
Media Transport in Packet Networks

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Why Not Use TCP?

◆ Four problems

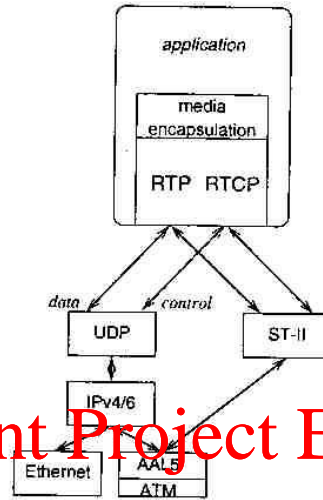
- » *No need for 100% reliability*
- » *Slow start*
- » *Retransmission delay*
- » *Window backoff*
- » *N participants -> N*N connections*
- » *So use UDP and IP multicast*

Need for RTP

- ◆ Loss, out of order: sequence number
- ◆ Loss, jitter, timing, timestamp, project Exam Help
- ◆ Source/payload identification <https://powcoder.com>
- ◆ Rate control: QoS feedback Add WeChat powcoder
- ◆ RTP provides functions to support these requirements for many real-time applications

RTP: de facto standard

- ◆ End-to-end transport of real-time data, such as audio and video; and data for non-real-time applications
- ◆ Does not guarantee QoS
- ◆ Does not address resource reservation along the path of a connection
- ◆ Requires the use of signaling protocol to setup the connection and negotiate media format to be used
- ◆ RTP is enhanced with RTCP which is part of the RTP specification
 - » *RTCP provides for end-to-end monitoring of data delivery and QoS*
- ◆ Independent of the underlying transport and network layers
 - » *most commonly used on UDP*
 - RTP assigned an even UDP port number
 - RTCP assigned the next higher UDP port (odd)
- ◆ Supports multiple destinations if network supports multicast distribution



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Data transport – RTP

Real-Time Transport Protocol (RTP) = data + control

data: timing, loss detection, content labeling, talkspurts, encryption

control: (RTCP) ⇔ periodic with $T \sim$ population

- QOS feedback
- membership estimation
- loop detection

Packet based delivery – basic issues

- ◆ Store and forward handling in routers – delay
 - » *IP switching (hardware technology) to reduce intermediate packet delays*
- ◆ High protocol overhead (RTP over UDP over IP over ATM) – extra bandwidth requirement
- ◆ To achieve bandwidth savings –
 - » *Header compression often prescribed and used*
 - » *statistical multiplexing – mixing of voice and data packets*
- ◆ Problem – not backward compatible with existing routers
 - » *new router designs*

Packet based delivery – basic issues

Uncompressed RTP/UDP/IP header = $12+8+20 = 40$ bytes

Encapsulation overhead for ATM (RFC1483) = 8 bytes

Payload = N bytes per packet

» *this represents up to 1500 bytes of payload + an ATM header of 5 bytes before first byte of payload is transmitted*

Minimum of 2 ATM cells is required for a single voice sample

Using header compression (RFC2809) 40 bytes is reduced to 2 bytes

Voice over RTP bandwidth calculations

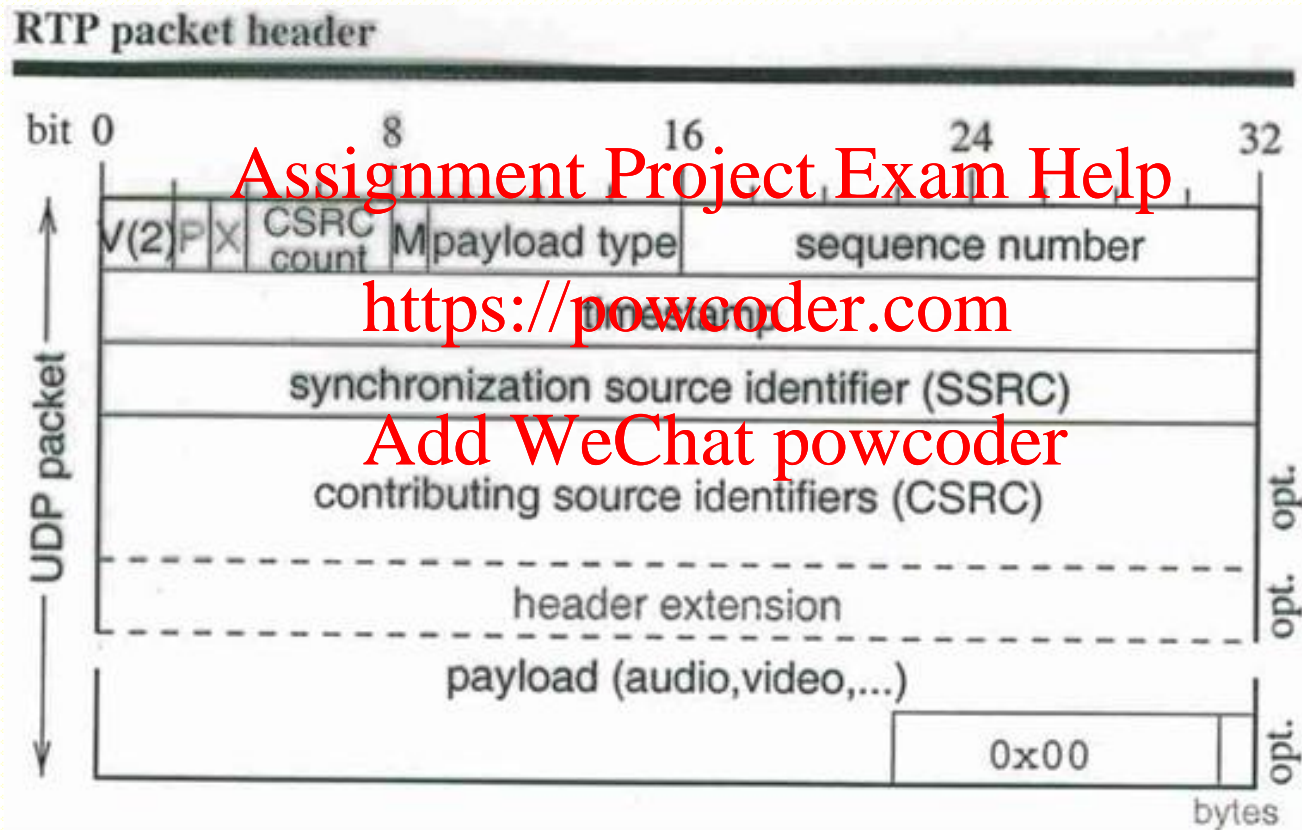
Payload format	Nominal rate	Packet rate (ms)	Payload size (bytes)	Required BW (Kbps)	
				uncompressed	compressed
G.711	64 Kbps	20	160	80	64.8
G.711		10	80	40	65.6
G.729	8 Kbps	20	20	24	8.8
G.729		10	10	40	9.6

