Real time Protocols

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Seminar of Mathematics and Theoretical Computer Science

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Sensitization

- demand for multi-media services
  - live reports
  - video on demand (VoD)
  - voice over IP (VoIP)
  - audio/video/whiteboard conferences
- HTTP and FTP $\rightarrow$ not prepared (no QoS, no streaming)

Existing IETF Protocols

- Real-Time Protocol (RTP)
- Real-Time Control Protocol (RTCP)
- RTP Profiles
  - Audio and Video Conferences
  - Secure RTP
- Resource ReSerVation Protocol (RSVP)
- Real-Time Streaming Protocol (RTSP)

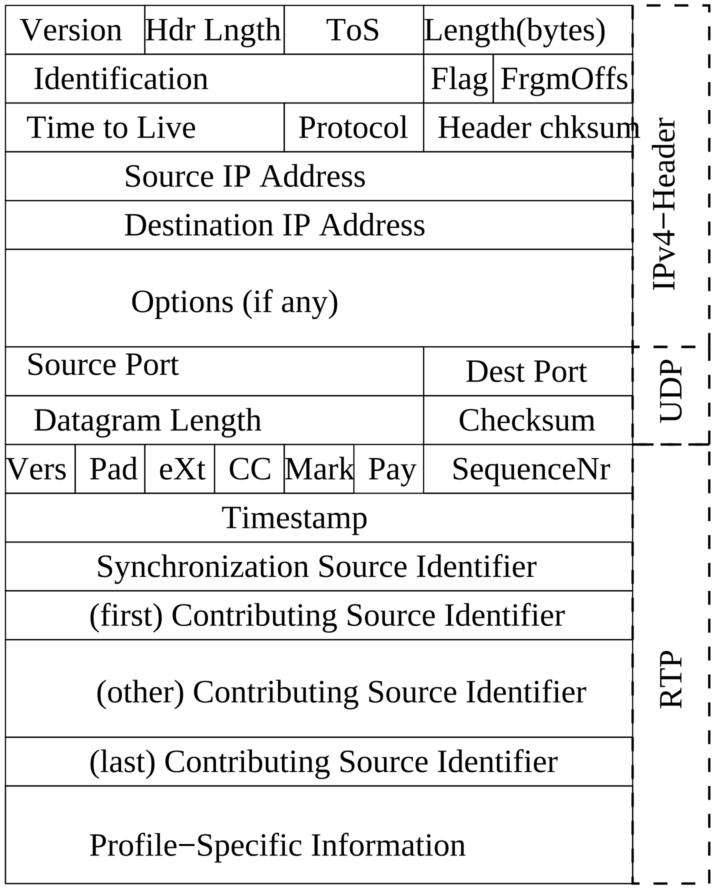
Real-Time Protocol

- runs on ST-II, UDP/IP, IPX or ATM AAL5
- UDP/IP widely used
  - unreliable
  - multicasting
- RTP inherits UDP “features” $\rightsquigarrow$ RTCP

Mixers vs. Translators

- Mixers
  - change the stream, eg. from high to low quality
  - generates new source identifier (SSRC)
  - keeps track of originating, contributing sources (CSRC)
- Translators
  - eg. firewalls: translate from multicast to unicast
  - translate from UDP/IP to ATM AAL5
  - forward RTP packets, keep SSRC intact

RTP/UDP/IP Header



RTP/UDP/IP Header: Details

- Payload Type (7 bits)
  - according to the profile
  - eg. GSM, MPEG-1 layer 3, JPEG, MPEG-2 video, H.261...
- Sequence Number (16 bits)
  - detect packet loss
  - restore packet sequence
  - randomly initialized (if not for server, then for possibly encrypting translators)

RTP/UDP/IP Header: Details 2

- Timestamp (32 bits)
  - monotonically and linearly clock
  - might be differently ordered like MPEG-2 I,B and P frames
  - clock frequency $>>$ sample rate
- Synchronization Source Identifier (SSRC) (32 bit)
- Contributing Source Identifiers (CSRC) (32 bit)

RTP/UDP/IP Header Compression

- 40 byte per RTP/UDP/IP header
- 20 ms packetization interval $\leadsto$ 16 kbit/s only for headers
- serial line compression down to 2 bytes (4 bytes with checksum)
- how?
  - only 50% of the fields change
  - other changes in a predictible way
  - generate 8 bit CIDs, store first packet and first-order difference
  - fall-back to full packets on other changes

