RTP Payloads

RTP Payload Types

- 7-bit payload type identifier
  - Some numbers statically assigned
  - Dynamic payload types identifiers for extensions – mapping to be defined outside of RTP (control protocol, e.g. SDP “a=rtpmap:”)

Payload formats defined for many audio/video encodings

- Conferencing profile document RFC 3551
  - Audio: G.711, G.722, G.723.1, G.728, GSM, CD, DVI, …
- In codec-specific RFCs
  - Audio: Redundant Audio, MP-3, ...
  - Video: JPEG, H.261, MPEG-1, MPEG-2, H.263, H.263+, BT.656
  - Others: DTMF, text, SONET, ...
- Generic formats
  - Generic FEC, (multiplexing)

IANA maintains a list
http://www.iana.org/assignments/rtp-parameters

Media Packetization Schemes (1)

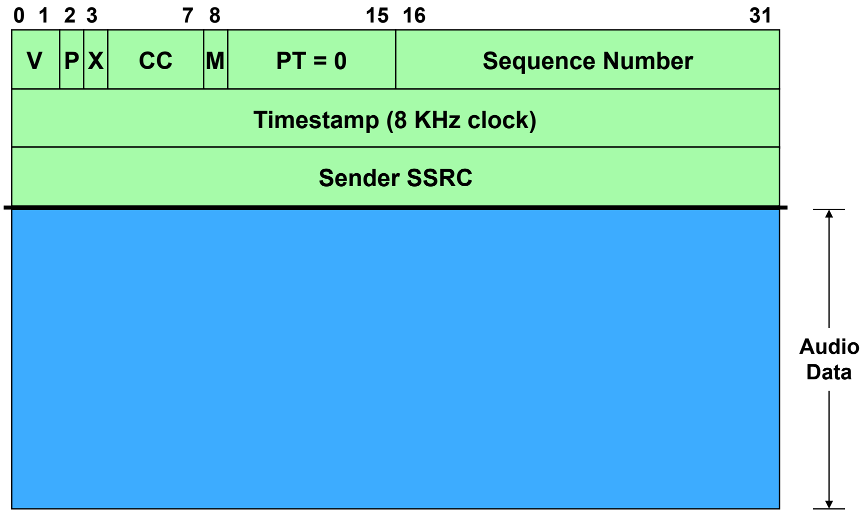
General principle:

- Payload specific additional header (if needed)
- Followed by media data
  - Packetized and formatted in a well-defined way
  - Trivial ones specified in RFC 3551
  - RFC 2029, 2032, 2035, 2038, 2190, 2198, 2250, 2343, 2429, 2431, RFC 2435, 2658, 2733, 2793, 2833, 2862, and many further ones
  - Guidelines for writing packet formats: RFC 2736
- Functionality
  - Enable transmission across a packet network
  - Allow for semantics-based fragmentation
  - Provide additional information to simplify processing and decoding at the recipient
  - Maximize possibility of independent decoding of individual packets

