CAPACITY BUILDING

QUALITY OF SERVICE TRAINING PROGRAMME (QoSTP)

Report

ITU Academy
empowering minds

Telecommunication Development Sector

Quality of service training programme (QoSTP)

December 2014

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This report was prepared by the Human Capacity Building Division within the Projects Support and Knowledge Management Department (PKM) of the International Telecommunication Union (ITU) Telecommunication Development Bureau (BDT). Substantive inputs to the report were provided by Dr Jan Holub.

Please consider the environment before printing this report.

Foreword

I am delighted to present the comprehensive report on our quality of service training programme developed under the framework of the ITU Academy.

This report outlines the study modules of the programme, provides an overview of each module, and identifies target audience and course duration. Further, it recommends various combinations and progression paths within the proposed study modules, leading to various levels of professional certification.

The information contained in this report will be of utmost importance to those who would like to work with ITU in the delivery and accreditation of this training product and establish partner relationships.

This report can be accessed at https://academy.itu.int/component/k2/item/1555-qostp

I hope that you will find this report useful and informative.

Brahima Sanou

Director

Telecommunication Development Bureau

Executive summary

This report puts forward a conceptual framework for a high-quality course comprising a set of study modules for a Quality of Service Training Programme (QoSTP). Once developed, QoSTP will offer the ITU membership a means for training staff in the theory and practice of quality of service (QoS) planning, monitoring and analysis in various areas of telecommunications.

The framework covers the development of a comprehensive QoSTP, including educational materials and resources for face-to-face training and for active-distance and self-paced learning. Importantly, the report recommends various combinations and progression paths within the proposed study modules towards different levels of professional certification.

QoSTP can create a unique and credible course, complementing existing professional QoS training options and promoting the harmonization of QoS practices. An essential differentiator for QoSTP will be formalized assessment of learning outcomes in terms of achieved professional skills and qualifications, supported by the award of a degree attesting to a recognized academic level, e.g. Master of Science.

The following main options are considered:

- **Option A**: QoSTP becomes a professional training course organized solely by the ITU Academy and delivered through training facilities and/or the ITU Academy platform under the aegis of the centres of excellence, possibly being driven by ITU staff, ITU-T study group experts and/or ITU-D experts, and with the possible cooperation of ITU-R industry partners or centre of excellence partners. Upon completion of the training (including the passing of all assessment tests/exams), trainees would receive a professional "Master of Quality of Service" certificate, awarded in the name of ITU/the ITU Academy.
- **Option B**: QoSTP becomes an international Master's course offered by a consortium of participating universities but organized in collaboration with ITU-D/ITU-T external experts (e.g. lecturing on actual case studies) and ITU-T industry partners, especially for practice and laboratory exercises, under the aegis of the ITU Academy. Such a course would result in the award of a Master's degree to successful trainees.
- **Option C**: QoSTP becomes an international course offered by a consortium of participating universities, under the aegis of the ITU Academy, as a part of regular university studies leading to a Master's degree.
- **Option D**: QoSTP becomes a guide for self-study and training through practical experience, in which case ITU/the ITU Academy would develop the training materials, offer student consultation opportunities and ultimately assume responsibility for the professional examination to determine whether the knowledge and skills acquired meet the requisite standard. For the latter task, ITU would need to develop and administer the final assessment tests for each module.

The justification for, and advantages and drawbacks of, the different options are described in the report, along with the proposed structure, content and duration of study modules.

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1 QoS training and awareness as an important aspect of contemporary telecommunication

In the early days of telecommunications, it used to be a privilege to have access to a telecommunication service and quality of service (QoS) was at a stable level, pre-determined by well-advanced planning together with the available technology. With the migration to packet-based technologies and introduction of social media, all this has changed dramatically. Under slogans such as "Better living with ICT", we are now seeing technologies that improve people's lives and environment, and digital living for the benefit of society and individuals.

Information and communication technologies (ICTs) assist us in many areas of our lives. The large-scale deployment of communication technologies has produced major changes in the way we communicate for social and business purposes. However, most of these deployments have been technology-led, without any prior assessment of social consequences, i.e. of the social and cultural contexts of the end users.

The ultimate goal is to improve the quality of life for all, and in this context it is important to address the user experience in a variety of ways, through “quality of service”, which refers to those characteristics of a telecommunication service that reflect its ability to satisfy the stated and implied needs of the service user, and “quality of experience”, which has to do with the overall acceptability of an application or service, as perceived subjectively by the end user.

With the rapid development of modern telecommunication services and emergence of new technologies, there is an increasing need to disseminate firm and robust QoS understanding to all stakeholders, such as operators, equipment manufacturers, administrations, regulatory bodies, end users and the latter’s representatives. Telecommunication services are an essential part of economies, business and social life, and inevitably come with user demands for QoS. Those demands relate to many factors, including not only technical measures and technology development but also user type and past experience, which is influenced by geographic location and time, etc. In cases where QoS is neglected or misunderstood, the economies, business and social life in question are liable to be severely compromised.

QoS is mentioned in several articles of the International Telecommunication Regulations (ITRs). For example:

Art. 3.1 – Members shall ensure that administrations cooperate in the establishment, operation and maintenance of the international network to provide a satisfactory quality of service.

Art. 3.4 – Subject to national law, any user, by having access to the international network established by an administration, has the right to send traffic. A satisfactory quality of service should be maintained to the greatest extent practicable, corresponding to relevant CCITT Recommendations.

Art. 4.3 – Subject to national law, Members shall endeavour to ensure that administrations provide and maintain, to the greatest extent practicable, a minimum quality of service corresponding to the relevant CCITT Recommendations with respect to:

a) access to the international network by users using terminals which are permitted to be connected to the network and which do not cause harm to technical facilities and personnel;

b) international telecommunication facilities and services available to customers for their dedicated use;

c) at least a form of telecommunication which is reasonably accessible to the public, including those who may not be subscribers to a specific telecommunication service; and

d) a capability for interworking between different services, as appropriate, to facilitate international communications.

It is for all these reasons that QoS training programmes are to be offered by ITU. They should be developed around the Recommendations and other ITU documents on QoS, which manifest the Union core competence in that field. As new telecommunication technologies emerge over time and the corresponding ITU-T Recommendations are elaborated, it might become desirable to revisit the present framework after a certain period of time.

2 Overview of existing QoS study resources and courses

Currently available QoS training options are reviewed in this section. The purpose of this analysis is to identify features which could be useful in designing QoSTP.