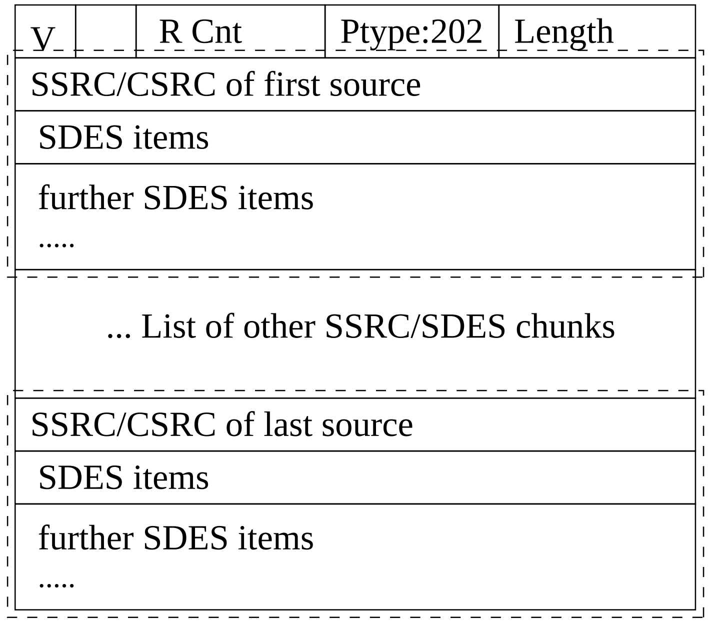
Real-Time Control Protocol

- four tasks
  - send information about QoS (lost packets, jitter...)
  - transfer clear-text information (eg. CNAME)
  - calculate RTCP packetization rates
  - keep track of all joined participants

- packet types
  - sender
  - receiver
  - SDES
  - BYE

V		Report Cnt	Ptype:200	Length
SSRC of Sender				
NTP Timestamp				
RTP Timestamp				
Sender’s Packet Count				
Sender’s Byte Count				
SSRC of first source				
% Lost	Cummulative Packets Lost			
Extended Highest Sequence Number Received				
Interarrival Jitter				
Time of last Sender Report				
Time since Last Sender Report				
..List of Sender Reports				
SSRC of last source				
% Lost	Cummulative Packets Lost			
Extended Highest Sequence Number Received				
Interarrival Jitter				
Time of last Sender Report				
Time since Last Sender Report				
Profile−specific Information				

V		R Cnt	Ptype:201	Length
SSRC of Sender				
SSRC of first source				
% Lost		Cummulative Packets Lost		
Extended Highest Sequence Number Received				
Interarrival Jitter				
Time of last Sender Report				
Time since Last Sender Report				
..List of Sender Reports				
SSRC of last source				
% Lost		Cummulative Packets Lost		
Extended Highest Sequence Number Received				
Interarrival Jitter				
Time of last Sender Report				
Time since Last Sender Report				
Profile−specific Information				


V
R Cnt
Ptype:202
Length
SSRC/CSRC of first source
SDES items
further SDES items
.....
... List of other SSRC/SDES chunks
SSRC/CSRC of last source
SDES items
further SDES items
.....

Where are we?

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RTP Profile: Audio and Video Conferences

- 20 ms packetization rate (or the formats natural frame size)
- sampling frequency out of 8000, 11025, 16000, 22050, 24000, 32000, 44100, 48000
- channel ordering from left-to-right
- audio encodings
  - eg. G.722, G.723, G.726, G.728, G.729, GSM, MPA, RED
- video encodings
  - Motion JPEG, H.261, H.263, MPV

