Question 1

$$48Kbps = 6000Bps$$

The sender generates data at 6000 Bytes per second.

Question 2

- RTP header size = 12 bytes
- UDP Header size = 8 bytes
- IP size = 20 Bytes
- Size of data chunk generated $6000Bps * 16ms = 96\ Bytes$
- RTP Packet = $12\ bytes + 96\ Bytes = 108\ Bytes$
- IP datagram size = $8B + 20B + 108B = 136\ Bytes$

The size of the IP-datagrams sent are 136Bytes.

Question 3.

The RTP-packet header has four main fields: Payload Type, Sequence number, Timestamp and Synchronization source identifier. With the addition of the payload this creates an RTP packet.

- Payload type (7 bits): This indicates what type of payload the packet includes. In our example it would be: 9 for G.722 encoded data.
- Sequence number (16 bits): This represents the sequence number of the packet; this will increment by one for each packet sent. As the packet is arbitrary, I chose number 20.
- Timestamp field (32 bits): The timestamp increments by one for each sampling period. The sampling clock in our example is 16 Khz, which means that our sampling period is 40 µs. And since our chunks consist of 16 ms of data. The timestamp should increment with 400 for each packet, as it consists of 400 encoded samples. This field will increment constantly, even if the source is inactive.
- Synchronization source identifier field (32 bits): Identifier for the source of the packet. This is not the IP-address of the sender, but a SSRC value. It is a random 32bit number assigned when a new stream is started.

Lastly the packet contains a payload, which is a series of bits, witch voice encoded data.

Question 4

FEC with redundant encoded chunks: Since a redundant packet chunk is generated every 5 chunks:

$$n = 5$$
$$incr = \frac{1}{5} = 20\%$$
$$6000Bps * 20\% = 7200\ Bps$$

The bandwidth is increase by 20% to 6687,5Bps.

FEC with redundant lower-resolution audio stream:

$$GSM\ bandwidth = 13kbps$$
$$13kbps\ = 1625\ Bps$$
$$\frac{1625Bps}{6000Bps} * 100 = 27.08\%$$
$$6000Bps * 27.08\ \% = 7624{,}8\ Bps$$

The bandwidth is increase by 27.08% to 7624,8Bps.

Interleaving does not increase the bandwidth but increases latency.

Question 5

FEC adds redundant information to the original packet stream. While this marginally increases the transmission rate, it also allows the redundant information to be used to reconstruct approximations or exact versions of some of the lost packets.

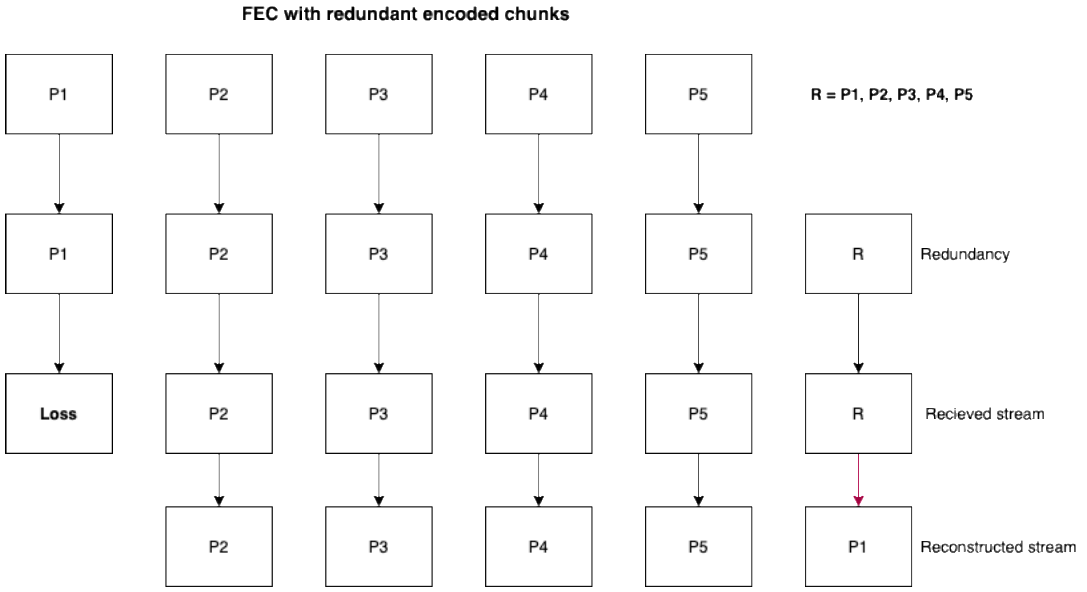
In FEC with redundant encoded chunks, the redundant chunk is obtained by exclusive OR-ing the n original chunks. Therefore, if the first packet of five is lost, the receiver can fully reconstruct that packet.