WRITTEN SOURCES

Written texts (online and offline) are a primary resource for any educational endeavour. Judiciously selected and (if necessary) adapted, they can enhance the content of QoSTP and be used in self-paced studies. With that in mind, the following sources, each with its own advantages and drawbacks, can be identified:

ITU-T RECOMMENDATIONS

The G- and P-series of ITU-T Recommendations are particularly relevant to QoS topics. QoSTP is based on the systematic study of chapters drawn up on the basis of those Recommendations, which are therefore set to play an essential part in establishing the scope and content of QoSTP.

ITU-T HANDBOOKS

ITU-T has developed two handbooks dealing in particular with QoS topics[1]. Also worth mentioning here is the ITU-T Handbook on Telephonometry. These handbooks are valuable sources of knowledge, both theoretical and practical, relating to QoS measurements and performance optimization. However, they are not a substitute for in-class learning, nor are they necessarily appropriate for use as course textbooks, a purpose for which they were not designed. They were intended to be reference sources, better suited to the occasional search for specific answers than as a source for the continuous presentation of didactic material unfamiliar to the students. Nevertheless, the ITU-T handbooks contain details of many practices that are not directly included in standards and which can be useful to students pursuing self-paced studies.

BOOKS PUBLISHED OUTSIDE ITU-T AND ITU-R

The literature of QoS is expanding but is focused primarily on voice QoS in packet networks[2]. The problem with most such books is that, like the ITU-T handbooks, they were not conceived as academic course textbooks but as reference sources for professionals. The most recent books tend to focus on a specific aspect, such as wireless or packet-based networks.

Such sources could and should be used as reference material when creating QoSTP content, and could also be used for some deeper studies, but they are unlikely to be suitable as course books. Their use is also problematic owing to their high purchase cost and limited availability in public libraries, which may make them difficult to obtain.

QoS COURSES

Most professionally-oriented QoS courses at academic institutions are designed for on-the-job training, and this has some advantages and drawbacks.

[1] ITU-T Handbook on Quality of Service and Network Performance, 2004. ITU-T Handbook on Practical Procedures for Subjective Testing , 2011

[2] E.g: Raake, A: *Speech Quality of VoIP: Assessment and Prediction*, Wiley, January 2007, ISBN: 978-0-470-03299-2. Marchese, M.: *QoS Over Heterogeneous Networks*, Wiley, 2007, ISBN 978-0-470-01752-4. Hardy, W.: *VoIP Service Quality: Measuring and Evaluating Packet-Switched Voice*, McGraw-Hill 2003, ISBN 978-0-071-41076-2. Wallace, K.: *Implementing Cisco Unified Communications Voice over IP and QoS* (CVOICE), CCIE 7945, ciscopress.com

ITU SEMINARS AND CoEs

There are activities for training by national SDOs, such as the German Association for Electrical, Electronic & Information Technologies (VDE), usually arranged in cooperation with local academic institutions (universities). BR and BDT, either directly or through ITU regional CoE partners, organize seminars and workshops aimed at disseminating knowledge of best practices, policies and processes in QoS. The most notable examples are QoS seminars[3], the majority of which have the advantage of being tailored to the needs and realities of a specific region. There are also benefits in addressing a narrow set of issues seen as topical at the time.

ITU workshops disseminate QoS know-how in a timely manner and promote regional discussion of topical challenges; however, occasional events cannot take the place of a comprehensive and formalized QoS training programme. Nor, in most cases, do such workshops include any evaluation of the effectiveness with which the knowledge was transferred and retained, despite the fact that a certificate of participation is usually awarded. The mere fact of participation cannot be taken as a measure of the level of professional qualification attained. The new course must therefore cover a full range of topics and have formal evaluation and certification procedures at its conclusion in order to differentiate QoSTP from existing ITU seminars and workshops.

COMMERCIAL QoS COURSES

Cisco Networking Academy

The Cisco Networking Academy[4], established in 1997, partners with a range of organizations worldwide to provide an e-learning environment catering for digital learners and their training requirements. Through a combined learning model using contemporary networking technologies, the Networking Academy combines in-classroom learning with curricula, assessments and tools to help students develop the knowledge and skills they require to be part of an effective workforce. Since its establishment, four million students have participated in Networking Academy courses at 10 000 academies.

Academies are located in high schools, secondary schools, colleges, technical schools, universities and community-based organizations in 165 countries. The programme leverages public-private partnerships with governments, academic institutions, NGOs and non-profit organizations to help students from different socio-economic backgrounds to develop the ICT and personal skills needed for entry-level career opportunities, continuing education and globally-recognized career certification.

Assessment in the Cisco Networking Academy

The Networking Academy programme includes instructor-led classroom sessions, web-based course content, interactive learning activities and tools, online assessments, hands-on labs with real equipment, and simulations. Online courses include assessment features whereby the programme can track student performance and collect data. Students receive personalized online feedback that enables them to monitor their progress through formative and summative assessments integrated throughout the curricula. The cloud-based, multilingual assessment infrastructure developed by the Networking Academy likewise provides immediate feedback that students can use to monitor their progress and learn from their mistakes, while generating automated data to help teachers evaluate students' knowledge and skills. Globally-consistent online assessments also allow instructors to compare their students' progress with similar Networking Academy classes around the world. The results help instructors to address individual learning needs in a timely manner, and course designers to improve the effectiveness of the curricula. The scalable and global online system delivers an average of one million student assessments each month. Depending on the objective and on the type of material being taught, assessment tasks range from true/false and

[3] www.itu.int/en/ITU-T/Workshops-and-Seminars/qos/201309/Pages/default.aspx
www.itu.int/en/ITU-T/Workshops-and-Seminars/qos/201307/Pages/default.aspx
www.itu.int/en/ITU-T/Workshops-and-Seminars/qos/201207/Pages/default.aspx
www.itu.int/ITU-T/worksem/qos/201007/index.html

[4] Cisco Networking Academy, www.cisco.com/web/learning/netacad/index.html

multiple-choice questions to complex troubleshooting activities on a simulated network, where answers may vary as a student's problem-solving and analytical thinking skills are tested. In some cases, assessment activities are provided solely to help students review their progress. Other assessments allow students to practice for hands-on summative course exams and industry certification exams.

The Cisco Networking Academy is only partially focused on QoS and most of this part is still geared towards packet-based network QoS, leaving relatively little space for voice QoS. Another clear disadvantage is that the content of the Cisco course curriculum is inevitably biased towards Cisco technology, policies and interests. However, it remains a great example of complex online and classroom tuition, and the Cisco Networking Academy already has an established reputation for course quality and reliable delivery of professional training, with a certificate that is highly recognized by professional community.

