

libSRTP 2.0.0 Overview and Reference Manual

David A. McGrew
mcgrew@cisco.com

Preface

The original implementation and documentation of libSRTP was written by David McGrew of Cisco Systems, Inc. in order to promote the use, understanding, and interoperability of Secure RTP. Michael Jerris contributed support for building under MSVC. Andris Pavenis contributed many important fixes. Brian West contributed changes to enable dynamic linking. Yves Shumann reported documentation bugs. Randell Jesup contributed a working SRTCP implementation and other fixes. Alex Vanzella and Will Clark contributed changes so that the AES ICM implementation can be used for ISMA media encryption. Steve Underwood contributed x86_64 portability changes. We also give thanks to Fredrik Thulin, Brian Weis, Mark Baugher, Jeff Chan, Bill Simon, Douglas Smith, Bill May, Richard Prestley, Joe Tardo and others for contributions, comments, and corrections.

This reference material in this documentation was generated using the `doxygen` utility for automatic documentation of source code.

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Chapter 1

Introduction to libSRTP

This document describes libSRTP, the Open Source Secure RTP library from Cisco Systems, Inc. RTP is the Real-time Transport Protocol, an IETF standard for the transport of real-time data such as telephony, audio, and video, defined by RFC 3550. Secure RTP (SRTP) is an RTP profile for providing confidentiality to RTP data and authentication to the RTP header and payload. SRTP is an IETF Proposed Standard, defined in RFC 3711, and was developed in the IETF Audio/Video Transport (AVT) Working Group. This library supports all of the mandatory features of SRTP, but not all of the optional features. See the [Supported Features](#) section for more detailed information.

This document is organized as follows. The first chapter provides background material on SRTP and overview of libSRTP. The following chapters provide a detailed reference to the libSRTP API and related functions. The reference material is created automatically (using the doxygen utility) from comments embedded in some of the C header files. The documentation is organized into modules in order to improve its clarity. These modules do not directly correspond to files. An underlying cryptographic kernel provides much of the basic functionality of libSRTP, but is mostly undocumented because it does its work behind the scenes.

1.1 License and Disclaimer

libSRTP is distributed under the following license, which is included in the source code distribution. It is reproduced in the manual in case you got the library from another source.

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1.2 Supported Features

This library supports all of the mandatory-to-implement features of SRTP (as defined by the most recent Internet Draft). Some of these features can be selected (or de-selected) at run time by setting an appropriate policy; this is done using the structure `srtp_policy_t`. Some other behaviors of the protocol can be adapted by defining an appropriate event handler for the exceptional events; see the [SRTP events and callbacks](#) section.

Some options that are not included in the specification are supported. Most notably, the TMMH authentication function is included, though it was removed from the SRTP Internet Draft during the summer of 2002.

Some options that are described in the SRTP specification are not supported. This includes

- the Master Key Index (MKI),
- key derivation rates other than zero,
- the cipher F8,
- anti-replay lists with sizes other than 128,
- the use of the packet index to select between master keys.

The user should be aware that it is possible to misuse this library, and that the result may be that the security level it provides is inadequate. If you are implementing a feature using this library, you will want to read the Security Considerations section of the Internet Draft. In addition, it is important that you read and understand the terms outlined in the [License and Disclaimer](#) section.

1.3 Installing and Building libSRTP

To install libSRTP, download the latest release of the distribution from `srtp.sourceforge.net`. The format of the names of the distributions are `srtp-A.B.C.tgz`, where `A` is the version number, `B` is the major release number, `C` is the minor release number, and `tgz` is the file extension¹. You probably want to get the most recent release. Unpack the distribution and extract the source files; the directory into which the source files will go is named `srtp`.

libSRTP uses the GNU `autoconf` and `make` utilities². In the `srtp` directory, run the configure script and then make:

¹The extension `.tgz` is identical to `.tar.gz`, and indicates a compressed tar file.

²BSD `make` will not work; if both versions of `make` are on your platform, you can invoke GNU `make` as `gmake`.

```
./configure [ options ]
make
```

The configure script accepts the following options:

- help** provides a usage summary.
- disable-debug** compiles libSRTP without the runtime dynamic debugging system.
- enable-generic-aesicm** compile in changes for ismacryp
- enable-syslog** use syslog for error reporting.
- disable-stdout** disables stdout for error reporting.
- enable-console** use `/dev/console` for error reporting
- gdoi** use GDOI key management (disabled at present).

By default, dynamic debugging is enabled and stdout is used for debugging. You can use the configure options to have the debugging output sent to syslog or the system console. Alternatively, you can define `ERR_REPORTING_FILE` in `include/conf.h` to be any other file that can be opened by libSRTP, and debug messages will be sent to it.

This package has been tested on the following platforms: Mac OS X (powerpc-apple-darwin1.4), Cygwin (i686-pc-cygwin), Solaris (sparc-sun-solaris2.6), RedHat Linux 7.1 and 9 (i686-pc-linux), and OpenBSD (sparc-unknown-openbsd2.7).

1.4 Applications

Several test drivers and a simple and portable srtp application are included in the `test/` subdirectory.

Test driver	Function tested
<code>kernel_driver</code>	crypto kernel (ciphers, auth funcs, rng)
<code>srtp_driver</code>	srtp in-memory tests (does not use the network)
<code>rdbx_driver</code>	rdbx (extended replay database)
<code>roc_driver</code>	extended sequence number functions
<code>replay_driver</code>	replay database
<code>cipher_driver</code>	ciphers
<code>auth_driver</code>	hash functions

The app `rtpw` is a simple rtp application which reads words from `/usr/dict/words` and then sends them out one at a time using `[s]rtp`. Manual srtp keying uses the `-k` option; automated key management using `gdoi` will be added later.

The usage for `rtpw` is

```
rtpw [[-d <debug>]* [-k <key> [-a][-e]] [-s | -r] dest_ip dest_port] | [-l]
```

Either the `-s` (sender) or `-r` (receiver) option must be chosen. The values `dest_ip`, `dest_port` are the IP address and UDP port to which the dictionary will be sent, respectively. The options are:

```

-s          (S)RTP sender - causes app to send words
-r          (S)RTP receive - causes app to receive words
-k <key>   use SRTP master key <key>, where the key is a hexadecimal value (without the leading "0x")
-e          encrypt/decrypt (for data confidentiality) (requires use of -k option as well)
-a          message authentication (requires use of -k option as well)
-l          list the available debug modules
-d <debug> turn on debugging for module <debug>

```

In order to get a random 30-byte value for use as a key/salt pair, you can use the `rand_gen` utility in the `test/` subdirectory.

An example of an SRTP session using two `rtpw` programs follows:

```

[sh1] set k=`test/rand_gen -n 30`
[sh1] echo $k
c1eec3717da76195bb878578790af71c4ee9f859e197a414a78d5abc7451
[sh1]$ test/rtpw -s -k $k -ea 0.0.0.0 9999
Security services: confidentiality message authentication
set master key/salt to C1EEC3717DA76195BB878578790AF71C/4EE9F859E197A414A78D5ABC7451
setting SSRC to 2078917053
sending word: A
sending word: a
sending word: aa
sending word: aal
sending word: aalii
sending word: aam
sending word: Aani
sending word: aardvark
...

[sh2] set k=c1eec3717da76195bb878578790af71c4ee9f859e197a414a78d5abc7451
[sh2]$ test/rtpw -r -k $k -ea 0.0.0.0 9999
security services: confidentiality message authentication
set master key/salt to C1EEC3717DA76195BB878578790AF71C/4EE9F859E197A414A78D5ABC7451
19 octets received from SSRC 2078917053 word: A
19 octets received from SSRC 2078917053 word: a
20 octets received from SSRC 2078917053 word: aa
21 octets received from SSRC 2078917053 word: aal
...

```

1.5 Secure RTP Background

In this section we review SRTP and introduce some terms that are used in libSRTP. An RTP session is defined by a pair of destination transport addresses, that is, a network address plus a pair of UDP ports for RTP and RTCP. RTCP, the RTP control protocol, is used to coordinate between the participants in an RTP session, e.g. to provide feedback from receivers to senders. An *SRTP session* is similarly defined; it is just an RTP session for which the SRTP profile is being used. An SRTP session consists of the traffic sent to the SRTP or SRTCP destination transport addresses. Each participant in a session is identified by a synchronization source (SSRC) identifier. Some participants may not send any SRTP traffic; they are called receivers, even though they send out SRTCP traffic, such as receiver reports.

