header that is compliant with reference [1].

3.2.2 The RTP header must start at byte 0 of the Bundle Protocol payload.

3.2.3 If stream signaling is used, all messages must be compliant with references [1], [3], and [4].

3.2.4 When transmitting RTP data over the Bundle Protocol, implementers must avoid the use of RTP fragmentation, instead relying upon BP fragmentation mechanisms.

3.2.5 If an RTP bundle is generated from an external source, the RTP concatenation mechanism from 3.3 must be used.

3.2.6 If RTP packets must traverse from Bundle-Protocol- to non-Bundle-Protocol-based networks, those packets must be decapsulated from the bundle payload. If required, fragmentation may be implemented, as outlined in 3.4.

3.3 RTP CONCATENATION

3.3.1 While some video formats may support aggregation, these mechanisms must not be used in place of the RTP concatenation process.

NOTE – H.264 and 265 support a form of packet aggregation. However, this mechanism involves the modification of data within the RTP PDU payload section. As a result, the RTP concatenator process would require a deep insight into the packet content. Additionally, it is assumed that such processes may be used by encoders and/or other non-DTN-aware parts of the pipeline, and such functionality should not be repeated.

3.3.2 The padding bit of the RTP PDU header may be set; if set, RTP concatenation must not be attempted.

3.3.3 If secure RTP is used, RTP concatenation must not be attempted.

3.3.4 All concatenated packets must have the same RTP timestamp. Therefore concatenation shall be triggered based on a monotonic increase of the RTP timestamp.

Optionally, the RTP sequence counter can be used in order to verify that there are no gaps prior to transmission of the concatenated data.

3.3.5 The marker bit for all packets within a single concatenated bundle must be set to the same value. If the marker changes (regardless of all other packet similarities), the previously concatenated bundle must be transmitted, and concatenation of a new bundle shall start with that marked packet.

3.3.6 The state of the eXtension (X) bit must remain constant for all concatenated packets in a single bundle. If the X bit is set, the extension data must remain constant across all packets. If the X bit is toggled or the extension data changes, then the bundle must be transmitted, and concatenation of a new bundle shall start with the changed packet.

3.3.7 All RTP packets destined for concatenation in a single bundle must be part of the same media stream, as defined by the payload type as well as the RTP SSRC and/or multicast port. This extends the best practices established in section 5.2 of reference [1].

3.3.8 The SSRC identifier must not change through the managed transmission path.

3.3.9 Subject to 3.3.4, if any RTP packets destined for concatenation have set the padding bit, the padding must be removed and the bit must be unset.

3.3.10 The sequence counter must increase by one for each outgoing concatenated RTP packet.

3.3.11 Bundles containing RTP packets must be transmitted as soon as concatenation is completed. Bitrate smoothing shall not be attempted.

NOTE – Figure 3-1 shows an example of a concatenated RTP bundle. The state of the marker bit is shown, while the sequence counters and timestamps are omitted.