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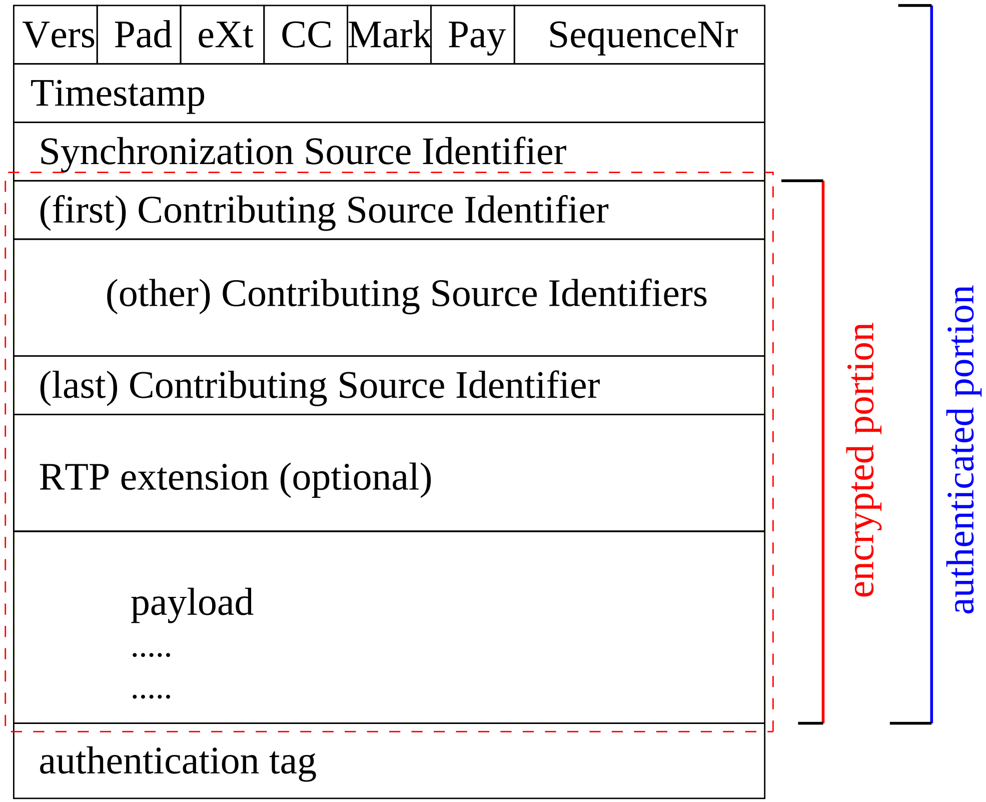
RTP Profile: Secure RTP

- provides privacy, message authentication, and replay protection
- additive AES compliant stream cypher in counter mode
- MAC over the whole packet
- contains data according to another profile
- normal RTP header format, adds a 4-byte authentication tag

Secure RTP Header



Cryptographic Context

- encryption key k_e (fixed for session)
- message authentication key k_m (fixed for session)
- 32-bit rollover counter r (which counts how many times the 16-bit *RTP sequence number* wrapped around 0xFFFF)
- the last authenticated sequence number s_l
- replay list L (only receiver side), keeps track of already processed packets

Cryptographic Background

- AES compliant symmectric key block-cypher
  - Advanced Encryption Standard
  - 128 bit block size
  - three key sizes of 128, 192, 256 bits
  - since October 2000: Winner is “Rijndael”

