[1] M. AL-Akhras, H. Zedan, R. John, and I. ALMomani. Non-intrusive speech quality prediction in VoIP networks using a neural network approach. *Neurocomputing*, 72(10-12):2595-2608, 2009. [bib | DOI]

Measuring speech quality in Voice over Internet Protocol (VoIP) networks is an increasingly important application for legal, commercial and technical reasons. Any proposed solution for measuring the quality should be applicable in monitoring live-traffic non-intrusively. The E-Model proposed by the International Telecommunication Union-Telecommunication Standardisation Sector (ITU-T) achieves this, but it requires subjective tests to calibrate its parameters. In this paper a solution is proposed to extend the E-Model to any new network conditions and for newly emerging speech codecs without the need for the time-consuming, expensive, hard to conduct subjective tests. The proposed solution is based on an artificial neural network model and is compared against the E-Model to check its prediction accuracy.

Keywords: Artificial neural network, Speech quality, VoIP, Non-intrusive, E-Model, Perceptual evaluation of speech quality

[2] J. Dolezal and L. Kencl. Improving QoE of SIP-based automated voice interaction in mobile networks. In *International Conference on Network and Service Management (CNSM)*, pages 329-335, 2012. [bib]

Mobile voice-assisted services are currently experiencing strong growth. However, occasionally low real-time quality of service within mobile networks could have significant negative impact on quality of experience of users interacting with automated voice services. Latency may grow to unacceptable levels and speech recognition and synthesis might suffer. We present a methodology of mitigating such effects by monitoring the immediate connection status and adapting various parameters of the SIP communication setup (buffer size, codec) in response, thus radically improving user experience. We demonstrate practical usability by implementation and testing in a real mobile network and by performing multiple test scenarios when interacting with a state-of-the-art automated voice platform.

Keywords: mobile radio;quality of service;signalling protocols;speech recognition;speech synthesis;telecommunication services;QoE improvement;SIP communication setup;SIP-based automated voice interaction;mobile networks;mobile voice-assisted services;real-time quality of service;session initiation protocol;speech recognition;speech synthesis;Codecs;Delay;Jitter;Mobile communication;Mobile computing;Quality of service

[3] S. Egger, R. Schatz, K. Schoenenberg, A. Raake, and G. Kubin. Same but different? - using speech signal features for comparing conversational VoIP quality studies. In *International Conference on Communications (ICC)*, pages 1320-1324, 2012. [bib | DOI]

In this paper we demonstrate how speech signal features can be used to detect and explain differences in human to human conversation tests. To this end, we compare the results of two conversational VoIP quality experiments designed to quantify the impact of network delay on perceived speech quality. Both studies followed the same procedures and used the same scenarios, but were conducted in two different labs. Our comparison shows that the two studies, despite having been executed correctly using the same test design, still can produce surprisingly different results regarding the users quality perception on a MOS scale. In this respect, speech signal features extracted from conversation recordings help identifying divergent participant behavior as plausible cause for such differences. Our in-depth analysis reveals how novel parameters developed by us like Intended and Unintended Interruption Rate (IIR, UIR) and the corrected Speaker Alternation Rate SARcorr can be used to successfully determine the extent to which the results of different conversational speech quality studies are directly comparable and thus eligible for pooling, or not.

Keywords: Internet telephony;feature extraction;speech processing;MOS scale;SAR;VoIP quality;conversation recordings;corrected speaker alternation rate;human-to-human conversation tests;intended interruption rate;network delay;perceived speech quality;speech signal feature extraction;unintended interruption rate;user quality perception;Delay;Feature extraction;Humans;Interrupters;Reliability;Speech;Telecommunications;Conversational Interactivity;Conversational VoIP Quality Studies;Delay;Speech Signal Features

[4] O. Hohlfeld, E. Pujol, F. Ciucu, A. Feldmann, and P. Barford. BufferBloat: How relevant? a QoE perspective on buffer sizing. Technical Report 2012-11, Fakultät IV - Elektrotechnik und Informatik, Technische Universität Berlin, Nov. 2012. [bib]

[5] B. Huntgeburth, M. Maruschke, and S. Schumann. Open-source based prototype for Quality of Service (QoS) monitoring and Quality of Experience (QoE) estimation in telecommunication environments. In *5th International Conference on Next Generation Mobile Applications, Services and Technologies (NGMAST)*, pages 161-168, 2011. [bib | DOI]

This paper describes an implementation for monitoring the Quality of Service (QoS) and expecting the Quality of Experience (QoE) of a voice communication in a Real-time Transport Protocol (RTP) based telecommunication environment. The resulting QoS parameters are evaluated, the QoE is determined with the E-Model and processed for graphical presentation. With the use of some open-source programming libraries, the presented prototype can be a helpful alternative for expensive measurement devices and is ready to be deployed in a widespread telecom environment at low cost.

