Cairo University
Faculty of Engineering
Electronics and Electrical Communications Department

Professional Masters Program – Major Telecommunications

ECP 610: Multimedia Communications

Part 2: VoIP Quality of Service (QoS)

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Topics

- What is VoIP?
- Why is VoIP Attractive?
- How Does it Work?
- Voice Coding
- VoIP Signaling Standards
- QoS Impairment Factors
- QoS Measurement Methods
- Design of VoIP Network Using SIP

VoIP Main Factors

- **VoIP Codecs** – G Series , AMR.
- **VoIP Signaling Protocols** – H.323 , SIP , H.248 , Megaco , SIGTRAN, BICC .
- **VoIP QoS Impairment Parameters** – Delay , Packet Loss , Jitter.
- **VoIP QoS Measurement Methods** – Subjective Methods (MOS) and Objective Methods (PESQ , E-Model)

VoIP Quality of Service (QoS)

VoIP QoS Impairment Parameters - 1

1. Delay:

- **The time taken by data to travel from the source to the destination is known as delay.**
- **The average time varies according to the amount of traffic being transmitted and the bandwidth available at that given moment. If traffic is greater than bandwidth available, packet delivery will be delayed**
- **Voice is a delay-sensitive application while most data applications are not. When voice packets are lost or arrive late they are discarded; the results are reduced voice quality.**
  - **E2E delay (Customer to Customer)**

    < 250ms (no echo canceling is required)
  - **Objective is < 150ms human ear starts to notice response delay above 150 ms**
  - **400 ms is unacceptable, except for satellite links**

VoIP QoS Impairment Parameters - 2

2. Jitter (Delay Variability):

Jitter is the variation in inter-packet arrival time as introduced by the variable transmission delay over the network.

- E2E should be < 40ms
- Delay variation: example of ETSI TIPHON
  - <10 ms class 1 = gold
  - 10 ms to 20 ms class 2 = silver
  - 20 to 40 ms class 3 = bronze

VoIP QoS Impairment Parameters - 3

3. Packet Loss:

IP networks cannot provide a guarantee that packets will be delivered at all, much less in order. Packets will be dropped under peak loads and during periods of congestion

- E2E packet loss for voice should be < 2%
- ETSI TIPHON (voice)
  - <0.5% class 1 = gold
  - 0.5% to 1% class 2 = silver
  - 1% to 2% class 3 = bronze

Summary of VoIP Network QoS Requirements

Optimal network QoS parameters

- Delay – one way <= 100ms
- Jitter <= 40ms
- Packet loss <= 1%

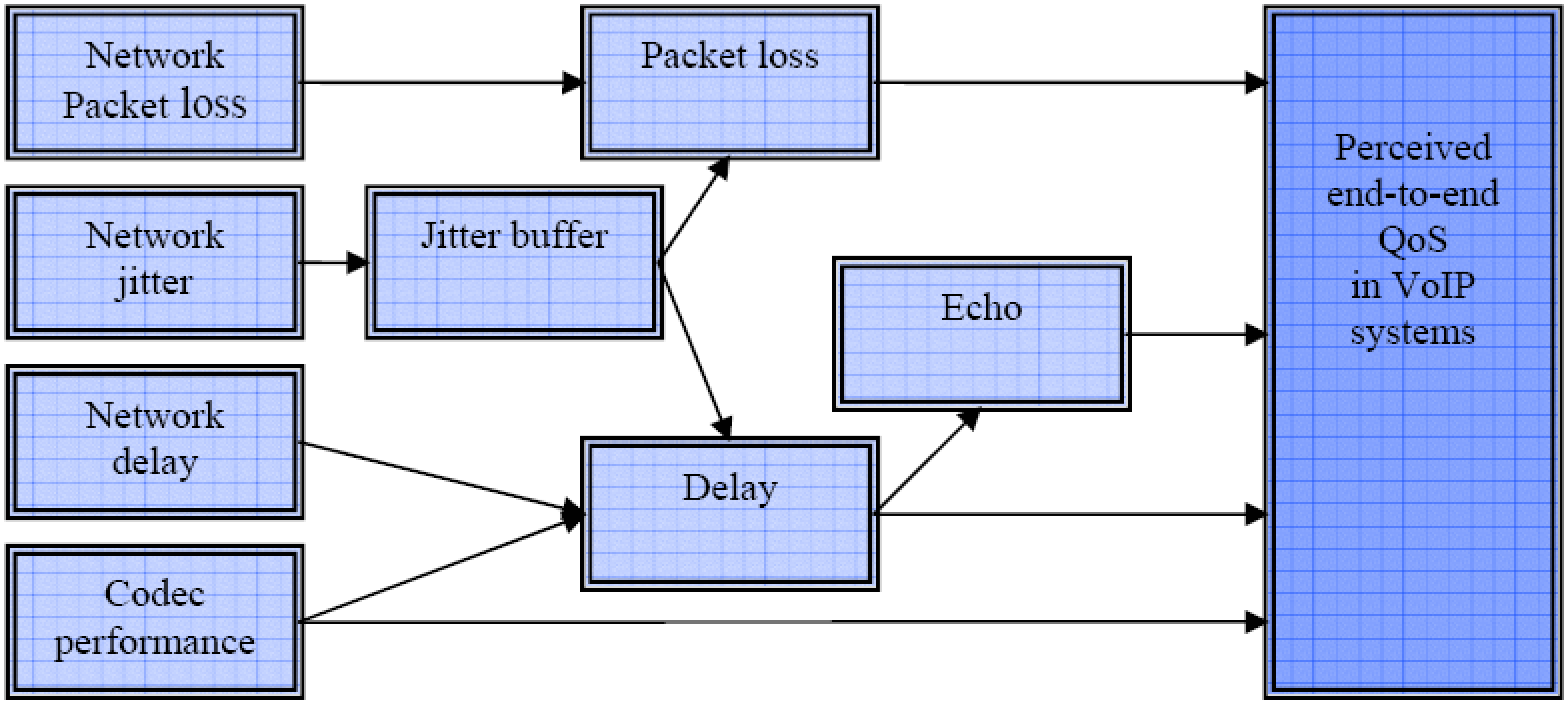
Limits of network QoS parameters

Delay – one way <= 150ms

Jitter <= 75ms

Packet loss <= 3%

Critical VoIP QoS parameters and their interrelationship



VoIP QoS Measurement Methods

VoIP QoS Measurement Methods

QoS MEASUREMENT METHODS

5.1 Introduction

5.2 Mean option score, MOS

5.3 Measurement methodologies

- 5.3.1 Subjective Methods
  - 5.3.1.1 Absolute Category Rating, ACR
  - 5.3.1.2 Degradation Category Rating, DCR
  - 5.3.1.3 Comparison Category Rating, CCR
  - 5.3.1.4 Conversational Tests
  - 5.3.1.5 Double Tests
  - 5.3.1.6 Talking and Listening Tests
  - 5.3.1.7 Listening Tests
  - 5.3.1.8 Interview and Survey Test
- 5.3.2 Objective Methods
  - 5.3.2.1 Intrusive Signal-Based Measuring
    - 5.3.2.1.1 Perceptual Speech Quality Measurement, PSQM
    - 5.3.2.1.2 Measuring Normalizing Blocks, MNB
    - 5.3.2.1.3 Perceptual Analysis Measurement System, PAMS
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    - 5.3.2.2.1 In-Service Non-Intrusive Measurement Device, INMD
    - 5.3.2.2.2 Call Clarity Index, CCI Model
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    - 5.3.2.2.4 Non–Intrusive Quality Assessment, NIQA
    - 5.3.2.2.5 Non-Intrusive Network Assessment, NINA
    - 5.3.2.2.6 Perceptual Single Ended Objective Measure, PSOM
  - 5.3.2.3 Objective Parameter-Based Measuring
    - 5.3.2.3.1 E-Model

VoIP QoS Measurement Methods

MOS Standard For Measuring Call Quality

- The leading subjective measurement of voice quality - Mean Opinion Score (MOS) – Recommendation ITU P.800 .

MOS	Quality Rating
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

VoIP QoS Measurement Methodologies

1. Subjective methods

- **Test phrases of speech chosen because of their sounds are recorded and listened to under controlled circumstances. The sounds are played to the test panel in specially designed rooms, with noise and other important environmental factors leveled in an appropriate way for the test.**
- **From a quality rating made by the listener, an average score can be produced.**
- **The score is normally presented as a mean opinion score, MOS ranging from 1 to 5, with 5 being the highest possible quality. The score can also be presented as Good or better, %GoB or poor and worse, %PoW, valued in a similar fashion.**
- **The most famous method is the MOS, ACR, DCR, CCR, Conversational tests,…etc**

VoIP QoS Measurement Methodologies - Subjective methods (ACR)

1.1 Absolute Category Rating, ACR:

- Listeners using an absolute category rating system, ACR are instructed to rank the absolute quality of test speech samples i.e. the perceived quality without comparison to a reference.

Quality of the Speech	Score
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

Listening quality scale for absolute category rating

VoIP QoS Measurement Methodologies - Subjective methods (DCR)

1.2 Degradation Category Rating, DCR :

- Degradation category rating is used when speech samples are of good quality and ACR is too inaccurate to detect small variances in quality.
- In DCR listeners are presented pairs (A-B) or repeated pairs (A-B-A-B) of speech samples, where A is a quality reference and B is a degraded version of A.
- Listeners are asked to rate the samples B according to a degradation category scale.
- The results are reported as degraded mean opinion scores, DMOS.

5	Degradation is inaudible
4	Degradation is audible but not annoying
3	Degradation is slightly annoying
2	Degradation is annoying
1	Degradation is very annoying

Opinion scale for degradation category rating

VoIP QoS Measurement Methodologies - Subjective methods (CCR)

1.3 Comparison Category Rating, CCR:

- Comparison category rating is similar to DCR except for the speech samples that come in a random order.
- The samples are rated according to an opinion scale.
- The CCR model can be used in systems where the processed sample is better than the reference.

3	Much better
2	Better
1	Slightly better
0	About the same
-1	Slightly worse
-2	Worse
-3	Much worse

Opinion scale for comparison category rating

VoIP QoS Measurement Methods - Subjective Methods

1.4 Conversational Tests

- Conversational-opinion tests are made in a laboratory environment involving experienced or untrained parties.
- The conversational test is the most realistic model of a real telephone interaction and is the only convincing way of considering the subjective effect of all joined parameters that have an impact on conversational quality.

1.5 Double Tests

- Double talk tests measures the transmission quality during double talk, which influences the naturalness of a conversation.
- The test is carried out with one person talking continuously while the other is interrupting.

1.6 Talking and Listening Tests

- Talking and listening tests consider all aspects that influence the transmission quality for the users while they are either talking or listening without having a conversational partner on the other end of the connection.
- It is a measurement specialized for talking related disturbances like echo and background noise transmission .

VoIP QoS Measurement Methods - Subjective Methods

1.7 Listening Tests

- Listening-opinion tests are mainly used for unidirectional speech transmission such as broadcast circuits.
- Listening-only tests use material that is recorded in advance for testing play-back.
- The technique is intended to estimate and compare individual performance factors of terminals, algorithm implementations or different measurement conditions in one test.

1.8 Interview and Survey Test

- Quality can also be measured by service observations i.e. by interviewing customers.
- However, this method takes a large effort and time, and the produced results usually end up being unreasonably variable and expensive

VoIP QoS Measurement Methods - Subjective Methods

- The drawbacks of using subjective quality scores are many. The results are depending on **<u>uncontrollable attributes</u>** such as experience, mood, attitude and culture.

- The methods are **<u>expensive</u>** due to the involvement of people and demand extensive elaboration with testing setups.

- ***They are inefficient and impractical for frequent testing such as needed for design, changes and routine monitoring.***

- Subjective methods do however involve real time user assessment and take the human perception into consideration, which is why it is a model very valuable for system evaluation.

VoIP QoS Measurement Methods - Objective Methods

2. Objective Methods

Objective methods for the evaluation of speech quality fall into three categories:

a) **Comparison Methods or Intrusive Signal Based Measuring:** Methods based on the comparison of transmitted speech signal and a known reference.
PSQM,(ITU), MNB , PAMS and PESQ.

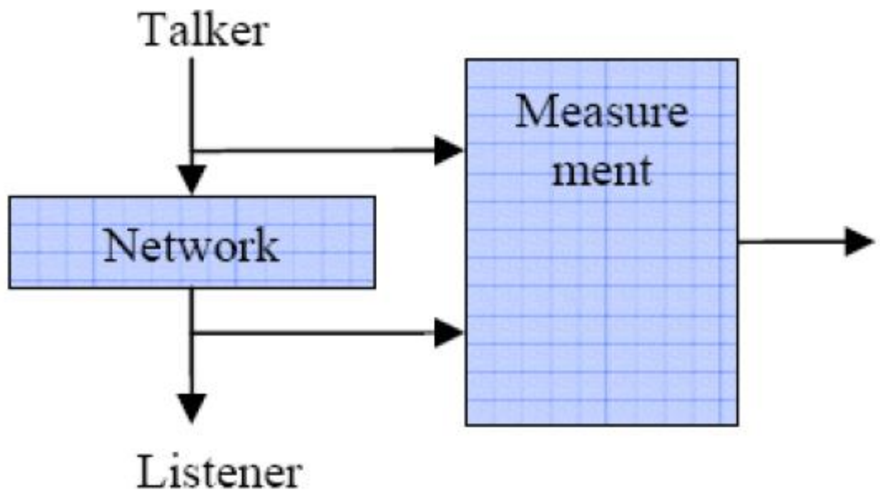
b) **Absolute Estimation Methods or Non Intrusive Signal Based Measuring:** Methods based on the absolute estimation of the speech quality (i.e. there is no known reference signal); e.g. INMD (ITU-T Recommendation P.561) , CCI Model , NIQA , NINA and PSOM.

c) **Transmission Rating Models or Parameter Based Measuring:** Methods that derive a value for the expected speech quality from knowledge about the network; e.g. ETSI Model (ETR 250, ITU-T Recommendation G.107) E-Model

VoIP QoS Measurement Methods - Objective Methods

1- Intrusive Signal-Based Measuring

- Intrusive signal-based measurements are, as the name suggests performed while interfering with the voice traffic transmitted during a connection.
- The methods set up test calls between two probes to compare the output signal with a reference signal.

