Performance Measurement of Telephone Network and Service

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Abstract

This paper summarizes my papers and books, which reports on studies concerning the performance measurement of telephone networks and services, particularly on mobile networks. In detail, the telephone network and service quality measurements are grouped into two, the first is a measurement that aims to measure the provider's customer satisfaction, performed by the provider for internal purposes, or to report to the regulator, and the second measurement goal is that the provider and/or by the vendor in relation to the quality assurance provided by the vendor to the provider. In detail, the first type of measurement uses a sampling technique to refer to the subjective method, otherwise, the second type performs aggregate measurements to refer to the Signaling Ladder Diagram of the telephone services.

1. Types of QoE and QoS Measurement of Phone Network and Service

Measurement of telephone service quality is grouped into two, the first is a measurement that aims to measure provider's customer satisfaction and the second measurement goal is that the provider and/or vendor is involved with the quality assurance provided by the vendor to the provider.

2. Quality Measurement of Telephony Service for Customer Satisfaction

In relation to customer satisfaction, telephone service quality measurements are carried out in a sampling of phone customers of the providers, thus the measured is QoE (Quality of Experience) rather than QoS (Quality of Service). In this type of measurement, the measured *parameter is the quality of the telephone conversation*, which is the indicator of the transmitted end-to-end telephony service.

This type of Quality of Experience is shown in the form of Mean Opinion Score (MOS), and the size of the MOS value is between 1.0 and 4.5 depending on the quality of the sound during conversation. In detail, MOS = 1.0 corresponds to a customer statement that feels absolutely unable to hear the conversation; instead, MOS = 4.5 is representing a very satisfied customer for the quality of the call.

Note: as a benchmark, there are some providers that provide MOS values in the range of 1 and 5 (not from 1 and 4.5).

The subjective measurement technique applies the statistical science and combines it with knowledge of ITU-T Recommendation P.830, 1996, which is a subjective method for quality assessment of broadband band and digital encoding aspects.

The application of statistical science should be accompanied by an in-depth understanding of the customer characteristics of the service provider, where the number of respondents interviewed is generally much more than the amount required for data processing.

For example, we only need data to be processed as many as 100 customers, then we should have 100 respondents who meet all the criteria of the type of customer provider, assume that is that meet the criteria for comparison of female and male customers, e.g. 55% versus 45%, and also meet the customer age criteria, e.g. 60% of youth and 40% of established age, also meet the criteria of income level, e.g. 65% lower and 35% middle, and should meeting the criteria of call intensity, e.g. 75% rarely call and 25% more than three times a day. As an

example of calculation, of all respondents who meet the criteria of the provider's customers, 20% stated that the quality of telephone conversations is very good, 50% said well, 15% said enough, and 10% said bad and 5% said the absolutely unable to hear the conversation. Then $MOS = 20\% * 4.5 + 50\% * 3.5 + 15\% * 2.5 + 10\% * 1.5 + 5\% * 1 = 3.225$ which state that the quality is relatively good.

3. Quality Measurement of the Network and Telephony Services in Relation between Provider and Vendor

Measurements of network performance and quality of telephone services in relation to vendors should be done based on the aggregate data, not a sampling, and should refer to the Signaling Ladder Diagram of the telephony services. The problem faced by the provider is almost the vendor's Signaling Ladder Diagram is not 100% the same as the Signaling Ladder Diagram compiled by International Standard ETSI and 3 GPP.

Some important parameters on this type of measurement are:

- 1) NER (Network Effectiveness Ratio)
- 2) Telephony Service Non-Accessibility
- 3) Telephony Setup Time = PDD (Post Dial Delay)
- 4) Telephony Speech Quality (objective method), and
- 5) Telephony Cut-off Call Ratio = Drop Call Ratio.

3.1. Signaling Ladder Diagram of Telephony Service on Cellular Network

SLD of telephony service on 3G cellular network refer to ETSI standard is shown in table 3.