Sample RTP Payload Types

Illustrate a variety of approaches to deal with packet loss in the Internet

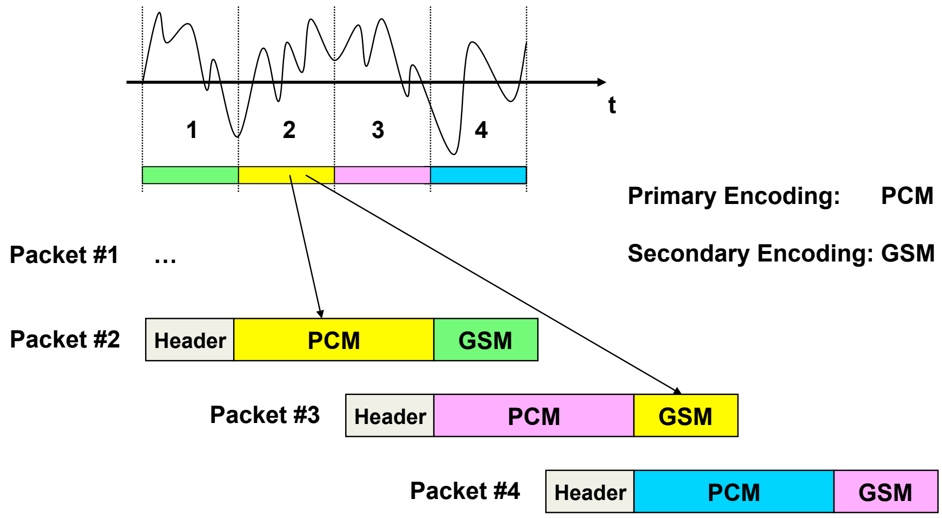
Audio over RTP: PCM



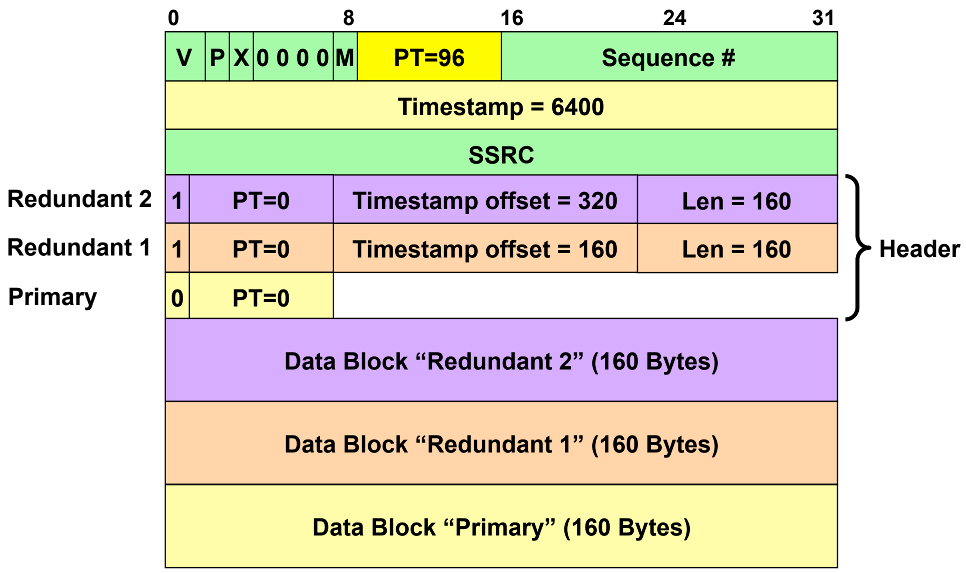
Audio Redundancy Coding (1)

- Audio Packets are small!
  - have to be because of interactivity
    - avoid large packetization delay
  - packet loss primarily depends on packet rate
    - rather than packet size
- Payloads for multiple time slots in one packet
  - send redundant information in packet n
    to reconstruct packets k, ..., n-1
  - redundant information typically sent at lower quality
  - details defined in RFC 2198
  - uses dynamic payload type
- Format specification, e.g. using SDP
  - m=audio 20002 RTP/AVP 96 0 0 0
  - a =rtpmap:96 red/8000/1

Audio Redundancy Coding (2)



Audio Redundancy Coding (3)

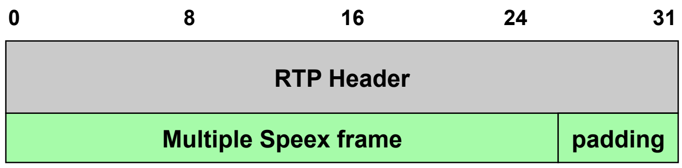


Standardized in RFC 5574

Speex

- It is based on Code Excited Linear Prediction [CELP]
- Opens source audio codec for VoIP applications
- When encoded at max bit-rate, a frame would be approx. 110 octets. Single frame is < path MTU.
- Speex frames can be combined and sent together, however no speex frame should be fragmented
- The padding begins with a 0 and is followed by a series of 1s to complete the octet