Keywords: Internet telephony;quality of service;telemetry;transport protocols;voice communication;QoE estimation;QoS monitoring;RTP;VoIP;graphical presentation;open-source based prototype;open-source programming library;quality of experience estimation;quality of service monitoring;real-time transport protocol;telecommunication environments;voice communication;voice over IP;Current measurement;Delay;Libraries;Monitoring;Protocols;Quality of service;Real time systems;QoE;QoS;VoIP;monitoring;open-source;prototype

[6] S. Jelassi, G. Rubino, H. Melvin, H. Youssef, and G. Pujolle. Quality of experience of VoIP service: A survey of assessment approaches and open issues. *IEE Communications Surveys Tutorials*, 14(2):491-513, 2012. [bib | DOI]

This survey gives a comprehensive review of recent advances related to the topic of VoIP QoE (Quality of user' Experience). It starts by providing some insight into the QoE arena and outlines the principal building blocks of a VoIP application. The sources of impairments over data IP networks are identified and distinguished from signal-oriented sources of quality degradation observed over telecom networks. An overview of existing subjective and objective methodologies for the assessment of the QoE of voice conversations is then presented outlining how subjective and objective speech quality methodologies have evolved to consider time-varying QoS transport networks. A description of practical procedures for measuring VoIP QoE and illustrative results is then given. Utilization methodology of several speech quality assessment frameworks is summarized. A survey of emerging single-ended parametric-model speech quality assessment algorithms dedicated to VoIP service is then given. In particular, after presenting a primitive single-ended parametric-model algorithm especially conceived for the evaluation of VoIP conversations, new artificial assessors of VoIP service are detailed. In particular, we describe speech quality assessment algorithms that consider, among others, packet loss burstiness, unequal importance of speech wave, and transient loss of connectivity. The following section concentrates on the integration of VoIP service over mobile data networks. The impact of quality-affecting phenomena, such as handovers and CODEC changeover are enumerated and some primary subjective results are summarized. The survey concludes with a review of open issues relating to automatic assessment of VoIP.

Keywords: IP networks;Internet telephony;mobile radio;quality of service;speech processing;voice communication;CODEC changeover;IP network;VoIP QoE;VoIP automatic assessment;VoIP conversation;VoIP service;artificial assessor;mobile data network;packet loss burstiness;quality degradation;quality of user experience;quality-affecting phenomena;signal-oriented sources;single-ended parametric-model algorithm;speech quality assessment algorithm;speech wave;telecom network;time-varying QoS transport network;utilization methodology;voice conversation;Degradation;Delay;Jitter;Laboratories;Quality assessment;Quality of service;Speech;Mobile heterogeneous networks;Parametric SQA algorithms;Quality of users' Experience (QoE);Speech quality assessment (SQA);VoIP

[7] S. Jelassi, H. Youssef, C. Hoene, and G. Pujolle. Single-ended parametric voicing-aware models for live assessment of packetized VoIP conversations. *Telecommunication Systems*, 49(1):17-34, 2012. [bib | DOI]

The perceptual quality of VoIP conversations depends tightly on the pattern of packet losses, i.e., the distribution and duration of packet loss runs. The wider (resp. smaller) the inter-loss gap (resp. loss gap) duration, the lower is the quality degradation. Moreover, a set of speech sequences impaired using an identical packet loss pattern results in a different degree of perceptual quality degradation because dropped voice packets have unequal impact on the perceived quality. Therefore, we consider the *voicing* feature of speech wave included in lost packets in addition to packet loss pattern to estimate speech quality scores. We distinguish between voiced, unvoiced, and silence packets. This enables to achieve better correlation and accuracy between *human-based subjective* and *machine-calculated objective* scores.