A small number of dedicated QoS courses are offered by private companies. These can be divided into three basic categories:

a) QoS courses focusing on general telecommunication training[5]. Typically, these are instructor-led courses lasting from one day to one week. As the companies in question are not dealing with QoS aspects on a daily basis, the training cannot reflect the very latest, including as yet unpublished, details of ongoing standardization and regulation matters. In addition, the daily fee of around EUR 800-1 500 is quite high and presents a barrier to many potential students, although this level of cost might be affordable for some committed employers and institutions.

b) Another option is QoS courses offered by manufacturers of switching and monitoring equipment[6]. To a certain extent, the Cisco Networking Academy also fits in here. Unlike in the previous category, such training is usually conducted by skilled experts from the QoS field. However, the obvious drawback is the strong focus on aspects related solely to the company current technology – for example, QoS algorithms that are implemented in the system are discussed in a positive light and in detail, while algorithms that are not currently implemented or expected to be implemented in the near future are mentioned only in passing, if at all.

c) A few companies with a strict focus on QoS offer highly specialized QoS courses[7] delivered by experts who are immersed in the topic at the theoretical and practical levels on a daily basis and who also have a solid training background. This approach combines the advantages of both previous categories while avoiding some of their drawbacks. However, the price of such courses tends to be even higher to enable the training companies to remain truly independent.

Another clear disadvantage of all privately-run QoS courses is that they concentrate a large volume of information into a short time-frame, thereby seriously challenging the student's ability to grasp and retain the intricacies of the material presented. There is no formal evaluation at the end of the course, and a mere certificate of participation is no guarantee that the desired level of proficiency has been attained.

In conclusion, the distinguishing features of QoSTP as compared to commercial QoS courses would be a more thorough and in-depth presentation of the material, perhaps interspersed with practical exercises to deepen students' understanding, with a formal examination and evaluation at the end of the course to ascertain the degree to which the requisite level of professional knowledge has been attained. The fees charged for commercial course offerings can be used as a benchmark for QoSTP pricing, where applicable.

[5] www.wraycastle.com/course/telecoms-training/ip-networks-and-protocols-training/ip-quality-of-service-161
www.technology-training.co.uk/qosinipandmplsnetworks_28.php
www.telecomsacademy.com/

[6] www.telecomsacademy.com/
www.opticom.de/products/services.php

[7] www.mesaqin.com/training.html

NATIONAL QoS TRAINING

In response to the limited choice of QoS training options and drawbacks of existing courses, some national administrations have been pursuing their own initiatives, with some of the larger ones organizing QoS classes to introduce students to the working practices in their own countries against the backdrop of international QoS. Other administrations conclude agreements with private companies to provide their staff with tailored, in-house classes similar to the example discussed earlier. In such cases, the administration may seek a discount on the tuition fee in view of the large class size, and will also save on expenses since the travel and accommodation costs of a few foreign tutors amount to less than the cost of sending a large group of students abroad.

Another type of QoS-related initiative with educational potential is seen in conferences and seminars dedicated to QoS, such as:

- the ETSI Workshops organized by ETSI STQ.
- IEEE ComSoc GLOBECOM 2003-2014 (partially).
- International Workshops on Perceptual Quality of Systems (PQS), 2003, 2006, 2010 and 2013, organized by the Telecommunications Research Center Vienna (FTW), the Technical University of Berlin and Dresden University of Technology.
- MESAQIN 2002-2011, organized by the Czech Technical University and MESAQIN.com Ltd. Consideration could be given to having these as partners of ITU, contributing to, among other things, the development of modules, either as writers or reviewers.

In terms of their design and composition, such seminars are similar to the ITU seminars described above, but by virtue of being organized independently, the scope and content of the presentations can be tailored very precisely to the audience’s needs. Given their occasional nature and precisely tailored, localized scope, the existing national initiatives do not overlap with the idea of a formalized international QoS course.

In conclusion, to complement and differentiate the above-mentioned offerings, QoSTP should be thoroughly international in terms of its curriculum, course content, faculty and tutoring partners. It should provide advanced, state-of-the-art knowledge and aspire to raising the professional level of students from developed and developing countries alike.

3 Certificate recognition

There are various options for the certificate that successful participants will receive at the end of the course and for the manner in which that certificate is recognized in the industrial and academic spheres. This is discussed in the following paragraphs.

ITU certification

The first option is to award a final certificate at the ITU Academy level. As ITU is a globally recognized standardization organization, this option should be suitable in most cases, being efficient from both the cost and organizational standpoints. No academic institutions or other partners are needed for this type of certification, which could prove highly fruitful, for example for purely online courses.

Additional industrial professional certification

The second option is to award a professionally certified and industry recognized diploma from non-ITU, non-academic institutions. There are several organizations with which cooperation can be established, including but not limited to the following:

INSTITUTION OF ENGINEERING AND TECHNOLOGY (IET) – UK BASED

The IET can provide either direct participation in the arrangement and organization of QoS courses, or IET endorsement can be negotiated, with such endorsement, provided in cooperation with the University of Surrey, serving to equip course providers with a professional code of practice whereby compliance with predefined guidelines is checked as part of the ongoing assessment. Feedback is also sought from course/event attendees and used for monitoring purposes. [8]

INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS (IEEE) – US BASED

One illustrative example of the IEEE certified examination programme[9] is its Wireless Communication Engineering Technologies Certification Programme (WCET), which is primarily a guide to the examination process, outlining the subjects of the certification exam. It also describes the knowledge required for each topic, how the exam is organized and how the certificates are awarded. Since this process is organized and run by the IEEE Communications Society branch (IEEE ComSoc), credibility and international recognition for the certification were achieved relatively quickly. Some larger employers, primarily in the United States, already require WCET certification as a precondition for hiring their senior wireless engineers. Consideration could be given to use of the WCET professional certification model, which should be easier to set up and administer by comparison with the task of organizing university degree certification or creating a longer and more comprehensive course than those offered by commercial firms.

University-certified commercial professional courses

Professional courses might be certified as part of a university postgraduate diploma based on cooperation with a duly accredited university. The first option here is to deploy one of several established approaches that are already being used by other training entities. Good examples are Informa Telecoms Academy and MEAD Education.