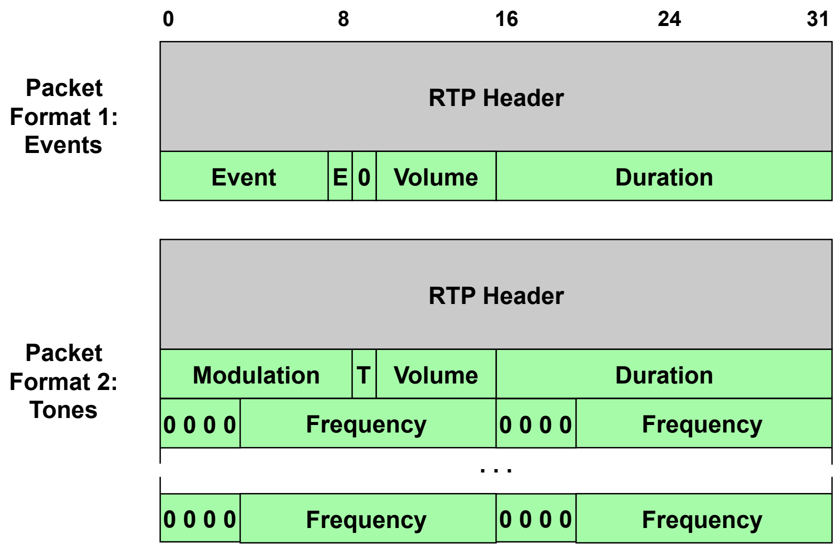
Example



DTMF over RTP (1)

- DTMF digits, telephony tones, and telephony signals
  - two payload formats
  - 8 kHz clock by default
  - audio redundancy coding for reliability
- Format 1: reference pre-defined events
  - 0 - 9 * # A - D (Hook)Flash [17]
  - modem and fax tones [18]
  - telephony signals and line events [43]
    - dial tones, busy, ringing, congestion, on/off hook, …
  - trunk events [44]
  - specified through identifier (8-bit value), volume, duration
- Format 2: specify tones by frequency
  - one, two, or three frequencies
  - addition, modulation
  - on/off periods, duration
  - specified through modulation, n x frequency, volume

DTMF over RTP (2)



Video Redundancy Coding (1)

- Video redundancy coding
  - For H.263+ video streams
  - Transmit several interleaved sequences of predicted frames (threads) instead of one
    - improves error resilience against packet loss
- Principle
  - create several (n) independently decodable streams
  - achieved by choosing different reference pictures
  - decode only streams with no packet losses
    - reduces temporal resolution by 1/n-th per affected stream
  - bit rate penalty due to larger deltas between frames
  - RFC 2429, revised version in progress

Video over RTP: H.261

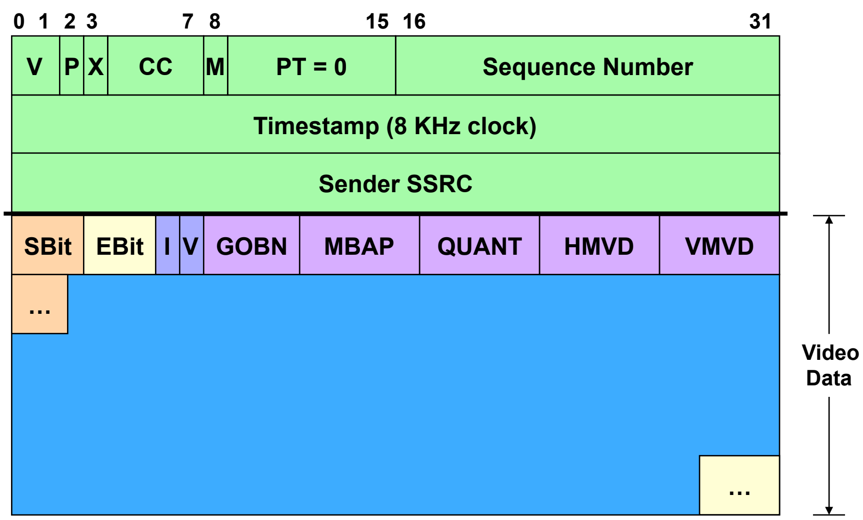
Additional payload-specific header preceeds payload