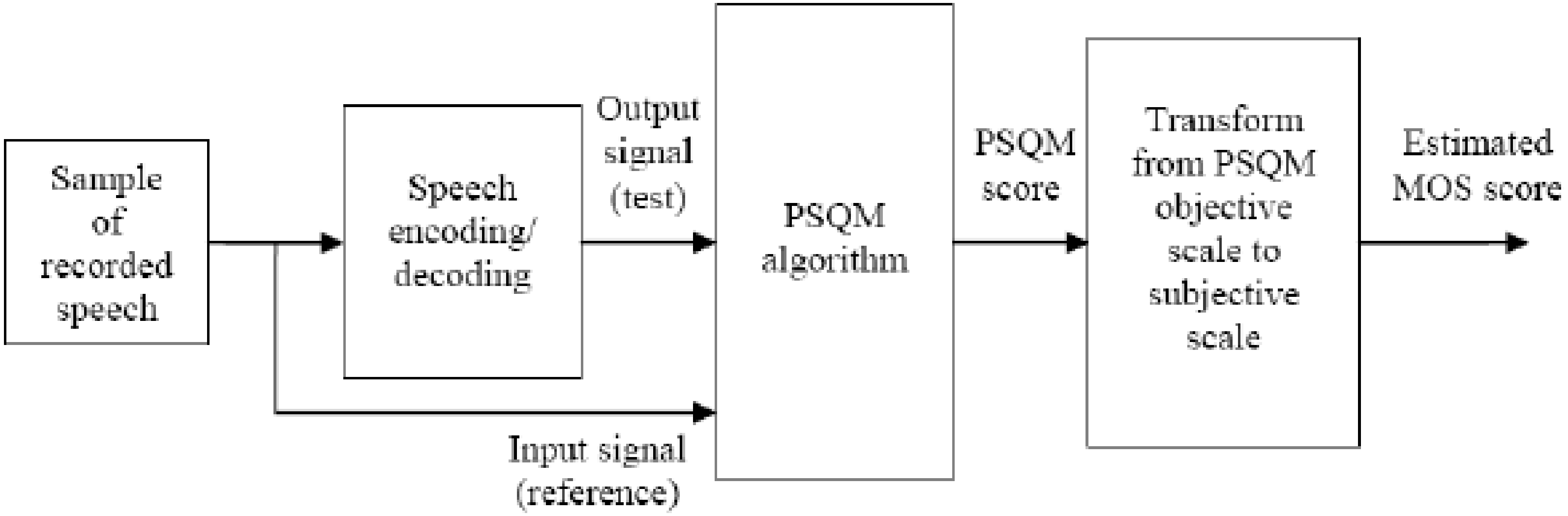


- The drawbacks with intrusive models are that they disturb regular voice traffic, use network resources and often need complicated test equipment to carry out measurements.
- The methods are however popular since they have shown good correlation with subjective quality scores.

VoIP QoS Measurement Methods - Objective Methods PSQM

1-1 Perceptual Speech Quality Measurement, PSQM

- The PSQM process consists of a mathematical algorithm that evaluates the difference between a signal distorted by a telephone system and a clean reference signal.
- The difference is used to calculate the noise disturbance, which is presumed to be directly related to the quality of speech
- The model is appropriate for narrowband telephone band signals (300-3400Hz) processed by low bit rate voice compression codecs or vocodecs.



The PSQM testing process

VoIP QoS Measurement Methods - Objective Methods PSQM

- The PSQM mathematical algorithm can be separated into the following three blocks:

1. Pre-processing where factors affecting human perception like loudness and frequency are taken into account.
2. Perceptual modeling: The perceptual modeling consists of a transformation of the values into a perceptual domain.
3. Cognitive modeling: The input and the output signals are compared using four operations: Loudness scaling, internal cognitive noise, asymmetry processing and silence interval processing
.

- **To perform a PSQM measurement, a sample of human speech, real or artificial, is processed with the codec used in the system. A score of 0 indicates a perfect match between the input and output voice signals. A higher value shows the amount of distortion in the connection. Normally a value of 15-20 indicates that the connection is extremely poor.**

- The PSQM model has shown satisfactory results but the values are depending on correct time synchronization prior to the PSQM process, which is considered a critical step.

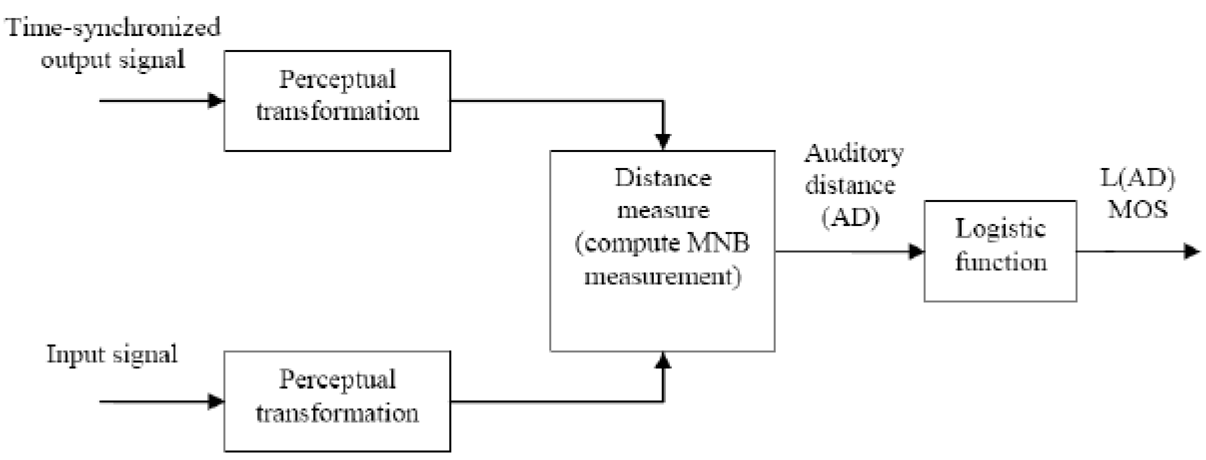
- **The PSQM method is not suitable for measurement of parameters like delay, time-clipping, packet loss or variable delay and the presence of packet delay variation can even damage the results of the process.**

- More advanced versions of the PSQM like PSQM+ and PSQM99 got popular in many systems like IP-telephony. However, today the method is no longer a valid official recommendation.

VoIP QoS Measurement Methods - Objective Methods MNB

1.2 Measuring Normalizing Blocks, MNB

- The method of measuring normalizing blocks, MNB was presented in 1997 as an alternative to the PSQM technique for measuring perceptual distance in a test call.
- The MNB model incorporates listener's capacity to adapt and react differently to results that span different time and frequency scales.
- The method can therefore either be used for calculation of the quality with respect to time or to frequency.



The MNB model

VoIP QoS Measurement Methods - Objective Methods MNB

- The first step in the model is the perceptual transformation where output and reference signals are time synchronized and level aligned before both of them are mapped to the frequency domain.

- All silent frames are removed and the power scale is transformed into a perceived loudness scale.

- Step two consists of a frequency measuring normalizing block, FMNB followed by a time measuring normalizing block, TMNB that create a set of integrated difference measurements and a normalized output signal.

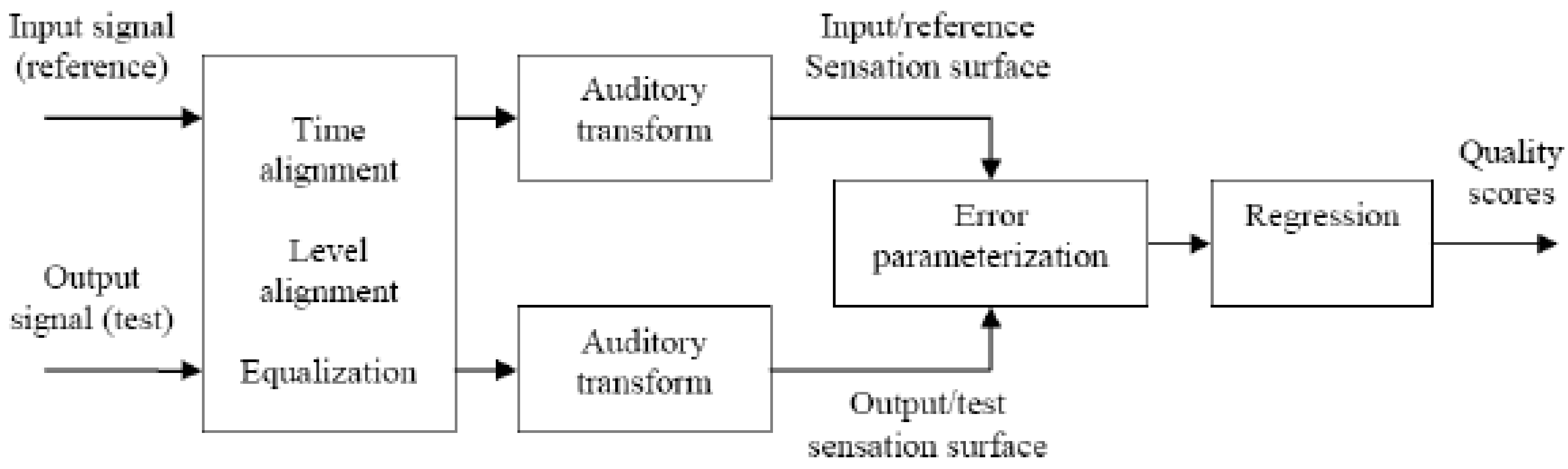
- The MNB algorithm then combines the two MNBs into a single nonnegative value that is called the auditory distance, AD.

- The AD value is representing the perceived quality difference between the two compared signals and is mapped onto a quality scale to obtain a subjective MOS score.

VoIP QoS Measurement Methods - Objective Methods PAMS

1.3 Perceptual Analysis Measurement System, PAMS

- PAMS is optimized for testing with artificial speech samples and measures one way speech quality over codecs as well as over a wide range of networks including fixed, mobile and IP-telephony systems.



The PAMS algorithm

VoIP QoS Measurement Methods - Objective Methods PAMS

The model can be divided into the following building blocks :

1. Pre-processing: The output and the input signals are time and level aligned in individual time segments to compensate for delay, small delay variation and overall system gain respectively. The signals are then equalized to remove any consequences from filtering.
2. Auditory transform: Both signals are modeled to include the human aspect of speech performance in the process.
3. Error parameterization: The difference between the sensation surfaces of the two signals is calculated to present an error surface that is carefully evaluated. Audible errors are then mapped onto subjective quality scores, presented as among others listening quality and Listening effort scores.

- PAMS became an important step in the development of objective measurement methods through the introduction of end to end measurements and perspective.

- The model can handle time varying conditions and enables fast and repeatable measurements. The reliability of the method depends critically on correct mapping from the error parameters to subjective quality scores.

VoIP QoS Measurement Methods - Objective Methods PESQ

1.4 Perceptual Evaluation of Speech Quality, PESQ

- During the end of the 1990s five different speech quality measurement methods were evaluated by the ITU. Two of the most successful ones, PSQM99 and PAMS were combined into a new further improved model, accepted as recommendation p.862 in 2001. The method was named the perceptual evaluation of speech quality, PESQ and rapidly became the most prominent standard in the area of objective, intrusive signal based measurement methods.

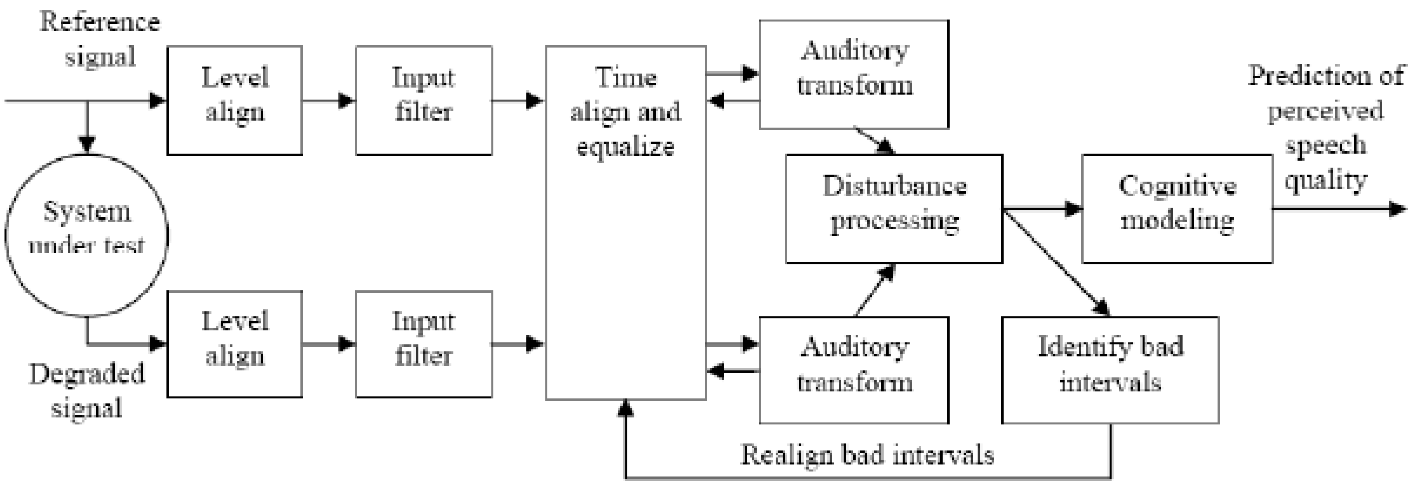
- Like PSQM and PAMS, PESQ is intended to be used for measuring one way quality on narrowband telephone signals. It joins the time alignment processes of PAMS with the perceptual modeling of PSQM99. On top of this PESQ adds new functions like transfer function equalization and a method for calculating average distortion over time. Both artificial and natural speech samples can be used for testing and the model is recommended for Codec evaluations, codec selection, live network testing and testing of imitated and prototype networks

VoIP QoS Measurement Methods - Objective Methods PESQ

- **PESQ has proven successful for measurements in the following areas :**
- Trans coding
- Transmission channel errors
- Speech input level at the codec
- Noise that is added by the system
- Packet loss

- **PESQ is unknown or not intended for measurements in the following areas:**
- Delay
- Listening levels and overall gain
- Multiple simultaneous talkers
- Bit-rate mismatching between encoder and decoder
- When background noise is present at the input signal
- Codecs < 4kb/s
- Listener echo and side tone

VoIP QoS Measurement Methods - Objective Methods PESQ



The PESQ methodology

The PESQ algorithm as shown in Figure can be described by the following stages:

1. Pre-processing: Level alignment is performed to keep the reference and the degraded output signal at the same constant power level. If essential, compensation is made for filtering effects.

