Audio/Video Transport Working Group:
Meeting Agenda
Overview of Revised Real-time Transport Protocol (RTPv2)

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Agenda for Tuesday, July 26, 1600–1800

- Status of Real-time Transport Protocol
- Overview of RTP revisions as incorporated into draft spec
- Report on implementations of revised RTP
- Discussion of revised RTP Internet-Draft
  - Agreement on choices made for “open issues”?
  - Rough consensus to submit as “proposed” RFC?
- Discussion of questions for Audio/Video Profile

Agenda for Wednesday, July 27, 1600–1800

- Continuation of discussion if needed
- Presentation of new encoding specifications over RTP
  - JPEG
  - MPEG1 / MPEG2
  - Cell-B
  - Revised H.261 spec (?)
- Other topics?
- Future working group activity

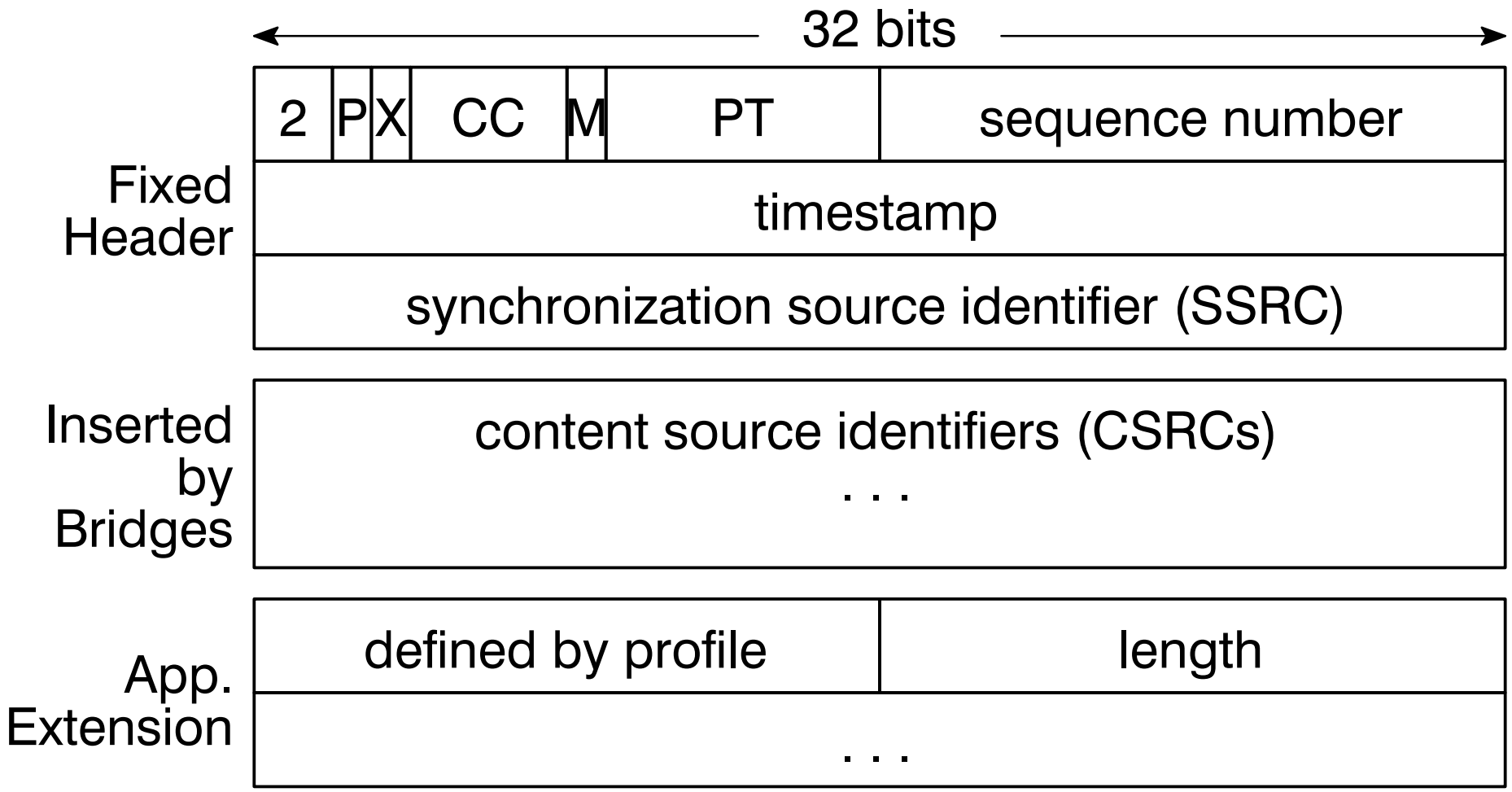
Status of RTP

- RTP version 1 submitted in October 1993
- Changes requested after Area Directorate review
- Proposed changes presented & discussed in March at Seattle IETF
- Details completed by Casner, Frederick, Jacobson, Schulzrinne, sent in June to rem-conf list for comments
- RTP version 2 spec draft-ietf-avt-rtp-05.txt is now posted (also available in PostScript form on gaia.cs.umass.edu: pub/hgshulz/rtp/draft-ietf-avt-rtp-05.ps)

Overview of Changes in RTP version 2

- Control and data on separate ports
- New control packet format; reception reports required
- No data options, CSRC count in fixed header
- Local 16-bit SSRC ID became 32-bit random global ID, always present (header now 12 octets)
- Media-specific timestamps (vs. fixed 65536 Hz rate)
- No Channel ID, use multiplexing in lower layers
- Application-specific sync marker bit
- Encryption covers whole packet; authentication omitted
- Translators can’t do unicast reverse path control packet
- BOS option eliminated, put info in encoding if needed
- Lengths changed to word counts, not counting length