- To avoid expensive bit shifting operations
  - Indicate # invalid bits in first (SBit) and last (EBit) octet of payload
- Indicate Intra encoding (I bit)
- Indicate the presence of motion vector data (V bit)
- Carry further H.261 header information to enable decoding in the presence of packet losses

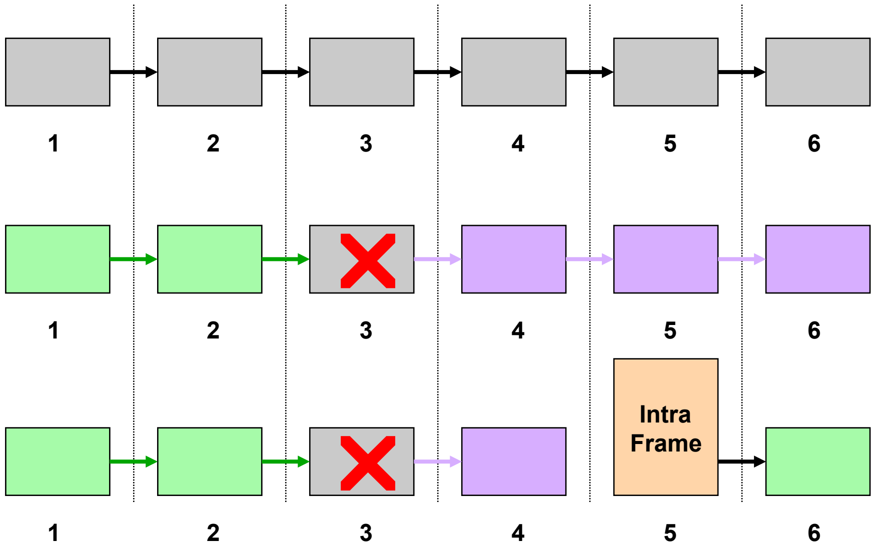
Further mechanisms for video conferencing

- FIR: Full Intra Request
  - Ask sender to send a full intra encoded picture
- NACK: Negative Acknowledgement
  - Indicate specific packet loss to sender

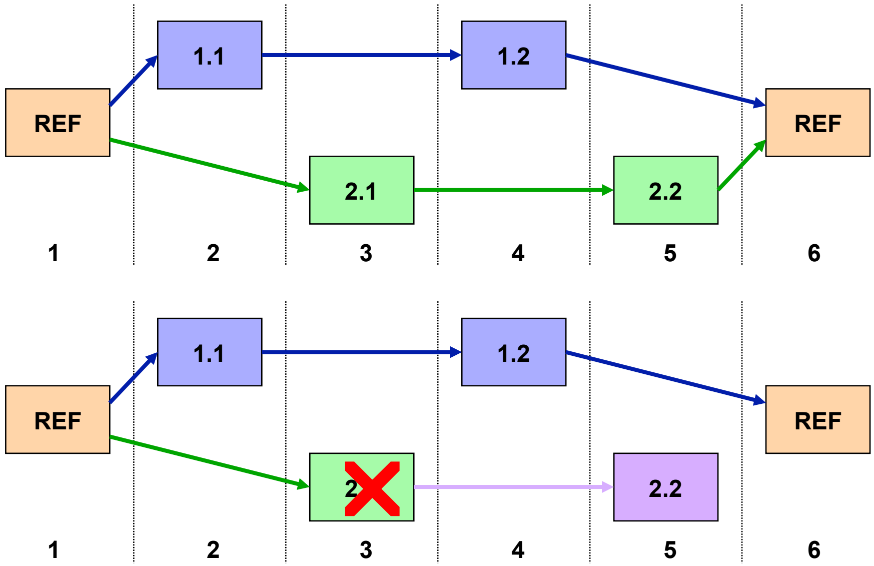
Video over RTP: H.261 (2)