Existing variable delay is eliminated by the system. For comparison reasons, the reference and the degraded signal are listed up by time alignment carried out in three steps: First, the delay is estimated, then the variable delay is detected in each separate speech sample and finally, bad delay intervals are realigned.

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VoIP QoS Measurement Methods - Objective Methods PESQ

2. Auditory transform: The auditory transform is where the human perception is included into the operation. Each signal is passed through a transform that imitates certain human hearing properties. The result is a representation of the perceived loudness of the signal, defined by time and frequency, called a sensation surface.

3. Disturbance processing: The audible difference between the reference and the degraded signal is presented as the divergence between the sensation surfaces, called the error surface. After evaluation of the surfaces, an absolute and an additive disturbance factor are calculated. These are defining the absolute audible error and the audible errors significantly louder compared to the reference signal.

4. As the last process, the two average values are linearly combined to calculate a correlated subjective quality score.

- The downside with the PESQ model is that it presents quality results that are difficult to comprehend. The model measures impacts caused by impairments like one way speech distortion and noise but leaves other impairments like echo and side tone out of consideration.

- It is therefore possible to get a high PESQ score even though the overall speech quality is bad. However, due to asymmetry processing and scaling, PESQ supplies improved correlation with subjective quality scores compared to earlier versions like PSQM and MNB.

VoIP QoS Measurement Methodologies - Objective Methods

2. Objective Non- Intrusive Signal-Based Measuring

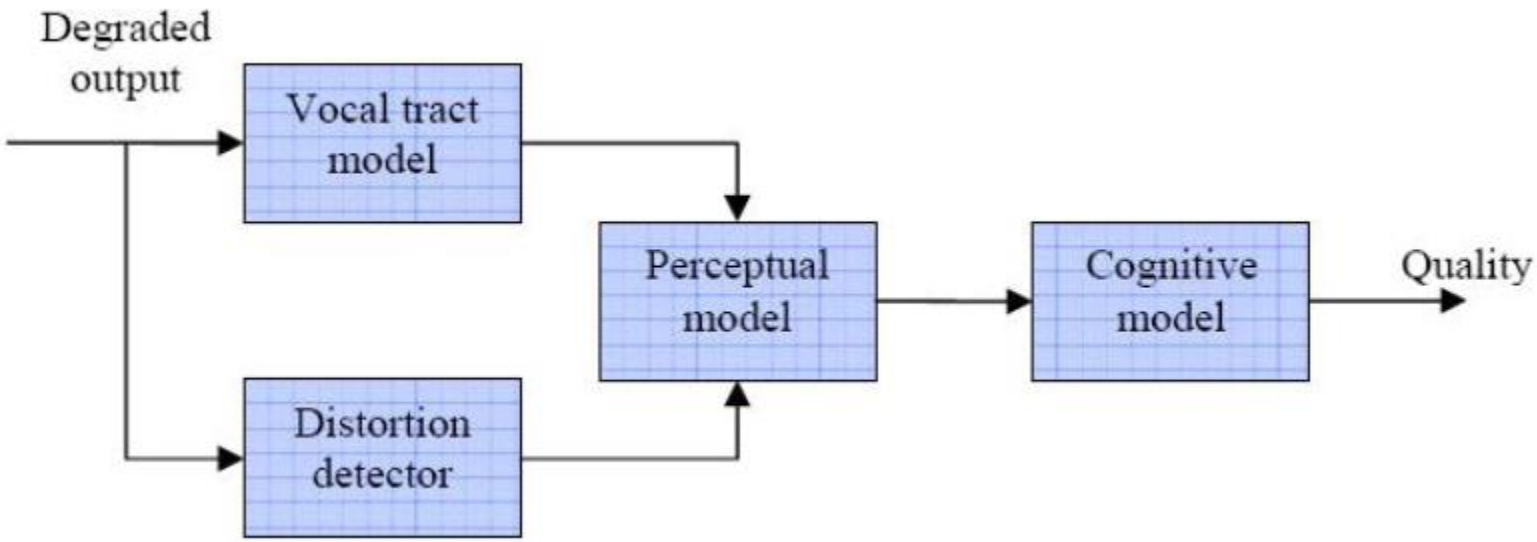
- Non intrusive or passive measurement methods were developed for performing measurements on real time traffic generated by actual users using only one measurement node. Compared to intrusive measurements methods, non-intrusive tests are simple and inexpensive. They are software based and do neither use network resources nor complicated test procedures.

- Unlike intrusive the non-intrusive methods perform signal analysis without a known reference signal and do not affect the actual traffic. Traditional non-intrusive measurements can be applied in circuit as well as packet domains and focus on impairments like packet loss, jitter and delay.

- These parameters are important since they indicate the performance of an IP network, relevant for transmitting voice. However, this information is not enough since it does not indicate the end user experience of speech quality.

- During the mid 90s, work to develop non-intrusive voice service measurements led to the recommendation P.561 **“in-service, non-intrusive measurement device, INMD”.** It permits the implementation of basic voice quality measurements in systems using live traffic and is considered being one of the best measurement systems today.

VoIP QoS Measurement Methods - Objective Methods INMD



Non-intrusive objective measurement method

- The in-service non-intrusive measurement device, INMD, was originally developed to measure parameters like speech level, noise level and echo loss in circuit switched networks. The method is today improved and intended for maintenance in both circuit and packet switched networks. It aims at locating and analyzing impairments that may affect the transmission performance of voice servers.

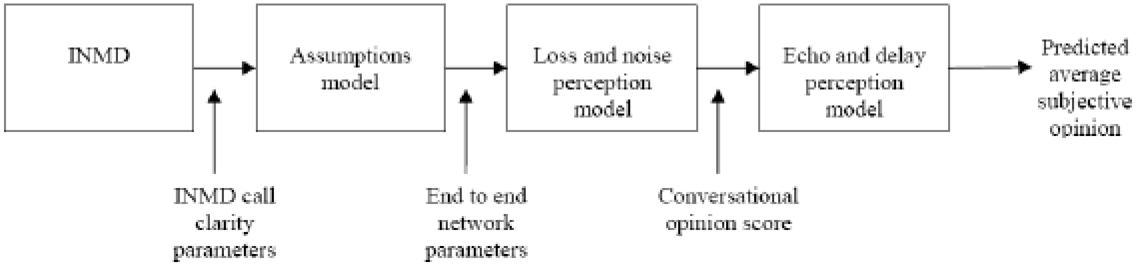
VoIP QoS Measurement Methods - Objective Methods CCI

2. 2 Call Clarity Index, CCI Model

- The call clarity index or the CCI model was developed by British Telecom for interpreting INMD measurement to predict voice quality on a call to call basis. It supervises call performance by searching for and ranking mismatches in delay, echo and noise in the network.

- The CCI method combines the measured data into a single call clarity index, CCI using a model of human perception that is calibrated against subjective quality scores. The index is expressed in a conversational quality scale and represents the mean opinion of the tested connection. The CCI algorithm can be loaded onto test equipment or deployed in network components like echo cancellers, voice quality enhancers and switching platforms.

VoIP QoS Measurement Methods - Objective Methods CCI



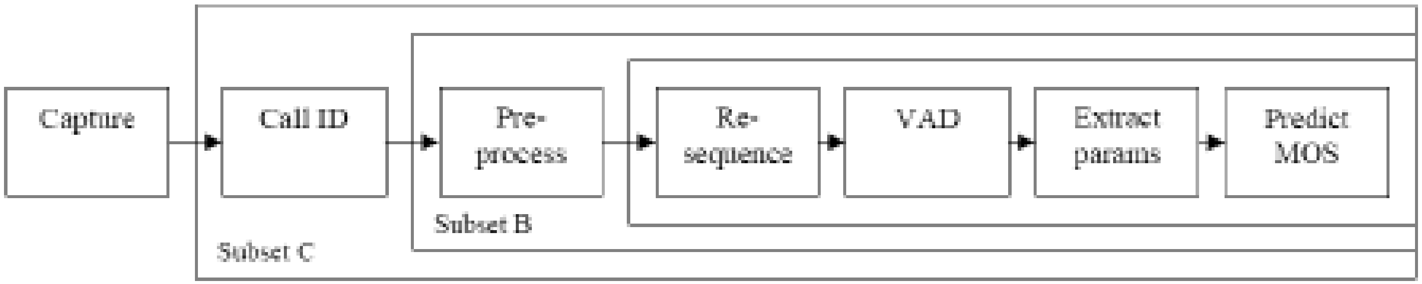
The call clarity model

- The model of CCI contains of three steps:

- The assumptions model: This first step makes estimates about parameters related to the network and users that can not be measured by the INMD. With these assumptions a complete model of the tested system can be made.
- The loss and noise model: This block takes human perception factors like frequency selectivity and noise on the connection into account. This stage also considers effects of side tone, noise masking and room noise. The output of this step is a single score representing conversational speech quality.
- Echo and delay: The next part of the model uses complex mathematical calculations to add the effects of echo and delay into a final output value.
- Multiple CCI values should always be used to assure an accurate statistical averaging. The mean value can then be expected to represent all degradations in a call correctly.

2.3 PsyVoIP

- When PsyTechnics started to develop PsyVoIP in 1999 the aim was to produce a model similar to the INMD methods but specifically created for VoIP systems. The PsyVoIP model is a non-intrusive method that can assess voice quality on a call by call basis, monitor calls continuously and measure quality as perceived by the users. The PsyVoIP model can be implemented anywhere along the VoIP connection from within a gateway.



- *The PsyVoIP model*

VoIP QoS Measurement Methods - Objective Methods PsyVoIP

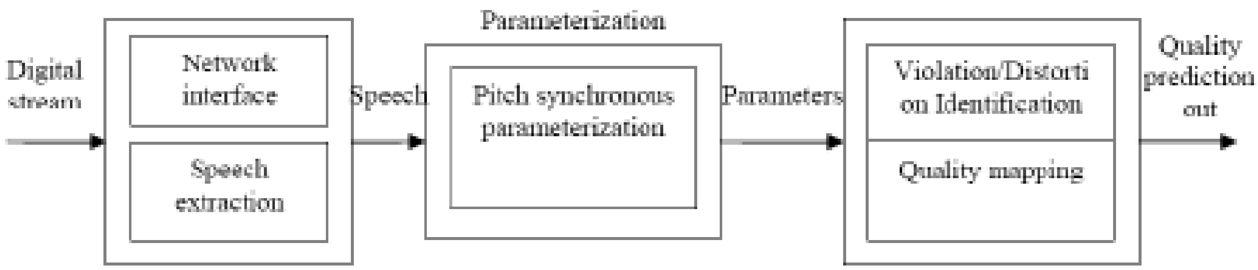
The model uses the following steps:
• Capture captures relevant packets needed for the analysis.
• Call-id Identifies packets to find if and what call they belong to.
• Pre-process Extracts necessary information from each packet and then discards it.
• Re-sequence Reorders out of sequence packets.
• VAD Distinguishes between silence and speech during a conversation.
• Extract Parameters Extracts the parameters needed for calculating quality Scores.
• Predict MOS When enough packets have been analyzed the model calculates a final quality score correlated to MOS.

- PsyVoIP is trying to consider every VoIP process that affects the experience when making a call and is a technique used together with the E-model to remove some of its guesswork. PsyVoIP is together with VQMon believed to be the candidates for future standards for voice quality evaluation in IP systems.

VoIP QoS Measurement Methods - Objective Methods NIQA

2.4 Non–Intrusive Quality Assessment, NIQA

- The non-intrusive quality assessment method, NIQA, was developed by PsyTechnics as an extension to the CCI method for the handling of a full range of distortions.
- The NIQA method can be implemented into gateways, switches, test equipment or voice quality enhancers practically anywhere in the network, in a single or in multiple locations.
- The first step is a speech production model that finds signal parts that do not originate in human voice. In the second step the signal is run through a speech perceptual and cognitive model that is similar to the intrusive model PESQ.
- The impact on the quality affected by distortions is predicted. The last parameterization stage monitors important information about the distortions present in the speech signal. After the quality calculation an output quality score is produced, correlated to the subjective MOS score .