RTP allows multiple sources to send RTP and RTCP traffic during the same session. The synchronization source identifier (SSRC) is used to distinguish these sources. In libSRTP, we call the SRTP and SRTCP traffic from a particular source a *stream*. Each stream has its own SSRC, sequence number, rollover counter, and other data. A particular choice of options, cryptographic mechanisms, and keys is called a *policy*. Each stream within a session can have a distinct policy applied to it. A session policy is a collection of stream policies.

A single policy can be used for all of the streams in a given session, though the case in which a single *key* is shared across multiple streams requires care. When key sharing is used, the SSRC values that identify the streams **must** be distinct. This requirement can be enforced by using the convention that each SRTP and SRTCP key is used for encryption by only a single sender. In other words, the key is shared only across streams that originate from a particular device (of course, other SRTP participants will need to use the key for decryption). libSRTP supports this enforcement by detecting the case in which a key is used for both inbound and outbound data.

1.6 libSRTP Overview

libSRTP provides functions for protecting RTP and RTCP. RTP packets can be encrypted and authenticated (using the [srtp_protect\(\)](#) function), turning them into SRTP packets. Similarly, SRTP packets can be decrypted and have their authentication verified (using the [srtp_unprotect\(\)](#) function), turning them into RTP packets. Similar functions apply security to RTCP packets.

The typedef `srtp_stream_t` points to a structure holding all of the state associated with an SRTP stream, including the keys and parameters for cipher and message authentication functions and the anti-replay data. A particular `srtp_stream_t` holds the information needed to protect a particular RTP and RTCP stream. This datatype is intentionally opaque in order to better separate the libSRTP API from its implementation.

Within an SRTP session, there can be multiple streams, each originating from a particular sender. Each source uses a distinct stream context to protect the RTP and RTCP stream that it is originating. The typedef `srtp_t` points to a structure holding all of the state associated with an SRTP session. There can be multiple stream contexts associated with a single `srtp_t`. A stream context cannot exist independent from an `srtp_t`, though of course an `srtp_t` can be created that contains only a single stream context. A device participating in an SRTP session must have a stream context for each source in that session, so that it can process the data that it receives from each sender.

In libSRTP, a session is created using the function [srtp_create\(\)](#). The policy to be implemented in the session is passed into this function as an [srtp_policy_t](#) structure. A single one of these structures describes the policy of a single stream. These structures can also be linked together to form an entire session policy. A linked list of [srtp_policy_t](#) structures is equivalent to a session policy. In such a policy, we refer to a single [srtp_policy_t](#) as an *element*.

An [srtp_policy_t](#) structure contains two `crypto_policy_t` structures that describe the cryptographic policies for RTP and RTCP, as well as the SRTP master key and the SSRC value. The SSRC describes what to protect (e.g. which stream), and the `crypto_policy_t` structures describe how to protect it. The key is contained in a policy element because it simplifies the interface to the library. In many cases, it is desirable to use the same cryptographic policies across all of the streams in a session, but to use a distinct key for each stream. A `crypto_policy_t` structure can be initialized by using either the `crypto_policy_set_rtp_default()` or `crypto_policy_set_rtcp_default()` functions, which set a crypto policy structure to the default policies for RTP and RTCP protection, respectively.

1.7 Example Code

This section provides a simple example of how to use libSRTP. The example code lacks error checking, but is functional. Here we assume that the value `ssrc` is already set to describe the SSRC of the stream that we are sending, and that the functions `get_rtp_packet()` and `send_srtp_packet()` are available to us. The former puts an RTP packet into the buffer and returns the number of octets written to that buffer. The latter sends the RTP packet in the buffer, given the length as its second argument.

```
srtp_t session;
srtp_policy_t policy;
uint8_t key[30];

// initialize libSRTP
srtp_init();

// set policy to describe a policy for an SRTP stream
crypto_policy_set_rtp_default(&policy.rtp);
crypto_policy_set_rtcp_default(&policy.rtcp);
policy.ssrc = ssrc;
policy.key = key;
policy.next = NULL;

// set key to random value
crypto_get_random(key, 30);

// allocate and initialize the SRTP session
srtp_create(&session, &policy);

// main loop: get rtp packets, send srtp packets
while (1) {
    char rtp_buffer[2048];
    unsigned len;

    len = get_rtp_packet(rtp_buffer);
    srtp_protect(session, rtp_buffer, &len);
    send_srtp_packet(rtp_buffer, len);
}
```

1.8 ISMA Encryption Support

The Internet Streaming Media Alliance (ISMA) specifies a way to pre-encrypt a media file prior to streaming. This method is an alternative to SRTP encryption, which is potentially useful when a particular media file will be streamed multiple times. The specification is available online at <http://www.isma.tv/specreq.nsf/SpecRequest>.

libSRTP provides the encryption and decryption functions needed for ISMAcryp in the library `libaesicm.a`, which is included in the default Makefile target. This library is used by the MPEG4IP project; see <http://mpeg4ip.sourceforge.net/>.

Note that ISMAcryp does not provide authentication for RTP nor RTCP, nor confidentiality for RTCP. ISMAcryp RECOMMENDS the use of SRTP message authentication for ISMAcryp streams while using ISMAcryp encryption to protect the media itself.

Chapter 2

Module Index

2.1 Modules

Here is a list of all modules:

Secure RTP	11
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data associated to a SRTP session.	33
SRTP events and callbacks	34
Error Codes	37
Cryptographic Kernel	38
Ciphers	39

Chapter 3

Data Structure Index

3.1 Data Structures

Here are the data structures with brief descriptions:

srtp_hdr_t	41
srtp_trailer_t	41
srtp_crypto_policy_t Srtp_crypto_policy_t describes a particular crypto policy that can be applied to an SRTP stream	41
srtp_debug_module_t	43
srtp_event_data_t Srtp_event_data_t is the structure passed as a callback to the event handler function	43
srtp_hdr_t	44
srtp_hdr_xtnd_t	44
srtp_policy_t Policy for an SRTP session	44
srtp_ssrc_t An srtp_ssrc_t represents a particular SSRC value, or a 'wildcard' SSRC	46

Chapter 4

Module Documentation

4.1 Secure RTP

libSRTP provides functions for protecting RTP and RTCP. See Section [libSRTP Overview](#) for an introduction to the use of the library.

Modules

- [Secure RTCP](#)
Secure RTCP functions are used to protect RTCP traffic.
- [data associated to a SRTP session.](#)
Store custom user data within a SRTP session.
- [SRTP events and callbacks](#)
libSRTP can use a user-provided callback function to handle events.

Data Structures

- struct [srtp_hdr_t](#)
- struct [srtp_hdr_xtnd_t](#)
- struct [srtcp_hdr_t](#)
- struct [srtcp_trailer_t](#)
- struct [srtp_crypto_policy_t](#)
[srtp_crypto_policy_t](#) describes a particular crypto policy that can be applied to an SRTP stream.
- struct [srtp_ssrc_t](#)
An [srtp_ssrc_t](#) represents a particular SSRC value, or a 'wildcard' SSRC.
- struct [srtp_policy_t](#)
represents the policy for an SRTP session.

Macros

- #define `SRTP_MAX_TRAILER_LEN` `SRTP_MAX_TAG_LEN`
the maximum number of octets added by `srtp_protect()`.
- #define `srtp_crypto_policy_set_aes_cm_128_hmac_sha1_80(p)` `srtp_crypto_policy_set_rtp_default(p)`
`srtp_crypto_policy_set_aes_cm_128_hmac_sha1_80()` sets a crypto policy structure to the SRTP default policy for RTP protection.