TELECOMS ACADEMY, A UNIT OF INFORMA TELECOMS & MEDIA GROUP – UK BASED

The Informa Telecoms Academy[10] offers several tracks leading to university certification, thanks to a partnership with the University of Derby corporate division. Students may choose among different combinations of physical classes (each up to one week in duration), followed by a work practice assignment or a complete remote training course lasting nine months and comprising a series of nine modules (some of them prerequisites, some voluntarily chosen), with formal assessment upon completion. Students successfully completing these tracks may receive a Postgraduate Certificate in Advanced Telecoms Management, which constitutes 60 English university credits at Level 7 (corresponding to the first year of a Master's level university course). For comparison, 180 credits are needed to receive a full Master's university degree in England. In other words, someone having received the Informa Telecoms Academy certificate and then wishing to obtain a full Master of Sciences qualification could continue earning credits at Derby (or another) university until reaching the required number of credits for a Master's degree.

The advertised fees for various courses at the Informa Telecoms Academy suggest that it would cost around EUR 8 000 to achieve the 60-credit university certification. In conclusion, the Informa Telecoms Academy represents an option whereby a professional course might be certified as part of a University postgraduate diploma. This could be considered as an option for QoSTP as well, depending on course size, duration and negotiated agreements with one or more partner universities.

MEAD EDUCATION – SWISS BASED

MEAD Education[11] organizes public discussions, case studies and lectures by reputable experts on the "Challenges of the 21st Century" in order to promote dialogue and contribute to offering decision-makers practical tools to further strengthen their decision-making ability. It cooperates with EPFL Lausanne in

[8] www.theiet.org/membership/career/index.cfm

[9] www.ieee.org/education_careers/education/accreditation/ctaa/ieee_role.html

[10] www.telecomsacademy.com/

[11] http://Mead.ch/mead

Switzerland, TU Delft in the Netherlands and UC Santa Cruz in California. Such a platform appears suitable for the hosting of ITU-T courses. In view of the commercial basis, successful students are awarded an industrial certificate.

University-certified academic courses

Many European countries have chosen to follow the European Credit Transfer and Accumulation System (ECTS)[12], originally set up in 1989 to facilitate the recognition of study periods undertaken abroad by mobile students, since when it has become a widely recognized and unified system for evaluating the different levels of university learning or other types of ongoing education, both formal and informal.

A key feature of ECTS is that it is learner based, i.e. the credits correspond to the average time it takes the learner/student to achieve certain learning objectives. Thus, one ECTS credit corresponds to 25-30 hours of learning (the precise value depending on the country), it being assumed that students taking a full-time formal course of education should be able to accumulate 60 ECTS credits in a typical academic year consisting of two semesters of three to four months' duration. On that basis, the first-level degree (Bachelor of Science) would normally require the accumulation of 180-240 ECTS credits (representing three to four years of full-time study), while the second-level degree (Master of Science) would require 90-120 ECTS credits (18 months to two years of full-time study). There is also provision for a second-level degree to have an interim level with at least 60 ECTS credits. This might be used in relation to the kind of postgraduate diploma in a specialized field that is especially popular in the United Kingdom, and which in other countries has titles such as Licentiate, Polytechnic Diploma or Master of Arts. The main distinguishing feature of a full Master's degree as opposed to a postgraduate diploma is, in many cases, that it is more theoretical and research oriented and requires the writing of a thesis. The postgraduate diploma, for its part, is seen purely as a diploma in applied sciences, where a thesis is not always required. As the presentation of a thesis usually represents 30 ECTS credits, this could make all the difference between 60 and 90 ECTS credits, which is the minimum required for a formal Master of Science degree according to ECTS principles.

Overall, given the wide adoption of ECTS as a reference system in Europe and beyond, it seems logical to consider using this system when designing the QoSTP programme. The only notable exception to ECTS in Europe is England, which uses a system called CATS. However, this system is readily convertible to ECTS, with one ECTS credit corresponding to two English CATS university credits.

An additional advantage of using a European system is that it also constitutes a good example of derivative international teaching programmes by university consortia, known as European Erasmus Mundus Master Courses[13]. Erasmus Mundus programmes offer international education options in many specialized subjects, so it might be fitting for an international Master's course on QoS to join this grouping. Most importantly, the Erasmus Mundus programmes and their students are eligible for some European funding for programme organization and student scholarships. This funding is also available to students from non-EU countries.

To sum up, then, the international Master's programmes offered today by university and industry consortia tend to follow the format of two academic years in full-time study, earning 120 ECTS credits. In general, it seems that the existing European Erasmus Mundus framework could be a good model for the creation of a university consortium to run an international Master's programme such as QoSTP.

Regarding the eventual academic composition, ECTS load and duration of the QoSTP programme, this would clearly be a subject for negotiation with the participating universities. However, it may be suggested that ITU could also offer a course of shorter duration, which would be more suitable for on-the-job training. It has to be borne in mind that any course lasting longer than one calendar year might be perceived as too lengthy and distracting to a career-oriented professional.

[12] http://ec.europa.eu/education/lifelong-learning-policy/doc/ects/guide_en.pdf

[13] http://eacea.ec.europa.eu/erasmus_mundus/results_compendia/selected_projects_action_1_master_courses_en.php

Overall conclusions from the analysis of existing training and certification options

QoSTP can successfully create a unique and credible platform, complementing the existing professional QoS training options, if it provides the distinguishing attributes set out below.

The programme should be broad in scope and international in character, with tutors and participating institutions of the highest quality, to ensure both its value and the reputation of its diploma. It should be comprehensive and well-rounded, addressing all the theoretical and practical aspects of QoS. It should offer a higher level of professional knowledge than is available from existing self-study resources or short courses. An essential differentiator for QoSTP would be the formalized assessment of learning outcomes in terms of the professional skills and qualifications attained, as certified primarily at the ITU Academy level.

Another alternative would be to consider it an independent professional (non-academic) qualification in the field of QoS, like the one offered by WCET under the auspices of the IEEE Communications Society. Consideration could be given to the elaboration or sponsorship of a professional certification programme with the ITU certificate, which would be recognized by administrations and the telecommunication industry without having ties to the university world or academic degrees.

Yet another alternative could be one whereby the awarded certificate represents partial completion of a university course of study with a certain number of transferrable credits that would count towards a full Master's qualification at a suitable institution of higher education (university or college). As the process of including QoSTP in the accredited university educational programme can take a significant amount of time (counted in years), the most fruitful approach may be to start with industrial courses provided at a CoE collocated with the university, and to work towards the course accreditation as a part of regular Master's telecommunication courses.