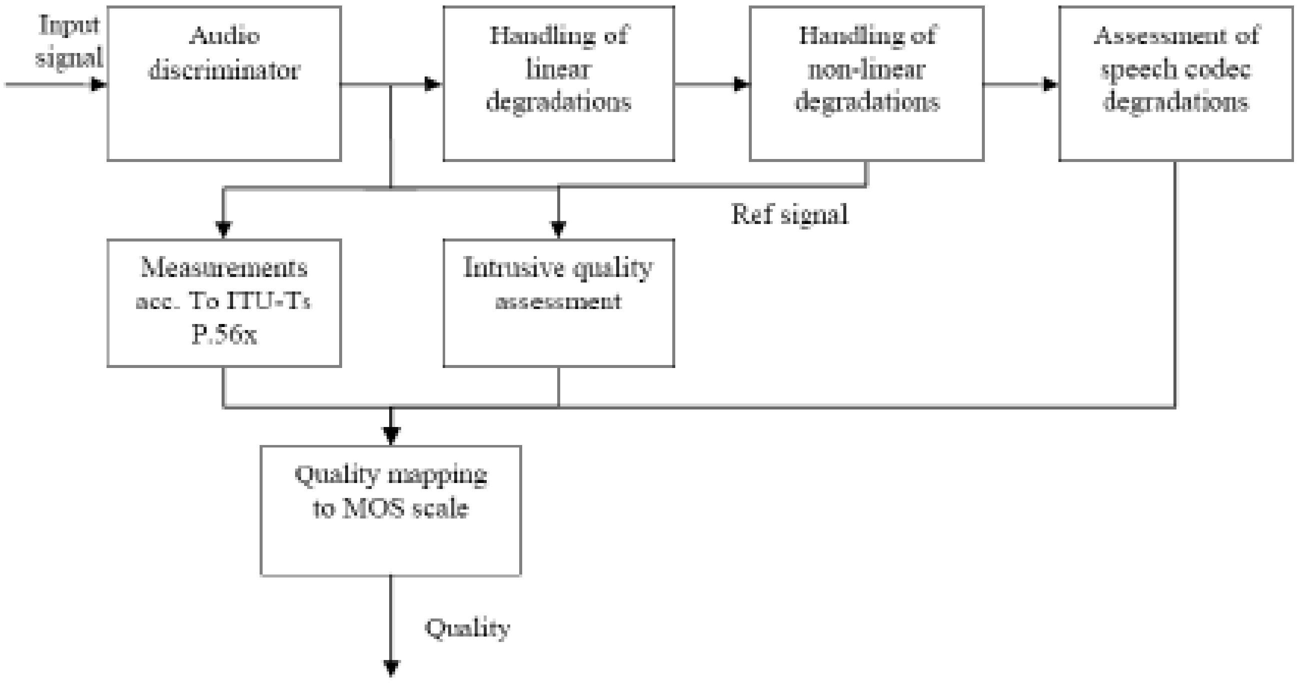


- *The NIQA model*

VoIP QoS Measurement Methods - Objective Methods NINA

2.5 Non-Intrusive Network Assessment, NINA

- The non-intrusive network assessment, NINA was developed by SwissQual and was included as an ITU recommendation in 2001.



The NINA model

VoIP QoS Measurement Methods - Objective Methods NINA

The NINA algorithm can be subdivided into the following steps:

1. Speech is separated from the input signal.
2. A reference signal is approximated by separating the degraded signal from the input signal.
3. Time clipping is detected and lost frames are recovered.

After finishing these steps the model contains of one degraded signal and one created reference signal.

4. The two signals are compared in an intrusive measurement model.
5. Impacts from parameters like background noise, echo and speech activity are calculated.
6. Impairments created by the speech codec are evaluated.
7. Finally the results are processed in a quality evaluation box to obtain a satisfactory speech quality degradation score, correlated with the subjective MOS.

In addition to quality scores, NINA generates results on time clipping, background noise, and echo and speech activity as is shown by the algorithm above

VoIP QoS Measurement Methods - Objective Methods PSOM

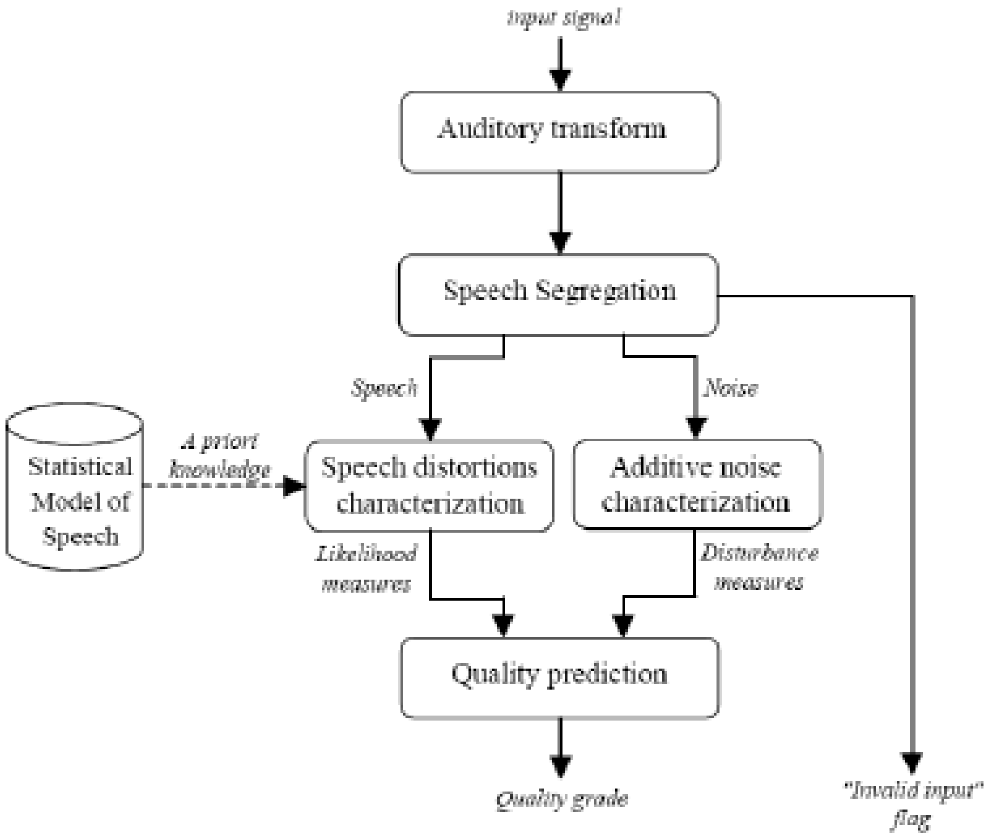
2.6 Perceptual Single ended objective measure, PSOM

- The perceptual single ended objective measure method, PSOM, was developed by France Telecom R&D. The model was presented as a contribution to the ITU-T SG12 in 2003 with the purpose of finding an estimation of perceptive speech quality on live traffic.

- In the PSOM method speech distortions are separately measured and analyzed through comparison with a statistical model for clean speech. The output is a likelihood of each parameter that also can be characterized as a numerical distance between a reference and a degraded output signal. The model does not provide any further analysis of degradations than the output quality score.

The PSOM algorithm is subdivided into the following steps:
1. The input signal is transformed to create a time-frequency representation.
2. The speech is separated from noise signals and other non-speech components.
3. Speech perception values are estimated with a statistical model as a reference.
4. Disturbance measures are calculated from the noise signal.
5. A quality prediction stage combines the likelihoods into a speech quality score.

VoIP QoS Measurement Methods - Objective Methods PSOM

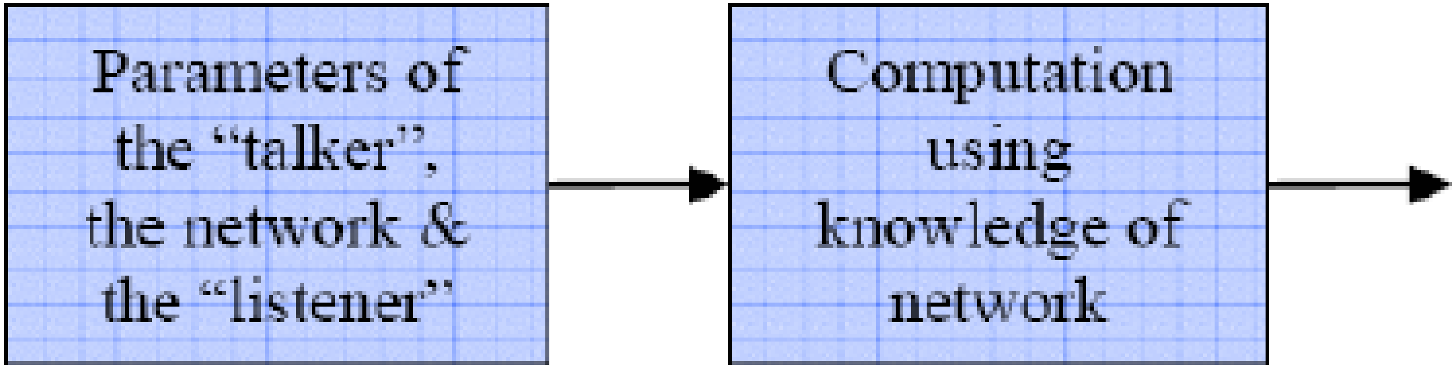


The PSOM model

VoIP QoS Measurement Methods - Objective Methods

3- Objective Parameter-Based Measuring

- Objective parameter-based measurement methods are, as the name suggests based on values derived from transmission parameters rather than the transmitted signals. The E-model and its extensions are dominant among the parameter-based models.

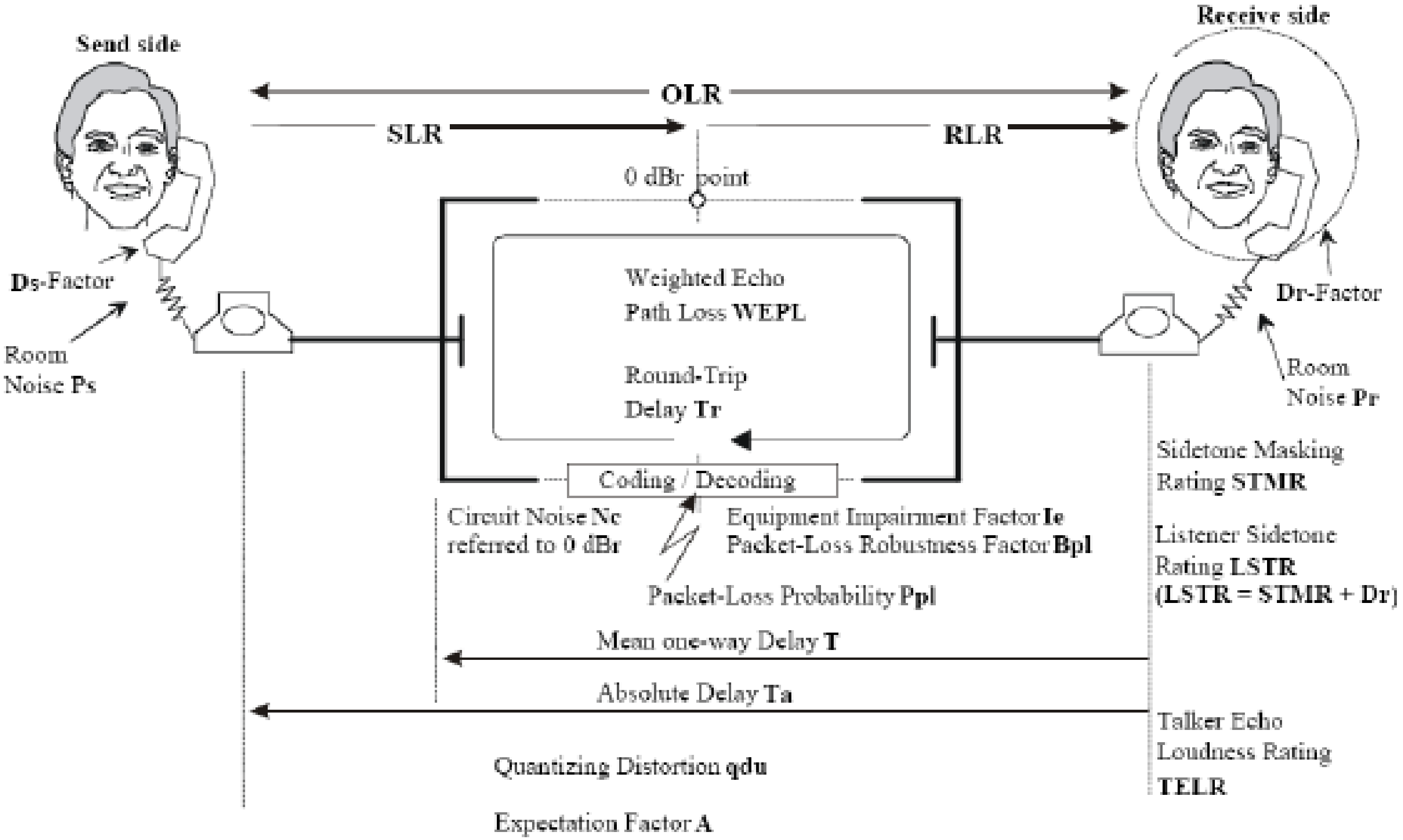


Parameter-based objective measurement method

E-Model

- The E-model is the idea of Nils-Olaf Johansson who was a Swedish expert working for ITU-T and later for ETSI.
- The E-model has been chosen by the ITU to become an international recommendation but differs from other objective measurement methods in that it is a planning tool.
- The input to the E-model consists of parameters that are available at the time of planning before and after the installation of the network. Some of these parameters are measured, others assumed or attained from standard documents.
- The primary output of the model is the transmission rating factor R, which is used to estimate the perceived quality of the system.
- The transmission factor R equation consists of the base quality value Ro, the impairment factors Is, Id and Ie and of an expectation factor A, also called the advantage factor.

E-Model



The E-model

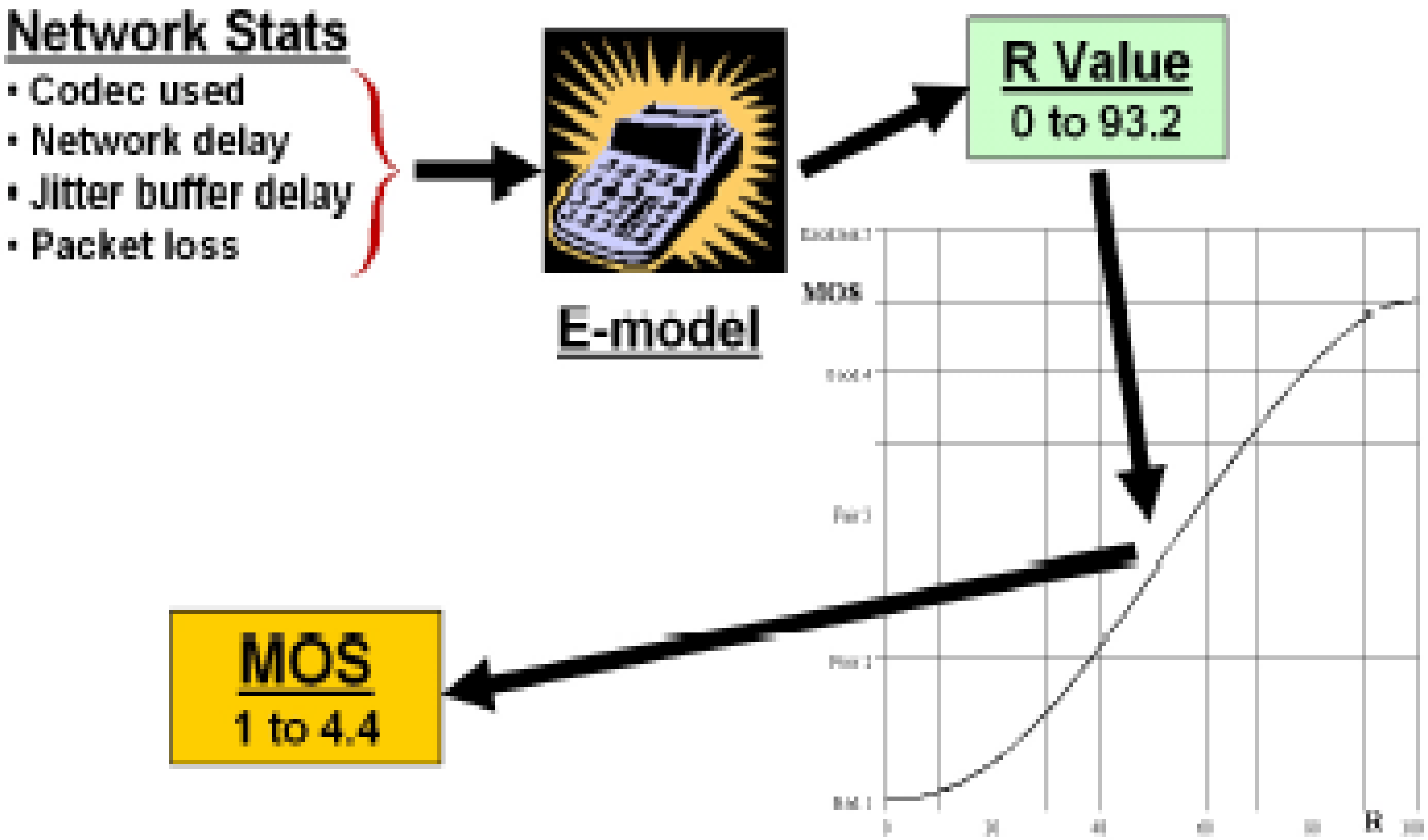
$$R = Ro - Is - Id - Ie + A$$

E-Model Parameters

$$R = Ro - Is - Id - Ie + A$$

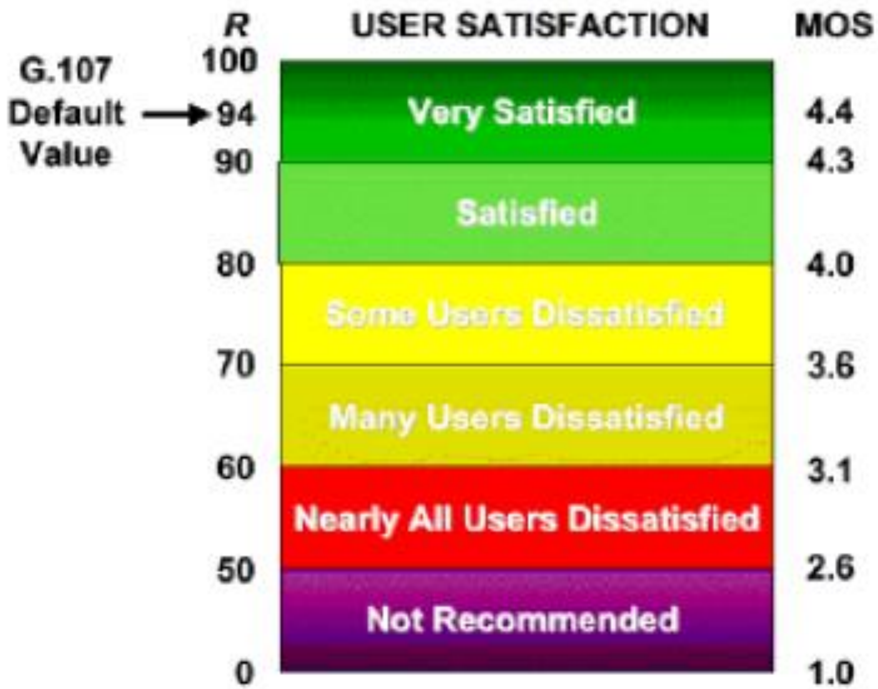
- **Ro** represents the signal to noise ratio. It should comprise the noise in the network and in the environment at the speaker and listener side.
- **Is** is the parameter for simultaneous impairments like **excessive loudness ratings**, **uncomfortable side tone levels** and **simple quantization distortions**.
- **Id** contains impairments due to echoes and delays e.g. talker and listener echoes and excessive delays.
- **Ie** represent the impairments from transmission equipment. Impacts from these impairments are normally difficult to measure, this is why values for **codecs ,packet loss** and other frequently used equipment have been measured and printed in recommendations by the ITU.
- **A** is a factor that adjusts the quality value in certain situations by considering non-technical aspects in the system or in the outside environment.
- The transmission factor **R** is shown at a scale from 0 to 100 but is typically ranging between 50 and 90. In the final step of the E-model the R value is transformed through a non-linear mapping into an equivalent MOS value.

R-Factor Values From the E-model and Corresponding MOS values



Mapping R factor Into MOS Scale

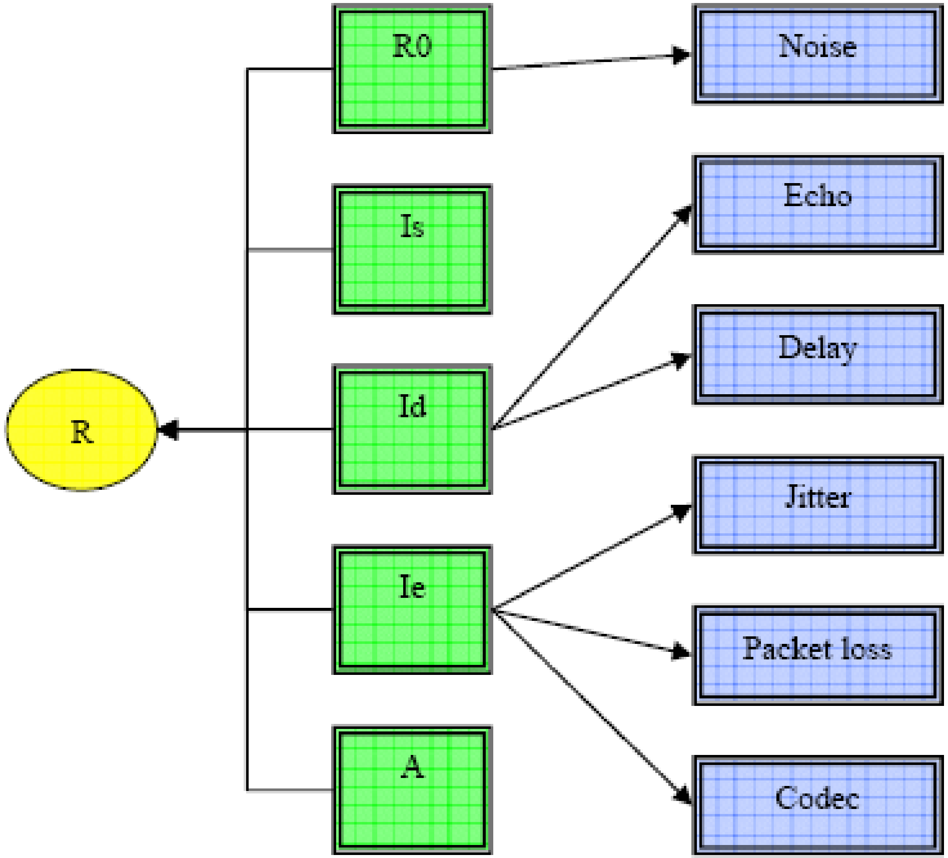
- We can map ***R*** into MOS scale by the following equations :
- For R < 0:
- MOS = 1
- For 0 <R <100:
- MOS = 1 + 0.035 R + 7.10-6 R(R-60)(100-R)
- For R > 100:
- MOS = 4.5



E-Model Related Quality and Satisfaction Categories

Range of E-Model Rating R	Speech transmission quality category	User satisfaction
$90 \le R < 100$	Best	Very satisfied
$80 \le R < 90$	High	Satisfied
$70 \le R < 80$	Medium	Some users dissatisfied
$60 \le R < 70$	Low	Many users dissatisfied
$50 \le R < 60$	Poor	Nearly all users dissatisfied
NOTE – Connections with E-Model Ratings R below 50 are not recommended.		

E-model Fault Diagnosis



E-Model Assumption

Due to the complexity of the E-Model, the approach used here is to try to identify which E-Model parameters are fixed and which parameters are not. In the context of this research the only parameters of the E-Model that are not fixed are:

- T, and Ta – Delay variables
- Ie – Equipment Impairment
- Where *T* is the mean one way delay of the echo path, Ta is the absolute delay in echo free conditions. In addition, parameters that affect delay Id and Ie are introduced:
- PL - Packet Loss %
- Background Link Utilization
- Coder Type

R = 93.2 – Id (Ta) – Ie (codec, packet loss)

Delay Impairment Factor, Id

- Id the impairment factor representing all impairments due to delay of voice signals is further subdivided into the three factors Idte, Idle and Idd :

$$Id = Idte + Idle + Idd$$

- The factor Idte gives an estimate for the impairments due to Talker Echo
- The factor Idle represents impairments due to Listener Echo
- The factor Idd represents the impairment caused by too-long absolute delay Ta, which occurs even with perfect echo canceling.
- For Ta ≤ 100 ms: $$Idd = 0$$
- For Ta > 100 ms:

$$Idd = 25\left\{\left(1 + X^{6}\right)^{\frac{1}{6}} - 3\left(1 + \left[\frac{X}{3}\right]^{6}\right)^{\frac{1}{6}} + 2\right\}$$

- With:

$$X = \frac{\log\left(\frac{Ta}{100}\right)}{\log 2}$$

Equipment Impairment Factor, Ie - 1

- The loss impairment **Ie** captures the distortion of the original voice signal due to **low-rate codec, and packet losses** in the network.
- The packet-loss dependent Effective Equipment Impairment Factor Ie-eff is derived using the codec specific value for the Equipment Impairment Factor at zero packet-loss Ie and the Packet-loss Robustness Factor Bpl,. With the Packet-loss Probability Ppl, Ie-eff is calculated using the formula

$$Ie\_eff = Ie + (95 - Ie)\frac{Ppl}{Ppl + Bpl}$$

- Ie is the equipment impairment factor.
- Bpl is called the packet-loss robustness factor, which depends on the Used codec.
- Ie,eff represents the packet loss dependent effective equipment Impairment factor, derived from the value of Ie depending on codec and at zero packet loss.
- Ppl is the packet loss probability

Equipment Impairment Factor, Ie - 2

Provisional planning values for the equipment impairment factor Ie in case of Ppl = 0 (no packet-loss)

Codec type	Reference	Operating rate kbit/s	Ie value
PCM	G.711	64	0
CS-ACELP	G.729-A	8	11
ACELP	G.723.1	5.3	19
MP-MLQ	G.723.1	6.3	15

Equipment Impairment Factor, Ie - 3

Provisional planning values for the equipment impairment factor Ie and for packet-loss robustness factor Bpl

Codec	Packet size	Ie	Bpl
G.723.1	30 ms	15	16.1
G.729A	20 ms (2 frames)	11	19.0
G.711	10 ms	0	25.1

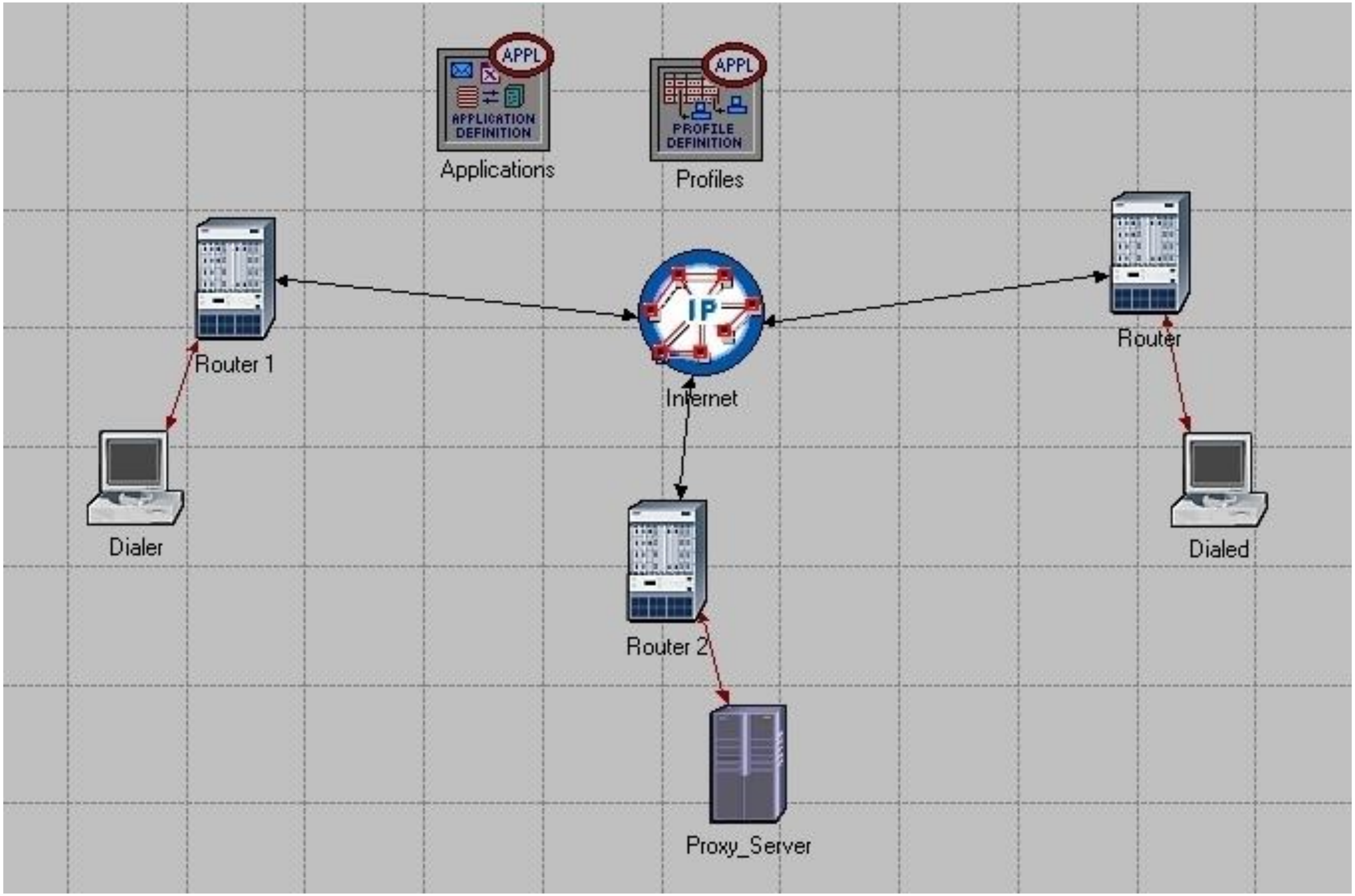
Advantage factor, A

Provisional examples for the advantage factor A

Communication system example	Maximum value of A
Conventional (wire bound)	0
Mobility by cellular networks in a building	5
Mobility in a geographical area or moving in a vehicle	10
Access to hard-to-reach locations, e.g., via multi-hop satellite connections	20

