Call Quality and Its Parameter Measurement in Telecommunication Networks

Akram Aburas and Khalid Al-Mashouq

***Abstract*—The measurement of call quality from end users perspective is emerging area of research on speech quality in telecommunication networks. Our idea is focused at deriving and developing a system to measure certain call parameters during the call and provide average scores at the end of the call. Call quality for the bundle of calls is derived based on the aggregation of successful call parameters which gives the overall call quality measure. The call parameters used in our research were Signal Strength, the successful call rate, normal drop call rate, handover drop rate. GPS coordinates are also used to locate the location and quality of the individual calls. Also a model using the sms feature for tackling the critical quality has been proposed and implemented. Finally the basic bandwidth quality measurement approach is presented which addresses the issue of low bandwidth quality with respect to both user and the operator.**

***Index Terms*—Call quality measurement, signal strength, successful call rate, normal drop call rate, handover drop rate, lac(location area code), bandwidth quality.**

I. Introduction

Traditional speech quality measurement techniques use the subjective listening tests called Mean Opinion Score (MOS). It's based on the human perceived speech quality based on the scale of 1 to 5, where 1 is the lowest perceived quality and 5 is the highest perceived quality.

Subjective listening tests are expensive, time consuming and tedious. So, currently most of the systems use objective evaluation of speech quality. Objective testing systems are use automated speech quality measurement techniques. The three well known objective measurement techniques are Perceptual Speech Quality Measure (PSQM), Perceptual Analysis Measurement System (PAMS) and Perceptual Evaluation of Speech Quality (PESQ).

Objective speech quality measurement techniques mostly are based on input-output approach [1]. In input-output objective measurement techniques basically works by measuring the distortion between the input and the output signal. The input signal would be a reference signal and output signal would be a received signal.

Input-output based speech quality assessment in objective speech quality measurement gave good correlations with reaches up to 99% in some cases [2]. Estimating the speech quality without the presence of input signal or reference signal is latest area of research.

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Input-output based speech quality assessment in objective speech quality measurement gave good correlations with reaches up to 99% in some cases [3]. The performance of objective measurement is basically achieved by correlating their results with the subjective quality measure.

Estimating the speech quality without the presence of input signal or reference signal is latest area of research.

Jin Liang and R. Kubichek [4] published a first paper on output-based objective speech quality using perceptually-based parameters as features. The results were quiet appreciable with 90% correlation. R. Kubichek and Chiyi Jin [5] used the vector quantization method with 83% correlation achievement.

An output based speech quality measurement technique using visual effect of a spectrogram is proposed in [6]. An output-based speech quality evaluation algorithm based on characterizing the statistical properties of speech spectral density distribution in the temporal and perceptual domains is proposed in [7]. The correlations results achieved with subjective quality scores were 0.897 and 0.824 for the training data and testing data set respectively.

A time-delay multilayer neural network model for measuring the output based speech quality was proposed by Khalid Al-Mashouq and Mohammed Al-Shayee in [8]. The correlation achieved for speaker and text independent was 0.87.

In this Paper we presented our work for determining the call quality parameters such as average signal strength, successful call rate and successful handover rate with respect to signal strength and successful rate. Then final call quality is computed from the extracted parameters.

This research is continuation of the work that has been proposed in [9]-[15]. The basic bandwidth quality measurement is proposed which can be used by both the operator and the user to evaluate the bandwidth quality of a particular operator.

II. Call Quality

The research is focused on call quality measurement. The system logs the signal strength information for every 5ms if there is change in the signal strength information. The system records the number of successful and un-successful call attempts made for every ten call attempts. The successful and un-successful call attempts are classified based on whether the call is successfully connected by the network. The call drop information such as normally dropped from either of the party or dropped due to handover during the cell change is also recorded.

The average signal strength of successful calls, normal dropped and handover dropped with their average scores are recorded. The overall successful call rate score is also derived based on below scale:

successful calls 9-10 score : 5(Excellent)
successful calls 7-8 score : 4
successful calls 5-6 score : 3
successful calls 3-4 score : 2
successful calls 1-2 score : 1(Very Bad)

Normally dropped call rate score is derived based on below scale:

Normal dropped calls >8 score: 5(Excellent)
Normal dropped calls < 7 & <8 score: 4
Normal dropped calls < 6 & < 7 score: 3
Normal dropped calls < 4 & < 6 score: 2
Normal dropped calls < 4 score: 1 (Very Bad)

The call quality is derived from the scores computed from the above parameters as below:

(Average signal strength score of all successful calls + successful call rate score + normal dropped calls rate score)/3 .

The landmarks that were marked with red colors are the calls dropped due to handover and the landmarks that were marked with green colors are normally dropped calls. The different colors landmarks help one to easily visualize and analyze the calls.

The system has the ability to send the signal strength information to the particular number. It has the provision of setting the mobile number, to which the sms would be sent automatically at the end of call. The system has the option of setting to send the sms always, less than bad etc. at the end of 10 calls the call statistics would also be sent as sms. The below Fig. 1. Signal meter flowchart illustrates the complete process of signal meter system.

