Advanced RTP/RTCP Toolkit

for transferring real-time media over UDP/IP

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Softil's Advanced RTP/RTCP Toolkit is designed to address the requirements of mature, production IP telephony applications. The Advanced RTP/RTCP Toolkit can seamlessly scale from small embedded platforms all the way up to densely-populated parallel-processing environments.

RTP/RTCP Basics

The Real-Time Transport Protocol (RTP) was designed to send real-time media, such as voice and video, over UDP/IP. It can also be used to transmit other types of data, such as DTMF, text and pointers. In addition, the protocol supplies information to enable the receiver to resynchronize media for lip syncing, and to display text at the correct time in relation to an image or word. Since RTP can be configured for low latency, it is useful for interactive conversations as well as for streaming media.

The Real-Time Transport Control Protocol (RTCP) is a companion protocol to RTP for gathering statistics on a media connection, as well as information such as bytes sent, packets sent, lost packets, jitter and round-trip delay. The application can use this information to judge the quality of its connections and make adjustments as required, such as changing from a low compression codec to a high compression codec, or using an adaptive jitter buffer.

For security, the RTP/RTCP data can be encrypted to enable improved privacy against eavesdropping. RTP/RTCP has been enhanced with the RFCs 3550 and 3551 for better monitoring, streaming capabilities and code support, as well as for serving Open Mobile Association (OMA) specifications for operating Push-to-talk Over Cellular services and 3GPP Mission Critical Communications over LTE (Release 13 and higher).

The Advanced RTP/RTCP Toolkit is used for developing applications such as:

Networking Equipment

- PoC Servers
- Video-on-Demand Servers
- Gateways
- Media Servers
- IP-PBXs
- IVR Systems
- MCUs
- Call Center Systems
- Softswitches

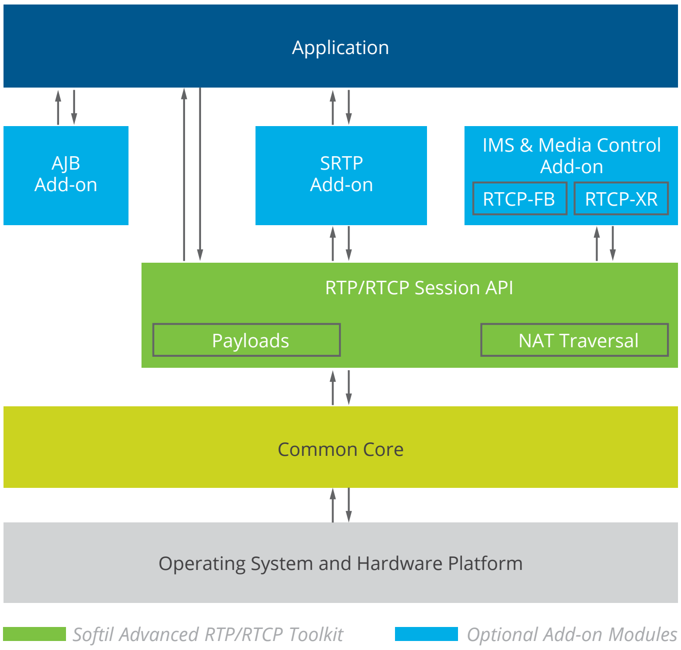
Terminals

- PoC Terminals
- VOD Terminals
- VC Terminals
- IP Phones
- WiFi Hand-held Devices
- Thin Clients

Chipset Solutions

- System on Chip (SoC) RTP/RTCP
- IP Phones, Videophones
- IADs, ATAs

Advanced RTP/RTCP Toolkit Architecture



Advanced RTP/RTCP Toolkit Overview

Softil's Advanced RTP/RTCP Toolkit is compliant with the IETF RFCs 3550, 3551, 2032, 2190, 3640, 3267, 2833 and 4733, as well as with OMA PoC requirements for RTP/RTCP, ISMA 2.0 and 3GPP Mission Critical Communications (MCC) over LTE. The Toolkit, designed for high-end applications implemented in real-time multi-threaded environments, is written in ANSI C. It is available on multiple operating systems and is portable to different environments. The Toolkit enables various run-time settings, such as filters and bandwidth, and provides both high-level and low-level/fine-grain APIs for maximum flexibility. In addition, both asynchronous and synchronous operation modes are supported.