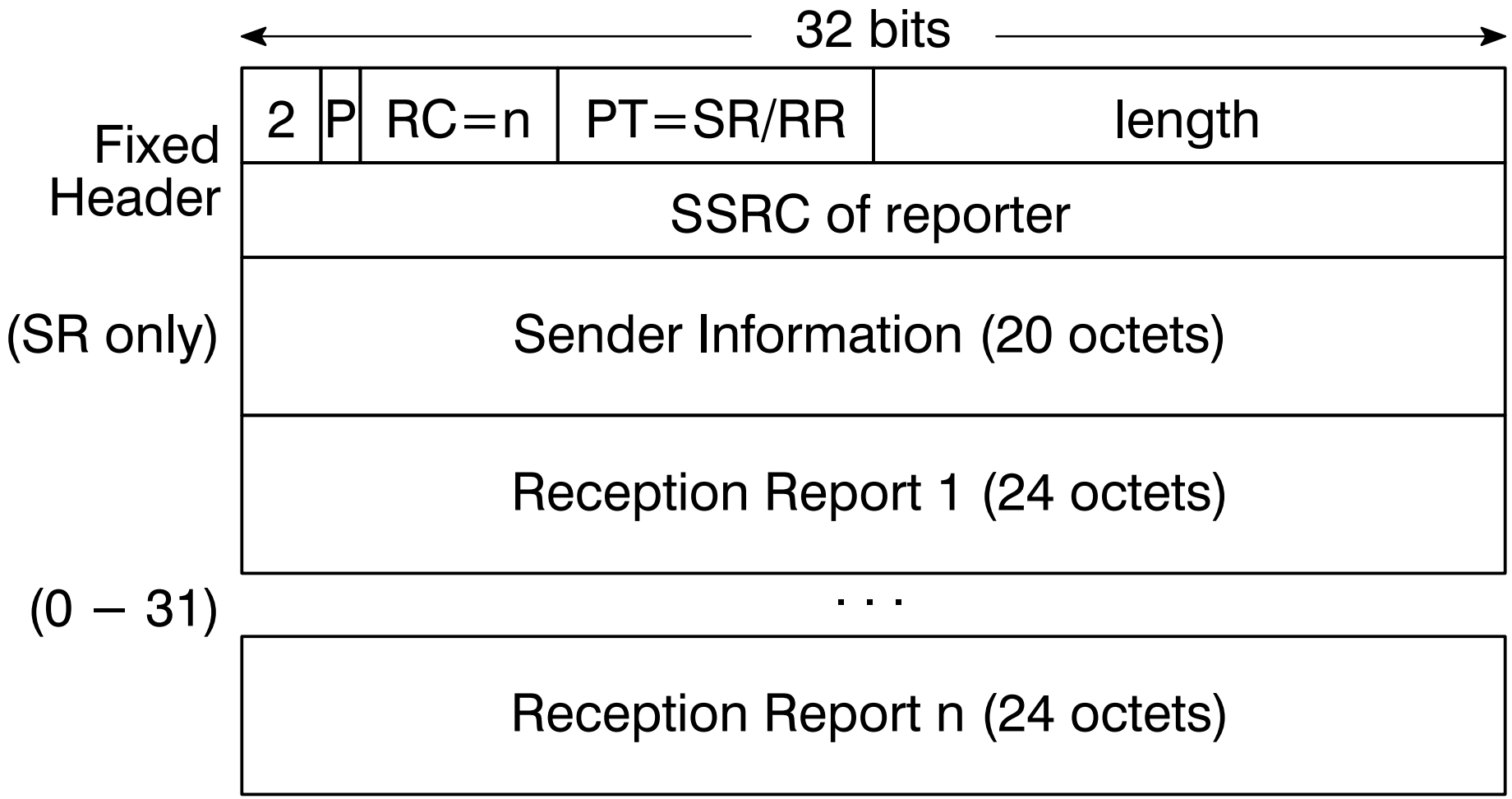
RTP Data Header



RTP Control Protocol (RTCP)

- Sent on separate “port” from data
- Variety of packet types “stackable” in one UDP packet
- For IP multicast, sender/receiver reports must be provided plus CNAME to provide meaningful binding
- Additional source description or control information may be sent as needed by the application
- Overall control traffic bandwidth should be limited to a small fraction (1−2%) of the data traffic through the calculation of an appropriate reporting interval
- For encryption, prefix stack with 32-bit random number

