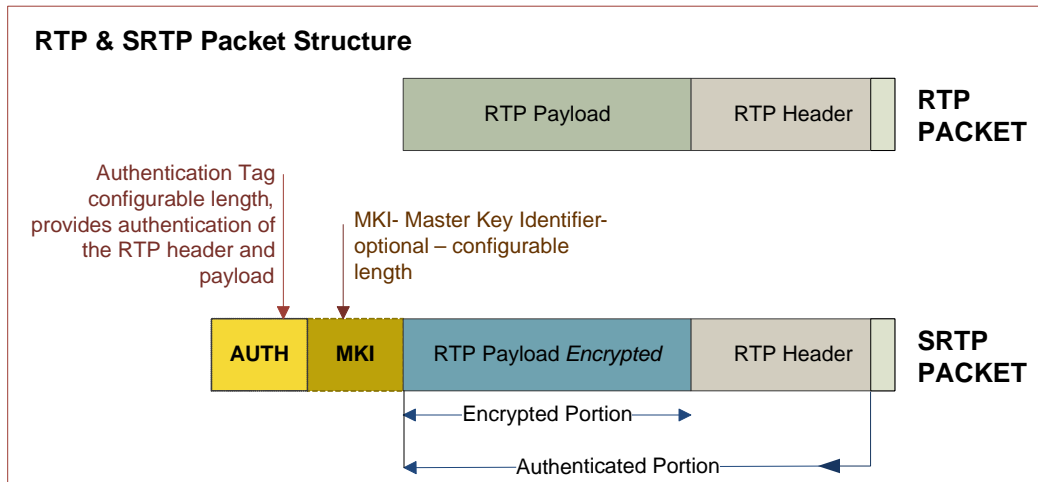


## RTP Series

The Real-Time Transport Protocol (RTP) is a networking protocol that is used to transport real-time media data streams such as voice and video over packet networks. This protocol is an industry standard that is defined by IETF RFC 3550.



## PRODUCT DESCRIPTION

Adaptive Digital Technologies' Real Time Protocol (RTP) software provides transport layer functionality for real-time applications communicating over an IP network. This product also contains a built in, configurable, jitter buffer to compensate for network delays, out-of-order packets, and lost packets.

Adaptive Digital's implementation of the RTP protocol is designed to provide fully re-entrant modules to allow multiple RTP streams to be processed concurrently. User supplied callback and support routines are used to allow the RTP software to be easily adapted to the application environment. Memory allocation for the packets stored within the jitter buffer is dynamic and is accomplished via a call to a user definable memory allocation support routine. To allow flexibility in interfacing with differing network stack mechanisms, the mechanism to send data over the network is accomplished via a user definable callback routine.

The built-in jitter buffer provides a storage mechanism for inbound packets. The RTP module stores incoming packets into the jitter buffer at the time of reception. The jitter buffer uses the RTP header timestamp and sequence number to position out-of-order packets correctly within the jitter buffer. Packets remain stored in the jitter buffer until they are ready for delivery to the receiving application.

## RTP, RTCP, RTCP-XR, AND THEIR SECURE COUNTERPARTS

The successful transport of real-time voice and video data necessitates the use of transport protocols that are different from the traditional protocols, such as TCP and UDP, that are typically used for the transport of non-real-time data. There are three primary differences:

Voice and Video need to be “played out” in a continuous fashion in spite of the bursty nature of packet networks.

When a packet is lost, late, or received in error, there is no time to request a retransmission.

Packets that are received out of sequence must be re-sequenced so that they can be played out in the correct order.

RTP is specifically designed to handle the play-out requirements of real-time media streams through the use of time stamps and jitter buffering. Due to the real-time nature of the data streams, where requesting retransmissions is too costly in time, RTP is typically used in conjunction with UDP to provide low-overhead network communications between two end-points.

RFC 3550 identifies two components to the real-time transport: data transport and control. Data transport is handled by RTP while Real Time Control Protocol (RTCP) handles control. RTCP, which can be used to help scale the network traffic to the available bandwidth, is optional.

An RTP packet identifies the media payload type (format) and its source. It also includes time stamps and sequence numbers that are used by the play-out side to handle lost or out of sequence packets. RTP provides for the use of multiple streams as in the case of a system that transmits both voice and video. The payload in an RTP payload contains the encoded voice or video information. The use of dynamically defined payload types allows RTP packets to carry virtually any type of media format.

RTCP\* is used to keep track of packet reception statistics and to provide supplementary information (user and domain name, e-mail address, phone number, etc.) about the source of the media data stream. It also assists with the synchronization of multiple RTP streams. While RTP packets are sent at high rates to handle the real-time media streams, RTCP packets can normally be sent much less often – a typical rate being every few seconds.

RTCP – Extended Report (RTCP-XR) is an extension to RTCP that enables a node to provide voice quality measurements (lost/dropped packet rates, signal levels, etc.) in a VoIP network. This extension is defined in IETF RFC 3611.

A secure transmission feature, as defined in IETF RFC 3711 is also available for these protocols. When security is used, an “S” prefaces the acronyms: SRTP, SRTCP, SRTCP-XR. When security is being used, the packet payloads are encrypted.

---

\* Adaptive Digital currently does not support RTCP.