This paper proposes novel *no-reference parametric* speech quality estimate models which account for the voicing feature of signal wave included in missing packets. Precisely, we develop separate speech quality estimate models, which capture the perceptual effect of removed voiced or unvoiced packets, using elaborated simple and multiple regression analyses. A new speech quality estimate model, which mixes voiced and unvoiced quality scores to compute the overall speech quality score at the end of an assessment interval, is developed following a rigorous multiple linear regression analysis. The input parameters of proposed voicing-aware speech quality estimate models, namely Packet Loss Ratio (PLR) and Effective Burstiness Probability (EBP), are extracted based on a novel *Markov model of voicing-aware packet loss* which captures properly the feature of packet loss process as well as the voicing property of speech wave included in lost packets. The conceived *voicing-aware packet loss model* is calibrated at run time using an efficient packet loss event driven algorithm. The performance evaluation study shows that our voicing-aware speech quality estimate models outperform voicing-unaware speech quality estimate models, especially in terms of accuracy over a wide range of conditions. Moreover, it validates the accuracy of the developed parametric no-reference speech quality models. In fact, we found that *predicted scores* using our speech quality models achieve an excellent correlation with *measured scores* (>0.95) and a small mean absolute deviation (<0.25) for ITU-T G.729 and G.711 speech CODECs.

Keywords: VoIP; Perceptual evaluation of voice quality; Voicing feature importance; Packet loss modeling

[8] F. Kuipers, R. Kooij, D. Vleeschauwer, and K. Brunnström. Techniques for measuring quality of experience. In *Wired/Wireless Internet Communications*, volume 6074 of *Lecture Notes in Computer Science*, pages 216-227. Springer Berlin Heidelberg, 2010. [bib | DOI]

Quality of Experience (QoE) relates to how users perceive the quality of an application. To capture such a subjective measure, either by subjective tests or via objective tools, is an art on its own. Given the importance of measuring users' satisfaction to service providers, research on QoE took flight in recent years. In this paper we present an overview of various techniques for measuring QoE, thereby mostly focusing on freely available tools and methodologies.

[9] K. Mitra, C. Ahlund, and A. Zaslavsky. Performance evaluation of a decision-theoretic approach for quality of experience measurement in mobile and pervasive computing scenarios. In *Wireless Communications and Networking Conference (WCNC)*, pages 2418-2423, 2012. [bib | DOI]

Measuring and predicting users quality of experience (QoE) in dynamic network conditions is a challenging task. This paper presents results related to a decision-theoretic methodology incorporating Bayesian networks (BNs) and utility theory for quality of experience (QoE) measurement and prediction in mobile computing scenarios. In particular, we show how both generative and discriminative BNs can be used to measure and predict users QoE accurately for voice applications under several wireless network conditions such as wireless signal fading, vertical handoffs, wireless network congestion and normal hotspot traffic. Through extensive simulation studies and results analysis, we show that our proposed methodology can achieve an average accuracy of 98.70% using three different types of Bayesian network.

Keywords: belief networks;computer network performance evaluation;decision theory;fading channels;mobile computing;mobility management (mobile radio);telecommunication traffic;Bayesian network;decision theoretic methodology;dynamic network condition;mobile computing;normal hotspot traffic;performance evaluation;pervasive computing;quality of experience measurement;user quality of experience;utility theory;vertical handoffs;voice applications;wireless network condition;wireless network congestion;wireless signal fading;Accuracy;Bayesian methods;Codecs;Delay;Fading;Mobile communication;Mobile computing

[10] S. Möller, W.-Y. Chan, N. Côté, T. Falk, A. Raake, and M. Wältermann. Speech quality estimation: Models and trends. *IEEE Signal Processing Magazine*, 28(6):18-28, 2011. [bib | DOI]

This article presents a tutorial overview of models for estimating the quality experienced by users of speech transmission and communication services. Such models can be classified as either parametric or signal based. Signal-based models use input speech signals measured at the electrical or acoustic interfaces of the transmission channel. Parametric models, on the other hand, depend on signal and system parameters estimated during network planning or at run time. This tutorial describes the underlying principles as well as advantages and limitations of existing models. It also presents new developments, thus serving as a guide to an appropriate usage of the multitude of current and emerging speech quality models.