Sender/Receiver Report packet format



Sender Information

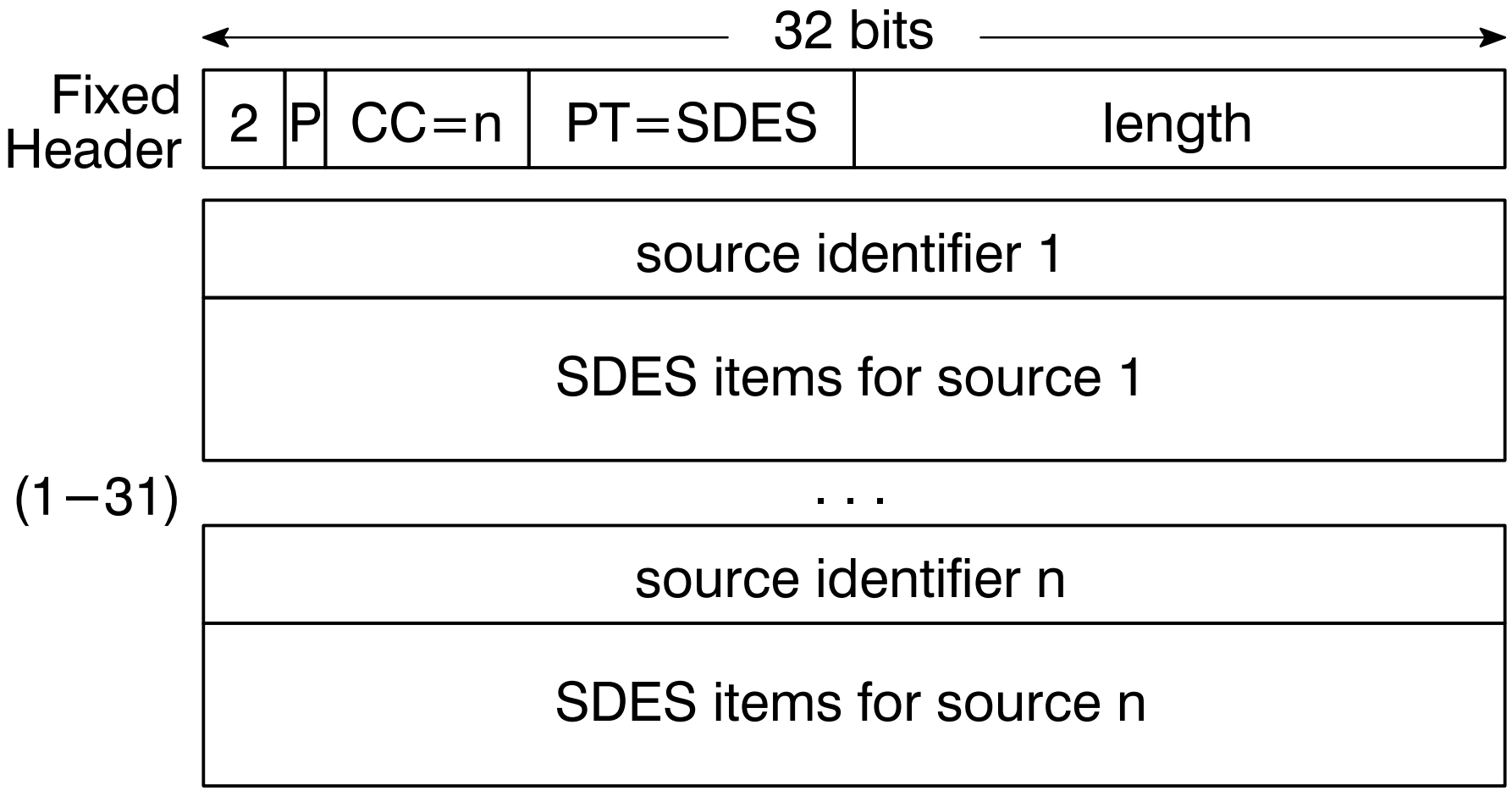
- Sent if data has been transmitted “recently”
  - NTP timestamp when report is sent (zero if unknown)
  - corresponding RTP media timestamp (calculated)
  - sender’s packet count (to calcluate packet rate)
  - sender’s byte count (to calculate data rate)

Reception Reports

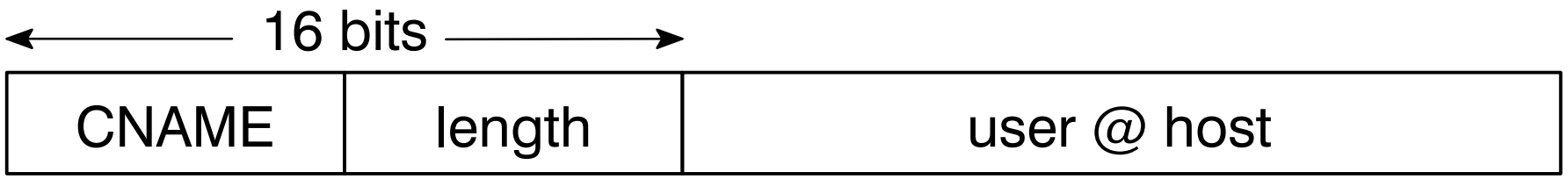
- Reception info from all sources heard since last report, or round robin for as many as will fit in allowed BW:
  - source identifier of source being reported
  - cumulative number of packets received
  - cumulative number of packets expected (from seq.)
  - interarrival jitter (algorithm TBD)
  - timestamp from last SR from this source
  - delay time since that SR was received

(last two items for round-trip propagation delay calculation)