- ◆ Packet rate is the frequency with which packets are formed and transmitted
 - either 10 ms or 20 ms for most applications
- ◆ Bits per 20 ms packetization interval for G.711 over RTP/UDP/IP/Encaps/AAL5 = minimum 5 ATM cells = 53×5 bytes = 2120 bits

◆ Compressed packet headers –

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- ◆ additional computational requirements in intermediate routing nodes
 - ◆ major issue with user multiplexing and stream mixing at intermediate points
 - ◆ header expansion and interpretation at intermediate nodes can be lengthy process – adds to end-to-end delay

RTP packet



RTP packet header

- ◆ Version (V, 2 bits)
- ◆ Padding (P, 1 bit) – for encryption with fixed block sizes
 - ◆ if set, last byte of the padding contains a count of how many padding bytes should be ignored
- ◆ Extension (X, 1 bit) –
 - ◆ if set, the fixed header is followed by exactly one header extension (2 bytes) to pass control information not to be interpreted by intermediate nodes
- ◆ CSRC count (CC, 4 bits) – for mixers
 - ◆ number of contributing sources to this packet
 - ◆ if only one SSRC in the stream, $CC = 0$

RTP packet header

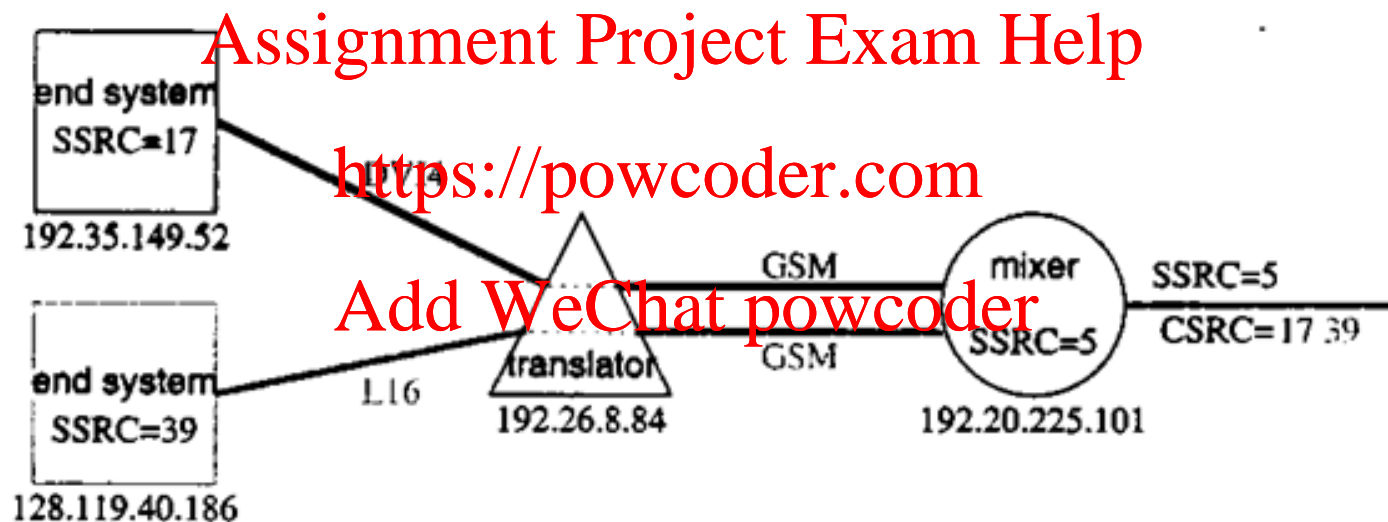
- ◆ Marker (M, 1 bit) – to mark frame boundaries
 - ◆ signifies the beginning/end of a talk spurt (to begin playout of comfort noise), or end of a filled frame
- ◆ Payload Type (PT, 7 bits) – format of RTP payload (e.g. G723)
 - ◆ once a session is set up, a sender cannot use different RTP payloads during the session
- ◆ Sequence Number (16 bits) – +1 for each packet
 - ◆ used by receiver to detect packet loss and to restore packet sequence
 - ◆ packet reordering very hard to do – costs in voice/video quality