4 Module organization, matrix of study tracks

The definition of the modules and their organization into study tracks based on the required specialization and level (basic, medium or advanced) of immersion is proposed below. Obligatory modules (OM) contain essential and overview knowledge relevant to the given study track, while elective modules (EM) are options designed to deepen the student's knowledge in specific areas of QoS.

The QoSTP modules are defined as the following set of OMs and EMs:

- OM1: Introduction - QoS and QoE (quality of experience)
- OM2: Subjective assessment of voice quality
- OM3: Objective assessment of voice quality
- OM4: QoS and QoE for multimedia and assessment methods
- EM1:
  - EM 1.1: Telephonometry
  - EM 1.2: Network performance and operation, administration and maintenance (OAM) for performance measurement
- EM2:
  - EM 2.1: Hands-free communication and user interfaces in vehicles
  - EM 2.2: Traffic management
- EM3:
  - EM 3.1: QoS for mobile services
  - EM 3.2: Bit-rate measurement of Internet connections

Such a structure allows for the achievement of different levels of immersion and different final specializations. The basic level is simply an overview of the entire topic of QoS and QoE and may serve as an optional or complementary course for students of other specializations (not directly focusing on QoS and QoE), while the medium level may be considered as corresponding to a Bachelor's degree, and the advanced level as corresponding to a Master's degree. For QoSTP, the final project/thesis constitutes the main difference between the medium and advanced levels, as can be seen from the matrix and list in Table 1.

Table 1: QoSTP basic, medium, and advanced levels

Study track	OM 1	OM 2	OM 3	OM 4	EM 1.1	EM 1.2	EM 2.1	EM 2.2	EM 3.1	EM 3.2	Thesis/ Project	Level
QoS/QoE General overview												basic
QoS/QoE Specialist												medium
End-user equipment QoS Specialist												medium
Fixed network QoS Specialist												medium
Mobile network QoS Specialist												medium
Advanced end-user equipment QoS Specialist												advanced
Advanced fixed network QoS Specialist												advanced
Advanced mobile network QoS Specialist												advanced

- QoS/QoE General overview (basic level): OM1
- *QoS/QoE Specialist* (medium level): OM1-OM2-OM3-OM4
- *End-user equipment QoS Specialist* (medium level): OM1-OM2-OM3-OM4-EM1.1-EM2.1-EM3.1
- *Fixed network QoS Specialist* (medium level): OM1-OM2-OM3-OM4-EM1.2-EM2.2-EM3.2
- *Mobile network QoS Specialist* (medium level): OM1-OM2-OM3-OM4-EM1.2-EM2.2-EM3.1
- *Advanced end-user equipment QoS Specialist* (advanced level): OM1-OM2-OM3-OM4-EM1.1-EM2.1-EM3.1-final project/thesis
- *Advanced fixed network QoS Specialist* (advanced level): OM1-OM2-OM3-OM4-EM1.2-EM2.2-EM3.2-final project/thesis
- *Advanced mobile network QoS Specialist* (advanced level): OM1-OM2-OM3-OM4-EM1.2-EM2.2-EM3.1-final project/thesis

Estimated number of experts

The number of experts needed to develop the content, and time needed to produce that content for each module, are recommended in Table 2.

Table 2: Number of experts needed

Code	Chapter Title	Number of experts	Preparation (man-days)
OM1	Introduction - Quality of Service and Quality of Experience	1-2	10
OM2	Subjective assessment of voice quality	2-3	15
OM3	Objective assessment of voice quality	2-3	20
OM4	QoS and QoE for multimedia and assessment methods	1-2	15
EM1.1	Telephonometry	1	15
EM1.2	Network performance and OAM for performance measurement	1-2	20
EM2.1	Hands-free communication and user interfaces in vehicles	1	10
EM2.2	Traffic management	1-2	15
EM3.1	QoS for mobile services	2-3	20
EM3.2	Bit-rate measurement of Internet connections	1	10

5 Programme structure, module time-frames and delivery options

QoSTP concept and programme outline

Based on the conclusions of the section on analysis of the existing QoS courses available on the market, this report proposes the following general concept for designing the QoSTP programme:

- Maximum duration of one calendar year for students taking the complete course from the basic level through the advanced level to its completion.
- If formal academic certification is desired, such as a university Master of Science degree, QoSTP should be planned so as to offer the equivalent of 60-90 ECTS credits:
  - 25 ECTS credits for OM1-OM4, which may, alternatively, be granted for a minimum of one year of active professional experience, subject to formal assessment of the professional competencies acquired;
  - 20-35 ECTS credits for EM1, EM2 and EM3.

Credits are awarded (20-30 ECTS) for the final thesis in the chosen area of professional specialization. The requisite assessment tools should be developed by the subject-matter experts.

Assuming 25 hours of studies for each ECTS credit, completion of the QoSTP would correspond to 1 500-2 250 hours of studying and work experience. The ITU regional centres of excellence and partner universities will be called upon to play an active role in the provision of support for the organization and coordination of QoSTP activities.

Target audiences

QoSTP will be designed for anyone wishing to enhance their professional knowledge while working in the field of QoS, for example in a national regulatory authority (NRA), wireless communications company or equipment manufacturer. It will focus on the broadening of existing skills in the complex field of QoS, and will therefore be available to any professional having previously graduated with a first-level university degree (Bachelor of Science).

Students entering QoSTP may be from different institutional levels on either the technical or managerial side, and from different backgrounds (engineering, economic, legal, etc.).

Course duration options

Since QoSTP is expected to be most attractive as an on-the-job training option for working professionals, the course could consist of a sequence of modules spread over no more than one calendar year, with each study module lasting four to five weeks, including, at the end of each module, formal assessment of the knowledge acquired.