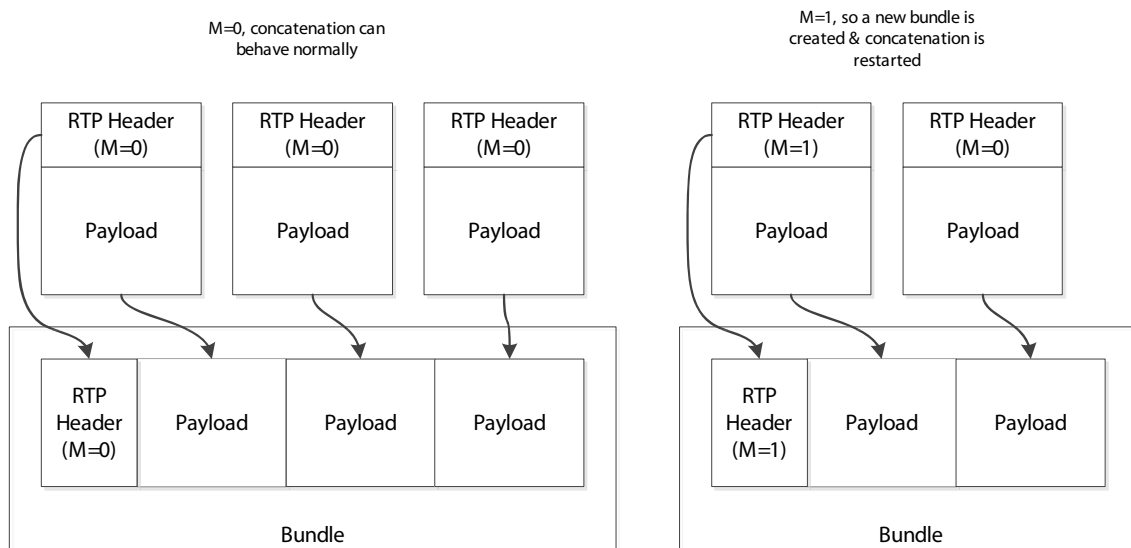


Figure 3-1: RTP Concatenation—Overview

3.4 BP-TO-NON-BP FRAGMENTATION

NOTE – This subsection refers primarily to the refragmentation of DTN-encapsulated RTP data destined for non-BP networks. On the receiver, some basic rules need to be followed. These rules also need to be modified depending on the technology in use, and are expanded in following paragraphs.

3.4.1 All incoming RTP payload data must be fragmented into acceptable segments, based on the underlying output network MTU, which may be set via management or (for IP-based networks) via Network Path MTU Discovery.

NOTE – Any retransmission of the bundle or bundle segments is expected to occur at lower levels. Therefore it is expected that fragmentation will only occur upon presentation of the complete bundle.

3.4.2 Each segment must be prepended with a RTP header and transmitted. With the exception of the sequence count, the RTP header of the outgoing packet must contain all data that was transmitted within the incoming RTP packet.

3.4.3 The sequence number of the outputted RTP packets shall be created independently from the input sequence number.

3.5 DTN TRANSMISSION OF RTP PACKETS

3.5.1 Each RTP source that is a component of one or more programs shall be transmitted on an individual EID.

3.5.2 If a bundle size limit is required, it shall be decided by the implementer.

3.6 SIGNALING DATA (SDP DESCRIPTION AND RTCP)

3.6.1 GENERAL

3.6.1.1 All signaling messages shall be transmitted with custody transfer enabled.

3.6.1.2 All signaling messages shall be transmitted at least once every 30 seconds.

3.6.1.3 Each type of service message shall have its own unique EID.

3.6.1.4 If bundle multicast is used, multiple programs may transmit their service messages on the same EID.

3.6.2 SDP

3.6.2.1 Unless the MPEG-TS RTP profile is used, the stream must be announced via SDP over the DTN network.

3.6.2.2 For all connection information parameters (c=), the *nettype* shall be set to 'DTN'.

3.6.2.3 For all connection information parameters (c=), the *addrtype* shall be set to 'BP'.

3.6.2.4 For all connection information parameters (c=), the address field shall be the URI representation of the CBHE node number (reference [5]) where the stream or service can be found. The scheme name shall be included, while the service number and proceeding period ('.') character must be omitted.

3.6.2.5 The address field of the connection parameter must specify a single DTN endpoint ID.

3.6.2.6 For each media descriptor (m=) in the SDP data, the *port* element must be changed to the service number of the BP EID that represents that media element.

NOTE – The receiving node needs to be able to determine unambiguously the EID of each media component by combining the address field of the connection information parameter with the port element of the media descriptor.

3.6.2.7 Other elements of the media description must be transmitted without alteration.

3.6.3 DISCUSSION

A connection parameter received from a (multicast-based) IP network can be formatted as:

c=IN IP4 224.1.0.1/255

The equivalent for a IPN-based BP network would be:

c=DTN BP ipn:1

Likewise, a media descriptor for a service transmitted on port 6000 of an IP multicast network would be formatted as:

m=video **6000** RTP/AVP 96

If CBHE service ID 2 is used for the outgoing media descriptor, this media descriptor would be formatted as:

m=video **2** RTP/AVP 96

Therefore a connection parameter describing *ipn:1* as the root of the media service can be combined with the media descriptor describing service ID 2 in order to form the IPN name of *ipn:1.2*

3.6.4 RTCP CONCATENATION

3.6.4.1 Transmission Concatenation and Receiver Refragmentation

NOTE – RTCP concatenation is slightly more complex than the RTP concatenation mechanism specified in 3.3. RTCP packets are designed to be concatenated and can be combined into a compound RTCP packet, as outlined in section 6.1 of reference [1]. The rules in this section are normative and override the suggestions provided within the RFC (reference [1]).

3.6.4.1.1 If any incoming RTCP packets contain padding, it must be stripped during the concatenation process. The padding bit of all RTCP packets within the concatenated packet must be set to '0'.

3.6.4.1.2 All non-Sender Report RTCP packets must be ignored during the concatenation process.

3.6.4.1.3 The SSRCs within the RTCP packets shall not be changed.

3.6.4.1.4 The payload of each bundle conveying concatenated RTCP data must contain the latest RTCP sender report from each SSRC received since the last elapse of the transmission interval; no other RTCP packets may be contained within a single bundle.