**Table 3. Signaling Ladder Diagram of 3G Network
(ETSI Standard, not 100 % fulfilled by the Vendors)**

Trigger Point	Logical Channel	Signal Name	Outgoing Network Element	Incoming Network Element
START of NER and PDD measurement: Push The Send button.	CCCH	RRC CONNECTION REQUEST	UE	RNC1

		RADIO LINK SETUP REQUEST	RNC1	NODE B1
		RADIO LINK SETUP RESPONSE	NODE B1	RNC1
		ESTABLISH REQUEST (AAL2)	RNC1	NODE B1
		ESTABLISH CONFIRM (AAL2)	NODE B1	RNC1
		DOWNLINK SYNCHRONISATION	RNC1	NODE B1
		UPLINK SYNCHRONISATION	NODE B1	RNC1
	CCCH	RRC CONNECTION SETUP	RNC1	UE1
		INSYNCH IND	UE1	NODE B1
		RADIO LINK RESTORE INDICATION	NODE B1	RNC1
	DCCH	RRC CONNECTION SETUP COMPLETE	RNC1	UE1
	DCCH	INITIAL DT (CM SERVICE REQUEST)	UE1	RNC1
		SCCP CONNECTION RQ [INITIAL MESSAGE (CM SERVICE REQUEST)]	RNC1	MSC/VLR1
		SCCP CONNECTION CONFIRM	MSC/VLR1	RNC1
		COMMON ID	RNC1	MSC/VLR1
		LOCATION REPORTING CONTROL	MSC/VLR1	RNC1
		SECURITY MODE COMMAND	MSC/VLR1	RNC1
	DCCH	SECURITY MODE COMMAND	RNC1	UE1
	DCCH	SECURITY MODE COMPLETE	UE1	RNC1
		SECURITY MODE COMPLETE	RNC1	MSC/VLR1
		DT [IDENTITY REQUEST] (IMEI)	RNC1	MSC/VLR1
	DCCH	DLDI [IDENTITY REQUEST] (IMEI)	RNC1	UE1
	DCCH	ULDI [IDENTITY RESPONSE] (IMEI)	UE1	RNC1
		DT [IDENTITY RESPONSE] (IMEI)	RNC1	MSC/VLR1
	DCCH	ULDI [SETUP]	UE1	RNC1
		DT [SETUP]	RNC1	MSC/VLR1
		DT [CALL PROCESSING]	MSC/VLR1	RNC1
	DCCH	DLDI [CALL PROCEEDING]	RNC1	UE1
		RAB ASSIGNMENT REQUESTS	MSC/VLR1	RNC1
		ESTABLISH REQUESTS (AAL2)	RNC1	MGW
		ESTABLISH CONFIRM (AAL2)	MGW	RNC1
		RADIO LINK RECONFIG PREPARE	RNC1	NODE B1
		RADIO LINK RECONFIG READY	NODE B1	RNC1
		ESTABLISH REQUESTS (AAL2)	RNC1	NODE B1
		ESTABLISH CONFIRM (AAL2)	NODE B1	RNC1
		RADIO LINK RECONFIG COMMIT	RNC1	NODE B1
	DCCH	RADIO BEARER SETUP	RNC1	UE1
		RAB ASSIGNMENT RESPONSE	RNC1	MSC/VLR1
	DCCH	RADIO BEARER SETUP COMPLETE	UE1	RNC1
		IAM	MSC/VLR1	MSC/VLR2
		PAGING	MSC/VLR2	RNC2
	PCH	PAGING TYPE 1	RNC2	UE2
	CCCH	RRC CONNECTION REQUEST	UE2	RNC2
		RADIO LINK SETUP REQUEST	RNC2	NODE B2
		RADIO LINK SETUP RESPONSE	NODE B2	RNC2