RTP packet header

- ◆ Timestamp (32 bits) – random starting value
 - ◆ increments by one for each sampling period for fixed-rate audio
 - ◆ If an audio application reads blocks covering 160 sampling periods from the input device, the timestamp would be increased by 160 for each such block
- ◆ SSRC (32 bits) – identifies synchronization source (sender)
 - ◆ value chosen randomly – no two senders have same SSRC identifier
 - ◆ collisions resolved by simple mechanisms in RTP
- ◆ CSRC list (0 to 15 items, 32 bits each) –
 - ◆ identifies contributing sources for payload contained in this packet
 - ◆ used for correct source identification when payloads played out at endpoint

Mixers and Translators

RTP mixers, translators, ...



RTP mixers, translators, ...

mixer: Assignment Project Exam Help

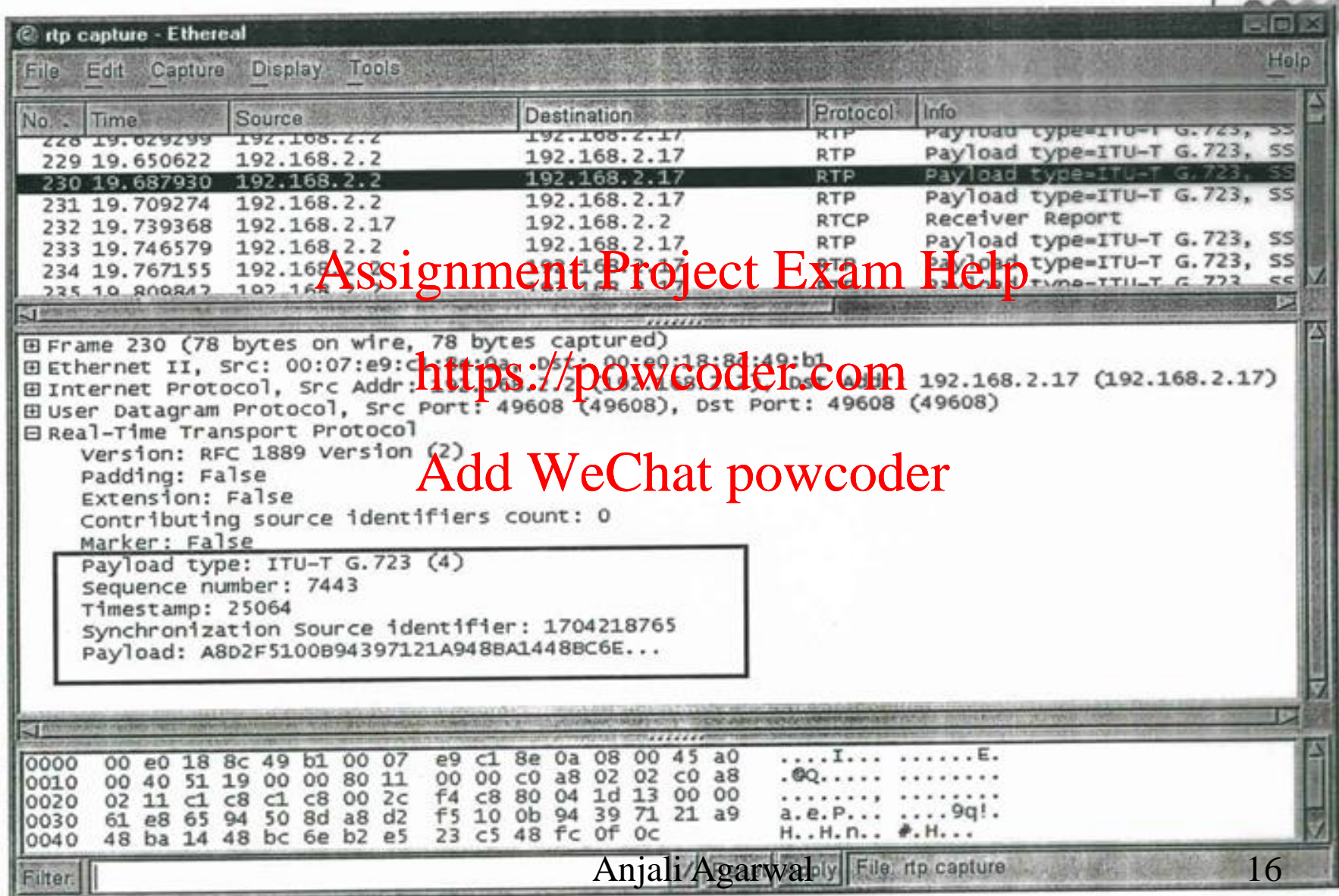
- several media stream \Rightarrow one new stream (new encoding)
- mixer: reduced bandwidth networks (dial-up)
- appears as new source, with own identifier

translator:

- single media stream
- *may* convert encoding
- protocol translation (native ATM \leftrightarrow IP), firewall
- all packets: source address = translator address

RTP Packet

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The screenshot displays the Wireshark interface for an RTP capture. The top pane shows a list of captured packets, with packet 230 selected. The middle pane provides a detailed view of the selected packet's structure, including Ethernet II, Internet Protocol, User Datagram Protocol, and Real-time Transport Protocol (RTP) fields. The bottom pane shows the raw packet data in hexadecimal and ASCII.

No.	Time	Source	Destination	Protocol	Info
228	19.629299	192.168.2.2	192.168.2.17	RTP	Payload type=ITU-T G.723, SS
229	19.650622	192.168.2.2	192.168.2.17	RTP	Payload type=ITU-T G.723, SS
230	19.687930	192.168.2.2	192.168.2.17	RTP	Payload type=ITU-T G.723, SS
231	19.709274	192.168.2.2	192.168.2.17	RTP	Payload type=ITU-T G.723, SS
232	19.739368	192.168.2.17	192.168.2.2	RTCP	Receiver Report
233	19.746579	192.168.2.2	192.168.2.17	RTP	Payload type=ITU-T G.723, SS
234	19.767155	192.168.2.2	192.168.2.17	RTP	Payload type=ITU-T G.723, SS
235	19.800847	192.168.2.2	192.168.2.17	RTP	Payload type=ITU-T G.723, SS

Frame 230 (78 bytes on wire, 78 bytes captured)

- Ethernet II, Src: 00:07:e9:c1:8e:0a, Dst: 00:00:18:8c:49:b1
- Internet Protocol, Src Addr: 192.168.2.2 (192.168.2.2), Dst Addr: 192.168.2.17 (192.168.2.17)
- User Datagram Protocol, Src Port: 49608 (49608), Dst Port: 49608 (49608)
- Real-time Transport Protocol
 - Version: RFC 1889 version (2)
 - Padding: False
 - Extension: False
 - Contributing source identifiers count: 0
 - Marker: False
 - Payload type: ITU-T G.723 (4)
 - Sequence number: 7443
 - Timestamp: 25064
 - Synchronization source identifier: 1704218765
 - Payload: A8D2F5100B94397121A948BA1448BC6E...

0000 00 e0 18 8c 49 b1 00 07 e9 c1 8e 0a 08 00 45 a0I... ..E.
 0010 00 40 51 19 00 00 80 11 00 00 c0 a8 02 02 c0 a8 .@Q.....
 0020 02 11 c1 c8 c1 c8 00 2c f4 c8 80 04 1d 13 00 00
 0030 61 e8 65 94 50 8d a8 d2 f5 10 0b 94 39 71 21 a9 a.e.P... ..9q!
 0040 48 ba 14 48 bc 6e b2 e5 23 c5 48 fc 0f 0c H..H.n..#.H..

Filter: File: rtp capture

IMPLEMENTATIONS OF RTP OVER THE INTERNET

“Internet telephones” (usually for PCs) available using proprietary audio coding and protocols, meant for point-to-point connections:

- ◆ Speak Freely for Unix and Windows
- ◆ Vocaltec Internet Phone)
- ◆ SoftFone by SilverSoft
- ◆ Digiphone
- ◆ Quarterdeck
- ◆ Internet Telephone Company
- ◆ Telescape Intercom by Telescape
- ◆ IBM Internet Connection Phone
- ◆ CuSeeMe (for Windows PC and the Macintosh)
- ◆ IRIS
- ◆ Lucent 4111 Multifunctional DCP Voice Terminal
- ◆ Audio/video directly over ATM:
 - ◆ Nemesys
 - ◆ FreeVue audio and video
 - ◆ NVAT
 - ◆ Ericsson LAN Phone
 - ◆ RealAudio (Microsoft Windows only)
 - ◆ AudioSoft
 - ◆ VDO

Real Time Control Protocol (RTCP)

- ◆ to convey end-to-end information about the quality of the session to each participant
 - » *like packet delay, jitter, packets received and lost are valuable to access network health in real-time*
- ◆ Based on periodic transmission of control packets to all participants
- ◆ RTP and RTCP may or may not be routed on the same end-to-end path