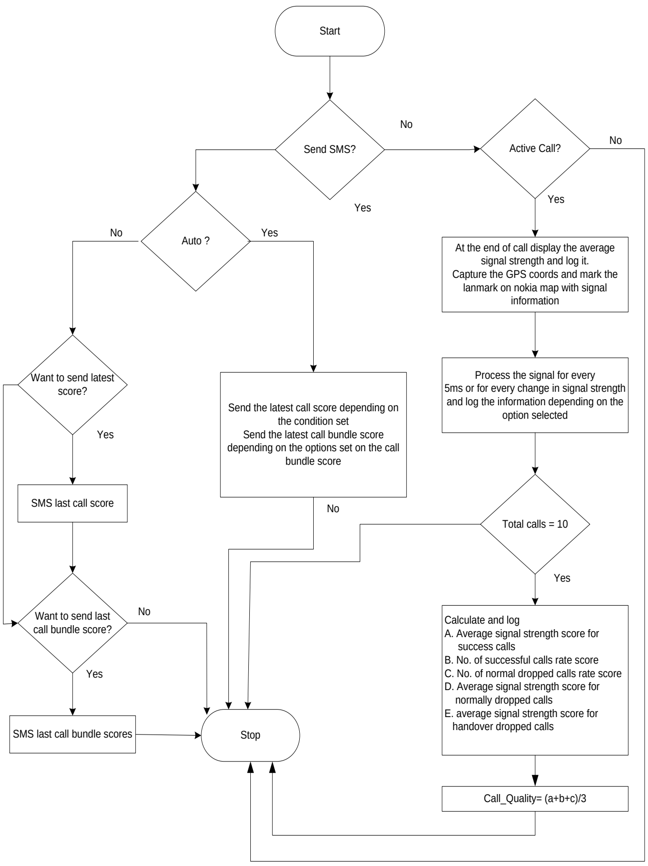

Fig. 1. Signal meter flowchart

A generic algorithm of our signal measure algorithm is presented below:

Signal_Measure( )

1) Get the preferences for log_change, log_location
2) Get total_calls, Call_attempts_failed, call_attempts_successfull, normal_dropped_calls, handover_dropped calls
3) if (total_calls =10) reset all variables to zeros
4) if (call_attempt = failed)
   total_calls=total_calls+1
   call_attempts_failed=call_attempts_failed+1
5) While (phone_status != idle && call_attempt = successful)
6) total_calls=total_calls+1
7) call_attempts_successful=call_attempts_successful +1
8) if(gps_coords available)
   Get the gps_coords
9) Get the date, time, cell_id.
10) Get the signal_strength
11) if (log_change= 5ms)
   write("signal_measure.log",date, time, gps_coords, cell_id, signal_strength)
12) if (log_change= when changed) and (signalstrengthchange=yes)
   Write("signal_measure.log",date, time, phone status, signal strength)
13) End of While
14) End of call
15) Calculate average_signal_strength
16) If (average_signal_strength <= -95 && average_signal_strength >= -120) SignalQuality= Extremely Bad
   Elseif (average_signal_strength <= -85 && average_signal_strength >-95) SignalQuality=Bad
   Elseif (average_signal_strength <= -75 && average_signal_strength > -85) SignalQuality=Average
   Elseif (average_signal_strength <=-65 && average_signal_strength > -75) SignalQuality=Good
   Elseif (average_signal_strength <= -55 && average_signal_strength > -65) SignalQuality=Very Good
17) Write("signal_measure.log",date, time, phone status, average_ signal_strength)
18) if (sendSMS = auto && whenSMSsend = always || sendSMS =auto && whenSMSsend < bad)
   sendSMS(average_signal_strength, SignalQuality,call_drop_information)
19) Write (SignalQuality)
20) If (GPS_Coords Available)
   If (call_dropped = Normal)
   Normal_dropped_calls=Normal_dropped_calls+1
   Landmark_color = green
   Else
   landmark_color = red
   handover_dropped_calls= handover_dropped_calls+1
   Open(nokia_map)
   Plot(gps_coords, landmark)
21) if (total_calls = 10)
   Score_handover_dropped=

sum(handover_dropped_quality)/handover_dropped_calls
Score_normal_dropped= sum(normal_dropped_quality)/normal_dropped_calls
score_successful_attempts= (sum(handover_dropped_quality+sum(normal_dropped_quality)) /total_successful_attempts

22) If (call_attempts_successfull< =2)
Score_successful_call_rate = 1
Elseif (call_attempts_successfull< =3 && average_signal_strength >=4)
Score_successful_call_rate = 2
Elseif (call_attempts_successfull< =5 && average_signal_strength >=6)
Score_successful_call_rate =3
Elseif (call_attempts_successfull< =7 && average_signal_strength >=8)
Score_successful_call_rate =4
Elseif (_attempts_successfull< =9 && average_signal_strength >=10)
Score_successful_call_rate =5
23) If (handover_success_calls< 40%)
Score_handover_success_calls_rate = 1
Elseif (handover_success_calls <40% && handover_success_calls >60)
Score_handover_success_calls_rate = 2
Elseif (handover_success_calls <60% && handover_success_calls >70)
Score_handover_success_calls_rate =3
Elseif (handover_success_calls <70% && handover_success_calls >80)
Score_handover_success_calls_rate =4
Elseif (handover_success_calls >80)
Score_handover_success_calls_rate =5
24) Calculate average_call_quality=(score_successful_attempts+score_successful_call_rate+score_handover_success_calls_rate)/3
25) Write("calls_stats", total_call_attempts_failed, total_call_attempts_successful, score_successful_attempts, normal_dropped_calls,score_normal_dropped, handover_dropped_calls,score_handover_dropped, score_successfull_call_rate,score_handover_success_calls_rate, average_call_quality)
26) if (sendSMS = auto && whenSMSstat_send = always || sendSMS =auto && whenSMS_call_failed < 5) || whenSMS_handover_dropped < 2)
sendSMS(num_calls_unsuccessful, num_calls_successful, num_of_calls_dropped_handover, num_normal_dropped)
27) If (log location = internal memory)
save signalmeter.log to c:/data
save calls_stats.log to c:/data
else save signalmeter.log to e:/data
save calls_stats.log to e:/data
28) if(sendSMS = Manual && want_to_send_sms= yes)
set(mobile_number)
sendSMS(signal_stength, SignalQuality,call_drop_information)
29) End of Program

TABLE I shows final call quality classification is based on the score for a bundle of 10 calls.

TABLE I: CALL QUALITY SCORE

Score	Classification
<1	Extremely Bad
1 - 2	Bad
2- 3	Average
3- 4	Good
4 - 5	Excellent

III. BANDWIDTH QUALITY

The bandwidth quality of internet provided by the mobile operators fluctuates drastically and frustrates the user sometimes. As the number of users increases in particular cell of the mobile network, the bandwidth decreases and hence there would be loss of revenue, if the situation continues for the operator. An attempt to measure the average bandwidth quality per individual and for bundle of ten download is calculated as portrayed in Fig. 2. The scores computed can be used by the user as well as the operator to evaluate the bandwidth quality.

The average bandwidth of data download is computed using the below procedure:

1) When a new download is initiated save the current time as "T_b".
2) When the download is finished save the current time as "T_e" and the file size as "Fs"
3) Calculate the average speed for this download as $S_i = F_s/(T_e – T_b)$.
4) The average bandwidth for "10" downloads is calculated as $S = (S_1 + S_2 + ... + S_{10}) /10$.

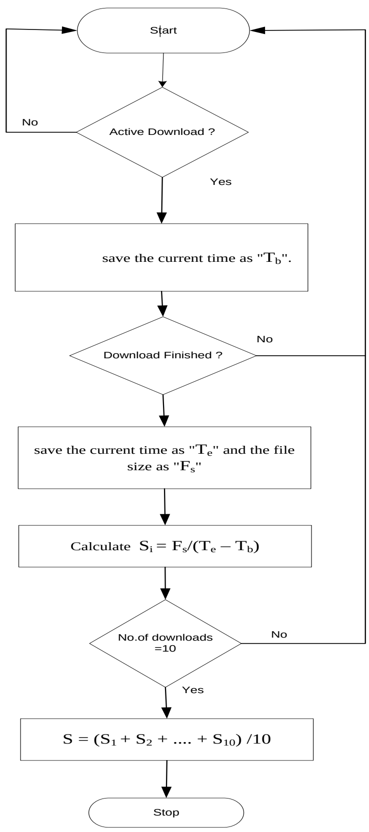

Fig. 2. Bandwidth calculation

The score for the average bandwidth is computed on the scale of 1 – 5 based on the below TABLE II:

TABLE II: BANDWIDTH SCORE

Bandwidth Range	Score
< 32 kbps	1 (Extremely Bad)
<32 & <64 kbps	2 (Bad)
<64 & < 128 kbps	3 (Average)
<128 & <256 kbps	4 (Good)
> 256	5 (Excellent)

The bandwidth quality scores will give the user and the operator the better insight into the usage of the bandwidth. The approach can further be enhanced by capturing the cell-id and sending the critical scores for analyzing to provide the better service.

IV. CONCLUSION

A conclusion section is not required. Although a conclusion may review the main points of the paper, do not replicate the abstract as the conclusion. A conclusion might elaborate on the importance of the work or suggest applications and extensions. The proposed research uses the parameters in measuring the call quality and bandwidth quality in mobile telecommunications networks. It can be used to benchmark the mobile network by the user and hence it can be used as a base for charging the customer by the operators. It can also be used by the telecom authorities to regulate and evaluate the mobile operators by regularly checking the network for knowing whether network operators are meeting the required license criteria of quality of network from end users perspective. Further, it can be used as consumer protection tool to ensure that tariffs correlate with call quality. The proposed bandwidth quality measurement approach can be used network operators to enhance the network in the particular areas and provide the better service.

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