Typedefs

- typedef `uint32_t srtp_cipher_type_id_t`
A `srtp_cipher_type_id_t` is an identifier for a particular cipher type.
- typedef `uint32_t srtp_auth_type_id_t`
An `srtp_auth_type_id_t` is an identifier for a particular authentication function.
- typedef struct `srtp_crypto_policy_t srtp_crypto_policy_t`
`srtp_crypto_policy_t` describes a particular crypto policy that can be applied to an SRTP stream.
- typedef struct
`srtp_ekt_policy_ctx_t * srtp_ekt_policy_t`
points to an EKT policy
- typedef struct
`srtp_ekt_stream_ctx_t * srtp_ekt_stream_t`
points to EKT stream data
- typedef struct `srtp_policy_t srtp_policy_t`
represents the policy for an SRTP session.
- typedef `srtp_ctx_t * srtp_t`
An `srtp_t` points to an SRTP session structure.
- typedef `srtp_stream_ctx_t * srtp_stream_t`
An `srtp_stream_t` points to an SRTP stream structure.

Enumerations

- enum `srtp_err_status_t` {
`srtp_err_status_ok = 0, srtp_err_status_fail = 1, srtp_err_status_bad_param = 2, srtp_err_status_alloc_fail = 3,`
`srtp_err_status_dealloc_fail = 4, srtp_err_status_init_fail = 5, srtp_err_status_terminus = 6, srtp_err_status_↵`
`auth_fail = 7,`
`srtp_err_status_cipher_fail = 8, srtp_err_status_replay_fail = 9, srtp_err_status_replay_old = 10, srtp_err_↵`
`status_algo_fail = 11,`
`srtp_err_status_no_such_op = 12, srtp_err_status_no_ctx = 13, srtp_err_status_cant_check = 14, srtp_err_↵`
`status_key_expired = 15,`
`srtp_err_status_socket_err = 16, srtp_err_status_signal_err = 17, srtp_err_status_nonce_bad = 18, srtp_err_↵`
`status_read_fail = 19,`
`srtp_err_status_write_fail = 20, srtp_err_status_parse_err = 21, srtp_err_status_encode_err = 22, srtp_err_↵`
`status_semaphore_err = 23,`
`srtp_err_status_pfkey_err = 24 }`
- enum `srtp_sec_serv_t` { `sec_serv_none = 0, sec_serv_conf = 1, sec_serv_auth = 2, sec_serv_conf_and_auth =`
`3 }`
`srtp_sec_serv_t` describes a set of security services.

- enum `srtp_ssrc_type_t` { `ssrc_undefined` = 0, `ssrc_specific` = 1, `ssrc_any_inbound` = 2, `ssrc_any_outbound` = 3 }
srtp_ssrc_type_t describes the type of an SSRC.

Functions

- `srtp_err_status_t srtp_init` (void)
srtp_init() initializes the srtp library.
- `srtp_err_status_t srtp_shutdown` (void)
srtp_shutdown() de-initializes the srtp library.
- `srtp_err_status_t srtp_protect` (`srtp_t` ctx, void *rtp_hdr, int *len_ptr)
srtp_protect() is the Secure RTP sender-side packet processing function.
- `srtp_err_status_t srtp_unprotect` (`srtp_t` ctx, void *srtp_hdr, int *len_ptr)
srtp_unprotect() is the Secure RTP receiver-side packet processing function.
- `srtp_err_status_t srtp_create` (`srtp_t` *session, const `srtp_policy_t` *policy)
srtp_create() allocates and initializes an SRTTP session.
- `srtp_err_status_t srtp_add_stream` (`srtp_t` session, const `srtp_policy_t` *policy)
srtp_add_stream() allocates and initializes an SRTTP stream within a given SRTTP session.
- `srtp_err_status_t srtp_remove_stream` (`srtp_t` session, unsigned int ssrc)
srtp_remove_stream() deallocates an SRTTP stream.
- `srtp_err_status_t srtp_update` (`srtp_t` session, const `srtp_policy_t` *policy)
srtp_update() updates all streams in the session.
- `srtp_err_status_t srtp_update_stream` (`srtp_t` session, const `srtp_policy_t` *policy)
srtp_update_stream() updates a SRTTP stream.
- void `srtp_crypto_policy_set_rtp_default` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_rtp_default() sets a crypto policy structure to the SRTTP default policy for RTP protection.
- void `srtp_crypto_policy_set_rtcp_default` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_rtcp_default() sets a crypto policy structure to the SRTTP default policy for RTCP protection.
- void `srtp_crypto_policy_set_aes_cm_128_hmac_sha1_32` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_aes_cm_128_hmac_sha1_32() sets a crypto policy structure to a short-authentication tag policy
- void `srtp_crypto_policy_set_aes_cm_128_null_auth` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_aes_cm_128_null_auth() sets a crypto policy structure to an encryption-only policy
- void `srtp_crypto_policy_set_null_cipher_hmac_sha1_80` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_null_cipher_hmac_sha1_80() sets a crypto policy structure to an authentication-only policy
- void `srtp_crypto_policy_set_null_cipher_hmac_null` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_null_cipher_hmac_null() sets a crypto policy structure to use no encryption or authentication.
- void `srtp_crypto_policy_set_aes_cm_256_hmac_sha1_80` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_aes_cm_256_hmac_sha1_80() sets a crypto policy structure to a encryption and authentication policy using AES-256 for RTP protection.
- void `srtp_crypto_policy_set_aes_cm_256_hmac_sha1_32` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_aes_cm_256_hmac_sha1_32() sets a crypto policy structure to a short-authentication tag policy using AES-256 encryption.
- void `srtp_crypto_policy_set_aes_cm_256_null_auth` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_aes_cm_256_null_auth() sets a crypto policy structure to an encryption-only policy
- void `srtp_crypto_policy_set_aes_gcm_128_8_auth` (`srtp_crypto_policy_t` *p)
srtp_crypto_policy_set_aes_gcm_128_8_auth() sets a crypto policy structure to an AEAD encryption policy.
- void `srtp_crypto_policy_set_aes_gcm_256_8_auth` (`srtp_crypto_policy_t` *p)

- srtp_crypto_policy_set_aes_gcm_256_8_auth()* sets a crypto policy structure to an AEAD encryption policy
- void `srtp_crypto_policy_set_aes_gcm_128_8_only_auth (srtp_crypto_policy_t *p)`
srtp_crypto_policy_set_aes_gcm_128_8_only_auth() sets a crypto policy structure to an AEAD authentication-only policy
- void `srtp_crypto_policy_set_aes_gcm_256_8_only_auth (srtp_crypto_policy_t *p)`
srtp_crypto_policy_set_aes_gcm_256_8_only_auth() sets a crypto policy structure to an AEAD authentication-only policy
- void `srtp_crypto_policy_set_aes_gcm_128_16_auth (srtp_crypto_policy_t *p)`
srtp_crypto_policy_set_aes_gcm_128_16_auth() sets a crypto policy structure to an AEAD encryption policy.
- void `srtp_crypto_policy_set_aes_gcm_256_16_auth (srtp_crypto_policy_t *p)`
srtp_crypto_policy_set_aes_gcm_256_16_auth() sets a crypto policy structure to an AEAD encryption policy
- `srtp_err_status_t srtp_dealloc (srtp_t s)`
srtp_dealloc() deallocates storage for an SRTP session context.
- `srtp_err_status_t srtp_crypto_policy_set_from_profile_for_rtp (srtp_crypto_policy_t *policy, srtp_profile_t profile)`
srtp_crypto_policy_set_from_profile_for_rtp() sets a crypto policy structure to the appropriate value for RTP based on an `srtp_profile_t`
- `srtp_err_status_t srtp_crypto_policy_set_from_profile_for_rtcp (srtp_crypto_policy_t *policy, srtp_profile_t profile)`
srtp_crypto_policy_set_from_profile_for_rtcp() sets a crypto policy structure to the appropriate value for RTCP based on an `srtp_profile_t`
- unsigned int `srtp_profile_get_master_key_length (srtp_profile_t profile)`
returns the master key length for a given SRTP profile
- unsigned int `srtp_profile_get_master_salt_length (srtp_profile_t profile)`
returns the master salt length for a given SRTP profile
- void `srtp_append_salt_to_key (unsigned char *key, unsigned int bytes_in_key, unsigned char *salt, unsigned int bytes_in_salt)`
appends the salt to the key

4.1.1 Detailed Description

4.1.2 Macro Definition Documentation

4.1.2.1 #define srtp_crypto_policy_set_aes_cm_128_hmac_sha1_80(p) srtp_crypto_policy_set_rtp_default(p)

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function `srtp_crypto_policy_set_aes_cm_128_hmac_sha1_80()` is a synonym for `srtp_crypto_policy_set_rtp_default()`. It conforms to the naming convention used in RFC 4568 (SDP Security Descriptions for Media Streams).