NOTE – Unlike RTP concatenation, RTCP packets from multiple SSRCs can be multiplexed.

3.6.4.1.5 An RTCP Sender Report, as identified by a tuple of SSRC and timestamp, must not be repeated in multiple bundles.

3.6.4.2 RTCP Decapsulation

If the encapsulated RTCP packets are ultimately destined for an IP-based network, the following additional constraints must be applied:

- a) The receiver must decapsulate and forward every RTCP packet from a received and concatenated bundle.
- b) Concatenated RTCP sender reports must be split into their component reports and transmitted to an odd-numbered port. Subject to the recommendations in section 11 of reference [1], this port must be one above that which is used for RTP transmission of a single media source.

NOTE – Since RTCP packets from multiple media sources may be concatenated, it is imperative that the implementer is aware that a single RTCP bundle may contain reports that will ultimately be destined for multiple ports on the IP network. Furthermore, as RTCP sender reports do not contain IP information, the only way to correlate the media sources from the RTCP packet with media sources on the network is via the SSRC of those component sources.

3.6.4.3 Packet Transmission Frequency

3.6.4.3.1 The packet transmission algorithm outlined in section 6.3.1 of reference [1] shall be ignored and replaced with a fixed transmission interval of less than or equal to 15 seconds.

NOTE – As shown in figure 3-2, the RTCP transmission interval determines the maximum period before real-time synchronization of RTP-based media streams with other sources (such as telemetry) can be achieved, and must be considered carefully. Decoding can begin at any time and is not affected by RTCP Sender Reports.

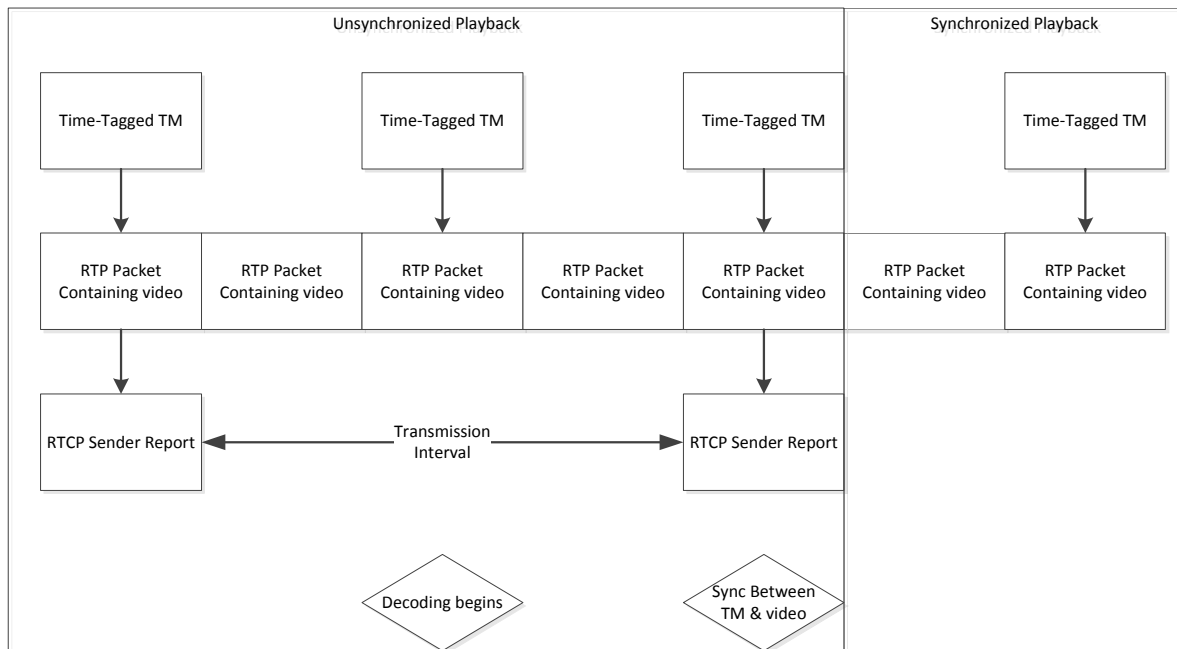


Figure 3-2: RTCP and Video Synchronization

3.6.4.3.2 After the transmission interval has elapsed, the concatenated bundle shall be transmitted only if non-empty.

NOTE – As a result of these rules, it is strongly encouraged that the receiver maintain additional state-related information, such as a table of SSRCs and respective multicast addresses. Therefore as stated in reference [1], the receiver will retransmit the RTCP packet on a multicast port that is one above the associated RTP stream. However, if the RTP and RTCP demultiplexers are different applications, the SSRC-port mapping from 2.5.3.4 will be used for enquiry.