Keywords: speech processing;telecommunication network planning;voice communication;acoustic interface;network planning;speech communication service;speech quality model estimation;speech signals measurement;speech transmission service;transmission channel;Quality assessment;Quality of service;Speech processing;Tutorials

[11] P. Paglierani and D. Petri. Uncertainty evaluation of objective speech quality measurement in VoIP systems. *IEEE Transactions on Instrumentation and Measurement*, 58(1):46-51, 2009. [bib | DOI]

This paper deals with the measurement uncertainty of the speech quality (SQ) achievable by an actual voice-over-Internet-protocol (VoIP) telephony network. The accuracy of end-to-end speech data provided by the perceptual estimation of SQ (PESQ) algorithm suggested in the International Telecommunication Union (ITU) recommendation P.862 is discussed. Then, the uncertainty of the PESQ results under different measurement conditions and real-life VoIP equipment (media gateway) is analyzed. This problem, in fact, has received very little attention in the literature, although many results related to other kinds of PESQ applications are available. Meaningful experimental data are reported and discussed mainly by means of statistical tools.

Keywords: Internet telephony;voice communication;International Telecommunication Union recommendation P.862;VoIP system;end-to-end speech data;media gateway;objective speech quality measurement;perceptual estimation algorithm;telephony network;uncertainty evaluation;voice-over-Internet-protocol;Communication system performance;Voice over Internet Protocol (VoIP);mean opinion score (MOS);network testing;speech codecs;speech quality (SQ)

[12] F. Rahdari and M. Eftekhari. Using bayesian classifiers for estimating quality of VoIP. In *International Symposium on Artificial Intelligence and Signal Processing (AISP)*, pages 348-353, 2012. [bib | DOI]

In real-time multi-media services, that uses internet infrastructure for transferring data traffics, the quality of service and consequently the level of user satisfaction are significant parameters. Our objective in this paper is to investigate the capability of Bayesian classifiers for estimating the quality of perceived voice in VoIP (Voice over IP) system. In this study, some quality parameters have been utilized to estimate the level of user satisfaction. The employed classifiers operate non-intrusively that means there is no need for original signal to estimate the quality of the perceived voice. For this purpose, a data set has been provided by simulation environment based on PESQ (that is an intrusive method that compares original and degraded signal for evaluating the quality of voice). Finally, we compare the performance of Bayesian classifiers with some other classification approaches in terms of estimating accuracy. For this purpose, the WEKA tool is used that contains implementation of many algorithms for classification problems. The results obtained, show the efficiency of Bayesian classifiers comparing to the other methods in terms of accuracy and computational time.

Keywords: Bayes methods;Internet telephony;computer network security;estimation theory;Bayesian classifiers;Internet infrastructure;VOIP estimating quality;Voice over IP system;WEKA tool;data traffics;intrusive method;multimedia services;quality parameters;simulation environment;user satisfaction;Bayesian methods;Codecs;IP networks;Predictive models;Speech;Bayesian classifier;PESQ;Voice Over IP;quality estimation;user satisfaction

[13] P. Reichl, S. Egger, R. Schatz, and A. D'Alconzo. The logarithmic nature of QoE and the role of the Weber-Fechner Law in QoE assessment. In *IEEE International Conference on Communications (ICC)*, pages 1-5, 2010. [bib | DOI]

The Weber-Fechner Law (WFL) is an important principle in psychophysics which describes the relationship be- tween the magnitude of a physical stimulus and its perceived intensity. With the sensory system of the human body, in many cases this dependency turns out to be of logarithmic nature. Re- cent quantitative QoE research shows that in several different scenarios a similar logarithmic relationship can be observed be- tween the size of a certain QoS parameter of the communication system and the resulting QoE on the user side as observed during appropriate user trials. In this paper, we discuss this surprising link in more detail. After a brief survey on the background of the WFL, we review its basic implications with respect to related work on QoE assessment for VoIP, most notably the recently published IQX hypothesis, before we present results of our own trials on QoE assessment for mobile broadband scenarios which confirm this dependency also for data services. Finally, we point out some conclusions and directions for further research.

Keywords: Internet telephony;mobile communication;quality of service;IQX hypothesis;QoE assessment;QoS parameter;VoIP;Weber-Fechner law;communication system;data services;human body sensory system;logarithmic nature;mobile broadband scenarios;quantitative QoE research;Communication industry;Communication systems;Communications Society;Delay effects;Delay systems;Humans;Joining processes;Psychology;Quality of service;Reliability theory

[14] R. Schatz, T. Hoßfeld, L. Janowski, and S. Egger. From packets to people: Quality of experience as a new measurement challenge. In E. Biersack, C. Callegari, and M. Matijasevic, editors, *Data Traffic Monitoring and Analysis*, volume 7754 of *Lecture Notes in Computer Science*, pages 219-263. Springer Berlin Heidelberg, 2013. [bib | DOI]

Over the course of the last decade, the concept of Quality of Experience (QoE) has gained strong momentum, both from an academic research and an industry perspective. Being linked very closely to the subjective perception of the end user, QoE is supposed to enable a broader, more holistic understanding of the qualitative performance of networked communication systems and thus to complement the traditional, more technology-centric Quality of Service (QoS) perspective.