		ESTABLISH REQUEST (AAL2)	RNC2	NODE B2
		ESTABLISH CONFIRM (AAL2)	NODE B2	RNC2
		DOWNLINK SYNCHRONISATION	RNC2	NODE B2
		UPLINK SYNCHRONISATION	NODE B2	RNC2
	CCCH	RRC CONNECTION SETUP	RNC2	UE2
		INSYNCH IND	UE2	NODE B2
		RADIO LINK RESTORE INDICATION	NODE B2	RNC2
	DCCH	RRC CONNECTION SETUP COMPLETE	UE2	RNC2
	DCCH	PAGING RESPONSE / DDCH: INITIAL DT	UE2	RNC2
		SCCP CONNECTION REQUEST	RNC2	MSC/VLR2
		UE CONNECTION ESTABLISH (PAGING RESPONSE)	RNC2	MSC/VLR2
		SCCP CONNECTION CONFIRM	MSC/VLR2	RNC2
		COMMON ID	RNC2	MSC/VLR2
		LOCATION REPORTING CONTROL	MSC/VLR2	RNC2
		SECURITY MODE COMMAND	MSC/VLR2	RNC2
	DCCH	SECURITY MODE COMMAND	RNC2	UE2
	DCCH	SECURITY MODE COMPLETE	RNC2	UE2
		SECURITY MODE COMPLETE	RNC2	MSC/VLR2
		DT [IDENTITY REQUEST] (IMEI)	MSC/VLR2	RNC2
		[IDENTITY REQUEST](IMEI)	RNC2	MSC/VLR2
	DCCH	[IDENTITY RESPONSE] (IMEI)	RNC2	UE2
	DCCH	DT [IDENTITY RESPONSE] (IMEI)	UE2	RNC2
		DT [IDENTITY RESPONSE] (IMEI)	MSC/VLR2	RNC2
		DT [SETUP]	RNC2	MSC/VLR2
	DCCH	DLDI [SETUP]	RNC2	UE2
	DCCH	DLDI [SETUP]	RNC2	UE2
	DCCH	ULDI [CALL CONFIRMED]	UE2	RNC2
		DT	MSC/VLR2	RNC2
		RAB ASSIGNMENT REQUESTS	MSC/VLR2	RNC2
		ESTABLISH REQUESTS (AAL2)	RNC2	MGW
		ESTABLISH CONFIRM (AAL2)	MGW	RNC2
		RADIO LINK RECONFIG PREPARE	RNC2	NODE B2
		RADIO LINK RECONFIG READY	NODE B2	RNC2
		ESTABLISH REQUESTS (AAL2)	RNC2	NODE B2
		ESTABLISH CONFIRM (AAL2)	NODE B2	RNC2
		RADIO LINK RECONFIG COMMIT	RNC2	NODE B2
	DCCH	RADIO BEARER SETUP	RNC2	UE2
		RAB ASSIGNMENT RESPONSE	RNC2	MSC/VLR2
	DCCH	RADIO BEARER SETUP COMPLETE	UE2	RNC2
	DCCH	ULDI [ALERTING]	UE2	RNC2
		DT [ALERTING]	RNC2	MSC/VLR2
		ACM	MSC/VLR2	MSC/VLR1
		DT [ALERTING]	MSC/VLR1	RNC2

STOP of NER and PDD measurement: Alerting tone is heard by the A-party coming from the B-party AND B-party rings.		DLDT [ALERTING]	RNC2	UE1
START of Drop Call Measurement: The CONNECT message on the DCCH logical channel is passed from the MSC to the UE to indicate that the connection has been established.	DCCH	DT [CONNECT]	MSC/VLR1	RNC1
		DLDT [CONNECT]	RNC1	UE1
	DCCH	ULDT [CONNECT ACK]	UE1	RNC1
		DT [CONNECT ACK]	RNC1	MSC/VLR1
STOP of Drop Call Measurement: A DISCONNECT message on the DCCH logical channel is sent from the UE (message sent when the user ends the call).	DCCH	ULDT [DISCONNECT]	UE1	RNC1

3.2. NER (Network Effectiveness Ratio) Measurement

Network Effectiveness Ratio is a measure of the quality of the network without applying the

A-party error and B-party busy.

Analysis: NER measurement data based on the counter in the MSC and or MGW which records all incoming signals, whether successfully processed or failed processing. The possibility of failure, other than those caused by a failure in the process of signal transmission and signal processing in network elements, is caused by the failure of these processes below:

- Radio Link setup and establish by node B1 (node B of A-party)
- Downlink and uplink synchronization by node B1 and RNC1
- RRC connection setup by node B1 and UE1
- SCCP connection by MSC1 & VLR1
- Location control by MSC1 & VLR1 and RNC1
- Authentication process of UE1, If it fails, it is not taken into account in the NER
- The same process as above to UE2 (= B-party), node B-2, RNC2 and MSC & VLR2

3.3. Measurement of the Telephony Service Non-Accessibility

The probability that the end user cannot access the Mobile Telephony Service when displaying of the network indicator on the mobile device is in good condition, which suggests that the availability and accessibility network is good. The terms of the QoS measurement should take into account that the B-party is not in the busy states, and no A-error states.

In cellular networks, the measurement begins when the A-party pressed the Send button. At that time the cellphone will send a signal "RRC CONNECTION REQUEST" to BTS or Node-B. Once the core network receives this signal, then do the preparation of the CCCH logical channel mapped to the RACH transport channel. It is possible that there is more than one RRC CONNECTION REQUEST per attempt, in this case only one signal that counts. It is possible that the RRC connection established because of the Location Update is wrong; in this case, test sample should be removed.