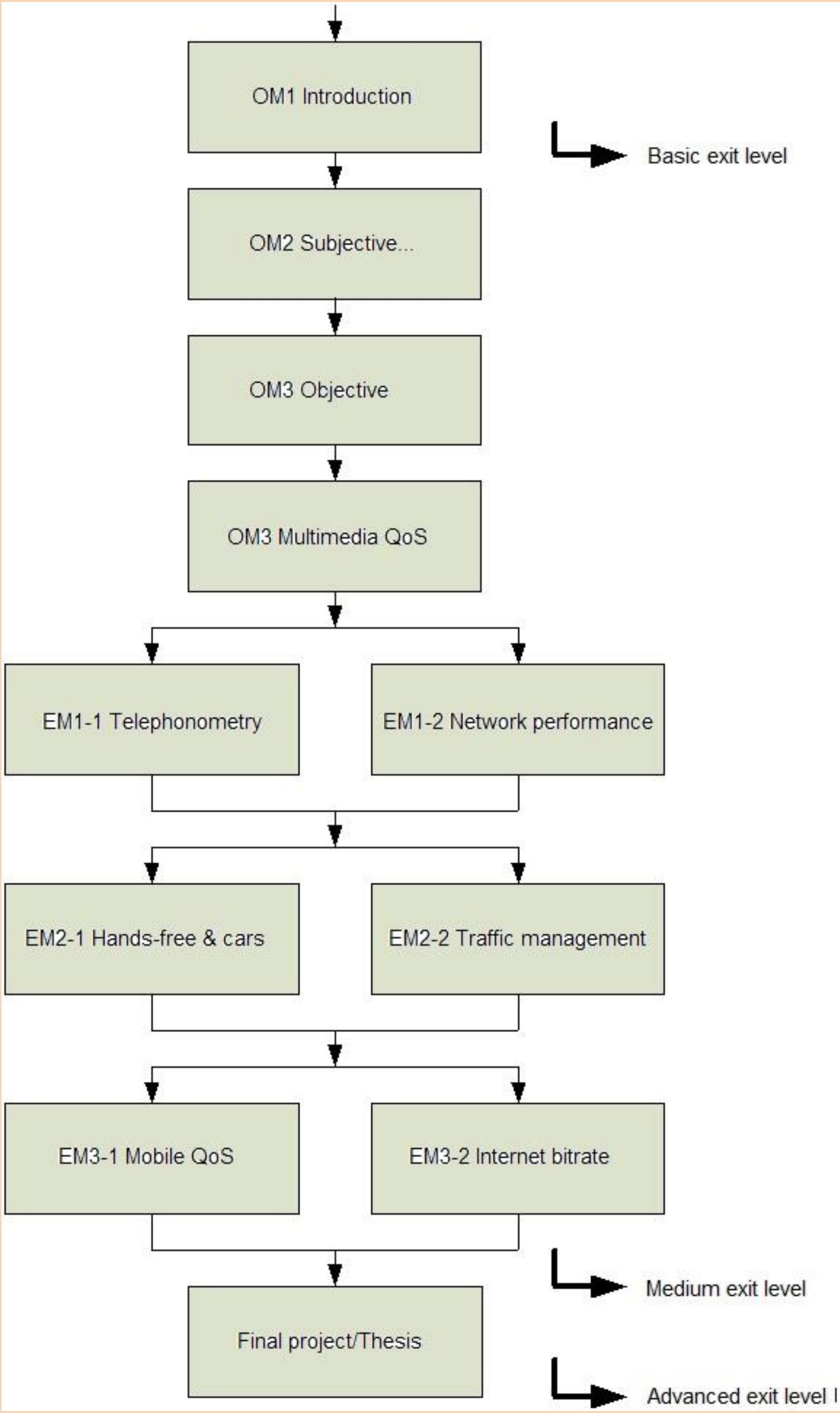
As a point of departure, and aiming for a course worth 60 to 90 ECTS, QoSTP could comprise ten training modules, each worth five to eight ECTS credits, complemented by a thesis worth 20 to 30 ECTS credits. To summarize this proposal, the overall course composition and different entry and exit points are illustrated in Figure 1.

It should be noted that the modular structure of QoSTP is well suited for constant updating of the modules initially proposed as well as for development of new specialized (elective) modules.

The overall QoSTP design concept proposed above would have the following benefits:

1. The sequential flow of modules would:
   a) be a convenient way to combine study options with daily work commitments, as students would need to focus on only one study topic at a time;
   b) make for simplification on the logistical side by minimizing the coordination needed between different partner institutions, with only the organization managing a particular study module being responsible for daily operations in any given time period;
   c) ensure predictability of time commitments for the teaching staff (assuming each tutor is assigned the same module in successive years and each module is offered annually in the same time-frame);
   d) have clear break-points allowing for different entry points; in the case of multiple sets of study paths and/or certification levels, the breaks would enable different combinations of entry and exit points. This option is discussed in more detail below.
2. A module duration of four to five weeks seems to strike a reasonable balance, being sufficiently short for focused study of one topic, but long enough to give instructors flexibility in planning their syllabuses. The said duration also allows for a succession of many different modules to fit within one year, with periods of self-directed study possibly coinciding with holidays, major industry events, work-related travel, etc.

Figure 1: Overall course composition



Source: ITU

3. Another notable feature of the above design is the deliberately changing proportion of obligatory and elective modules at different levels. This is proposed because the course including the basic entry level should be broad in scope and is hence formed by obligatory modules imparting knowledge in subject areas that must be familiar to any QoS professional. The advanced levels, on the other hand, will involve more specialized study and therefore have a higher proportion of elective modules.

Bearing in mind the above principles and observations, the QoSTP design concept recommended in this report would appear to provide a robust yet versatile course structure that can be easily adapted to the composition of the consortium of participating institutions and to any subsequent modification of the objectives. For instance, as was discussed earlier, different options can be chosen according to the various QoSTP certification objectives, necessitating a possible change in the course delivery format while maintaining the same overall design as described above. The following main options may be considered:

- **Option A**: QoSTP becomes a professional training course organized solely by the ITU Academy and delivered through training facilities and/or the ITU Academy platform under the aegis of the centres of excellence, possibly being driven by ITU staff, ITU-T study group experts and/or ITU-D experts, and with the possible cooperation of ITU-R industry partners or centre of excellence partners. Upon completion of the training (including the passing of all assessment tests/exams), trainees would receive a professional "Master of Quality of Service", certificate, awarded in the name of ITU/the ITU Academy.
- **Option B**: QoSTP becomes an international Master's course offered by a consortium of participating universities but organized in collaboration with ITU-D/ITU-T external experts (e.g. lecturing on actual case studies) and ITU-T industry partners, especially for practice and laboratory exercises, under the aegis of the ITU Academy. Such a course would result in the award of a Master's degree to successful trainees.
- **Option C**: QoSTP becomes an international course offered by a consortium of participating universities, under the aegis of the ITU Academy, as a part of regular university studies leading to a Master's degree.
- **Option D**: QoSTP becomes a guide for self-study and training through practical experience, in which case ITU/the ITU Academy would develop the training materials, offer student consultation opportunities and ultimately assume responsibility for the professional examination to determine whether the knowledge and skills acquired meet the requisite standard. For the latter task, ITU would need to develop and administer the final assessment tests for each module.