3.6.4.4 In-Band SPS/PPS Transmission of H264/H265 Data

If in-band transmission of H.264/H.265 SPS/PPS data is to be used for video source, SDP data must only be transmitted if multiple media sources exist within a single program and/or RTCP is used.

ANNEX A

IMPLEMENTATION CONFORMANCE STATEMENT PROFORMA

(NORMATIVE)

A1 INTRODUCTION

A1.1 OVERVIEW

This annex provides the Implementation Conformance Statement (ICS) Requirements List (RL) for an implementation of CCSDS 766.3-R-1. The ICS for an implementation is generated by completing the RL in accordance with the instructions below. An implementation claiming conformance must satisfy the mandatory requirements referenced in the RL.

A1.2 ABBREVIATIONS AND CONVENTIONS

The RL consists of information in tabular form. The status of features is indicated using the abbreviations and conventions described below.

Item Column

The item column contains sequential numbers for items in the table.

Feature Column

The feature column contains a brief descriptive name for a feature. It implicitly means ‘Is this feature supported by the implementation?’

Status Column

The status column uses the following notations:

- M mandatory;
- O optional;
- C conditional;
- X prohibited;
- I out of scope;
- N/A not applicable.

Support Column Symbols

The support column is to be used by the implementer to state whether a feature is supported by entering Y, N, or N/A, indicating:

- Y Yes, supported by the implementation.
- N No, not supported by the implementation.
- N/A Not applicable.

The support column should also be used, when appropriate, to enter values supported for a given capability.

A1.3 INSTRUCTIONS FOR COMPLETING THE RL

An implementer shows the extent of compliance to the Recommended Standard by completing the RL; that is, the state of compliance with all mandatory requirements and the options supported are shown. The resulting completed RL is called an ICS. The implementer shall complete the RL by entering appropriate responses in the support or values supported column, using the notation described in A1.2. If a conditional requirement is inapplicable, N/A should be used. If a mandatory requirement is not satisfied, exception information must be supplied by entering a reference X_i , where i is a unique identifier, to an accompanying rationale for the noncompliance.

A2 ICS PROFORMA FOR CCSDS 766.3-R-1

A2.1 GENERAL INFORMATION

A2.1.1 Identification of ICS

Date of Statement (DD/MM/YYYY)	
ICS serial number	
System Conformance statement cross-reference	

A2.1.2 Identification of Implementation Under Test

Implementation Name	
Implementation Version	
Special Configuration	
Other Information	

A2.1.3 Identification of Supplier

Supplier	
Contact Point for Queries	
Implementation Name(s) and Versions	
Other information necessary for full identification, for example, name(s) and version(s) for machines and/or operating systems;	
System Name(s)	

A2.1.4 Identification of Specification

CCSDS 766.3-R-1	
Have any exceptions been required?	Yes [] No []
NOTE – A YES answer means that the implementation does not conform to the Recommended Standard. Non-supported mandatory capabilities are to be identified in the ICS, with an explanation of why the implementation is non-conforming.	

A2.2 REQUIREMENTS LIST

Item	Description	Reference	Status	Support
101	IETF RTP Header used	3.2.1, 3.2.2	m	
102	RTP Concatenation	3.3	o	
103	BP to Non-BP fragmentation	3.4	o	
104	DTN Transmission of RTP packets	3.5	m	
105	Signaling data	3.6	o	
106	SDP	3.6.2	o	
107	RTCP Concatenation	3.6.4	o	
108	RTCP decapsulation	3.6.4.2	o	
109	Packet Transmission Frequency	3.6.4.3	c5	
110	SPS/PPS transmission for H264/H265	3.6.4.4	o	

ANNEX B

SECURITY, SANA, AND PATENT CONSIDERATIONS

(INFORMATIVE)

B1 SECURITY CONSIDERATIONS

B1.1 SECURITY CONCERNS WITH RESPECT TO THE CCSDS DOCUMENT

This specification does not provide any built-in facilities for the securing of RTP data; instead, this specification relies on the use of secure RTP (reference [C4]) and/or Streamlined Bundle Security Protocol (SBSP) (reference [C5]) on the underlying DTN network.

B1.1.1 Data Privacy

This specification does not make any provision to privatize video data. It is the responsibility of the mission developer to, if required, incorporate secure RTP and/or SBSP into the mission infrastructure. If either of these security mechanisms are used, the relevant primitives for key management must be added to the Management Information Base (MIB) for the video hardware and/or avionics.