Returns

void.

4.1.2.2 #define SRTP_MAX_TRAILER_LEN SRTP_MAX_TAG_LEN

`SRTP_MAX_TRAILER_LEN` is the maximum length of the SRTP trailer (authentication tag and MKI) supported by libSRTP. This value is the maximum number of octets that will be added to an RTP packet by `srtp_protect()`.

4.1.3 Typedef Documentation

4.1.3.1 `typedef uint32_t srtp_auth_type_id_t`

An `srtp_auth_type_id_t` is an integer that represents a particular authentication function type, e.g. HMAC-SHA1. A `SRTP_NULL_AUTH` is available; this authentication function performs no computation, and can be selected to indicate that no authentication is to take place.

4.1.3.2 `typedef uint32_t srtp_cipher_type_id_t`

A `srtp_cipher_type_id_t` is an integer that represents a particular cipher type, e.g. the Advanced Encryption Standard (AES). A `SRTP_NULL_CIPHER` is available; this cipher leaves the data unchanged, and can be selected to indicate that no encryption is to take place.

4.1.3.3 `typedef struct srtp_crypto_policy_t srtp_crypto_policy_t`

A `srtp_crypto_policy_t` describes a particular cryptographic policy that can be applied to an SRTP or SRTCP stream. An SRTP session policy consists of a list of these policies, one for each SRTP stream in the session.

4.1.3.4 `typedef struct srtp_policy_t srtp_policy_t`

A single `srtp_policy_t` struct represents the policy for a single SRTP stream, and a linked list of these elements represents the policy for an entire SRTP session. Each element contains the SRTP and SRTCP crypto policies for that stream, a pointer to the SRTP master key for that stream, the SSRC describing that stream, or a flag indicating a 'wildcard' SSRC value, and a 'next' field that holds a pointer to the next element in the list of policy elements, or NULL if it is the last element.

The wildcard value `SSRC_ANY_INBOUND` matches any SSRC from an inbound stream that for which there is no explicit SSRC entry in another policy element. Similarly, the value `SSRC_ANY_OUTBOUND` will matches any SSRC from an outbound stream that does not appear in another policy element. Note that wildcard SSRCs cannot be used to match both inbound and outbound traffic. This restriction is intentional, and it allows libSRTP to ensure that no security lapses result from accidental re-use of SSRC values during key sharing.

Warning

The final element of the list **must** have its 'next' pointer set to NULL.

4.1.3.5 `typedef srtp_stream_ctx_t* srtp_stream_t`

The typedef `srtp_stream_t` is a pointer to a structure that represents an SRTP stream. This datatype is intentionally opaque in order to separate the interface from the implementation.

An SRTP stream consists of all of the traffic sent to an SRTP session by a single participant. A session can be viewed as a set of streams.

4.1.3.6 typedef srtp_ctx_t* srtp_t

The typedef `srtp_t` is a pointer to a structure that represents an SRTP session. This datatype is intentionally opaque in order to separate the interface from the implementation.

An SRTP session consists of all of the traffic sent to the RTP and RTCP destination transport addresses, using the RTP/SAVP (Secure Audio/Video Profile). A session can be viewed as a set of SRTP streams, each of which originates with a different participant.

4.1.4 Enumeration Type Documentation

4.1.4.1 enum srtp_err_status_t

Enumerator

`srtp_err_status_ok` nothing to report
`srtp_err_status_fail` unspecified failure
`srtp_err_status_bad_param` unsupported parameter
`srtp_err_status_alloc_fail` couldn't allocate memory
`srtp_err_status_dealloc_fail` couldn't deallocate properly
`srtp_err_status_init_fail` couldn't initialize
`srtp_err_status_terminus` can't process as much data as requested
`srtp_err_status_auth_fail` authentication failure
`srtp_err_status_cipher_fail` cipher failure
`srtp_err_status_replay_fail` replay check failed (bad index)
`srtp_err_status_replay_old` replay check failed (index too old)
`srtp_err_status_algo_fail` algorithm failed test routine
`srtp_err_status_no_such_op` unsupported operation
`srtp_err_status_no_ctx` no appropriate context found
`srtp_err_status_cant_check` unable to perform desired validation
`srtp_err_status_key_expired` can't use key any more
`srtp_err_status_socket_err` error in use of socket
`srtp_err_status_signal_err` error in use POSIX signals
`srtp_err_status_nonce_bad` nonce check failed
`srtp_err_status_read_fail` couldn't read data
`srtp_err_status_write_fail` couldn't write data
`srtp_err_status_parse_err` error parsing data
`srtp_err_status_encode_err` error encoding data
`srtp_err_status_semaphore_err` error while using semaphores
`srtp_err_status_pfkey_err` error while using pfkey

4.1.4.2 enum srtp_sec_serv_t

A `srtp_sec_serv_t` enumeration is used to describe the particular security services that will be applied by a particular crypto policy (or other mechanism).

Enumerator

- `sec_serv_none`** no services
- `sec_serv_conf`** confidentiality
- `sec_serv_auth`** authentication
- `sec_serv_conf_and_auth`** confidentiality and authentication

4.1.4.3 enum srtp_ssrc_type_t

An `srtp_ssrc_type_t` enumeration is used to indicate a type of SSRC. See [srtp_policy_t](#) for more information.

Enumerator

- `ssrc_undefined`** Indicates an undefined SSRC type.
- `ssrc_specific`** Indicates a specific SSRC value
- `ssrc_any_inbound`** Indicates any inbound SSRC value (i.e. a value that is used in the function [srtp_unprotect\(\)](#))
- `ssrc_any_outbound`** Indicates any outbound SSRC value (i.e. a value that is used in the function [srtp_protect\(\)](#))

4.1.5 Function Documentation

4.1.5.1 srtp_err_status_t srtp_add_stream (srtp_t session, const srtp_policy_t * policy)

The function call `srtp_add_stream(session, policy)` allocates and initializes a new SRTP stream within a given, previously created session, applying the policy given as the other argument to that stream.

Returns

values:

- `srtp_err_status_ok` if stream creation succeeded.
- `srtp_err_status_alloc_fail` if stream allocation failed
- `srtp_err_status_init_fail` if stream initialization failed.

4.1.5.2 void srtp_append_salt_to_key (unsigned char * key, unsigned int bytes_in_key, unsigned char * salt, unsigned int bytes_in_salt)

The function call `srtp_append_salt_to_key(k, klen, s, slen)` copies the string `s` to the location at `klen` bytes following the location `k`.

Warning

There must be at least `bytes_in_salt + bytes_in_key` bytes available at the location pointed to by `key`.

4.1.5.3 `srtp_err_status_t srtp_create (srtp_t * session, const srtp_policy_t * policy)`

The function call `srtp_create(session, policy)` allocates and initializes an SRTP session context, applying the given policy.

Parameters

<i>session</i>	is a pointer to the SRTP session to which the policy is to be added.
<i>policy</i>	is the srtp_policy_t struct that describes the policy for the session. The struct may be a single element, or it may be the head of a list, in which case each element of the list is processed. It may also be NULL, in which case streams should be added later using srtp_add_stream() . The final element of the list must have its 'next' field set to NULL.

Returns

- `srtp_err_status_ok` if creation succeeded.
- `srtp_err_status_alloc_fail` if allocation failed.
- `srtp_err_status_init_fail` if initialization failed.

4.1.5.4 `void srtp_crypto_policy_set_aes_cm_128_hmac_sha1_32 (srtp_crypto_policy_t * p)`**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_cm_128_hmac_sha1_32(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use policy AES_CM_128_HMAC_SHA1_32 as defined in RFC 4568. This policy uses AES-128 Counter Mode encryption and HMAC-SHA1 authentication, with an authentication tag that is only 32 bits long. This length is considered adequate only for protecting audio and video media that use a stateless playback function. See Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Warning

This crypto policy is intended for use in SRTP, but not in SRTCP. It is recommended that a policy that uses longer authentication tags be used for SRTCP. See Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

Returns

void.