In FEC with redundant lower-resolution audio stream the sender can create a nominal audio stream and a corresponding low-resolution and low-bit audio stream. In this case GSM is used for the redundant, low-quality stream. Here the sender can append the (n-1)st chunk from the redundant stream to the nth chunk from the nominal stream. In the case of losing the first in a group of five packets, the receiver can conceal this loss by playing out the low-bit rate encoded chunk that arrives with the second packet, at the expense of the audio quality.

In Interleaving the sender resequences the units of audio data so that if a packet I lost, the overall gaps in audio data are smaller and more spread out, as opposed to a noninterleaved stream that may contain a single large gap in audio data. If the first packet is lost while interleaving, the loss inn data will be spread out over a longer period and be hardly noticeable.

While interleaving give good audio quality, the best audio quality will in this case be provided by FEC with redundant encoded chunks because this scheme can completely reconstruct the lost packet.

Question 6



Question 7.

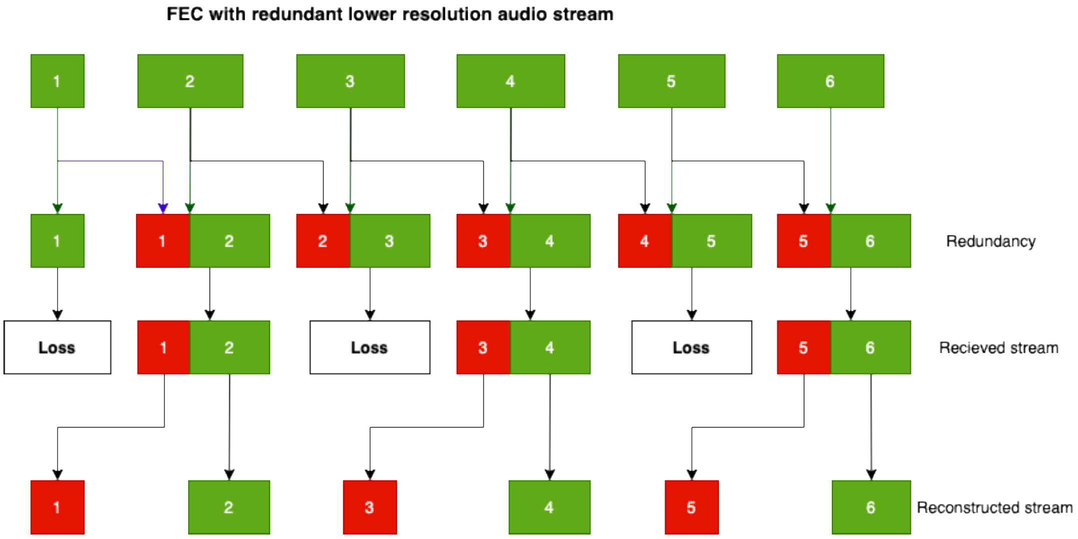
For FEC with redundant encoded chunks there are a lot of missing packets in this case. Because of this, the scheme can´t perfectly reconstruct the missing packets and therefore has to make an approximation of the lost packets, which lowers quality.

The FEC with redundant lower resolution audio stream still works by using the appended lower quality stream for the second packet in every group but will lead to a poorer audio quality overall.

The interleaving scheme will resequencing the audio data but because of the frequent loss of packets, the number of gaps in missing audio data increases to the point that lowers the quality significantly.

The better audio quality will therefore be provided by FEC with redundant lower resolution audio stream, since despite its reduce in quality, it will still provide audio, while use of the two other schemes might result in missing audio

Question 8



Question 9

The FEC with redundant encoded chunks requires all n chunks to be received before playback. Since the application collects encoded data in 16 millisecond chunks and n = 5, the delay is 5 packets, or 16ms* 5=80ms.

The FEC with redundant lower-resolution audio only needs to receive two packets before playback and therefore the playback delay will be 16ms* 2=32ms.

When it comes to interleaving, one of its biggest disadvantages is that it increases latency. This is because it requires all the packets back to correctly sequence and reconstruct the stream. This limits its use for conversational applications; like VoIP.Interleaving: