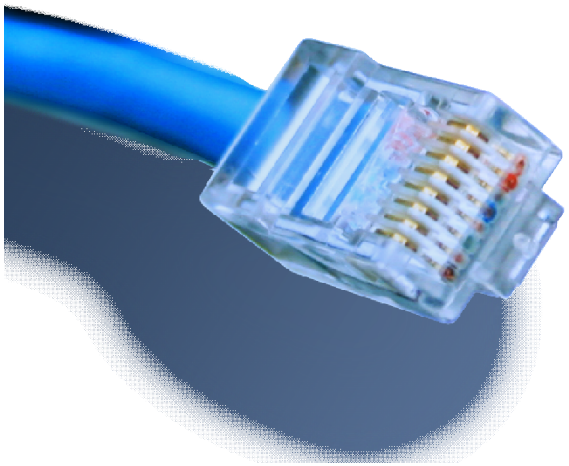
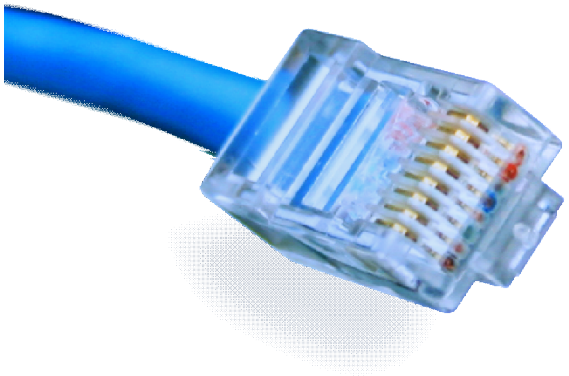
KARAKTERISASI DELAY DAN PACKET LOSS PADA RURAL NEXT GENERATION NETWORK (R-NGN)



Faktor-faktor QoS pada VoIP

- Speech Encoding
- Delay
- Jitter
- Packet Loss



Speech Encoding

- menjelaskan bagaimana suatu aliran sinyal analog suara di-digitalisasi dan dikompresi menjadi suatu bentuk digital sehingga siap dikirimkan ke dalam jaringan paket
- Diimplementasikan dengan *codec*

Compression Method	Bit Rate (kbps)	Processing Power (MIPS)	Framing Size (ms)	MOS Score
G.711 PCM	64	0.34	0.125	4.1
G.726 ADPCM	32	14	0.125	3.85
G.728 LD-CELP	16	33	0.625	3.61
G.729 CS-ACELP	8	20	10	3.92
G.729 x2 Encoding	8	20	10	3.27
G.729 x3 Encoding	8	20	10	2.68
G.729a CS-ACELP	8	10.5	10	3.7
G.723.1 MPMLQ	6.3	16	30	3.9
G.723.1 ACELP	5.3	16	30	3.65

Delay

- Waktu yang dibutuhkan oleh sebuah paket data terhitung dari saat pengiriman oleh *transmitter* sampai saat diterima oleh *receiver*.

  - Accumulation Delay

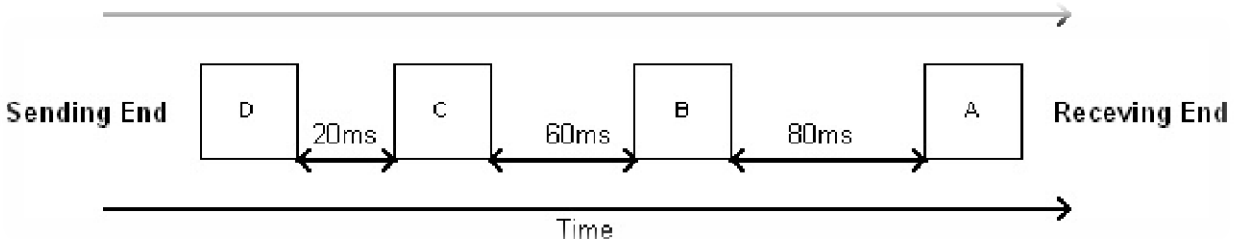
    Delay yang terjadi karena sinyal suara harus dikumpulkan terlebih dahulu sebelum di proses. Delay ini tergantung oleh codec yang digunakan
  - Processing Delay

    Delay ini terjadi karena proses pengumpulan dan pengkodean sample suara menjadi paket data yang siap untuk ditransmisikan dalam jaringan. Delay ini tergantung pada kecepatan processor dan algoritma yang digunakan.
  - Network Delay

    Delay ini disebabkan oleh medium fisik jaringan dan protokol yang digunakan untuk transmisi data. → queuing delay, transmission delay, propagation delay

- Standard :
  - Good (0ms-150ms)
  - Acceptable (150ms-400ms)
  - Poor (> 400ms)

Jitter



- variasi *delay*, yaitu perbedaan selang waktu kedatangan antar paket di terminal tujuan
- Disebabkan oleh kongesti, kurangnya bandwidth, varisasi ukuran paket, ketidakurutan paket
- Standard :
  - Good (0ms-20ms)
  - Acceptable (20ms-50ms)
  - Poor (> 50ms).