VoIP Assignment using OPNET

SIP Network Topology



SIP Network Design

- The simulation is done using OPNET simulation tool IT Guru Academic Edition 9.1 for a VoIP network using SIP Protocol.
- The network consists of IP-Telephones connected to the Internet by routers which act as IP gateway; the network is managed by the SIP proxy server which uses the SIP protocol to establish the voice calls on the IP network.
- The links between the routers and the Internet are T1 with link speed 1.544 Mbps and the links between the dialer, dialed, Proxy Server and the routers are 1000 Base-x
- The E-Model, is a model that allows users to relate network impairments to voice quality. This model allows impairments to be introduced and voice quality to be assessed. Three cases are considered to demonstrate the effectiveness of optimizing the E-Model.
- The idea is to configure the network with a certain parameters and run the simulation then getting from the tool the result values which used in E-Model equations to measure the Quality of service Factor R.

E-Model Optimization

The objective function for all cases is to maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (R>70). The cases considered are:

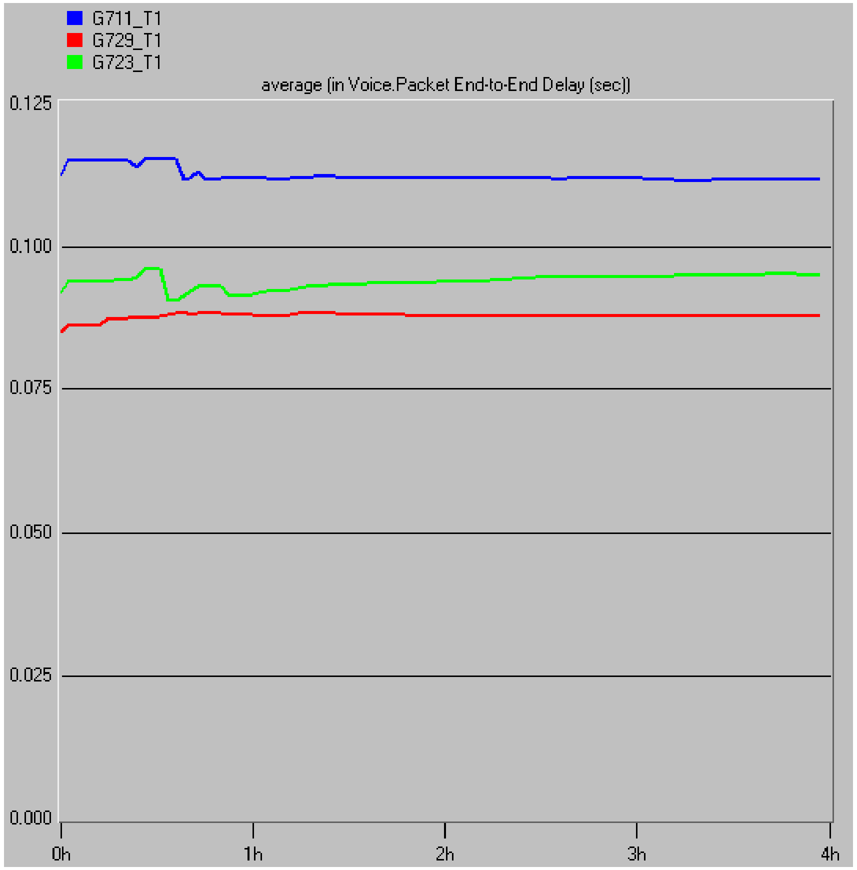
- Find the optimal voice coder given link bandwidth, packet loss level, and background link utilization level.
- Find the optimal voice coder and the optimal packet loss level given link bandwidth and background link utilization.
- Find the optimal voice coder and the optimal background link utilization level given link bandwidth and packet loss level.

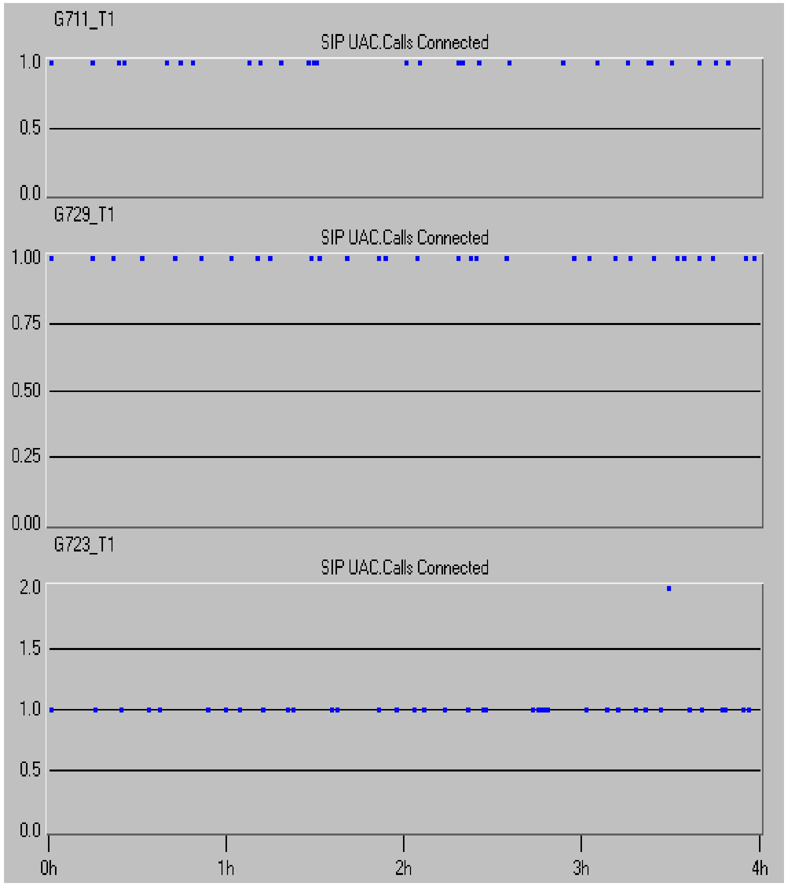
Case 1 - Optimizing for Coder Selection - 1

- Find the optimal voice coder given link bandwidth, packet loss level, and background link utilization level.

Standard	Type	Codec Bit Rate (kbps)	Voice Frame Length (ms)	Look ahead (ms)	Frame length (ms) Packet Length	Number of Voice Frames per Packet	I_E No PL
G.711	PCM	64	20	0	20	1	0
G.729	CS-ACELP	8	10	5	20	2	11
G.723.1	MP-MLQ	6.3	30	7	30	1	15

Case 1 - Optimizing for Coder Selection - 2

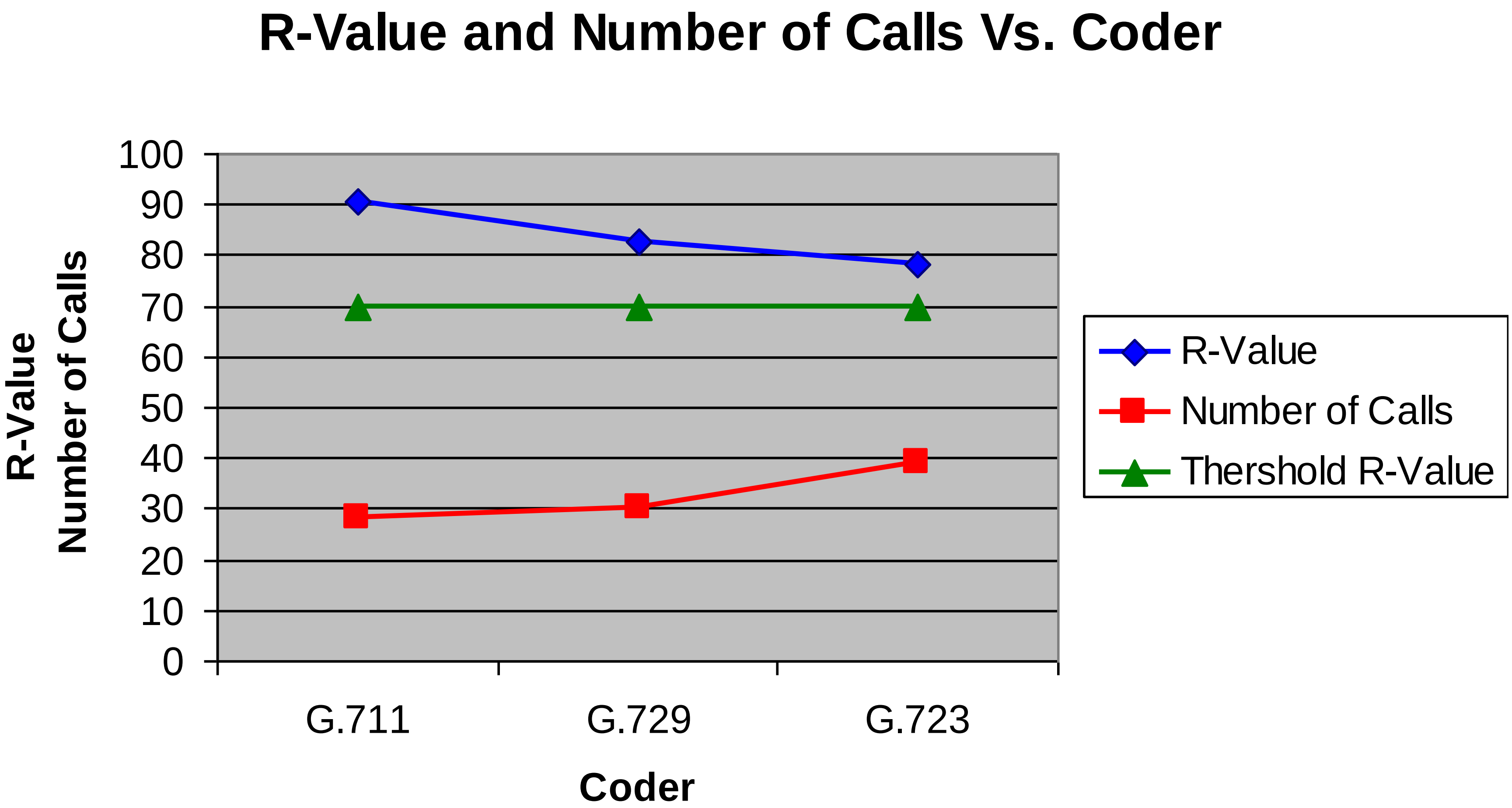




Case 1 - Optimizing for Coder Selection - 3

PL Ratio	CODEC	Delay (T_A) (ms)	I_D	I_E	R	MOS	Calls
0 %	G.711	115	2.76	0	90.44	4.33	28
	G729a	89	0	11	82.776	4.02	30
	G723.1	97	0	15	78.2	3.86	39

Case 1 - Optimizing for Coder Selection - 4



Case 2 - Optimizing for Coder and Packet Loss Level Selection - 1

- Find the optimal voice coder and the optimal packet loss level given link bandwidth and background link utilization.