The above options are just the main ones, as further derivations are also possible, for example by allowing certain combinations. For instance, completion of the entire course as proposed in Figure 1 might lead to a Master's degree, whereas completion of only some part of it may result in the award of a professional certificate at the corresponding level. The proposed course structure, comprising general and specialized modules, allows for a number of combinations to provide for the offering of different certification types, knowledge attainment levels and specializations. The set of training paths ultimately to be offered may be decided upon after the objectives have been agreed and negotiations have taken place with all the parties wishing to form the QoSTP teaching consortium.

Course delivery options

As QoSTP is conceived as an advanced training programme, it can rely on a substantial amount of instructor-led self-study (eLearning). Most people become accustomed to self-study during their undergraduate university years, so this self-discipline could be used to significant advantage in postgraduate on-the-job training. Self-paced eLearning (under the supervision of an instructor and with ad hoc consulting services) is easier to adapt to the changing time constraints imposed by the fluctuating daily workload of the student's primary job, in addition to which it enables tutors to focus more efficiently both on advising those students and on the topics that are subject to the highest levels of learning friction.

All modules might benefit from exploiting different delivery modes tailored by the tutor. For example, the time allocated to a module might be spread among:

- classroom teaching and physically attended seminars and workshops;
- practical exercises, whenever possible (e.g. with monitoring equipment, software tools);
- instructor-led remote (live or prerecorded streamed audiovisual) lectures;

- self-study of textbooks and reference material;
- interactive seminars and discussions with tutor and peers using web tools.

Classroom instruction could be coordinated by the ITU Academy and carried out at ITU CoEs or at other partner institutions such as universities.

Partner institutions

The ITU Academy is a convenient coordinating point for devising and then implementing QoSTP. However, it would need to draw in skilled partners to prepare the teaching content, decide how and what to test for in the certification procedure, design entry-level examinations that might confer academic credit for work experience, and ultimately determine how to staff, deliver and manage the programme. Some may suggest that there is no need to predefine at the outset the types of partner institutions liable to be considered for participation in QoSTP, but suitable institutional partners could surely be found among the following and other categories:

- ITU-T and its study groups (primarily SG12);
- ITU-R and its study groups (e.g. those dealing with relevant QoS standards);
- organizations participating in the running of the ITU CoEs in various regions;
- NRAs, especially those which already have their national QoS training programmes and facilities;
- regional telecommunication organizations (e.g. CEPT);
- universities and research centres;
- organizations and companies currently running educational programmes for QoS (such as mesaqin.com and others reviewed in the "Overview of existing QoS study resources and courses" section of this report);
- sector-specific industry associations (e.g. IETF, 3GPP);
- companies prominent in the field of telecommunication equipment, especially producers of QoS software tools, QoS monitoring equipment and wireless network (e.g. drive-test) equipment;
- other stakeholders.

All potential partners should be consulted in order to gauge their interest in QoSTP, with regard both to the possibility of their participating in the training and to the potential demand for QoSTP-certified specialists. The level of demand will surely influence the number of students accepted into the programme, which will in turn influence the budget and staffing requirements. The consultations will help in forming the consortium by providing a better understanding of the types of institution that are interested in participating and their level of engagement, as well as of the overall scale of the undertaking.

Another important consideration in terms of support and participation should be the establishment of a large pool of well-qualified QoS experts, who could initially be involved in the process of developing the teaching content and assessment exams, after which some may continue their involvement through the delivery of the QoSTP classroom-based modules and the remote tutoring of students, as well as supervision of their progress and eventual Master's thesis.

6 Possible topics related to performance, QoS and QoE for the ITU academy

The following section outlines the topics that constitute the entire QoSTP Programme. The topics are structured in ten modules as already described.

6.1 Obligatory Module OM1: Introduction and overview - QoS and QoE

Module objectives

This module will impart an understanding of the concepts of QoS and QoE. It will cover the different viewpoints of the various types of stakeholder involved in telecommunication and ICT and a high-level comparison of standardized assessment methods for QoS and QoE in ITU-T and other SDOs, if required.

Target audience

Since this module conveys the basics of QoS and QoE, the target audience can be quite broad and should include, but be not limited to, regulators, ICT/telecommunication providers and students in their final year of engineering or ICT-related courses, as well as user organizations and those providing consultation in the telecommunication/ICT field.

Learning outcomes

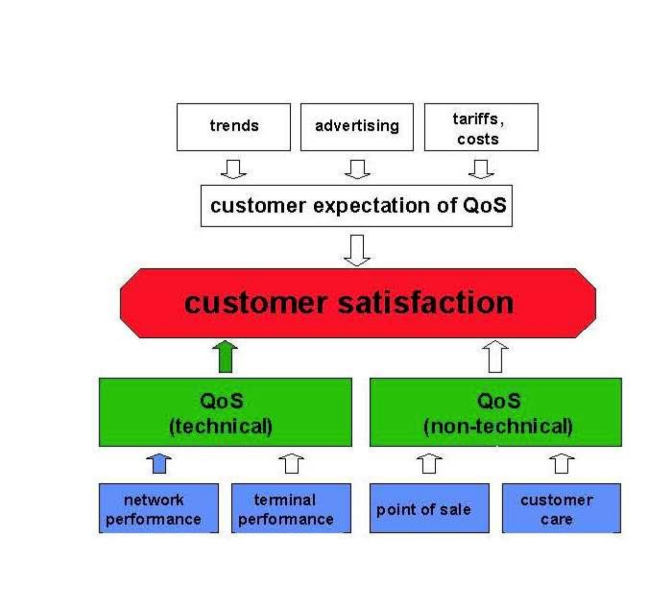
Those following this module will acquire knowledge which is essential for any technical work or decision making related to the quality of ICT media.

QoS is defined as the totality of characteristics of a telecommunications service that bear on its ability to satisfy stated and implied needs of the user of the service (Recommendation ITU-T E.800). Those characteristics can be measured by objective means (e.g. by a level meter or delay counter). QoS is frequently confused with elements of network performance (NP) because (signalling) functions inside networks are sometimes referred to as "services"; for example, IETF uses QoS to describe the performance of functional services in network layer models. In order to avoid that confusion, QoS is often more precisely referred to as "end-to-end QoS".

Another confusion needs to be avoided when network counters and key performance indicators (KPI) are brought into the discussion about QoS. Network counters are vendor-specific NP parameters which cannot be standardized owing to their proximity to specific implementations. The majority of standardized KPIs describe NP parameters, and only a very limited number of such KPIs are truly related to end-to-end QoS.