SRTP Add-on Module

Supports RFC 3711 (Secure RTP)

Key Features

- Confidentiality
- Packet authentication
- Replay protection
- Enhanced privacy and security through implementation of key derivation

IMS & Media Control Add-on Module: RTCP-XR

Supports RFC 3611 (RTCP Extended Reports)

- Required for IMS and PSS deployments
- Enables QoS optimizations and improved error analysis

Provides

- Packet-by-packet reports on received/lost RTP packets
- Reference time information between RTP participants
- Metrics related to packet receipts

IMS & Media Control Add-on Module: RTCP-FB

Supports RFC 4585 (RTCP Feedback)

- Required for IMS and video deployments
- Enables adaptation to network conditions using an immediate feedback mechanism in RTCP

Key Features

- Feedback channel piggybacked on RTCP reports
- Sends and receives RTCP-FB-based messages
- Maintains low RTCP bandwidth usage

Adaptive Jitter Buffer (AJB) Add-on Module

Enables smooth playback of media in packet-based reception over error-prone networks, such as RTP-based VoIP systems. Addresses issues such as out-of-order arrival of packets, packet loss and variation in delays that result in poor quality of media. A jitter buffer is used to compensate for jitter in packet arrival and out-of-order packets.

Key Features

- Suitable for use with different media types, including audio and video streams
- Asynchronous interface, enabling inserting and extracting buffers from different threads
- Automatic collection of video packets into frames based on timestamps and sequence numbers
- Packet reordering
- Packet dropping based on preferred strategy
- Statistics collection for network behavior analysis
- Flexible configuration

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Advanced RTP/RTCP Toolkit API

The Toolkit consists of several modular and layered APIs, as shown in the Architecture diagram. The following APIs are included:

- RTP/RTCP Session Management
- Dual Tone Multi-Frequency (DTMF)
- RTP Payloads
- RTP Key
- RTP Encryption
- RTCP Events
- RTCP APP
- RTP/RTCP Session
- RTP/RTCP H.460.19 Multiplexing
- OS Abstraction (Common Core)

RTP/RTCP Session Management API—The RTP/RTCP Session Management API is the highest-level API in the Toolkit. This API enables initiating and terminating RTP/RTCP sessions. In addition, the API enables building and sending, as well as receiving and extracting data.

Dual Tone Multi-Frequency (DTMF) API—The Advanced RTP/RTCP Toolkit provides APIs for transporting DTMF Digits, Telephony Tones and Telephony Signals over RTP in accordance with RFC 4733. These APIs simplify the transport of in-call events and signals over RTP.

RTP Payloads API—The RTP Payloads API provides built-in RTP codec payloads for pre-defined codecs: G.711 a-law/µ-law, G.723.1, G.728, G.729, iLBC, EVRC, SMV, GSM-AMR, H.261, H.263, H.263+, MPEG-2, MJPEG, MPEG-4 and H.264/AVC.

RTP Key Plug-In—The RTP Key Plug-In provides management of several keys and enables setting up separate keys for different RTP sessions. Different keys can be defined for encoding/decoding on the same session.

RTP Encryption API—The RTP Encryption API is used to integrate external encryption functions such as DES and 3DES, and user-defined encryption functions such as AES (the user has to supply the implementation of the encryption algorithm).

RTCP Events API—The RTCP Events API enables the application to register RTCP events with callback functions.

RTP APP API—The RTP APP API enables creating RTCP APP reports, sending and receiving report PDUs and extracting reported information.

RTP/RTCP Session API—The RTP/RTCP Session API mediates between the upper layers (see the Architecture diagram) and the OS Abstraction layer (Common Core). The API handles each session as a separate object instance in terms of memory handling and RTP/RTCP protocol state. In addition, this API can generate automatic RTCP reports. The reports are generated in time intervals as defined in RFC 3550, calculated automatically based on RTCP session parameters such as bandwidth to the application, for QoS/Monitoring purposes.

RTP/RTCP H.460.19 Multiplexing API—The RTP/RTCP H.460.19 Multiplexing API enables applications to support the H.460.19 standard for NAT/FW traversal using H.323 entities. This support includes multiplexing of several RTP or RTCP sessions over the same socket, allowing a single pinhole for RTP and a single pinhole for RTCP in the firewall to handle all media needs of an enterprise or home user.