Source Description (SDES) packet format

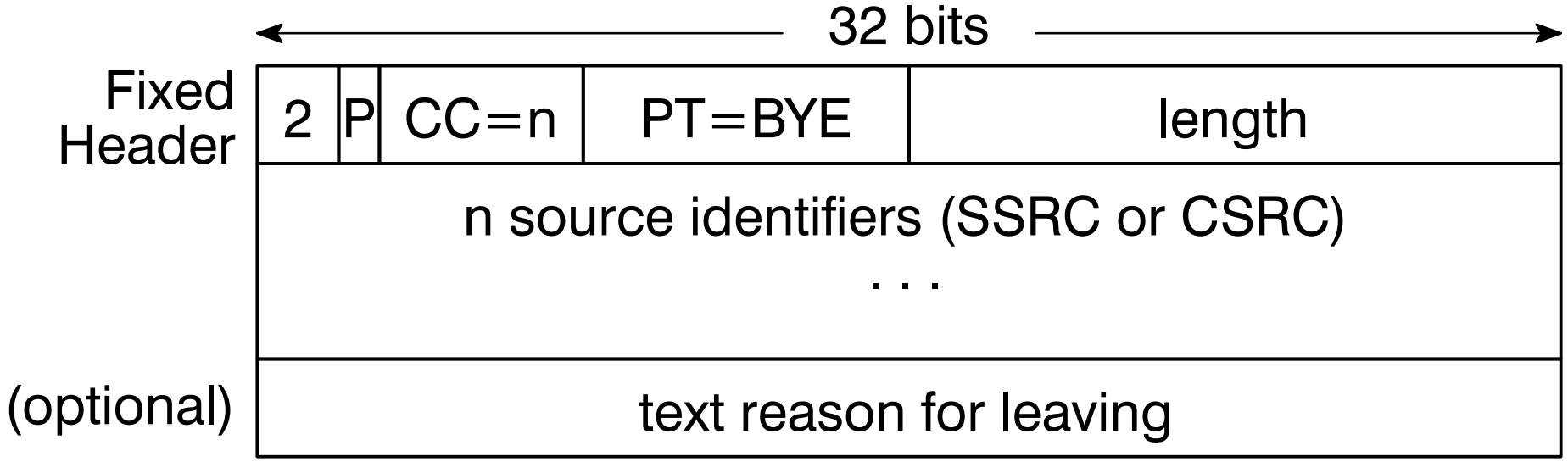


SDES Items

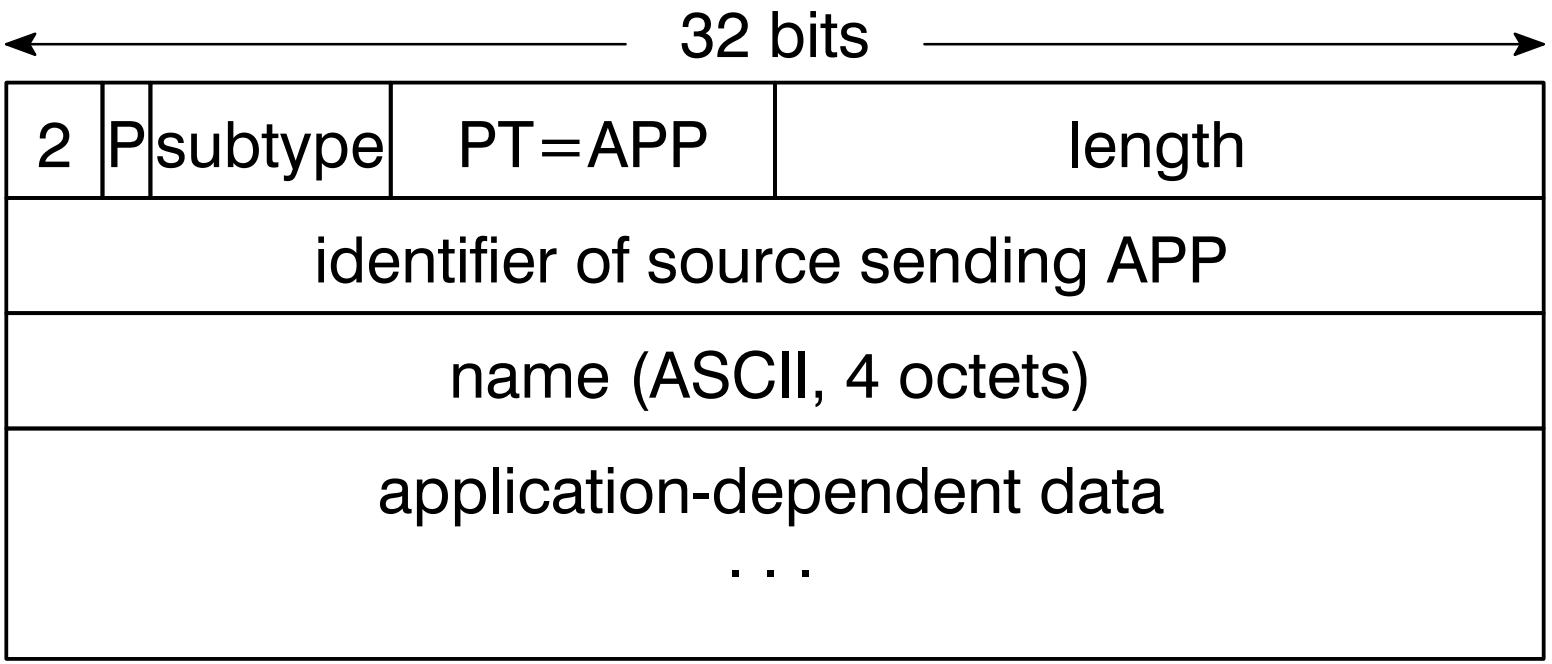


- CNAME (canonical name) is required as unique identifier to provide a binding among media tools and over time
- Optional additional information about a source: NAME, EMAIL, LOC (location), TXT (e.g. status)
- Items are arbitrary octet length, unaligned
- Terminator after last item pads to 32-bit boundary