The purpose of this chapter is twofold: firstly, it introduces the reader to QoE by discussing the origins and the evolution of the concept. Secondly, it provides an overview of the current state of the art of QoE research, with focus on work that particularly addresses QoE as a measurement challenge on the technology as well as on the end-user level. This is achieved by surveying the different streams of QoE research that have emerged in the context of Video, Voice and Web services with respect to the following aspects: fundamental relationships and perceptual principles, QoE assessment, modeling and monitoring.

[15] H. P. Singh, S. Singh, S. Khan, and J. Singh. VoIP: State of art for global connectivity-A critical review. *Journal of Network and Computer Applications*, 2013. [bib | DOI]

The Internet has revolutionized the telecommunication systems by supporting new applications and services. Voice over Internet Protocol (VoIP) is one of the most prominent telecommunication services based on the Internet Protocol (IP). The signal quality of the VoIP system depends on several factors such as networking conditions, coding processes, speech content and error correction schemes. The work in the present paper reviewed these issues, used for providing toll-quality communication service to the users over VoIP system. From the very beginning of transferring the voice data over packet switched networks, the journey of the packet based communications to modern VoIP and advancements to improve the service of the VoIP system has been summarized in this work.

Keywords: VoIP, IP telephony, Speech coder, Voice quality, Digital signal processing (DSP)

[16] H. P. Singh, S. Singh, R. Sarin, and J. Singh. Evaluating the perceived voice quality on VoIP network using interpolated FIR filter algorithm. *International Journal of Electronics*,

99(10):1449-1469, 2012. [bib | DOI]

Voice over Internet Protocol (VoIP) is a popular communication service nowadays. VoIP reduces the cost of call transmission by passing voice and video packets through the available bandwidth for data packets through Internet protocol. The quality of the VoIP signal is degraded due to the various network impairments. The proposed scheme, interpolated finite impulse response, is implemented as post-processor after decoding the signal in VoIP system. The performance of the proposed scheme is evaluated for various network conditions. The results of the proposed scheme are measured with the objective measurement methods for signal quality evaluation. The performance of the proposed system is compared with the existing techniques for quality improvement in VoIP system. The results show much improvement in speech quality with the proposed scheme in comparison to other similar schemes.

[17] L. Sun and E. Ifeachor. Voice quality prediction models and their application in VoIP networks. *IEEE Transactions on Multimedia*, 8(4):809-820, 2006. [bib | DOI]

The primary aim of this paper is to present new models for objective, nonintrusive, prediction of voice quality for IP networks and to illustrate their application to voice quality monitoring and playout buffer control in VoIP networks. The contributions of the paper are threefold. First, we present a new methodology for developing perceptually accurate models for nonintrusive prediction of voice quality which avoids time-consuming subjective tests. The methodology is generic and as such it has wide applicability in multimedia applications. Second, based on the new methodology, we present efficient regression models for predicting conversational voice quality nonintrusively for four modern codecs (G.729, G.723.1, AMR and iLBC). Third, we illustrate the usefulness of the models in two main applications - voice quality prediction for real Internet VoIP traces and perceived quality-driven playout buffer optimization. For voice quality prediction, the results show that the models have accuracy close to the combined ITU PESQ/E-model method using real Internet traces (correlation coefficient over 0.98). For playout buffer optimization, the proposed buffer algorithm provides an optimum voice quality when compared to five other buffer algorithms for all the traces considered

Keywords: IP networks;Internet telephony;regression analysis;Internet;VoIP networks;conversational speech quality;playout buffer control;regression models;voice quality monitoring;voice quality prediction models;voice-over-IP;Communication system control;Intelligent networks;Internet telephony;Monitoring;Predictive models;Quality of service;Speech analysis;Sun;Telecommunication traffic;Testing;Conversational speech quality;E-model;jitter buffer optimization;nonintrusive;perceptual evaluation of speech quality (PESQ);regression model;voice over IP;voice quality prediction

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