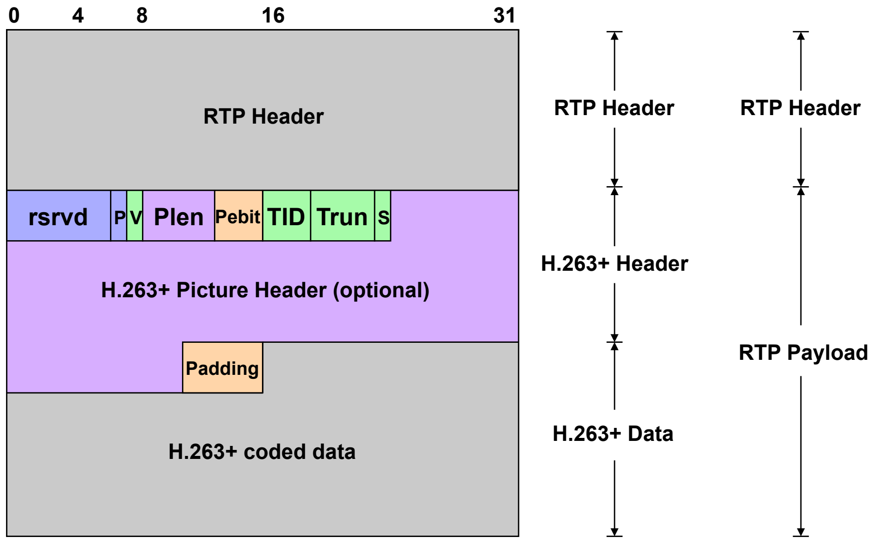
Video Redundancy Coding (2)



Video Redundancy Coding (3)

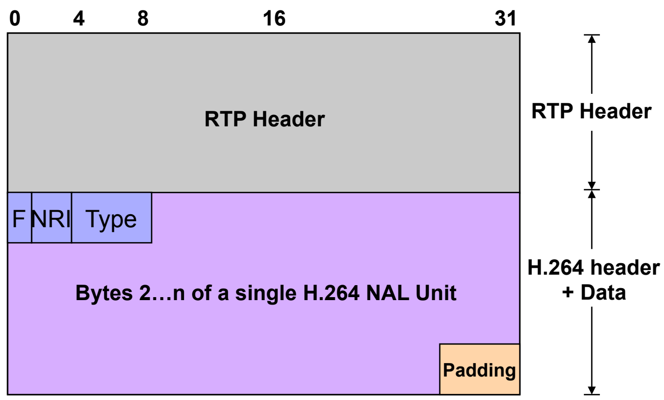


Video Redundancy Coding (4)



H.264/AVC

- AVC stands for Advanced Video Coding
- RTP profile for H.264 AVC is standardized in RFC 3984
  - Updated draft 3984bis is being worked on



F is a flag for bit-error checking

NRI is 2 bits and indicates that usefulness of the NALU for reference frame reconstruction

Type is 5 bits and indicates NALU type. They are:

Single Unit (1-23), Single time aggregation packet(24, 25), Multi-time aggregation packet (26,27), Fragmentation unit (28-29).

SVC

- RTP profile for scalable Video coding is under final review will become an RFC soon
- SVC is an extension, Annex. G of H.264 standard
- Packet loss can cause artifacts in high quality video but by using more computational complexity an SVC compliant codec can achieve the same playback quality but at lower image quality.
- Extended NALU Type 14, 15 and 20 of H.264 AVC and defined as an 24 bit header.