Packet Loss

- Banyaknya paket yang hilang selama proses transmisi ke tujuan

- Disebabkan oleh :
  - Terjadinya *overload* trafik didalam jaringan,
  - Kongesti dalam jaringan,
  -Error yang terjadi pada media fisik,

- Standard :
  - Good (0%-0.5%)
  - Acceptable (0.5%-1.5%)
  - Poor (> 1:5%)

Pengukuran Kualitas Suara pada VoIP

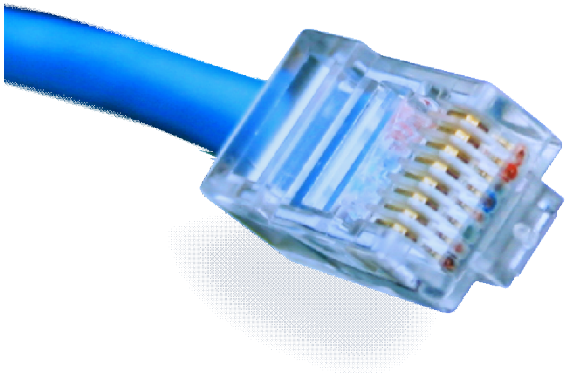
- Metoda Subjektif

  Kualitas suara ditentukan oleh persepsi user yang dikonversikan pada skala *Mean Opinion Score* (MOS)

- Metoda Objektif

  Kualitas suara ditentukan dari hasil pengukuran parameter-parameter QoS sistem kemudian dikonversikan pada skala *Mean Opinion Score* (MOS)

  Menggunakan “E-Model” (ITU-T G.107)



MOS Rankings

Quality Scale	Score	Listening Effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with reasonable effort



Objective Method

- **PSQM** (***Perceptual Speech Quality Measure***) is an ITU-T P.861 recommended objective technique for determining the quality of speech codecs.

- **PAMS (*Perceptual Analysis Measurement System),*** *is a technique* that enables researchers to asses the quality of a voice conversation over a wide range of networks

- **PESQ (P*erceptual evaluation of speech quality)*** *is the result* of integrating PAMS and PSQM. The ITU-T has Withdrawn P.861 and replaced it with **P.862 (PESQ)** which contains an improved speech assessment algorithm.

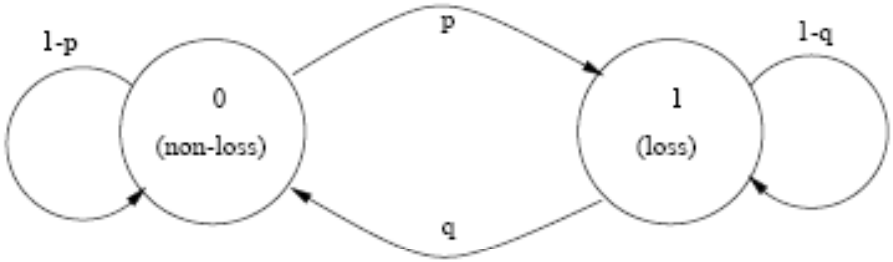
Basic Formula for the E-model

- Developed by ETSI → Standardized : ITU-T G.107 defines an objective model known as E-Model based on Network, Speech, Terminal/Device parameters to estimate/predict the perceived quality of VoIP session.

- R-value = Ro - Is - Id - Ie + A
  - Ro
    - the basic **signal-to-noise ratio** based on sender and receiver loudness ratings and the circuit and room noise
  - Is
    - the sum of real-time or **simultaneous speech transmission impairments**,e.g. loudness levels, sidetone and PCM quantizing distortion
  - Id
    - the sum of **delay impairments** relative to the speech signal, e.g., talker echo, listener echo and absolute delay
  - Ie
    - the **equipment impairment factor** for special equipment, e.g., low bit-rate coding (determined subjectively for each codec and for each % packet loss and documented in ITU-T Recommendation G.113) I_e *covers the negative effects that are* due to distortion (by, for example, using a low bit rate codec).
  - A
    - the **advantage factor** describes the decrease in the total rating R that users will tolerate compared to traditional PSTN telephone, for example ease of use, ease of access, etc.

Gilbert Model

- 2-State Markov model



- *p* is the probability that the next packet is lost, provided by the previous one has arrived. *(1-q)* is the conditional loss probability (clp). Normally *p+q < 1*. If *p+q = 1*, Gilbert model reduced to Binomial model
- Gilbert model represent the mean arrival and loss probability