B1.1.2 Data Integrity

This specification does not provide any mechanism to ensure end-to-end data integrity, instead relying upon the underlying Bundle Protocol implementation to provide these primitives. The SBSP Bundle Integrity Block (BIB) may be used.

B1.1.3 Authentication of Communicating Entities

This specification does not provide any mechanism to authenticate the source of video data, instead relying upon the underlying Bundle Protocol implementation to provide these primitives. For example, LTP authentication (see reference [C6]) and/or the BIB may be used.

B1.1.4 Control of Access to Resources

This specification assumes that access control is a task that is better performed by the underlying network, either via the use of encryption and/or network topologies. If encryption is used, individual video streams may be encrypted with unique keys; interested parties must possess those keys in order to access the video streams.

B1.1.5 Availability of Resources

As this standard is intended to run within a heterogeneous DTN network, it is foreseen that other mechanisms, such as Bundle Protocol priorities, will be utilized to ensure reliability. If prioritization is used, the MIB must be extended to allow the control of priority for all video streams within the mission.

B1.1.6 Auditing of Resource Usage

Auditing shall only be provided if an ancillary security mechanism is used. In this case, the security mechanism must present an audit trail.

B1.2 POTENTIAL THREATS AND ATTACK SCENARIOS

Several potential threat vectors exist for this standard: First, a malicious actor may inject non-compliant RTP data into the network, which could cause hardware decoders to crash. More significantly, since it is foreseen that the transition from IP to DTN to IP will be performed by software, this malicious RTP data may be used to exploit the underlying infrastructure (via a buffer overflow attack or other software-vector).

Video distribution networks are also susceptible to Denial of Service attacks, a risk that is increased when multicast is in use. In a Denial of Service attack, the attacker transmits malicious data and relies upon the underlying network topology to ‘amplify’ the data. This may cause an overload of the network.

Finally, RTP data may convey other data besides video; if a malicious actor has access to the underlying RTP encapsulation system, they may use non-video RTP packets to exfiltrate data from internal networks.

B1.3 CONSEQUENCES OF NOT APPLYING SECURITY TO THE TECHNOLOGY

If confidentiality is not implemented, imagery might be visible to unauthorized entities resulting in disclosure of sensitive or private information.

Without source authentication or integrity verification, valid imagery could be corrupted or invalid imagery substituted in its place. Without access controls, authorized entities might be able to redistribute sensitive or proprietary information to unauthorized third parties.

B2 SANA CONSIDERATIONS

The recommendations of this document do not require any action from SANA. It is expected that the DTN node numbers used for network entities as well as the requisite multicast group identifiers will be managed as required, in accordance with other CCSDS specifications, such as DTN Network Management.

B3 PATENT CONSIDERATIONS

The RTP standard has been provided to the IETF community and therefore is not patent encumbered.

ANNEX C

INFORMATIVE REFERENCES

(INFORMATIVE)

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CCSDS, May 2015.

ANNEX D

ABBREVIATIONS AND ACRONYMS

(INFORMATIVE)

<u>Term</u>	<u>Meaning</u>
BIB	SBSP Bundle Integrity Block
CBHE	Compressed Bundle Header Encoding
COTS	commercial off-the-shelf
DLNA	Digital Living Network Alliance
DTN	Delay/Disruption Tolerant Networking
EID	endpoint identifier
EVA	extravehicular activity
ICS	implementation conformance statement
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPN	interplanetary network
IPTV	Internet Protocol television
LTP	Licklider Transmission Protocol
MIB	management information base
MPEG-TS	MPEG transport stream
MTU	maximum transmission unit
NAL	network abstraction layer
PDU	protocol data unit

DRAFT CCSDS RECOMMENDED STANDARD FOR REAL-TIME TRANSPORT PROTOCOL
OVER DELAY TOLERANT NETWORKING FOR VIDEO APPLICATIONS

PPS	picture parameter set
PT	payload type
RTCP	Real-Time Control Protocol
RTP	Real-Time Transport Protocol
RTP/AVP	RTP Profile for Audio and Video
RTSP	Real Time Streaming Protocol
SAP	Session Announcement Protocol
SBSP	Streamlined Bundle Security Protocol
SDI	serial data interface
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SMPTE	Society of Motion Picture and Television Engineers
SPS	sequence parameter set
SR	sender report
SSRC	synchronization source
URI	Uniform Resource Identifier
VLC	VideoLAN client
VPN	virtual private network
X	extension