OS Abstraction (Common Core) API—The Common Core is an OS abstraction layer that provides OS services such as sockets, threading and locking, and memory management to the Softil Stack levels. The Common Core has been ported to a wide range of real-time operating systems (RTOS) and non-RTOS, and comes with a comprehensive Porting Guide to enable porting to additional operating systems. Services, such as DNS address resolver, IPv6/IPv4 and flexible logging API, are also provided.

The Advanced RTP/RTCP Toolkit is delivered with:

- Source code
- Sample application
- Release notes
- Detailed documentation

Operating Systems Supported*

- MS Windows
- Monta Vista Linux (Embedded Linux)
- Linux, Red Hat
- Solaris
- VxWorks
- Nucleus
- Integrity
- Wind River Linux
- Google Android
- Apple iOS and OSX

* Inquire about support for additional operating systems

Additional Softil Solutions

SIP Server Platform—A comprehensive SIP server development solution with complete standards-based functionality of Proxy, Redirect and Registrar servers.

Professional Services—A full range of design, integration and deployment consulting services, including kernel mode driver development for various operating systems.

SIP Developer Suite—The award-winning SIP Developer Suite is a powerful and highly versatile set of tools for dramatically accelerating SIP application development.

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Advanced RTP/RTCP Toolkit Features

- High performance
- Written in ANSI C
- Thread-safe
- Supports RTP OMA PoC 1.0 requirements
- IPv4/IPv6 support (including address translation)
- Security profiles support
- Mode 1: RFC 3550 Section 9 Encryptions
- Mode 2: H.235 Annex D (Ciphertext stealing)
- Mode 3: H.235 Annex D (Padding)
- SRTP
- Encryption profiles supported
- DES
- 3DES
- User-defined (for example, AES)
- UDP Unicast/Multicast support
- Blocking and non-blocking operation modes
- Synchronous/Asynchronous operation modes
- Extensible codec interface (for supported codecs)
- Payloads API (enables supporting new codecs)
- Built-in codec payloads API for Voice: G.711 a-law/μ-law, G.722.x, G.723.1, G.728, G.729, AMR, EVRC, SMV, Opus
- Video: H.261, H.263, H.263+, MPEG-4 (narrow band/wide band), H.264, MJPEG, MPEG-2
- RTCP Reports API
- RTCP Extended Reports (RTCP-XR)
- Support for compounded reports
- Support for manual APP & BYE messages
- RTCP events API
- IP TOS/Diffserv setting for improved QoS (if OS provides these services)
- Extensible SSRC generation mechanism
- Full 3DES support
- Adaptive RTCP report interval
- Support for RTCP APP messages
- Session logging capabilities
- Multiple Stack instance per process space
- Platform independent (OS abstraction layer-based)
- Support for general extension in RTP header
- Provides statistics for adaptive jitter buffer
- Support for multiple NAT traversal techniques (H.460, STUN, TURN)

Standards Compliance

- IETF RFC 3550 - Transport protocol for Real-Time Applications (replaces RFC 1889)
- IETF RFC 3551 - Profile for audio and video conferences with minimal control (replaces RFC 1890)
- RTP/RTCP OMA PoC 1.0
- RTP/RTCP ISMA 2.0
- IETF RFC3771 - Secure RTP (SRTP Add-on)
- IETF RFC 3611 - RTCP Extended Reports (RTCP-XR Add-on)
- IETF RFC 3556 bandwidth modifiers support
- IETF RFC 4733–RTP payload for DTMF digits, telephony tones, and telephony signals
- IETF RFC 4573 - RTP payload format for Far End Camera Control (FECC)
- ITU-T H.235 Annex D Security (Media Encryption)
- ITU-T H.460.19 FW/NAT Traversal
- 3GPP Mission Critical Communications over LTE
- IETF RFC 2032
- IETF RFC 2190–RTP payload format for H.263 video streams
- IETF RFC 2429–RTP payload format for H.263+
- IETF RFC 3640–RTP payload format for MPEG-4 payload
- IETF RFC 3267–RTP payload format for AMR narrow band and AMR wide band payload
- IETF RFC 7587–RTP payload format for the Opus speech and audio codec
- IETF RFC 3984–RTP payload format for H.264/AVC
- IETF RFC 2429–RTP payload format for H.263
- IETF RFC 3558–RTP payload format for EVRC and SMV
- IETF RFC 3389–RTP payload format for Comfort Noise (CN)
- IETF RFC 5371–RTP payload format for JPEG2000
- IETF RFC 4585–RTCP Feedback Report

For more information, contact Softil at info@softil.com

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