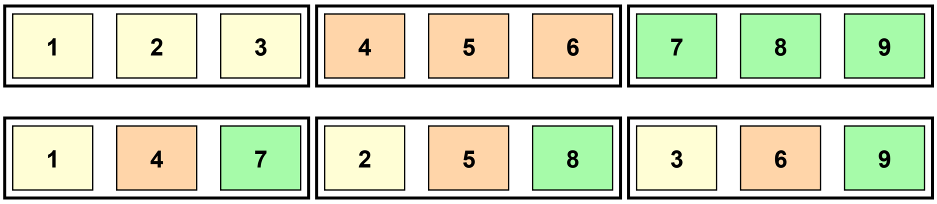
Media Packetization Schemes (2)

<u>Error-resilience for real-time media</u>

- Input: Observation on packet loss characteristics
- Generic mechanisms (RFC 2354)
  - Retransmissions
    - in special cases only (e.g. with no interactivity!)
  - Interleaving
  - Forward Error Correction (FEC)
    - media-dependent vs. media-independent
    - Generic FEC: RFC 2733
- Feedback loops for senders
  - based upon generic and specific RTCP messages
  - adapt transmission rate, coding scheme, error control, ...

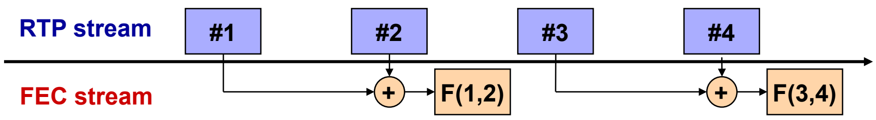
RTP Interleaving

- Distribute packets or packet contents for transmission
  - Avoid consecutive packet erasures in case of (burst) losses
  - Avoid loss of large consecutive data portions in case of single packet losses
- Motivations
  - Human perception tolerates individual losses better (with error concealment)
  - Make simple FEC schemes work better with burst losses (e.g. XOR)
- Drawback
  - Re-ordering causes additional delay

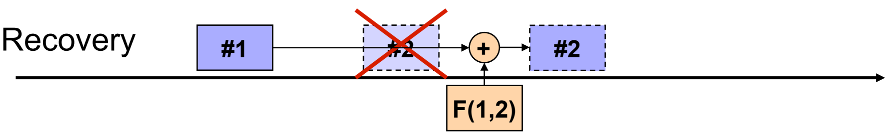


RTP FEC (RFC 2733)

- Forward Error Correction scheme for RTP packets
  - Media-independent, flexible FEC (that can be enhanced)
- Simple XOR-based (parity) FEC
  - P_fec = P1 XOR P2 XOR P3 XOR … XOR Pn
    - Allows reconstruction of any single missing packets of P1, …, Pn, P_fec
- RTP FEC stream transmitted independently of RTP stream
  - Separate transport address (IP address, port number)
  - Different SSRC

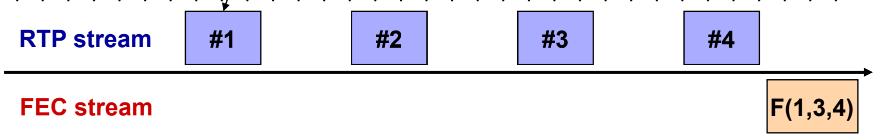


- Recovery



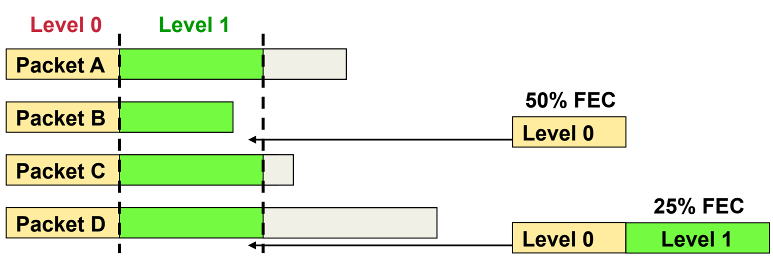
RTP Parity FEC Packet format

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|   FMT   |       PT      |             length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of packet sender                        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  SSRC of media source                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|            SN base    = 1     |          length recovery      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|E| PT recovery |          mask   = 10110000 00000000 00000000   |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                          TS recovery                          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
:                                                               :
:         XOR of Payloads indicated by SN Base and mask         :
:                                                               :
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```