Codec	Rate (Kbps)	Packet Size (ms)	Ie With no PL	Bpl
G 711	64	10	0	25.1
G.729A+VAD	8	20	11	19.0
G.723.1+VAD	6.3	30	15	16.1

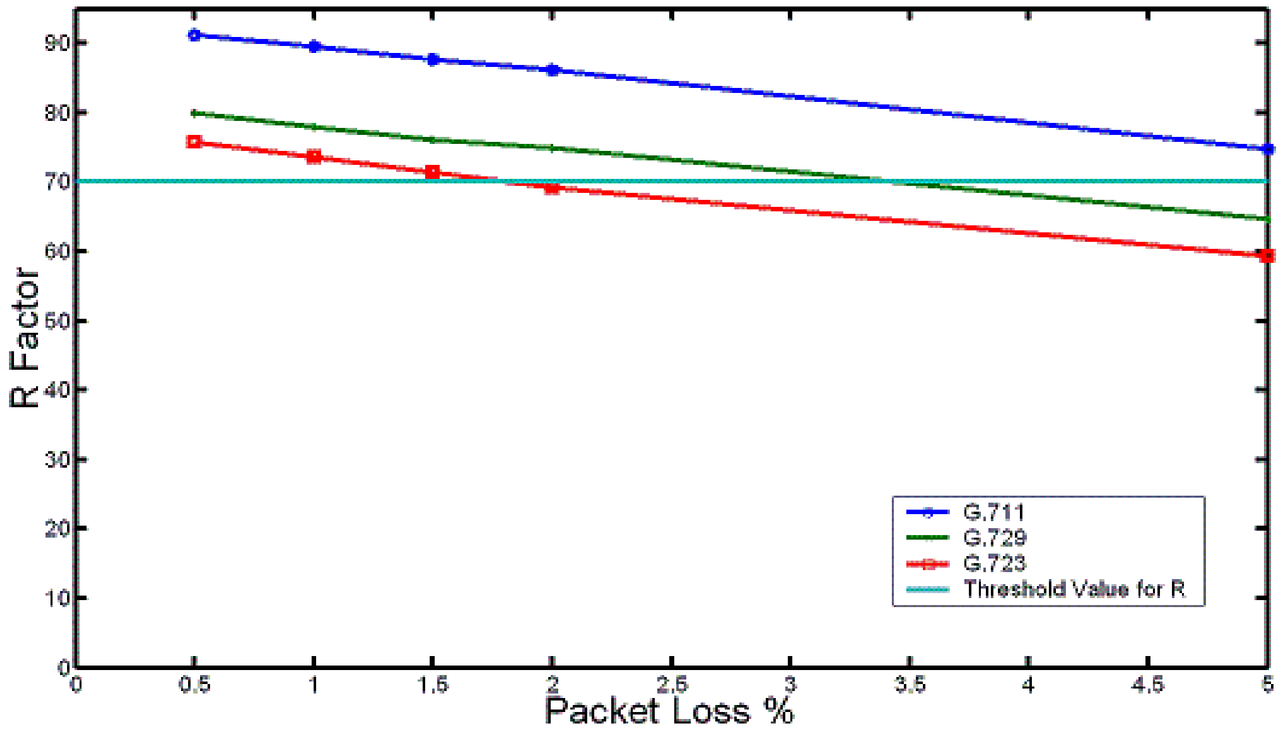
Case 2 - Optimizing for Coder and Packet Loss Level Selection - 2

PL Ratio	CODEC	Delay (Ta) (ms)	I_D	I_E	R	MOS	Calls
0.50%	G.711	142	0.054	1.855	91.29	4.36	27
	G729a	86	0	13.15	80.05	4.02	28
	G723.1	91	0	17.4	75.8	3.7	29
1.00%	G.711	141	0.053	3.64	89.51	4.3	27
	G729a	86	0	15.2	78	3.8	28
	G723.1	91.4	0	19.678	73.522	3.6	35
1.5%	G.711	141	0.053	5.36	87.78	4.2	25
	G729a	87	0	17.15	76.05	3.8	29
	G723.1	93	0	21.82	71.38	3.6	33
2%	G.711	140	0.053	7.011	86.136	4.2	27
	G729a	87	0	19	74.99	3.7	29
	G723.1	92	0	23.839	69.153	3.5	34
5%	G.711	137	0.052	15.78	74.732	3.7	28
	G729a	86	0	28.5	64.7	3.3	26
	G723.1	91	0	33.95	59.25	3.04	27

Case 2 - Optimizing for Coder and Packet Loss Level Selection - 3

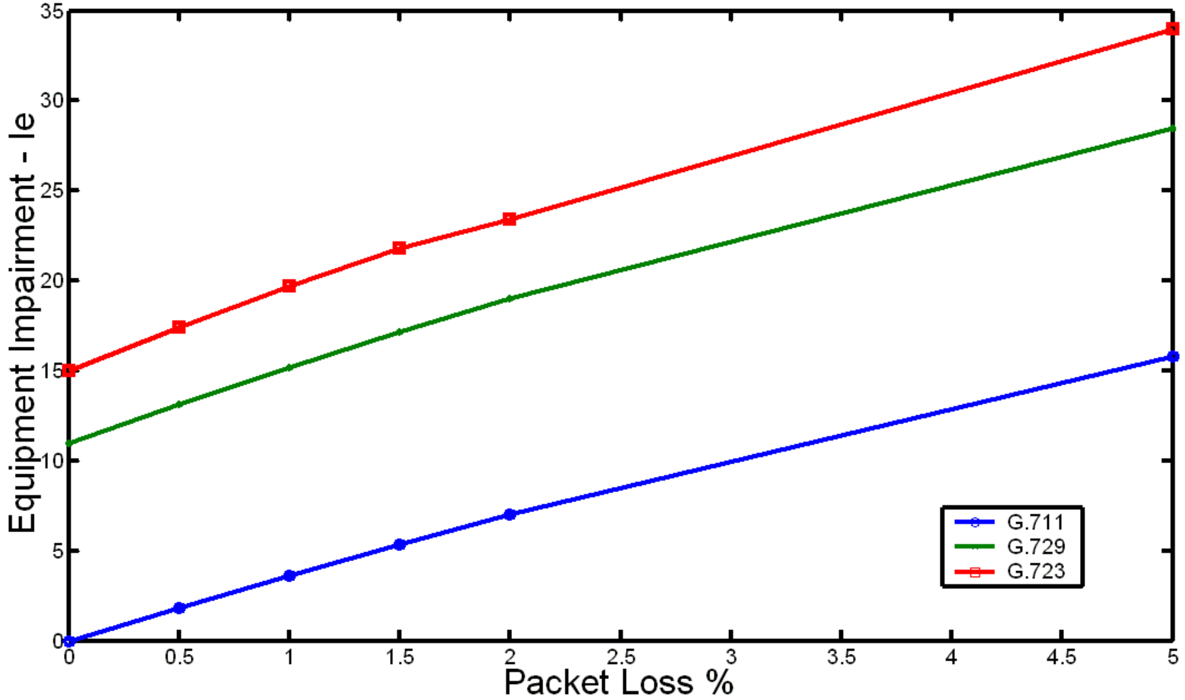
- For the 3 coders G.711.G.729 and G.723 with different values of packet loss ratio (0.5 %, 1 %, 1.5 %, 2 % and 5 %) knowing that the maximum allowable ratio is 3 % but the simulation was run for PL% equal 5% to observe the network behavior in case of crisis.
- The test was run with a link speed of 1.544 Mbps. The maximum number of calls was 29 calls. G.723.1 with packet loss of 0.5% was the combination chosen and the same combination was chosen till packet loss of 1.5 %.
- When packet loss ratio reached 2 %, G.723.1 became not feasible as its R value is less than 70 and G.729 with packet loss 2% was the combination chosen.
- For packet loss more than 3.5 % G.723.1 and G.729 became not feasible and the only feasible coder is G.711.G.711 with packet loss more than 3.5 % was the combination chosen.

Case 2 - Optimizing for Coder and Packet Loss Level Selection - 4



R Value and PL % vs. Coder – case (2)

Case 2 - Optimizing for Coder and Packet Loss Level Selection - 5

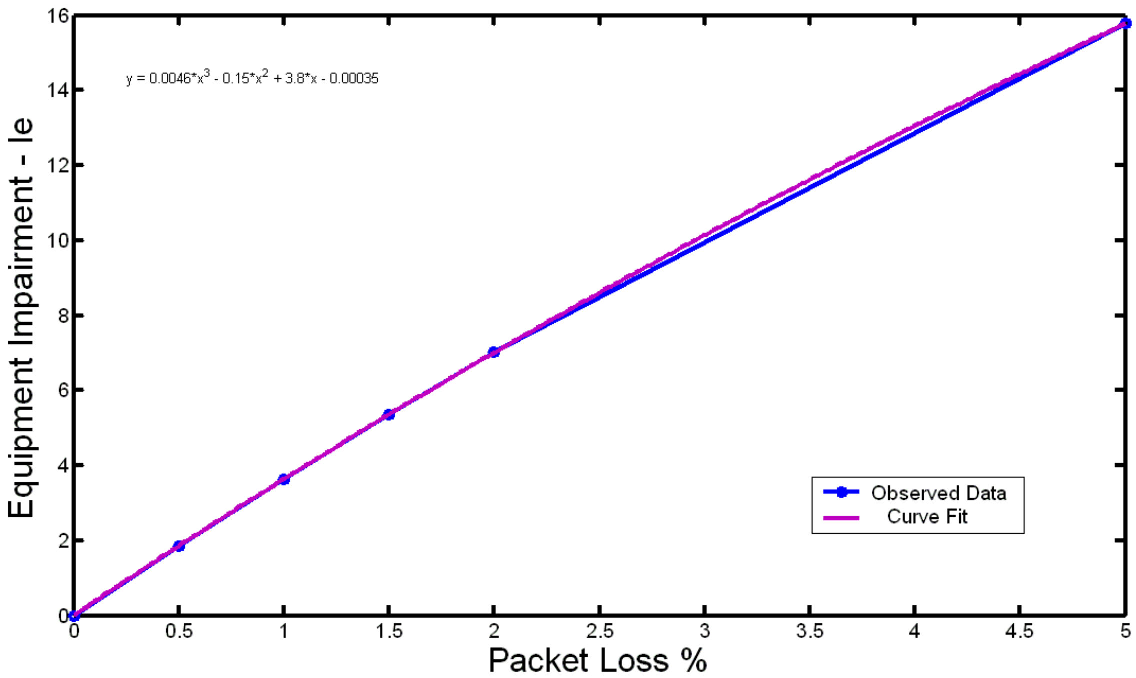


PL % and Ie vs. Coder – case (2)

Case 2 - Optimizing for Coder and Packet Loss Level Selection - 6

For G.711, the following polynomial was generated, where x represents the level of packet loss and y represents the level of impairment (Ie).

$$y = 0.0046\,x^3 - 0.156x^2 + 3.8x - 0.00035$$

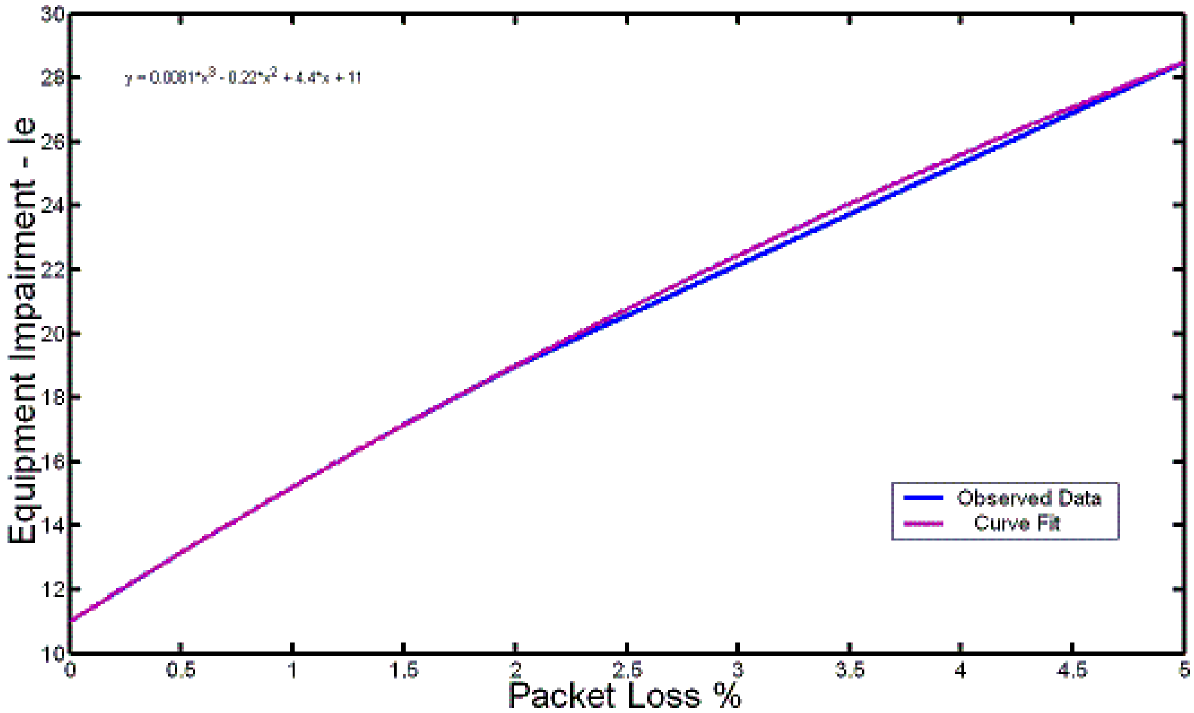


G.711 Polynomial Fit

Case 2 - Optimizing for Coder and Packet Loss Level Selection - 7

For G.729A, the following polynomial was generated:

$$y = 0.0081x^3 - 0.22x^2 + 4.4x + 11$$

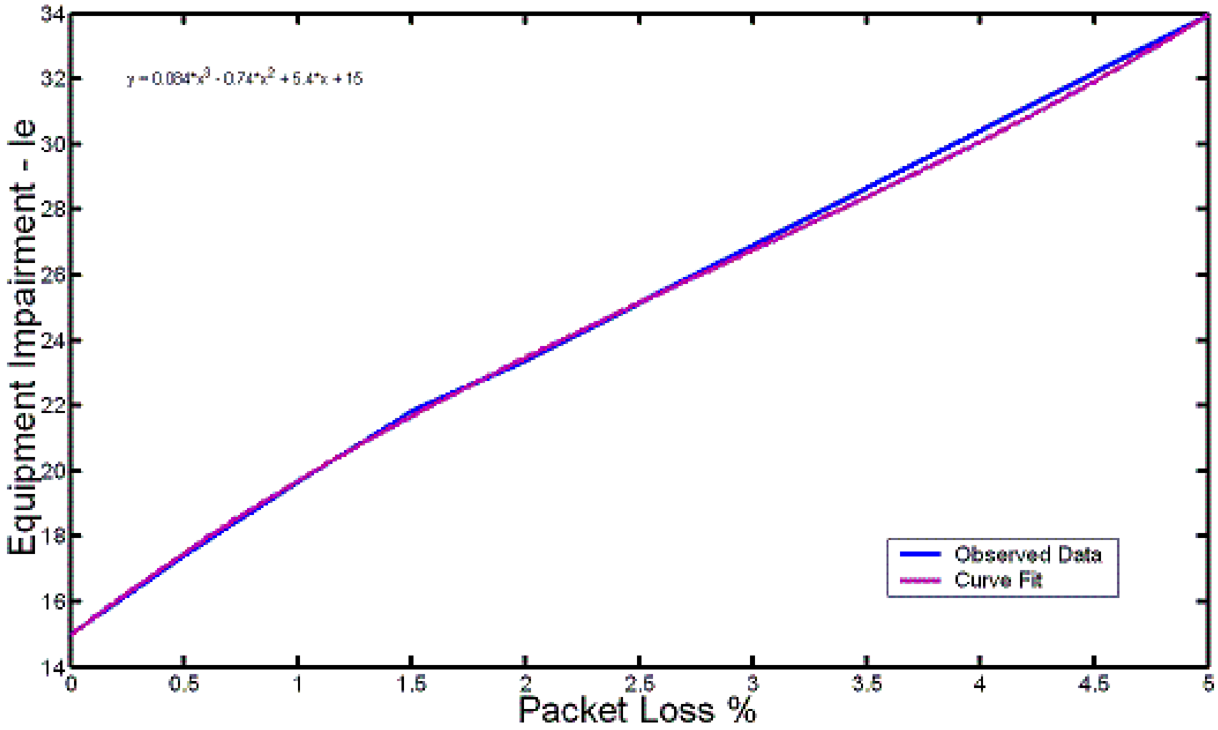


G.729 Polynomial Fit

Case 2 - Optimizing for Coder and Packet Loss Level Selection - 8

For G.723.1, the following polynomial was generated:

$$y = 0.084x^3 - 0.74x^2 + 5.2348x + 15$$



G.723.1 Polynomial Fit

Case 2 - Optimizing for Coder and Packet Loss Level Selection - Results

Case #	Packet loss%	Optimum Coder
1	0.5%	G.723.1
2	1%	G.723.1
3	1.5%	G.723.1
4	2%	G.729
5	> 3.5%	G.711

Case 3 – Optimizing for Coder and Background Link Utilization - 1

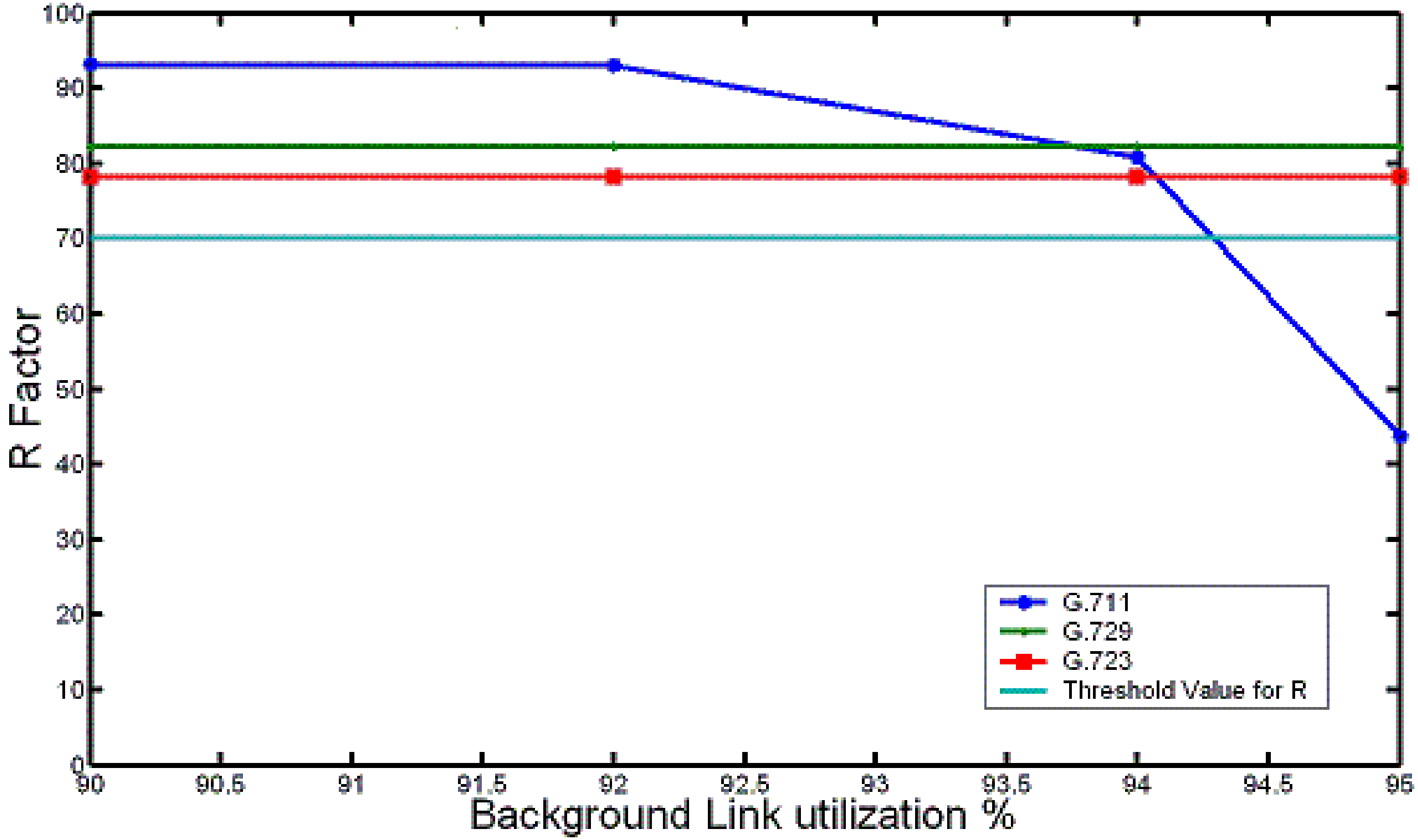
- Find the optimal voice coder and the optimal background link utilization level given link bandwidth and packet loss level.

Standard	Type	Codec Bit Rate (kbps)	Voice Frame Lenght (ms)	Look ahead (ms)	Frame length (ms) Packet Length	Number of Voice Frames per Packet	I_E No PL
G.711	PCM	64	20	0	20	1	0
G.729	CS-ACELP	8	10	5	20	2	11
G.723.1	MP-MLQ	6.3	30	7	30	1	15

Case 3 – Optimizing for Coder and Background Link Utilization - 2

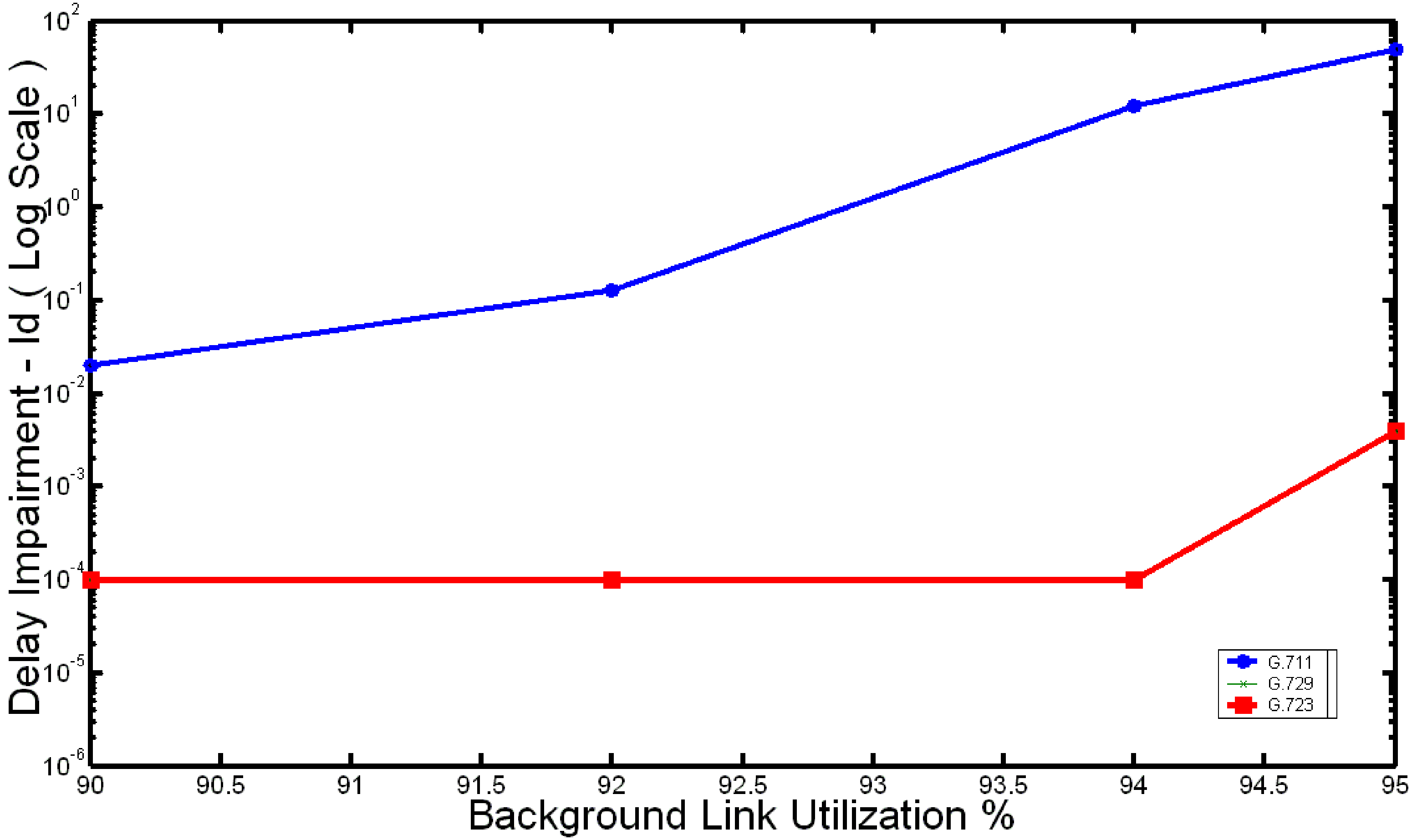
Background Link Utilization %	Codec	Delay (Ta) (ms)	I_D	I_E	R	MOS	Calls
90%	G.711	132	0.02	0	93.18	4.4	13
	G729A	91	0	11	82.2	4.1	13
	G723.1	97	0	15	78.2	3.9	13
92%	G.711	148	0.13	0	93.07	4.4	14
	G729A	94	0	11	82.2	4.1	18
	G723.1	104	0	15	78.2	3.9	19
94%	G.711	278	12.28	0	80.9	4	17
	G729A	97	0	11	82.2	4.1	14
	G723.1	106	0	15	78.2	3.9	18
95%	G.711	2326	49.4	0	43.79	2.2	22
	G729A	123	0.004	11	82.196	4.1	8
	G723.1	124	0.004	15	78.196	3.9	15

Case 3 – Optimizing for Coder and Background Link Utilization - 3



R Value, Background link Utilization % vs. Coder)

Case 3 – Optimizing for Coder and Background Link Utilization - 4



Background Link Utilization % and Id vs. Coder – case (3)

Case 3 – Optimizing for Coder and Background Link Utilization - 5

Case #	Background Link Utilization	Optimum Coder
1	90%	G.711
2	92%	G.723
3	94%	G.723
4	95%	G.723

Discussion of E-Model Optimization Results - 1

- This discussion utilized the E-Model to assist with the selection of parameters important to Design VoIP Network.

- These include the voice coder, allowable packet loss and the allowable background link utilization. It was based on the concept that maximization of the link usage with respect to the number of calls which is important to the user.

- All three cases found that G.723.1 is optimal depending on the Circumstances. G.723.1 looks more favorable due to the fact that G.723.1 uses less bandwidth per audio stream.

- Case 2 found that G.723.1 with 0.5%, 1% and 1.5% packet loss was optimal but with packet loss 2 % it was not feasible and G.729 was the optimum coder.

- Case3 G.711 coder was selected in case of background link utilization of 90% but in all other cases till 95% G.723.1 was the optimal coder giving the maximum number of calls with R Value more than 70%.

Discussion of E-Model Optimization Results - 2

Case #	Variables	Optimum Solution
1	Coder	G.723.1
2	Coder, Packet Loss %	G.723.1 with 0.5% PL
3	Coder, Packet Loss %	G.723.1 with 1% PL
4	Coder, Packet Loss %	G.723.1 with 1.5% PL
5	Coder, Packet Loss %	G.729 with 2% PL
6	Coder, Packet Loss %	G.711 with 5% PL
7	Coder, Background Link utilization	G.711 With Link Utilization 90%
8	Coder, Background Link utilization	G.723 With Link Utilization 92%
9	Coder, Background Link utilization	G.723 With Link Utilization 94%
10	Coder, Background Link utilization	G.723 With Link Utilization 95%

Discussion of E-Model Optimization Results - 3

- Although this model has the potential to be very useful in selecting coders and allowable parameters like utilization and packet loss, it is not without drawbacks. In all three tests, G.729A and G.723.1 were near enough to an R Value of 70 that any error in the estimates for delay or packet loss may cause an error in the optimization. In these situations, the optimization is considered to be sensitive to changes in those parameters.

- The ability to analyze various coders, delay, packets loss and the effect of background link utilization is vital to the design of VoIP network. The optimization of the E-Model provides a tool that is useful for this purpose.

Assignment 2

Assignment 2

- Build a VoIP Network with SIP Protocol using OPNET, the QoS tool used is E-Model.
- The objective function for all cases is to maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (R>70).
- The cases considered are:
- Find the optimal voice coder given link bandwidth, packet loss level, and background link utilization level.
- Find the optimal voice coder and the optimal packet loss level given link bandwidth and background link utilization.
- Find the optimal voice coder and the optimal background link utilization level given link bandwidth and packet loss level.