RTCP packet format

- ◆ SR: Sender Report

- » *for transmission and reception statistics from active sender participants*

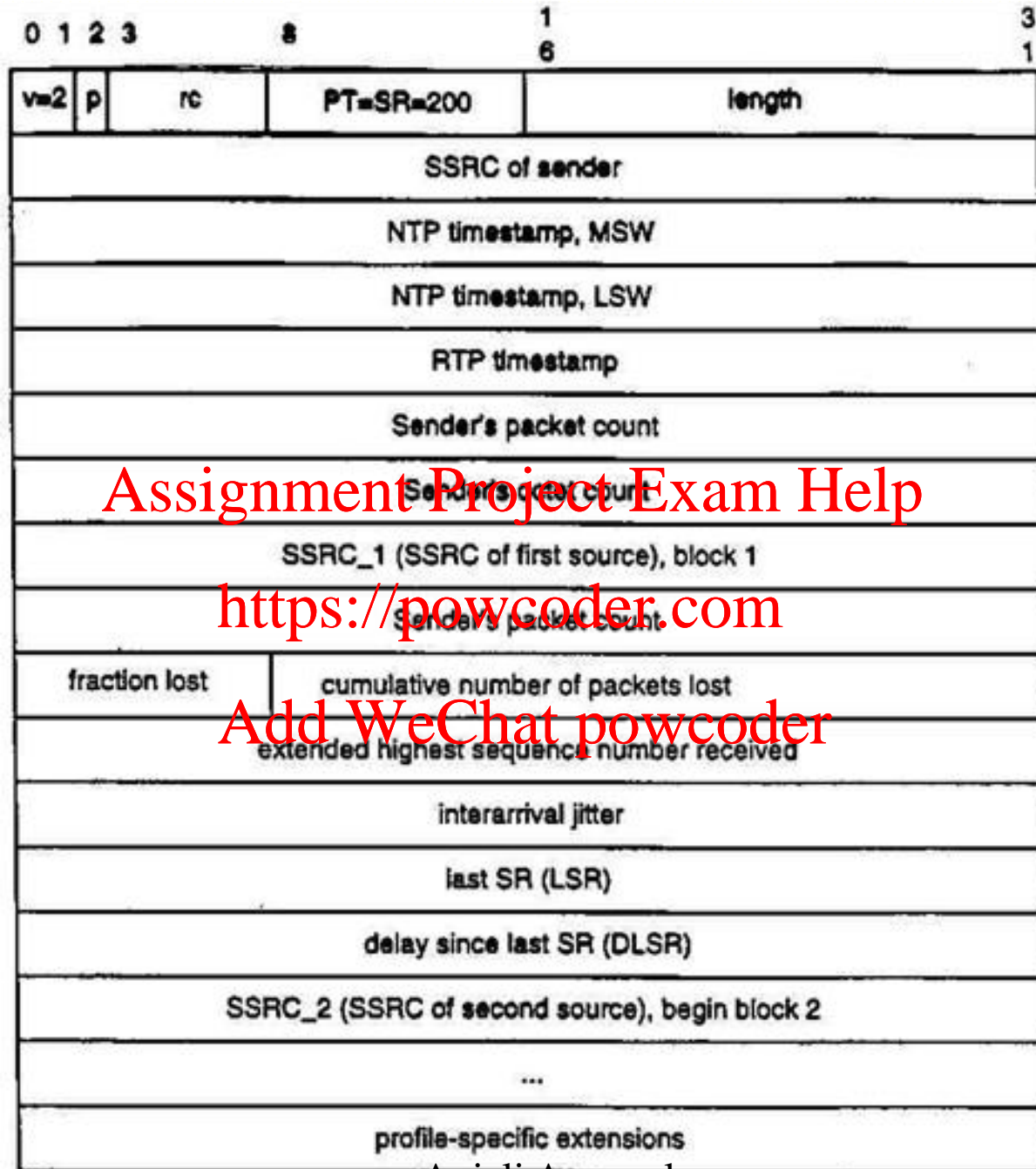
- ◆ RR: Receiver Report

- » *for reception statistics from not active senders*

ietf-avt-rtp-new-03.txt (work in progress)

- » *suggests 5% of session bandwidth be allocated for RTCP packets*

- 1.25% go to senders
 - 3.75% be allocated to receivers



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RTCP Sender Report

- ◆ SSRC of sender: identifies sending source of SR
- ◆ NTP timestamp: when report was send
- ◆ RTP timestamp: corresponding RTP time
- ◆ Sender's packet count: total RTP packets sent
- ◆ Sender's octet count: total octets sent
- ◆ rc: number of reception report counts (max 32) for which statistics are included in the packet
- ◆ FL: fraction of packets lost since last SR was send
- ◆ Cumulative packets lost: since beginning of session
- ◆ Highest sequence number received from SSRCn
- ◆ interarrival jitter as measured at the receiver
- ◆ LSR: time last SR heard
- ◆ DLSR: delay since last SR