4.1.5.5 void srtp_crypto_policy_set_aes_cm_128_null_auth (srtp_crypto_policy_t * p)

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_cm_128_null_auth(&p)` sets the `srtp_crypto_policy_t` at location `p` to use the SRTP default cipher (AES-128 Counter Mode), but to use no authentication method. This policy is NOT RECOMMENDED unless it is unavoidable; see Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the `srtp_crypto_policy_t` datatype.

Warning

This policy is NOT RECOMMENDED for SRTP unless it is unavoidable, and it is NOT RECOMMENDED at all for SRTCP; see Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

Returns

void.

4.1.5.6 void srtp_crypto_policy_set_aes_cm_256_hmac_sha1_32 (srtp_crypto_policy_t * p)**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_cm_256_hmac_sha1_32(&p)` sets the `srtp_crypto_policy_t` at location `p` to use policy `AES_CM_256_HMAC_SHA1_32` as defined in draft-ietf-avt-srtp-big-aes-03.txt. This policy uses AES-256 Counter Mode encryption and HMAC-SHA1 authentication, with an authentication tag that is only 32 bits long. This length is considered adequate only for protecting audio and video media that use a stateless playback function. See Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the `srtp_crypto_policy_t` datatype.

Warning

This crypto policy is intended for use in SRTP, but not in SRTCP. It is recommended that a policy that uses longer authentication tags be used for SRTCP. See Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

Returns

void.

4.1.5.7 void srtp_crypto_policy_set_aes_cm_256_hmac_sha1_80 (srtp_crypto_policy_t * p)

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_cm_256_hmac_sha1_80(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use policy `AES_CM_256_HMAC_SHA1_80` as defined in `draft-ietf-avt-srtp-big-aes-03.txt`. This policy uses AES-256 Counter Mode encryption and HMAC-SHA1 authentication, with an 80 bit authentication tag.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.8 void srtp_crypto_policy_set_aes_cm_256_null_auth (srtp_crypto_policy_t * p)**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_cm_256_null_auth(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use the SRTP default cipher (AES-256 Counter Mode), but to use no authentication method. This policy is NOT RECOMMENDED unless it is unavoidable; see Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Warning

This policy is NOT RECOMMENDED for SRTP unless it is unavoidable, and it is NOT RECOMMENDED at all for SRTCP; see Section 7.5 of RFC 3711 (<http://www.ietf.org/rfc/rfc3711.txt>).

Returns

void.

4.1.5.9 void srtp_crypto_policy_set_aes_gcm_128_16_auth (srtp_crypto_policy_t * p)**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_gcm_128_16_auth(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use the SRTP default cipher (AES-128 Galois Counter Mode) with 16 octet auth tag. This policy applies confidentiality and authentication to both the RTP and RTCP packets.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.10 void srtp_crypto_policy_set_aes_gcm_128_8_auth (srtp_crypto_policy_t * p)**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_gcm_128_8_auth(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use the SRTP default cipher (AES-128 Galois Counter Mode) with 8 octet auth tag. This policy applies confidentiality and authentication to both the RTP and RTCP packets.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.11 void srtp_crypto_policy_set_aes_gcm_128_8_only_auth (srtp_crypto_policy_t * p)**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_gcm_128_8_only_auth(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use the SRTP default cipher (AES-128 Galois Counter Mode) with 8 octet auth tag. This policy applies confidentiality and authentication to the RTP packets, but only authentication to the RTCP packets.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.12 void srtp_crypto_policy_set_aes_gcm_256_16_auth (srtp_crypto_policy_t * p)

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_gcm_256_16_auth(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use the SRTP default cipher (AES-256 Galois Counter Mode) with 16 octet auth tag. This policy applies confidentiality and authentication to both the RTP and RTCP packets.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.13 void srtp_crypto_policy_set_aes_gcm_256_8_auth (srtp_crypto_policy_t * p)**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_gcm_256_8_auth(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use the SRTP default cipher (AES-256 Galois Counter Mode) with 8 octet auth tag. This policy applies confidentiality and authentication to both the RTP and RTCP packets.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.14 void srtp_crypto_policy_set_aes_gcm_256_8_only_auth (srtp_crypto_policy_t * p)**Parameters**

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_aes_gcm_256_8_only_auth(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use the SRTP default cipher (AES-256 Galois Counter Mode) with 8 octet auth tag. This policy applies confidentiality and authentication to the RTP packets, but only authentication to the RTCP packets.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.15 `srtp_err_status_t srtp_crypto_policy_set_from_profile_for_rtcp (srtp_crypto_policy_t * policy, srtp_profile_t profile)`

Parameters

<code>p</code>	is a pointer to the policy structure to be set
----------------	--

The function call `srtp_crypto_policy_set_rtcp_default(&policy, profile)` sets the `srtp_crypto_policy_t` at location `policy` to the policy for RTCP protection, as defined by the `srtp_profile_t` `profile`.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the `srtp_crypto_policy_t` datatype.

Returns

values

- `srtp_err_status_ok` no problems were encountered
- `srtp_err_status_bad_param` the profile is not supported

4.1.5.16 `srtp_err_status_t srtp_crypto_policy_set_from_profile_for_rtp (srtp_crypto_policy_t * policy, srtp_profile_t profile)`

Parameters

<code>p</code>	is a pointer to the policy structure to be set
----------------	--

The function call `srtp_crypto_policy_set_rtp_default(&policy, profile)` sets the `srtp_crypto_policy_t` at location `policy` to the policy for RTP protection, as defined by the `srtp_profile_t` `profile`.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the `srtp_crypto_policy_t` datatype.

Returns

values

- `srtp_err_status_ok` no problems were encountered
- `srtp_err_status_bad_param` the profile is not supported

4.1.5.17 `void srtp_crypto_policy_set_null_cipher_hmac_null (srtp_crypto_policy_t * p)`

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_null_cipher_hmac_null(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use no encryption and no authentication. This policy should only be used for testing and troubleshooting.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Warning

This policy is NOT RECOMMENDED for SRTP unless there is a requirement to forego encryption and authentication.

Returns

void.

4.1.5.18 void srtp_crypto_policy_set_null_cipher_hmac_sha1_80 (srtp_crypto_policy_t * p)

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_null_cipher_hmac_sha1_80(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to use HMAC-SHA1 with an 80 bit authentication tag to provide message authentication, but to use no encryption. This policy is NOT RECOMMENDED for SRTP unless there is a requirement to forego encryption.

This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Warning

This policy is NOT RECOMMENDED for SRTP unless there is a requirement to forego encryption.

Returns

void.

4.1.5.19 void srtp_crypto_policy_set_rtcp_default (srtp_crypto_policy_t * p)

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `srtp_crypto_policy_set_rtcp_default(&p)` sets the [srtp_crypto_policy_t](#) at location `p` to the SRTP default policy for RTCP protection, as defined in the specification. This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the [srtp_crypto_policy_t](#) datatype.

Returns

void.

4.1.5.20 void srtp_crypto_policy_set_rtp_default (srtp_crypto_policy_t * p)

Parameters

<i>p</i>	is a pointer to the policy structure to be set
----------	--

The function call `crypto_policy_set_rtp_default(&p)` sets the `crypto_policy_t` at location `p` to the SRTP default policy for RTP protection, as defined in the specification. This function is a convenience that helps to avoid dealing directly with the policy data structure. You are encouraged to initialize policy elements with this function call. Doing so may allow your code to be forward compatible with later versions of libSRTP that include more elements in the `crypto_policy_t` datatype.

Returns

void.

4.1.5.21 srtp_err_status_t srtp_dealloc (srtp_t s)

The function call `srtp_dealloc(s)` deallocates storage for the SRTP session context `s`. This function should be called no more than one time for each of the contexts allocated by the function [srtp_create\(\)](#).