Successful call attempt is marked by the presence of Alerting tone is heard by the A-party, AND B-party rings.

Abstract Equation of Telephony Service Non-Accessibility:

$$\text{TelephonyService Non - Accessibility(\%)} = \frac{\text{unsuccessfull call attempts}}{\text{all call attempts}} \times 100 \quad (1)$$

3.4. Measurement of the Telephony Setup Time = PDD (Post Dial Delay)

Telephony Setup Time describes the duration of customer presses the send button (= mobile station sending the complete address information to the network) and customer receive the call set-up notification (receive the connect established message).

Abstract Equation of Telephony Setup Time:

$$\text{TelephonySetup Time} = t_{\text{connect_established}} - t_{\text{sending of complete address in}} \quad (2)$$

Analysis: Measurement of telephony setup time is done based on the data at the timer in the MSC / MGW which records the time of arrival signal and the release signals. In general, if all signaling goes normal, the processing time of the entire signal, plus the propagation time is only a few hundred milliseconds. But, the telephony setup time may be measured in a few seconds, this is happening cause of the processing time of tens of signals. The waiting time will be greater if the resource-pool is available in limited quantities. Telephony setup time does not take into account the elapsed time by a signaling failure. Signaling that fail will only affect the performance of the NER and other performance-measured in the counter.

3.5. Measurement the Telephony Speech Quality

Based on the recommendations of the ITU-T, there are two ways to measure the value of MOS objectively, including:

- a) Refer to ITU-T P.862 (ITU-T Recommendation P.862, 2001), which use the concept of Perceptual Evaluation of Speech Quality (PESQ) is an objective method for assessing the quality of the conversation end-to-end narrow-band telephone networks and speech codecs
- b) Refer to the Recommendation ITU-T G.107 (E-model) (ITU-T Recommendation G.107, 2014), which is a calculation of predictable sound quality as perceived by customers for telephone connection end-to-end using passive methods.

3.5.1. ITU-T Recommendation G.107 (E-model).

The calculation of the predicted sound quality as perceived by customers for telephone connection end-to-end using passive methods can be done using a transmission planning E-model. E-model takes into account various telephony disorders, particularly due to interference from the low bit rate coding devices, one-way delay, and disruption of classic telephony is loss, noise, and echo. E-model can be used to calculate the R factor, known as the transmission rating factor, which is a simple measure of the quality of the sound. The R factor can be transformed into a quality index as Mean Opinion Score (MOS), Percentage Good or Better (Gob), or Percentage Poor or Worse (POW).

3.5.2. Telephony voice quality calculation procedure (using the R factor)

1. Calculate the R factor use the formula

$$R = R_o - I_s - I_d - I_e - \text{eff} + A \quad (3)$$

Where:

- R_o represents the SNR (signal-to-noise-ratio), including noise sources such as noise circuit and room noise.
- I_s is a combination of all the distractions that occur simultaneously with the voice signal?
- I_d Factor is a disorder caused by the delay.
- I_e -eff (effective equipment impairment factor) represents the interference caused by the low bitrate codec, and the disruption caused by loss of packet at random.
- The A factor is an advantage that allows the compensation of the factors that caused the disruption another advantage of the user access.

2. After obtaining the R factor, then calculate the factor E, use the formula:

$$E = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^x e^{-\frac{t^2}{2}} dt \quad [18] \quad (4)$$

3. After obtaining the E factor, then the sound quality telephony is calculated use the formula:

- Percentage Good or Better (Gob) is calculated using the formula:

$$GoB(\%) = \frac{R - 60}{16} * 100 \quad [18] \quad (5.a.)$$

- Percentage Poor or Worse (POW) is calculated using the formula

$$PoW(\%) = \frac{45 - R}{16} * 100 \quad [18] \quad (5.b.)$$

○ MOSCQE (MOS Conversational Quality Estimate) is calculated using the formula

- for $R < 0$, then: MOSCQE = 1
- for $0 < R < 100$, then: MOSCQE = $1 + 0.035R + 0.000001 R*(R-60)*(100-R)*7$
- for $R > 100$, then: MOSCQE = 4.5

Figure 1. E Models (ITU-T Rec. G107)

Where:

- SLR (Send Loudness Rating)
- RLR (Receive Loudness Rating)
- OLR (Overall Loudness Rating)
- STMR (Side tone Masking Rating)
- LSTR (Listener Side tone Rating)
- Ds (D-value of telephone at send-side)
- Dr (D-value of telephone at receive-side)
- TELR (Talker Echo Loudness Rating)
- WEPL (Weighted Echo Path Loss)
- T (Mean one way delay of the echo path)
- Tr (Round-trip delay in a closed 4-wire loop)
- Ta (Absolute one-way delay in echo free connections)
- qdu (Number of quantization distortion units)
- Ie (Equipment impairment factor)
- Ppl (Random packet-loss probability)
- Bpl (Packet-loss robustness factor)
- Ne (Circuit noise referred to the 0 dBr-point)

- Nfor (Noise floor at the receive-side)
- Ps (Room noise at the send-side)
- Pr (Room noise at the receive-side)
- A (Advantage factor)

Table 1. Default Parameters value

Parameter	Abbr.	Unit	Default value	Permitted range
Send loudness rating	SLR	dB	+8	0 ... +18
Receive loudness rating	RLR	dB	+2	-5 ... +14
Sidetone masking rating	STMR	dB	15	10 ... 20
Listener sidetone rating	LSTR	dB	18	13 ... 23
D-Value of telephone, send side	Ds	–	3	-3 ... +3
D-Value of telephone, receive side	Dr	–	3	-3 ... +3
Talker echo loudness rating	TELRL	dB	65	5 ... 65
Weighted echo path loss	WEPL	dB	110	5 ... 110
Mean one-way delay of the echo path	T	ms	0	0 ... 500
Round-trip delay in a 4-wire loop	Tr	ms	0	0 ... 1000
Absolute delay in echo-free connections	Ta	ms	0	0 ... 500
Number of quantization distortion units	qdu	–	1	1 ... 14
Equipment impairment factor	Ie	–	0	0 ... 40
Packet-loss robustness factor	Bpl	–	4.3	4.3 ... 40
Random packet-loss probability	Ppl	%	0	0 ... 20
Burst ratio	BurstR	–	1	1 ... 8
Circuit noise referred to 0 dBm-point	Nc	dBmOp	-70	-80 ... -40
Noise floor at the receive side	Nfor	dBmp	-64	–
Room noise at the send side	Ps	dB(A)	35	35 ... 85

Table 2. The Correlation between Satisfactions with 4 Parameters

R-value (lower limit)	MOS _{CQE} (lower limit)	GoB (%) (lower limit)	PoW (%) (upper limit)	User satisfaction
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

3.6. Measurement the Telephony Cut-off Call Ratio (= Drop Call Ratio)

Telephony Cut-off Call Ratio calculation of the probability of a successful call is not terminated by the normal termination.

The normal termination of telephone connections only occur when:

- 1) The A-party hang-up the connection, and/or
- 2) The A-party still holds on, but the B-party hang-ups, then the normal termination occur in 30 seconds after the B-party hangs up.

Telephony cut-off call occurs when termination occurs abnormally.

The measurement of the Telephony Cut-off Call Ratio begins when alerting or busy tone heard by the A-party coming from B-party. In cellular networks, START: The CONNECT message on the DCCH logical channel is passed from the MSC to the MS to indicate that the connection has been established. NORMALLY, STOP: Release of connection directly by A- or B-party, marked by A DISCONNECT message on the DCCH logical channel is sent from the UE (message sent when the user ends the call).

Abstract Equation of Telephony Cut-off Call Ratio:

$$\text{Telephony cut - off call ratio} = \frac{\text{unintentionally terminated telephony calls}}{\text{all successful telephony call attempts}} \quad [6]$$

Practical experience shows that the event of the cut-off of the phone call is in progress when the SLD shows this reason.

4. Measurement of the Network and Telephone Service on 4G and 5G Cellular Networks

The 4G and 5G cellular networks have characteristics very different from their predecessors, where there are two very principal differences:

- a) The 4G and 5G mobile networks are fully IP-based or end to end IP, as a result, SLD from the network is **simpler** than that of 3G networks because 4G and 5G networks no longer have network-elements that separate the circuit-switched and packet switched as found on 3G networks.
- b) Telco should develop the IP-based telephone service, otherwise if it does not work, then the over-the-top (OTT) provider's telephone service will dominate, so the network performance measurement will be done separately with service quality measurement, where the network performance is the responsibility of Telco, on the other hands, the service quality depends on the quality of the OTT provider.

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