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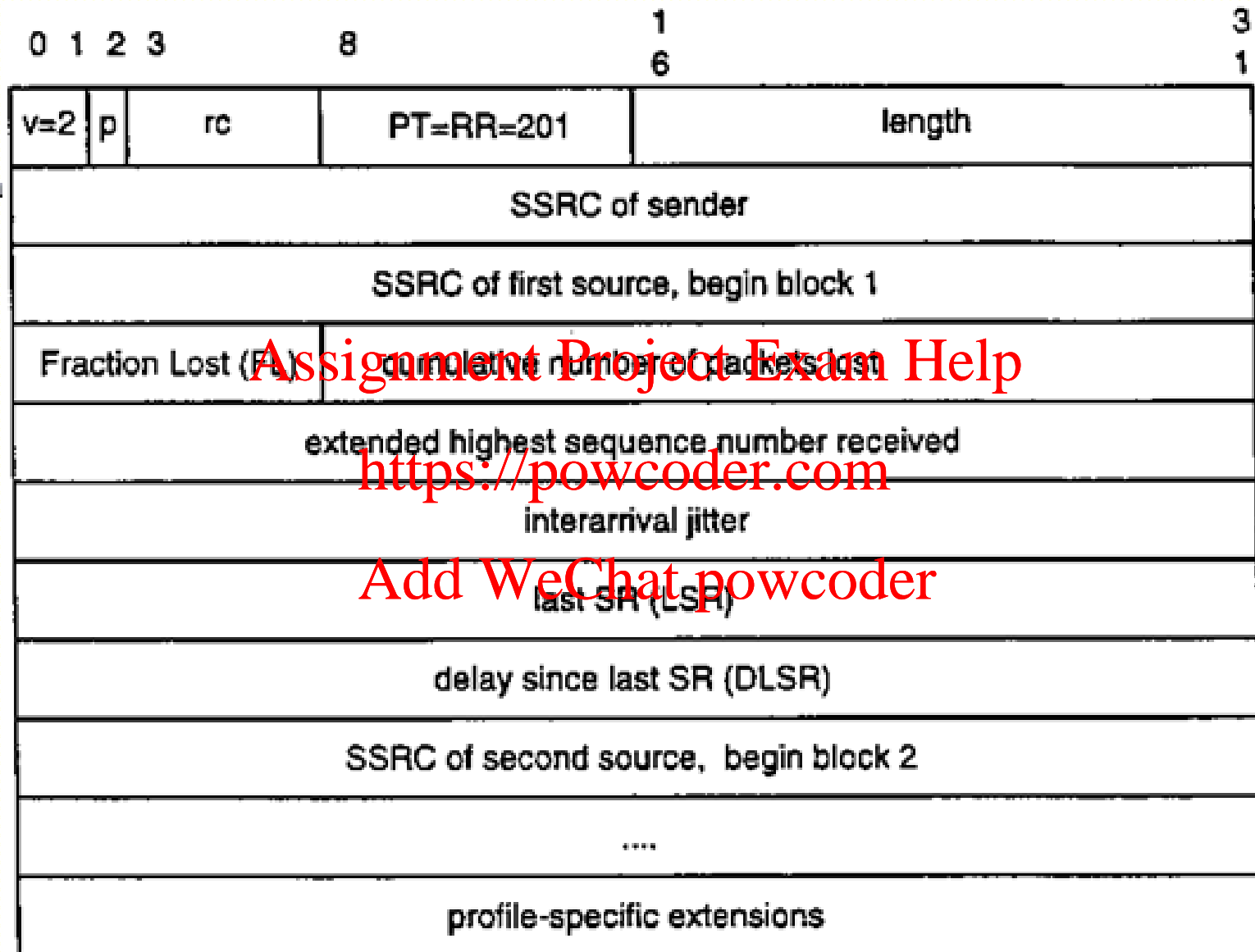
RTCP Receiver Report

- ◆ Conveys identical information as SR except sender information
- ◆ An endpoint can send RR or SR but SR contain overall packet and byte counts not reported in RR.