Parameters

<i>s</i>	is the <code>srtp_t</code> for the session to be deallocated.
----------	---

Returns

- `srtp_err_status_ok` if there no problems.
- `srtp_err_status_dealloc_fail` a memory deallocation failure occurred.

4.1.5.22 srtp_err_status_t srtp_init (void)

Warning

This function **must** be called before any other srtp functions.

4.1.5.23 srtp_err_status_t srtp_protect (srtp_t ctx, void * rtp_hdr, int * len_ptr)

The function call `srtp_protect(ctx, rtp_hdr, len_ptr)` applies SRTP protection to the RTP packet `rtp_hdr` (which has length `*len_ptr`) using the SRTP context `ctx`. If `srtp_err_status_ok` is returned, then `rtp_hdr` points to the resulting SRTP packet and `*len_ptr` is the number of octets in that packet; otherwise, no assumptions should be made about the value of either data elements.

The sequence numbers of the RTP packets presented to this function need not be consecutive, but they **must** be out of order by less than $2^{15} = 32,768$ packets.

Warning

This function assumes that it can write the authentication tag into the location in memory immediately following the RTP packet, and assumes that the RTP packet is aligned on a 32-bit boundary.

This function assumes that it can write `SRTP_MAX_TRAILER_LEN` into the location in memory immediately following the RTP packet. Callers **MUST** ensure that this much writable memory is available in the buffer that holds the RTP packet.

Parameters

<i>ctx</i>	is the SRTP context to use in processing the packet.
<i>rtp_hdr</i>	is a pointer to the RTP packet (before the call); after the function returns, it points to the srtp packet.
<i>len_ptr</i>	is a pointer to the length in octets of the complete RTP packet (header and body) before the function call, and of the complete SRTP packet after the call, if <code>srtp_err_status_ok</code> was returned. Otherwise, the value of the data to which it points is undefined.

Returns

- `srtp_err_status_ok` no problems
- `srtp_err_status_replay_fail` rtp sequence number was non-increasing
- *other* failure in cryptographic mechanisms

4.1.5.24 srtp_err_status_t srtp_remove_stream (srtp_t session, unsigned int ssrc)

The function call `srtp_remove_stream(session, ssrc)` removes the SRTP stream with the SSRC value `ssrc` from the SRTP session context given by the argument `session`.

Parameters

<i>session</i>	is the SRTP session from which the stream will be removed.
<i>ssrc</i>	is the SSRC value of the stream to be removed.

Warning

Wildcard SSRC values cannot be removed from a session.

Returns

- `srtp_err_status_ok` if the stream deallocation succeeded.
- [other] otherwise.

4.1.5.25 `srtp_err_status_t srtp_shutdown (void)`**Warning**

No `srtp` functions may be called after calling this function.

4.1.5.26 `srtp_err_status_t srtp_unprotect (srtp_t ctx, void * srtp_hdr, int * len_ptr)`

The function call `srtp_unprotect(ctx, srtp_hdr, len_ptr)` verifies the Secure RTP protection of the SRTP packet pointed to by `srtp_hdr` (which has length `*len_ptr`), using the SRTP context `ctx`. If `srtp_err_status_ok` is returned, then `srtp_hdr` points to the resulting RTP packet and `*len_ptr` is the number of octets in that packet; otherwise, no assumptions should be made about the value of either data elements.

The sequence numbers of the RTP packets presented to this function need not be consecutive, but they **must** be out of order by less than $2^{15} = 32,768$ packets.

Warning

This function assumes that the SRTP packet is aligned on a 32-bit boundary.

Parameters

<i>ctx</i>	is the SRTP session which applies to the particular packet.
<i>srtp_hdr</i>	is a pointer to the header of the SRTP packet (before the call). after the function returns, it points to the rtp packet if <code>srtp_err_status_ok</code> was returned; otherwise, the value of the data to which it points is undefined.

<i>len_ptr</i>	is a pointer to the length in octets of the complete srtp packet (header and body) before the function call, and of the complete rtp packet after the call, if srtp_err_status_ok was returned. Otherwise, the value of the data to which it points is undefined.
----------------	---

Returns

- srtp_err_status_ok if the RTP packet is valid.
- srtp_err_status_auth_fail if the SRTP packet failed the message authentication check.
- srtp_err_status_replay_fail if the SRTP packet is a replay (e.g. packet has already been processed and accepted).
- [other] if there has been an error in the cryptographic mechanisms.

4.1.5.27 srtp_err_status_t srtp_update (srtp_t session, const srtp_policy_t * policy)

The function call srtp_update(session, policy) updates all the streams in the session applying the given policy and key. The existing ROC value of all streams will be preserved.

Parameters

<i>session</i>	is the SRTP session that contains the streams to be updated.
<i>policy</i>	is the srtp_policy_t struct that describes the policy for the session. The struct may be a single element, or it may be the head of a list, in which case each element of the list is processed. The final element of the list must have its 'next' field set to NULL.

Returns

- srtp_err_status_ok if stream creation succeeded.
- srtp_err_status_alloc_fail if stream allocation failed
- srtp_err_status_init_fail if stream initialization failed.
- [other] otherwise.

4.1.5.28 srtp_err_status_t srtp_update_stream (srtp_t session, const srtp_policy_t * policy)

The function call srtp_update_stream(session, policy) updates the stream(s) in the session that match applying the given policy and key. The existing ROC value of all stream(s) will be preserved.

Parameters

<i>session</i>	is the SRTP session that contains the streams to be updated.
<i>policy</i>	is the srtp_policy_t struct that describes the policy for the session.

Returns

- `srtp_err_status_ok` if stream creation succeeded.
- `srtp_err_status_alloc_fail` if stream allocation failed
- `srtp_err_status_init_fail` if stream initialization failed.
- [other] otherwise.

4.2 Secure RTCP

Secure RTCP functions are used to protect RTCP traffic.

Functions

- [srtp_err_status_t srtp_protect_rtcp](#) ([srtp_t](#) ctx, void *rtcp_hdr, int *pkt_octet_len)
srtp_protect_rtcp() is the Secure RTCP sender-side packet processing function.
- [srtp_err_status_t srtp_unprotect_rtcp](#) ([srtp_t](#) ctx, void *srtp_hdr, int *pkt_octet_len)
srtp_unprotect_rtcp() is the Secure RTCP receiver-side packet processing function.

4.2.1 Detailed Description

RTCP is the control protocol for RTP. libSRTP protects RTCP traffic in much the same way as it does RTP traffic. The function [srtp_protect_rtcp\(\)](#) applies cryptographic protections to outbound RTCP packets, and [srtp_unprotect_rtcp\(\)](#) verifies the protections on inbound RTCP packets.

A note on the naming convention: [srtp_protect_rtcp\(\)](#) has an [srtp_t](#) as its first argument, and thus has 'srtp_' as its prefix. The trailing '_rtcp' indicates the protocol on which it acts.

4.2.2 Function Documentation

4.2.2.1 [srtp_err_status_t srtp_protect_rtcp](#) ([srtp_t](#) ctx, void * rtcp_hdr, int * pkt_octet_len)

The function call [srtp_protect_rtcp](#)(ctx, rtp_hdr, len_ptr) applies SRTCP protection to the RTCP packet rtp_hdr (which has length *len_ptr) using the SRTP session context ctx. If [srtp_err_status_ok](#) is returned, then rtp_hdr points to the resulting SRTCP packet and *len_ptr is the number of octets in that packet; otherwise, no assumptions should be made about the value of either data elements.

Warning

This function assumes that it can write the authentication tag into the location in memory immediately following the RTCP packet, and assumes that the RTCP packet is aligned on a 32-bit boundary.

This function assumes that it can write `SRTP_MAX_TRAILER_LEN+4` into the location in memory immediately following the RTCP packet. Callers MUST ensure that this much writable memory is available in the buffer that holds the RTCP packet.

Parameters

<i>ctx</i>	is the SRTP context to use in processing the packet.
------------	--

<i>rtcp_hdr</i>	is a pointer to the RTCP packet (before the call); after the function returns, it points to the srtp packet.
<i>pkt_octet_len</i>	is a pointer to the length in octets of the complete RTCP packet (header and body) before the function call, and of the complete SRTCP packet after the call, if <code>srtp_err_status_ok</code> was returned. Otherwise, the value of the data to which it points is undefined.