Unequal Error Protection

- Observation: not all parts of a packet are equally important
  - Beginning of packet contains headers/parameters, more relevant contents
  - Holds for both audio and video
- Uneven Level Protection (ULP)
  - Create independent parity packets for different parts of packets
  - Allows for selectively more overhead for the more important parts



- Related thoughts: partial checksums
  - Live with bit errors in the less important parts (rather than dropping a packet)

ULP FEC payload format

```
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                RTP Header (12 octets or more)                 |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    FEC Header (10 octets)                     |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      FEC Level 0 Header                       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                     FEC Level 0 Payload                       |
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      FEC Level 1 Header                       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                     FEC Level 1 Payload                       |
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                            Cont.                              |
|                                                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

- FEC header is constructed with one ore more level of FEC encoding

defined in RFC 5104

New FEC packet

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|E|L|P|X|  CC   |M| PT recovery |            SN base            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                          TS recovery                          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|        length recovery        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

- Standardized so that it is agnostic to Audio/Video data as much as possible
- E and L are flags for extension bits (E), long mask (L)
- P, X, CC, M and PT are the recovery fields for the corresponding fields in the FEC‘ed packets.

FEC level header

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|       Protection Length       |             mask              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|              mask cont. (present only when L = 1)             |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

- FEC level header indicates which packets are associated with the FEC packet
- The L flag specifies if the mask is 16 bits or 48 bits.
- The rules for masking are based on the levels of protection.
- So a media packet protected at LEVEL “P” should also be protected at LEVEL “P-1” in any FEC packet

RTP Payload Type Overview (1)

- RFC 3551 Collection of simple packetization formats (formerly RFC 1890)
- RFC 2029 Sun CellB Video encoding
- RFC 2032,4587 H.261 video
- RFC 2435 JPEG video (was RFC 2035)
- RFC 2250 MPEG-1/MPEG-2 video (was RFC 2038)
- RFC 2190 H.263 video (historic)
- RFC 2343 Bundled MPEG
- RFC 2429 H.263+ video & video redundancy support
- RFC 2431 BT.656 video
- RFC 2658 PureVoice audio
- RFC 2793,4103 Text conversation
- RFC 2833 DTMF, telephony tones, and telephony signals
- RFC 2862 Real-time Pointers
- RFC 3016 MPEG-4 Audio/visual streams
- RFC 3047 G.722.1 audio
- RFC 3119 Loss-tolerant format for MP3
- RFC 3189 DV video
- RFC 3190 12-bit DAT and 20-/24-bit linear audio

RTP Payload Type Overview (2)

- RFC 3267,4352 Adaptive Multirate (AMR, AMR-WB+) audio
- RFC 3389 Comfort noise
- RFC 3497 SMPTE 292M video
- RFC 3557 ETSI Distributed speech recognition (ES 201 108)
- RFC 3558 Enhanced variable rate codecs and selectable mode vocoders
- RFC 3640 MPEG-4 elementary streams
- RFC 3952 Low Bit Rate Codec (iLBC) Speech
- RFC 3984 H.264 Video
- RFC 4040 64 kbit/s Transparent Call
- RFC 4060 Distributed speech recognition encoding (ES 202 050/211/212)
- RFC 4175,4421 Uncompressed Video
- RFC 4184,4598 AC-3 Audio, Enhanced AC-3
- RFC 4298 BroadVoice Speech codec
- RFC 4348,4424 Variable Rate Multimodal Wideband Audio (VMR-WB)
- RFC 4351 Text conversation interleaved with audio stream
- RFC 4396 3GPP Timed Text
- RFC 4425 Video Codec 1 (VC-1)
- RFC 4588 Retransmission payload format

Many more to come…