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RTCP packet format

- ◆ SDES: Source DEscription

- » *binds SSRC in RTP with actual user identification*

- user's name, email address, login ID

- optional items as telephone #, user location

- » *send at beginning of the session to explicitly identify each participant*

- ◆ BYE: ends user participation in a call

- » *tells all participants that sending user is departing*

- » *should contain the reason for statistics*

- ◆ APP: application specific RTCP packet

- » *none yet (work in progress)*

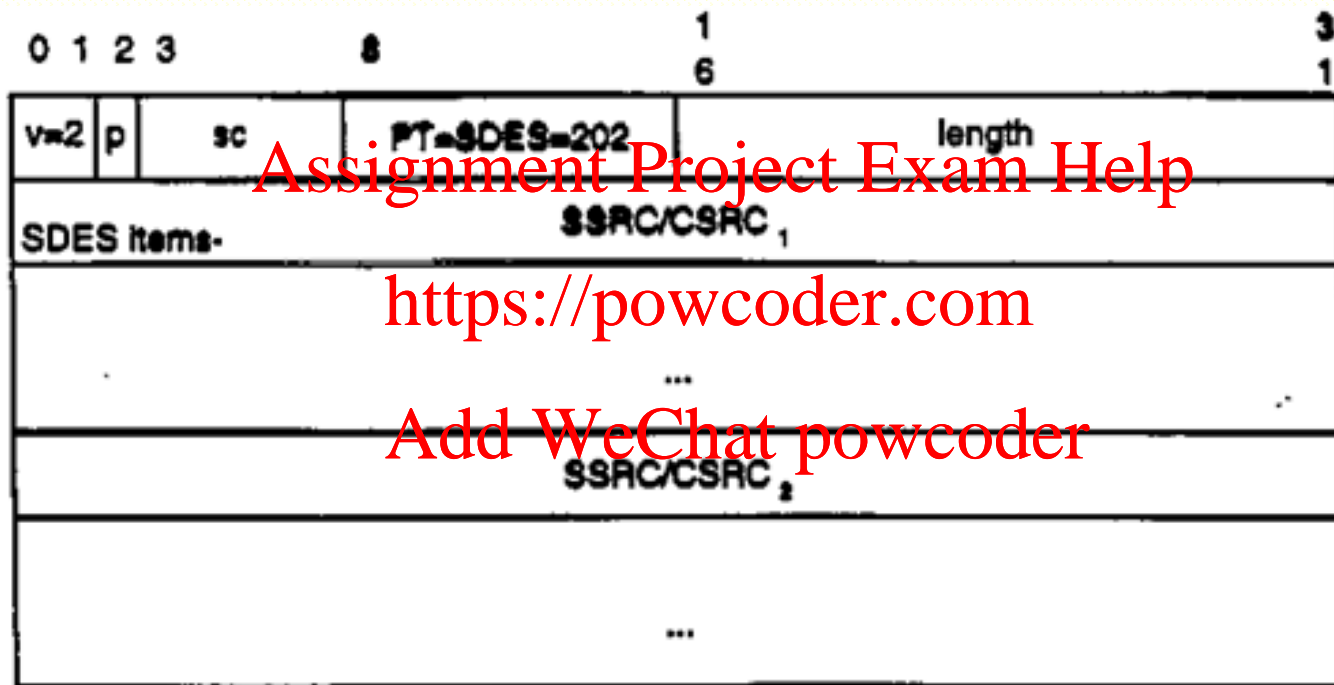
RTCP SDES packets

- ◆ binds an SSRC to sender's real identification
- ◆ up to 32 participants identified
- ◆ SDES items contain item id, length of item, and item itself
- ◆ Only 8 SDES items currently proposed

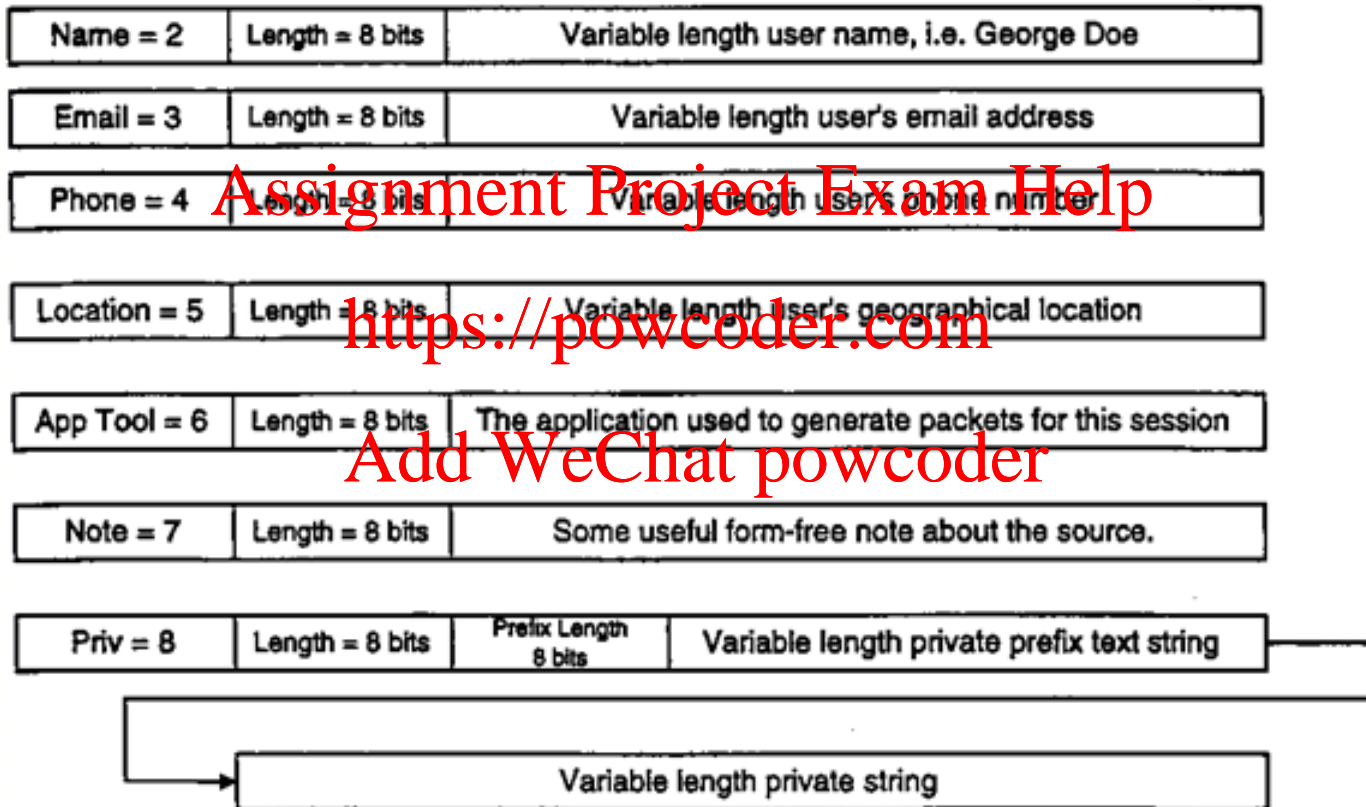
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RTCP Source Description (SDS) Packet Header