Returns

- `srtp_err_status_ok` if there were no problems.
- [other] if there was a failure in the cryptographic mechanisms.

4.2.2.2 `srtp_err_status_t srtp_unprotect_rtcp (srtp_t ctx, void * rtcp_hdr, int * pkt_octet_len)`

The function call `srtp_unprotect_rtcp(ctx, rtcp_hdr, len_ptr)` verifies the Secure RTCP protection of the SRTCP packet pointed to by `rtcp_hdr` (which has length `*len_ptr`), using the SRTP session context `ctx`. If `srtp_err_status_ok` is returned, then `rtcp_hdr` points to the resulting RTCP packet and `*len_ptr` is the number of octets in that packet; otherwise, no assumptions should be made about the value of either data elements.

Warning

This function assumes that the SRTCP packet is aligned on a 32-bit boundary.

Parameters

<i>ctx</i>	is a pointer to the <code>srtp_t</code> which applies to the particular packet.
<i>rtcp_hdr</i>	is a pointer to the header of the SRTCP packet (before the call). After the function returns, it points to the rtp packet if <code>srtp_err_status_ok</code> was returned; otherwise, the value of the data to which it points is undefined.
<i>pkt_octet_len</i>	is a pointer to the length in octets of the complete SRTCP packet (header and body) before the function call, and of the complete rtp packet after the call, if <code>srtp_err_status_ok</code> was returned. Otherwise, the value of the data to which it points is undefined.

Returns

- `srtp_err_status_ok` if the RTCP packet is valid.
- `srtp_err_status_auth_fail` if the SRTCP packet failed the message authentication check.
- `srtp_err_status_replay_fail` if the SRTCP packet is a replay (e.g. has already been processed and accepted).
- [other] if there has been an error in the cryptographic mechanisms.

4.3 data associated to a SRTP session.

Store custom user data within a SRTP session.

Functions

- void [srtp_set_user_data](#) ([srtp_t](#) ctx, void *data)
[srtp_set_user_data\(\)](#) stores the given pointer into the SRTP session for later retrieval.
- void * [srtp_get_user_data](#) ([srtp_t](#) ctx)
[srtp_get_user_data\(\)](#) retrieves the pointer to the custom data previously stored with [srtp_set_user_data\(\)](#).

4.3.1 Detailed Description

4.3.2 Function Documentation

4.3.2.1 void* srtp_get_user_data (srtp_t ctx)

This function is mostly useful for retrieving data associated to a SRTP session when an event fires. The user can then get such a custom data by calling this function with the session field of the [srtp_event_data_t](#) struct as argument.

Parameters

<i>ctx</i>	is the srtp_t context in which the given data pointer was stored.
------------	---

Returns

void* pointer to the user data.

4.3.2.2 void srtp_set_user_data (srtp_t ctx, void * data)

Parameters

<i>ctx</i>	is the srtp_t context in which the given data pointer is stored.
<i>data</i>	is a pointer to the custom information (struct, function, etc) associated with the SRTP session.

Returns

void.

4.4 SRTP events and callbacks

libSRTP can use a user-provided callback function to handle events.

Data Structures

- struct [srtp_event_data_t](#)
srtp_event_data_t is the structure passed as a callback to the event handler function

Typedefs

- typedef struct [srtp_event_data_t](#) [srtp_event_data_t](#)
srtp_event_data_t is the structure passed as a callback to the event handler function
- typedef void([srtp_event_handler_func_t](#))(srtp_event_data_t *data)
srtp_event_handler_func_t is the function prototype for the event handler.

Enumerations

- enum [srtp_event_t](#) { [event_src_collision](#), [event_key_soft_limit](#), [event_key_hard_limit](#), [event_packet_index_limit](#) }
srtp_event_t defines events that need to be handled

Functions

- [srtp_err_status_t](#) [srtp_install_event_handler](#) ([srtp_event_handler_func_t](#) func)
sets the event handler to the function supplied by the caller.
- const char * [srtp_get_version_string](#) (void)
Returns the version string of the library.
- unsigned int [srtp_get_version](#) (void)
Returns the numeric representation of the library version.
- [srtp_err_status_t](#) [srtp_set_debug_module](#) (char *mod_name, int v)
srtp_set_debug_module(mod_name, v)
- [srtp_err_status_t](#) [srtp_list_debug_modules](#) (void)
srtp_list_debug_modules() outputs a list of debugging modules

4.4.1 Detailed Description

libSRTP allows a user to provide a callback function to handle events that need to be dealt with outside of the data plane (see the enum [srtp_event_t](#) for a description of these events). Dealing with these events is not a strict necessity; they are not security-critical, but the application may suffer if they are not handled. The function [srtp_set_event_handler\(\)](#) is used to provide the callback function.

A default event handler that merely reports on the events as they happen is included. It is also possible to set the event handler function to NULL, in which case all events will just be silently ignored.

4.4.2 Typedef Documentation

4.4.2.1 typedef struct srtp_event_data_t srtp_event_data_t

The struct [srtp_event_data_t](#) holds the data passed to the event handler function.

4.4.2.2 typedef void(srtp_event_handler_func_t)(srtp_event_data_t *data)

The typedef [srtp_event_handler_func_t](#) is the prototype for the event handler function. It has as its only argument an [srtp_event_data_t](#) which describes the event that needs to be handled. There can only be a single, global handler for all events in libSRTP.

4.4.3 Enumeration Type Documentation

4.4.3.1 enum srtp_event_t

The enum [srtp_event_t](#) defines events that need to be handled outside the 'data plane', such as SSRC collisions and key expirations.

When a key expires or the maximum number of packets has been reached, an SRTP stream will enter an 'expired' state in which no more packets can be protected or unprotected. When this happens, it is likely that you will want to either deallocate the stream (using [srtp_remove_stream\(\)](#)), and possibly allocate a new one.

When an SRTP stream expires, the other streams in the same session are unaffected, unless key sharing is used by that stream. In the latter case, all of the streams in the session will expire.

Enumerator

event_ssrc_collision An SSRC collision occurred.

event_key_soft_limit An SRTP stream reached the soft key usage limit and will expire soon.

event_key_hard_limit An SRTP stream reached the hard key usage limit and has expired.

event_packet_index_limit An SRTP stream reached the hard packet limit (2^{48} packets).

4.4.4 Function Documentation

4.4.4.1 srtp_err_status_t srtp_install_event_handler (srtp_event_handler_func_t func)

The function call [srtp_install_event_handler\(func\)](#) sets the event handler function to the value *func*. The value NULL is acceptable as an argument; in this case, events will be ignored rather than handled.

Parameters

<i>func</i>	is a pointer to a fuction that takes an srtp_event_data_t pointer as an argument and returns void. This function will be used by libSRTP to handle events.
-------------	--

4.4.4.2 `srtp_err_status_t srtp_set_debug_module (char * mod_name, int v)`

sets dynamic debugging to the value v (0 for off, 1 for on) for the debug module with the name mod_name

returns err_status_ok on success, err_status_fail otherwise

4.5 Error Codes

Error status codes are represented by the enumeration `srtp_err_status_t`.

4.6 Cryptographic Kernel

Modules

- [Ciphers](#)

A generic cipher type enables cipher agility, that is, the ability to write code that runs with multiple cipher types. Ciphers can be used through the crypto kernel, or can be accessed directly, if need be.

4.6.1 Detailed Description

All of the cryptographic functions are contained in a kernel.

4.7 Ciphers

A generic cipher type enables cipher agility, that is, the ability to write code that runs with multiple cipher types. Ciphers can be used through the crypto kernel, or can be accessed directly, if need be.

Functions

- `err_status_t cipher_type_alloc` (`cipher_type_t *ctype`, `cipher_t **cipher`, unsigned `key_len`)
Allocates a cipher of a particular type.
- `err_status_t cipher_init` (`cipher_t *cipher`, const `uint8_t *key`)
Initialized a cipher to use a particular key. May be invoked more than once on the same cipher.
- `err_status_t cipher_set_iv` (`cipher_t *cipher`, void `*iv`)
Sets the initialization vector of a given cipher.
- `err_status_t cipher_encrypt` (`cipher_t *cipher`, void `*buf`, unsigned int `*len`)
Encrypts a buffer with a given cipher.
- `err_status_t cipher_output` (`cipher_t *c`, `uint8_t *buffer`, int `num_octets_to_output`)
Sets a buffer to the keystream generated by the cipher.
- `err_status_t cipher_dealloc` (`cipher_t *cipher`)
Deallocates a cipher.

