Datarate

The calculated data rate includes IP, UDP & RTP headers.

Datarate (bit / sek) : dr = (samplerate/(8*audioPacketSize))* (40*8 + dataSize*(8*audioPacketSize))

dataSize:	8, 12 or 16 bits
audioPacketSize:	1 to 20 units of 8 samples each
constraint:	dataSize*8*audioPacketSize

Example 1: dr = (8000/(8*20)) * (40*8 + 8*(8*20))
dr = 80000 bit/sek

Example 2: dr = (8000/(8*5)) * (40*8 + 16*(8*5))
dr = 448000 bit/sek

Example 3: dr = (16000/(8*5)) * (40*8 + 16*(8*5))
dr = 896000 bit/sek

Example 4: dr = (24000/(8*1)) * (40*8 + 16*(8*1))
dr = 1344000 bit/sek

NOTE:
Constraints regarding APS (audio packet size) related to the implementation.
Any greater value in the settings will fall back to highest allowed.

Datasize (bits)	APS (sample units)
8	1-20
12	1-6
16	1-10

Table of Approximate datarate

(APS - audio packet size)

8 kHz

APS	pl (b)	oh (b)	/s	bw8 (kbps)	bw16 (kbps)
20	160	40	50	80	-
10	80	40	100	96	160
5	40	40	200	128	192
2	16	40	500	224	288
1	8	40	1k	384	448

16 kHz

APS	pl (b)	oh (b)	/s	bw8 (kbps)	bw16 (kbps)
20	160	40	100	160	-

10	80	40	200	192	320
5	40	40	400	256	384
2	16	40	1k	448	576
1	8	40	2k	768	896

24 kHz

APS	pl (b)	oh (b)	/s	bw8 (kbps)	bw16 (kbps)
20	160	40	150	240	-
10	80	40	300	288	480
5	40	40	600	384	576
2	16	40	1500	672	864
1	8	40	3k	1152	1344

Delay

The delay between recoding at one end and playback at the other end comes from several sources.

* Codec internal delay in both AD and DA converters.
* Propagation delay in the network.
* Buffering to allow for an uninterrupted audio stream.

Total delay = codec delay + network delay + buffering delay

Codec internal delays

Solely dictated by the samplerate.

Samplerate (kHz)	Delay (AD + DA) (ms)
8	6.25
12	4.2
16	3.125
24	2.1

Network delay

Depends on the network over which the audio is transmitted.
At best it is down at sub milli second lengths, but can extend indefinitely depending on the constitution of the network.

Buffering delay

Buffering delay: Jitter delay * Audio packet size*8/samplerate

Approximate minimum delays

Measured real values including codec delays and minimum amount of network delay.

```
Jitter buf size:    2
Jitter delay:         1
Audio packet size:  1

Audio quality     Delay
            (ms)
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0           10-11
1           10-11
2           10-11

3           7.5-8.5
4           7.5-8.5
5           7.5-8.5

6           5-6
7           5-6
8           5-6

9           3.75-4.25
10          3.75-4.25
11          3.75-4.25
```

Glosary
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Audio quality
: Refers to user selectable modes of operation, with varying samplerates and data encoding, and thereof following signal bandwidth and resolution.

Jitter buf size
: Maximum number of audio packets buffered. Dictates maximum playback delay.

Jitter delay
: Number of packets received and buffered before beginning playback. Dictates minimum playback delay.

Audio packet size
: Number of samples within an audio packet. Always a multiple of 8 samples. Some audio quality modes allow 1 to 20, other will automatically limit the size to maximum allowed by internal structures.