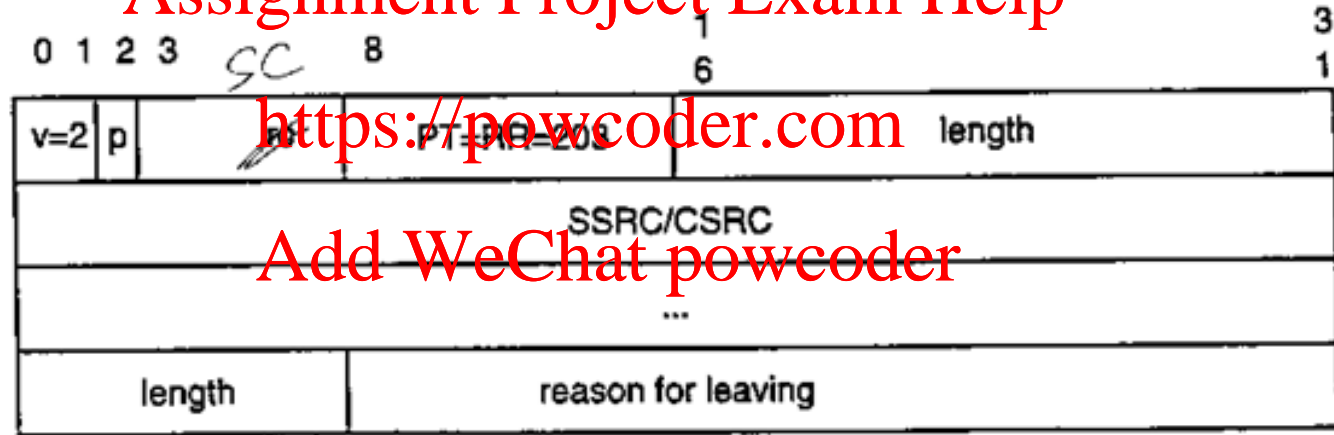
SDS Item Description

Bye packet

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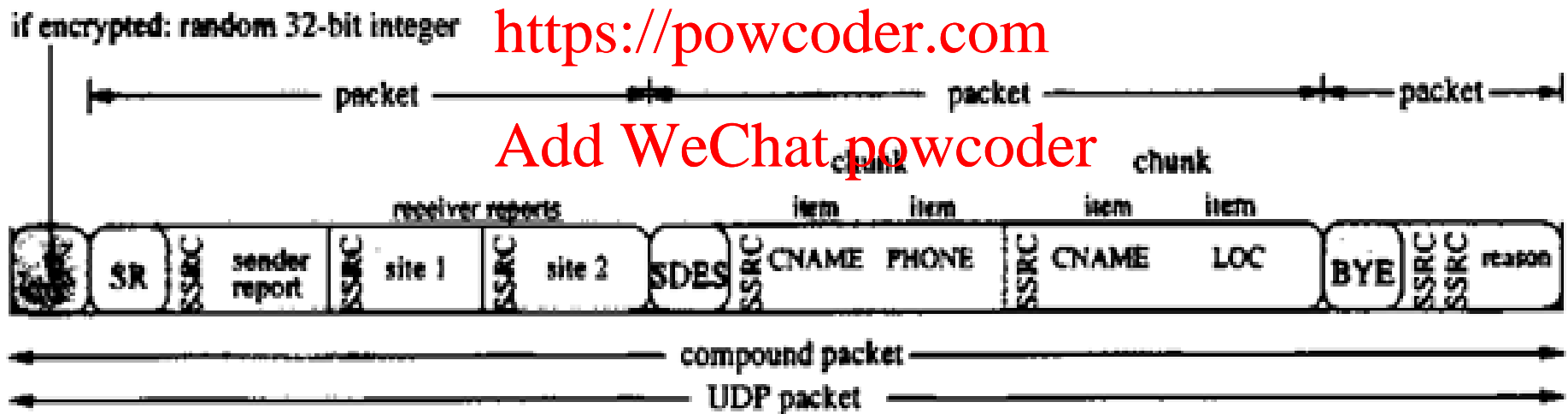


RTCP BYE Packet

RTCP packet

RTCP packet structure

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Multimedia protocol stack

