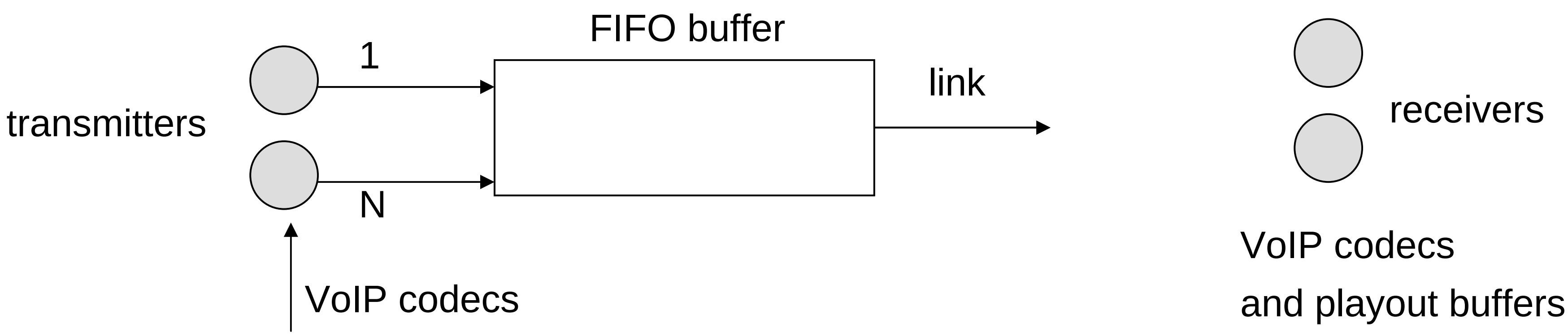
MOS-oriented design of the VoIP service

- N VoIP flows are multiplexed on a transmissionb link through a FIFO buffer
- Calculate the MOS performance of the telephone service with N=200 VoIP flows, assuming that the adopted VoIP codec is the G.726@32 kbit/s with Voice Activity Detection and and the link is a 10 Mbit/s Ethernet with the IEEE 802.1q VLAN option



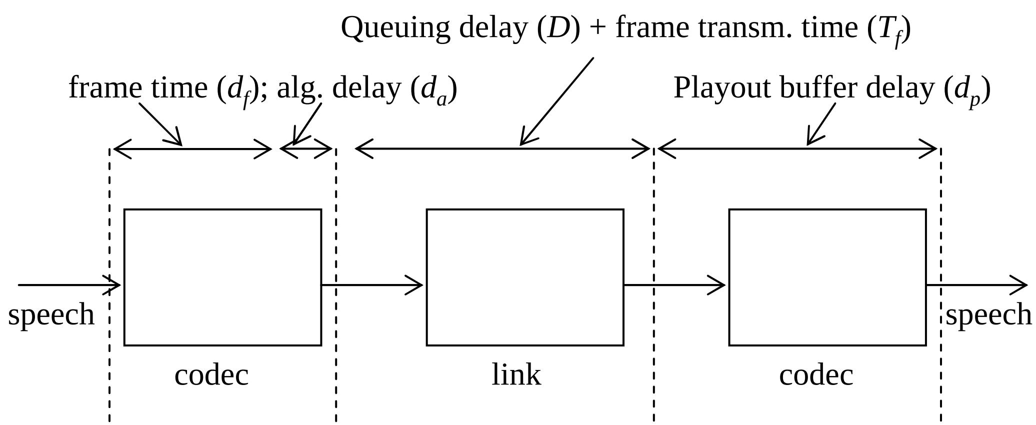
MOS values of the G.726 codec as a function of average delay and packet loss

G.726 codec, 32 kbit/s, voice activity detection						
	Packet loss (%)					
delay (ms)	0.00	0.01	0.05	0.10	0.50	1.00
0.00	4.23	4.21	4.10	3.95	2.93	2.17
50.00	4.19	4.16	4.04	3.86	2.86	2.10
100.00	4.15	4.12	4.00	3.85	2.80	2.04
150.00	4.11	4.08	3.96	3.80	2.74	1.99
200.00	3.98	3.94	3.81	3.63	2.55	1.82

Note that packet loss is the probability that a packet arrives at
The receiver after its playout time

MOS-oriented design of the VoIP service

- The delay components are
  - Frame time d_f: 10 ms for the G.726 codec
  - algorithmic delay d_a: 0.125 ms for the G.726 codec
  - Queuing delay D, to be calculated with statistical network calculus
  - Transmission time t_f: to be calculated according to link speed and protocol overhead
  - Playout buffer time, d_p



Calculation of average delay

$$d = d_f + d_a + D + t_f + d_p$$

$$E(d) = d_f + d_a + E(D) + t_f + d_p$$

Calculation of average delay

- In general, given a SRD traffic, the average delay is equal to

$$E(D)=\frac{Nrb}{2C(C-Nr)}$$

thus

$$E(d)=d_f+d_a+\frac{Nrb}{2C(C-Nr)}+t_f+d_p$$

Calculation of average delay

- In turn, given the two-state model of the variable bit rate VoIP traffic:

$$E(d) = d_f + d_a + \frac{2N \frac{\lambda\mu}{(\lambda+\mu)^3} P^2}{2C\left(C - N\frac{\lambda}{\lambda+\mu}P\right)} + t_f + d_p$$

Where P is the actual rate of a single VoIP flow, at the Phisical layer

Calculation of P

- The calculation of the actual peak rate *P* of voice depends on protocol overheads; with RTP/UDP/IP over Ethernet (with the IEEE 802.1q VLAN option):
  - RTP HEADER: 12 bytes
  - UDP HEADER: 8 bytes
  - IP HEADER: 20 bytes
  - ETHERNET PREAMBLE: 7 bytes
  - ETHERNET START FRAME DELIMITER: 1 byte
  - ETHERNET DESTINATION ADDRESS: 6 bytes
  - ETHERNET SOURCE ADDRESS: 6 bytes
  - ETHERNET FRAME TYPE LENGTH: 2 bytes
  - ETHERNET 802.1Q VLAN: 2 bytes
  - ETHERNET FRAME CHECK SEQUENCE: 4 bytes
  - ETHERNET INTERFRAME GAP: 12 bytes
- Thus, the per-IP packet ovehead is 80 bytes

Calculation of P

- With the G.726@32 kbit/s codec, with a frame time of 10 ms, the number of payload bytes per packet is equal to 40, thus, the effective peak rate of the codec is P = 8 x (40 + 80) / 0.01 = 96 kbit/s

Other numerical parameters

- Frame time d_f: 10 ms for the G.726 codec
- algorithmic delay d_a: 0.125 ms for the G.726 codec
- Transmission time t_f: to be calculated according to link speed and protocol overhead
  - T_f=120x8/10,000,000=0.096 ms
- Playout buffer time: $d_p=20\ ms$
- μ=2.8571 s^{-1}
- λ=1.538 s^{-1}

Calculation of packet loss

- A packet is lost when the network delay is larger than the playout buffer delay, thus: $p=\Pr(D>d_p)$, that is:

$$p = \Pr\left(D > d_p\right) \approx \exp\left(-2 \frac{C\left(C - N\frac{\lambda}{\lambda+\mu}P\right)}{2N\frac{\lambda\mu}{\left(\lambda+\mu\right)^3}P^2} d_p \right)$$

Numerical values

- With N=200:
  - E(d)=53 ms
  - p=0.1%
- From the MOS table: MOS = 3.86