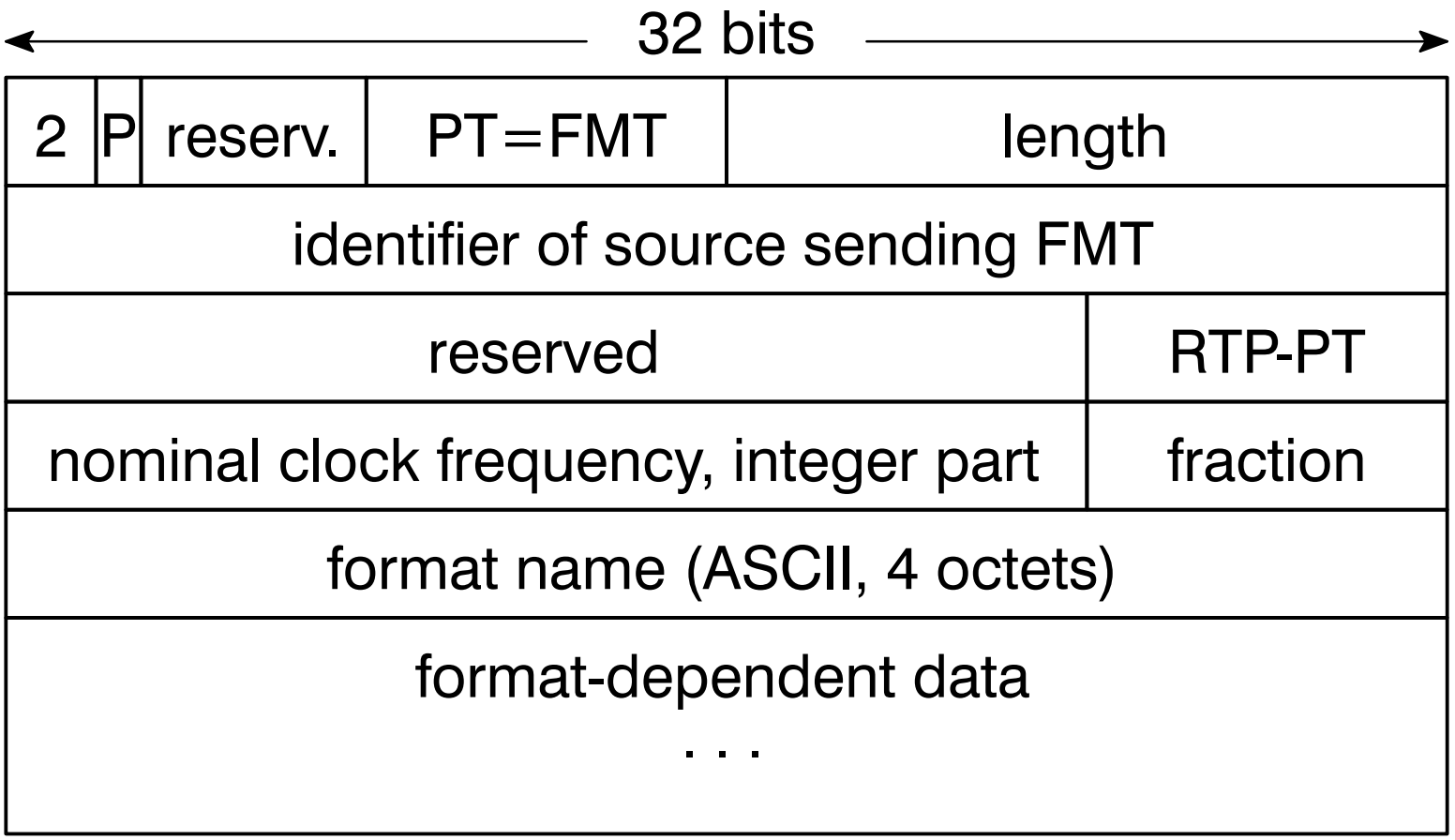
Goodbye (BYE) packet format



Application-Specific (APP) packet format



Payload type mapping (FMT) packet format



Steve McCanne: RTPv2 implementation in vic

- Suggests defining FMT in a profile when it is needed rather than in the base spec.
- Agrees with change of lengths to word count and elimination of check for 0 by not including the length of the length field itself.
- SDES parser was somewhat cumbersome, but dealing with variable length things is always a pain.
- Would like a bit that lets you distinguish between RTCP and RTP packets, i.e., to simplify archival and multiplexing onto a single transport channel.

June List of “Open Issues”

- Provision for (testing) Wakeman/Turletti/Bolot scheme, comparison of performance with Jacobson scheme
- Provision for (testing) unicast feedback mechanism
- Need to define encapsulations for ST−II or ATM AAL5?
- Shall FMT packet type be kept? (−> profile issue?)
- Authentication has been omitted: OK?
- Need to clarify when sender reports may be omitted
- Right tradeoff in bit allocations for data header?
- Lengths have been changed to word counts
- Other issues?

Other items to be completed...

- Specify CNAME for binding across multiple machines, or for multiple sources in one medium on one machine? What about machines with non-unique addresses?
- Assign profile-specific RTCP packet type space?
- Is the description of collision resolution sufficient?
- Need descriptions of algorithms for:
  - report interval calculation
  - interarrival jitter calculation
  - header validity check (for encryption)
- Specify how authentication might be done

Questions for Audio/Video Profile

- One or two marker bits for audio? video?
- Which default encryption method: entire packet, or payload types indicating encrypted encodings?
- Should we specify that control port = data port + 1?
- Should we include the FMT control packet type?

Issues left from November

- Need details from PC and Mac folks for L8 encoding (linear, 8-bit audio)
- Andrew Cherenson's additional formats and status info
- How to calculate interarrival jitter if video packets of one frame arrive in a clump (all will have same timestamp)
- Fengmin asked: if a translator performs lower-level protocol translation, might segmentation info be needed in RTP?
- What else is needed in the profile?