## FEATURES

- Multi-channel capable.
- Functions are C-callable.
- Support of RTP version 2 protocol as defined in RFC 3550
- Independence from underlying protocol stack.
- Minimum count of consecutively increasing sequence numbers prior to playout.
- Built in jitter buffer
- Automatic timestamp synchronization.
- Multiple independent RTP streams
- Re-entrant routines callable by multiple processing threads
- eXpressDSP™ Algorithm Interoperability Standard (xDIAS) Compliant
- Secure variants include support for:
  - Authentication Algorithm Types: HMAC-SHA1 and MD5
  - Key Definition Schemes: PSK, MKI, and FT
  - Encryption types: CM, F8



## AVAILABILITY

ADT RTP is available in transportable “C” source code format as well as in library object format on all the Texas Instruments TMS320™ DSPs, and TNETV™ family of VoIP processors.

Product	Platform	Memory Model	Endian	Code Gen Tool Version
ADT_rtp_c64x	TI TMS320C64x	L3	Little	N/R
ADT_rtp_c64xp	TI TMS320C64x+	L3	Little	N/R
ADT_rtpdll_win32	Win32	N/A	Little	VS2010
ADT_rtpplib_win32	Win32	N/A	Little	VS2010
ADT_rtp_arm9e	Arm9e	N/A	Little	Code Sourcery Linux 2011_09-70
ADT_rtp_cortex-a8	Arm cortex-a8	N/A	Little	Code Sourcery Linux 2011_09-70
ADT_rtp_i686	i686	N/A	Little	gcc

## SPECIFICATIONS

### TI TMS320C6000

#### CPU UTILIZATION

Peak CPU utilization occurs when a new master encryption key is required. Normal voice over IP applications require only a single key for the duration of a conversation; in this case, the peak occurs only at the start of an RTP stream.

#### C64x & C64x+ MIPS

#### CPU UTILIZATION

Processor	Direction	Frame Rate (ms)					
		Avg			Peak		
		10	20	30	10	20	30
C64	Rx	.26	.14	.10	.43	.22	.15
C64	Tx	.2	.1	.07	.2	.1	.07
C64+	Rx	.18	.09	.05	.54	.27	.26
C64+	Tx	.11	.06	.04	.21	.11	.07

**MEMORY REQUIREMENTS**

Program memory is shared between all streams. All function calls are re-entrant and all stream share the same program memory.

**C64x & C64x+****MEMORY REQUIREMENTS**

All Memory usage is given in units of byte.

Processor Type	Memory (bytes)	
	Program	Instance
C64	9652	172
C64+	8324	172

**ARM****ARM CORTEX-A8****CPU UTILIZATION & MEMORY REQUIREMENTS**

All Memory usage is given in units of bytes.

	Average MIPS	Program Memory	Instance Data Memory
Tx	.20	8.6k	172
Rx	.20		

**ARM9e****CPU UTILIZATION & MEMORY REQUIREMENTS**

All Memory usage is given in units of bytes.

	Average MIPS	Program Memory	Instance Data Memory
Tx	.20	8.6k	172
Rx	.10		

**PC/Windows****Win Static Lib****CPU UTILIZATION & MEMORY REQUIREMENTS**

All Memory usage is given in units of bytes.

	Average MIPS	Program Memory	Instance Data Memory
Tx	<0.33*	8.6k	172
Rx	<0.33*		

\*Value too small to measure accurately, less than 0.33

**Win DLL****CPU UTILIZATION & MEMORY REQUIREMENTS**

All Memory usage is given in units of bytes.

	Average MIPS	Program Memory	Instance Data Memory
Tx	<0.33	38k	172
Rx	<0.33		

**LINUX****Linux i686****CPU UTILIZATION & MEMORY REQUIREMENTS**

All Memory usage is given in units of bytes.

	Average MIPS	Program Memory	Instance Data Memory
Tx	<0.33	8.6k	172
Rx	<0.33		

We specify MIPS (Millions of Instructions Per Second) as MCPS (Millions of Instruction Cycles Per Second). Unless otherwise specified, peak MIPS are indicated.

## TERMINOLOGY

---

### Acronyms

RTP - Real-time Transport Protocol

SRTP - Secure Real-time Transport Protocol

AES CM - Advanced Encryption Standard counter mode

AES-f8 - AES in f8-mode, Universal Mobile Telecommunications System (UMTS) 3G mobile networks use AES-f8.

MKI - Master key identifier

HMAC - Hashed message authentication

MD5 - Message Digest 5 is a widely used cryptographic hash function with a 128-bit hash value

### Deliverables

The deliverable items are platform dependent. In general, there is one library. (Sometimes multiple variants of the library are included in the deliverables.) There are also header files, some of which are specific to the product and others are common across many of Adaptive Digital's products. Also included in the deliverables is product documentation, which includes a users guide and usually includes release notes and a data sheet. Sample/test code may be included as well.

*Adaptive Digital is a member of the Texas Instruments Developer Network, and ARM Connected Community.*

### CONTACT INFORMATION

Web: [www.adaptivedigital.com](http://www.adaptivedigital.com)  
 Email: [information@adaptivedigital.com](mailto:information@adaptivedigital.com)  
 Tel: 610.825.0182 ~ Toll Free: 1.800.340.2066  
 Fax: 610.825.7616  
 Address: 525 Plymouth Road, Suite 316  
 Plymouth Meeting, PA 19462



**IMPORTANT NOTICE:** Data subject to change, for the most up to date information visit our website. Customers are advised to obtain the most current and complete information about Adaptive Digital products and services before placing orders.  
 All trademarks are property of their respective owners.