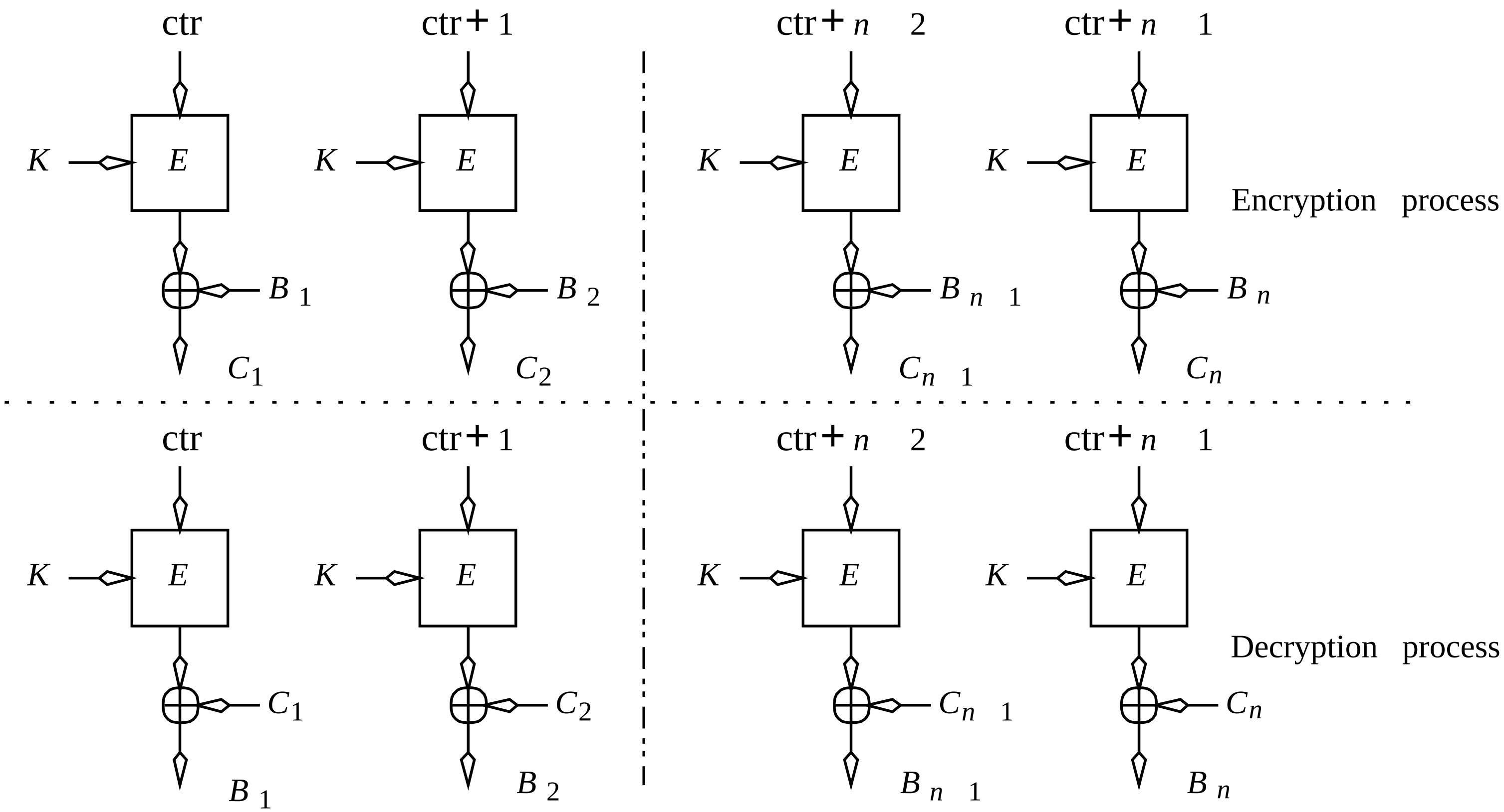
Cryptographic Background: Rijndael

- data block is partitioned into an array of bytes
- each cypher operation is byte oriented
- multiple rounds (10, 12 or 14; depends on key size)
- one round consists of four layers
  - first layer: 8x8 S-Box applied to each byte
  - second and third: shifting of array rows, mixing columns
  - fourth layer: subkey bytes XORed with each byte of array

Cryptographic Background: Counter Mode

- resistancy to redundant plaintext attacks
- cypher block chaining (CBC) mode
  - encrypt last block with key and XOR with new block
  - dependancy on all packets
- segmented integer counter mode (SIC)
  - encrypt counter with key and XOR with new block
  - small hamming distance for ctr and $ctr + 1 \leadsto$ only problematic for differentially weak ciphers

Cryptographic Background: SIC



Cryptographic Background: SIC in SRTP

- $ctr = [(r * 65536) + seq] * 4096 + i$
- seq is the RTP sequence number, i is a counter for each 128 bit block
- IP packet size $=$ 64 KB $\leadsto$ 4096 blocks
- Jumboframes are unlikely to be used for multimedia traffic
- ctr has to be unique over the session life time
- maximum of $2^{48} = 281,474,976,710,656$ SRTP packets
- 20 ms packetization time $\leadsto$ 178,510 years

Cryptographic Background: UMAC

- Message Authentication Code (MAC) $\rightarrow$ UMAC
- fast (eg. one clock cycle for one byte)
- extra security by key and a “nonce” (our counter ctr)
- $UMAC = E_{AES}(k_m, ctr) \oplus UHASH(k_m, B_i)$
- UMAC-OUTPUT-LEN $=$ 4 bytes

Cryptographic Background: UHASH

- three layers
  - bulk hash function $NH \leadsto$ speed optimized, compresses block
  - polynomial hash $\leadsto$ 16 byte
  - inner-product hash $\leadsto$ 2 byte
- repeat layers with slightly different keys to get additional 2 bytes
- repetition is independent $\leadsto$ trade authenticity with speed
- allows quick processing to survive DoS attacks

Replay Detection

- bit field over the last SRTP_WINDOW_SIZE
- $SRTPseq - SRTP\_WINDOW\_SIZE$ and $SRTPseq$
- $SRTPseq = r * 65,536 + seq$
- log all older packets and replayed packets (paranoia?)

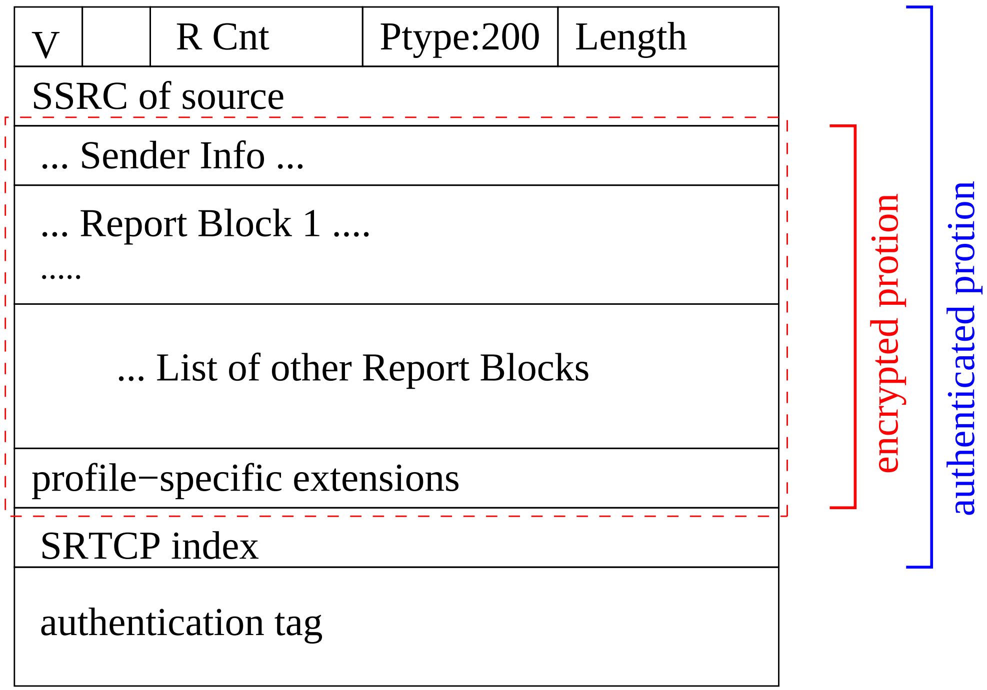
Replay Detection: SRTP WINDOW SIZE

- how big should SRTP_WINDOW_SIZE be?
  - receiver buffer size $\rightarrow$ "real" real time
  - unidirectional com (eg. radio) $\rightarrow$ 3 sec
  - video conferences $\rightarrow$ not more than 400 ms
  - 20 ms packetization rate $\rightsquigarrow$ 50 packets/sec
  - 500 ms buffer $\rightsquigarrow$ max. 25 packets (opt. 32 packets)

Secure RTCP

- based on same ideas as for SRTP
- 32 bit authentication tag
- 32 bit SRTCP index
  - $2^{32} = 4,294,967,296$ packets for one session
  - has to be terminated with RTCP BYE packet

Secure RTCP Header



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Real-Time Resource ReSerVation Protocol (RSVP)

- IP: best effort
- RSVP adds rate-sensitive and delay-sensitive QoS
- soft state over routers
- tunneling for non-RSVP networks

Real-Time Streaming Protocol (RTSP)

- may use RTP
- “VCR-style” remote conrol functionality
- similar syntax to HTTP/1.1
- typical session: DESCRIBE $\rightarrow$ SETUP $\rightarrow$ PLAY $\rightarrow$ PAUSE $\rightarrow$ TEAR-DOWN

Conclusion

- what did we discuss?
  - Real-Time Protocol (RTP)
  - Real-Time Control Protocol (RTCP)
  - RTP Profiles
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- $\exists$ base for real-time streaming of multi-media data, not widely used

  $\oplus$ RealNetworks uses RTP, but favors RDP

  $\oplus$ RSVP is supported by modern routers (eg. Cisco)