4.7.1 Detailed Description

4.7.2 Function Documentation

4.7.2.1 `err_status_t cipher_dealloc (cipher_t * cipher)`

Warning

May be implemented as a macro.

4.7.2.2 `err_status_t cipher_encrypt (cipher_t * cipher, void * buf, unsigned int * len)`

Warning

May be implemented as a macro.

4.7.2.3 `err_status_t cipher_init (cipher_t * cipher, const uint8_t * key)`

Warning

May be implemented as a macro.

4.7.2.4 `err_status_t cipher_output (cipher_t * c, uint8_t * buffer, int num_octets_to_output)`

Warning

May be implemented as a macro.

4.7.2.5 `err_status_t cipher_set_iv (cipher_t * cipher, void * iv)`

Warning

May be implemented as a macro.

4.7.2.6 `err_status_t cipher_type_alloc (cipher_type_t * ctype, cipher_t ** cipher, unsigned key_len)`

Warning

May be implemented as a macro.

Chapter 5

Data Structure Documentation

5.1 `srtcp_hdr_t` Struct Reference

The documentation for this struct was generated from the following file:

- `srtcp.h`

5.2 `srtcp_trailer_t` Struct Reference

The documentation for this struct was generated from the following file:

- `srtcp.h`

5.3 `srtcp_crypto_policy_t` Struct Reference

[srtcp_crypto_policy_t](#) describes a particular crypto policy that can be applied to an SRTP stream.

Data Fields

- [srtcp_cipher_type_id_t](#) `cipher_type`
- `int` `cipher_key_len`
- [srtcp_auth_type_id_t](#) `auth_type`
- `int` `auth_key_len`
- `int` `auth_tag_len`
- [srtcp_sec_serv_t](#) `sec_serv`

5.3.1 Detailed Description

A [srtp_crypto_policy_t](#) describes a particular cryptographic policy that can be applied to an SRTP or SRTCP stream. An SRTP session policy consists of a list of these policies, one for each SRTP stream in the session.

5.3.2 Field Documentation

5.3.2.1 `int srtp_crypto_policy_t::auth_key_len`

The length of the authentication function key in octets.

5.3.2.2 `int srtp_crypto_policy_t::auth_tag_len`

The length of the authentication tag in octets.

5.3.2.3 `srtp_auth_type_id_t srtp_crypto_policy_t::auth_type`

An integer representing the authentication function.

5.3.2.4 `int srtp_crypto_policy_t::cipher_key_len`

The length of the cipher key in octets.

5.3.2.5 `srtp_cipher_type_id_t srtp_crypto_policy_t::cipher_type`

An integer representing the type of cipher.

5.3.2.6 `srtp_sec_serv_t srtp_crypto_policy_t::sec_serv`

The flag indicating the security services to be applied.

The documentation for this struct was generated from the following file:

- `srtp.h`

5.4 srtp_debug_module_t Struct Reference

The documentation for this struct was generated from the following file:

- err.h

5.5 srtp_event_data_t Struct Reference

[srtp_event_data_t](#) is the structure passed as a callback to the event handler function

Data Fields

- [srtp_t session](#)
- [srtp_stream_t stream](#)
- [srtp_event_t event](#)

5.5.1 Detailed Description

The struct [srtp_event_data_t](#) holds the data passed to the event handler function.

5.5.2 Field Documentation

5.5.2.1 srtp_event_t srtp_event_data_t::event

An enum indicating the type of event.

5.5.2.2 srtp_t srtp_event_data_t::session

The session in which the event happend.

5.5.2.3 srtp_stream_t srtp_event_data_t::stream

The stream in which the event happend.

The documentation for this struct was generated from the following file:

- srtp.h

5.6 srtp_hdr_t Struct Reference

The documentation for this struct was generated from the following file:

- [srtp.h](#)

5.7 srtp_hdr_xtnd_t Struct Reference

The documentation for this struct was generated from the following file:

- [srtp.h](#)

5.8 srtp_policy_t Struct Reference

represents the policy for an SRTP session.

Data Fields

- [srtp_ssrc_t](#) `ssrc`
- [srtp_crypto_policy_t](#) `rtp`
- [srtp_crypto_policy_t](#) `rtcp`
- unsigned char * `key`
- [srtp_ekt_policy_t](#) `ekt`
- unsigned long `window_size`
- int `allow_repeat_tx`
- int * `enc_xtn_hdr`
- int `enc_xtn_hdr_count`
- struct [srtp_policy_t](#) * `next`

5.8.1 Detailed Description

A single [srtp_policy_t](#) struct represents the policy for a single SRTP stream, and a linked list of these elements represents the policy for an entire SRTP session. Each element contains the SRTP and SRTCP crypto policies for that stream, a pointer to the SRTP master key for that stream, the SSRC describing that stream, or a flag indicating a 'wildcard' SSRC value, and a 'next' field that holds a pointer to the next element in the list of policy elements, or NULL if it is the last element.

The wildcard value `SSRC_ANY_INBOUND` matches any SSRC from an inbound stream that for which there is no explicit SSRC entry in another policy element. Similarly, the value `SSRC_ANY_OUTBOUND` will matches any SSRC from an outbound stream that does not appear in another policy element. Note that wildcard SSRCs &b cannot be

used to match both inbound and outbound traffic. This restriction is intentional, and it allows libSRTP to ensure that no security lapses result from accidental re-use of SSRC values during key sharing.

Warning

The final element of the list **must** have its 'next' pointer set to NULL.

5.8.2 Field Documentation

5.8.2.1 int srtp_policy_t::allow_repeat_tx

Whether retransmissions of packets with the same sequence number are allowed. (Note that such repeated transmissions must have the same RTP payload, or a severe security weakness is introduced!)

5.8.2.2 srtp_ekt_policy_t srtp_policy_t::ekt

Pointer to the EKT policy structure for this stream (if any)

5.8.2.3 int* srtp_policy_t::enc_xtn_hdr

List of header ids to encrypt.

5.8.2.4 int srtp_policy_t::enc_xtn_hdr_count

Number of entries in list of header ids.

5.8.2.5 unsigned char* srtp_policy_t::key

Pointer to the SRTP master key for this stream.

5.8.2.6 struct srtp_policy_t* srtp_policy_t::next

Pointer to next stream policy.

5.8.2.7 srtp_crypto_policy_t srtp_policy_t::rtcp

SRTCP crypto policy.

5.8.2.8 srtp_crypto_policy_t srtp_policy_t::rtp

SRTP crypto policy.

5.8.2.9 srtp_ssrc_t srtp_policy_t::ssrc

The SSRC value of stream, or the flags SSRC_ANY_INBOUND or SSRC_ANY_OUTBOUND if key sharing is used for this policy element.

5.8.2.10 unsigned long srtp_policy_t::window_size

The window size to use for replay protection.

The documentation for this struct was generated from the following file:

- srtp.h

5.9 srtp_ssrc_t Struct Reference

An [srtp_ssrc_t](#) represents a particular SSRC value, or a 'wildcard' SSRC.

Data Fields

- [srtp_ssrc_type_t](#) type
- unsigned int [value](#)

5.9.1 Detailed Description

An [srtp_ssrc_t](#) represents a particular SSRC value (if its type is `ssrc_specific`), or a wildcard SSRC value that will match all outbound SSRCs (if its type is `ssrc_any_outbound`) or all inbound SSRCs (if its type is `ssrc_any_inbound`).

5.9.2 Field Documentation

5.9.2.1 srtp_ssrc_type_t srtp_ssrc_t::type

The type of this particular SSRC

5.9.2.2 unsigned int srtp_ssrc_t::value

The value of this SSRC, if it is not a wildcard

The documentation for this struct was generated from the following file:

- srtp.h

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