Furthermore, the end-to-end QoS of services at the user interface, which basically summarizes the characteristics of the underlying in-service media streams, should not be mistaken for the QoS of non-utilization stages of ICT services, which describe the (customer) "service" surround of ICT services offered by service providers outside the actual usage of services that are of interest and concern to the users, e.g. quality and content of information on a service and its features, the contractual conditions offered by the service provider, provisioning facilities, documentation, and service support after contract with customers (Recommendation ITU-T E.803 - Quality of service parameters for supporting service aspects).

Figure 2: Factors influencing customer satisfaction[14]



Source: ITU

The (average) user perception of end-to-end QoS can be assessed by subjective testing, which is very costly, and for this reason objective methods have been developed that help to predict user perception of QoS with the aid of objective measurement tools. The most prominent example has recently been standardized in Recommendation ITU-T P.863 (Perceptual objective listening quality assessment) – the successor to Recommendation ITU-T P.862 (Perceptual evaluation of speech quality) – and is intended for QoS assessment of voice services. Objective measurement methods for other services, for example video and data, are currently under development.

The user's perception of quality is, however, not limited to the objective characteristics at the man-machine interface, as summarized in the QoS concept. What counts for end users is the quality they personally experience during their use of a telecommunication service. QoE therefore takes account of additional subjective parameters stemming from user expectations and from the context in which the user is embedded during use of the service, typical examples of context-related influences being personal mood and environment. Also, QoE covers the potential discrepancy between the service offered and individual users reading additional features into the service.

Global challenges

With the move from traditional networks, which were based on dedicated service channels and/or separate networks for each service, to integrated (transport) services on a single packet-based transport infrastructure delivering all (transport) services via a single network access point, an access network and a so-called unified backbone, the predefined transmission planning of QoS has become a major challenge.

In traditional networks, the allocation of transmission impairments was based on a simple but effective concept, with resources being divided into the so-called international chain and the two terminating national networks (including terminals), with heavy regulation in place, whereas modern packet-based network quality parameter requirements are relatively undefined, and the impression is that the responsibility for end-to-end QoS has been lost. Basically, in an IP environment, services must be considered as applications executed in the terminal devices; IP networks cannot provide for self-standing end-to-end QoS, but only for transport classes, which enable QoS differentiation.

[14] www.itu.int/md/T09-CWG.WCIT12-INF-0005/en

The way in which QoS-related challenges are viewed depends strongly on the role of the stakeholders involved:

- **SDOs** such as ITU-T, the European Telecommunications Standards Institute (ETSI) or the Internet Engineering Task Force (IETF) have the collective knowledge and expertise to address the QoS-related problems inherited with the change of paradigms in networks and terminals and also with the aspects of planning and possible regulation of end-to-end QoS. However, SDOs are contribution-driven, which means that if stakeholders opt to rely on industry standards rather than globally recognized standards, and if they wish to keep control of their intellectual property and not to invest resources in globally recognized standards, then there is not very much for SDOs to do, other than try to convince industry leaders, for example through dedicated events such as conferences.
- **Network equipment manufacturers** basically have to rely on QoS-related performance requests (relating to network and system functions) from network operators and service providers. Ideally, they would participate in the QoS work of SDOs in order to standardize the QoS and performance requirements between the various parties involved in the network business. Unfortunately, however, for many network equipment manufacturers there is no visible short-term incentive that would lead them to participate in the end-to-end QoS work of SDOs, given that the return on investment from such an engagement is not readily visible.
- **Terminal device manufacturers** are today confronted with a mass market. In the past, terminal standards targeted, for example, minimum attachment requirements, which were designed not to harm the network. Nowadays, there are terminal standards which target the possibility of provision to the customer of high-level end-to-end QoS. This is a challenge for terminal equipment manufacturers, since the acceptance of terminals in the market is based, at least initially, on factors other than end-to-end QoS, such as price, additional functions (MP3 player, GPS, etc.), applications (including games), branding, and even colour ("kids prefer the pink phone!").
- **Network operators and service providers** are faced with the need to invest heavily in both infrastructure and access technology. They are liable to react in part by investing in new capacity, while at the same time rationing existing capacity. From their perspective, traffic management tools play an important role, increasing the efficiency with which operators can manage existing network capacity. The appropriateness of different approaches to traffic management is at the heart of the net neutrality debate. Given the controversial nature of this debate, it is important to bear in mind that traffic management always has beneficial aspects. It is commonly used, for example, to protect safety-critical traffic such as calls to the emergency services. The question, therefore, is not whether traffic management is acceptable in principle, but whether particular approaches to traffic management cause concern.

  But there also remains the question as to whether network operators and service providers may or may not use traffic management as an acceptable means of suppressing competition from the so-called "unmanaged" (i.e. not differentiating between traffic types, source or destination points) Internet or of inhibiting the opportunities for content or application providers with which it competes from introducing new, innovative products. Opening access and core packet networks as pure bit pipes will probably not provide revenues to match the aforementioned huge investments. Network operators and service providers will, therefore, seek to provide services on top of the bit stream itself. Since the earliest point in the development of next-generation networks (NGNs), which began in the mid-1990s with the ETSI TIPHON project, the outcome of which was ultimately harmonized with ITU work in the Next-Generation Networks Global Standards Initiative (NGN-GSI), network operators and service providers have claimed that the so-called "guaranteed QoS" (which is no more than a statistical guarantee) requires service differentiation in the networks. In fact, for the network, the differentiation would have more to do with traffic class, with different services then requesting a certain transport class from the network.

- **Regulators and administrations in general** are challenged, with their responsibility for consumer protection being affected by the rapid introduction of vendor-specific new services which they have to take into account. In addition, they are also required to strike an appropriate balance between service competition and infrastructure competition to address the challenges associated with QoS in the network. In the early days of the move towards end-to-end services being no longer provided on a fixed, familiar platform, it still seemed relatively easy to require that the new technology provide QoS "not less than in the ISDN era" (quote from the Austrian regulator). Today, however, it is easy to lose the overview of the proprietary services provided by various network operators and service providers "on-net" and the corresponding QoS. The real problem seems to be that services are not standardized, which would mean that for interconnection scenarios (one of the ITU major responsibilities, and one of the main purposes of the ITRs), one would need specific service agreements for each network-to-network interface (NNI).

  By contrast, regulators and administrations have in the recent past seen that the unmanaged Internet has led to the creation of new services offered "over the top" (e.g. Skype), which, like network operators and service providers, are an important factor contributing to the economic well-being of their respective countries. Services on the Internet can be created, improved, judged and used by each individual, subject to the legislation in force, without restriction.

  Consequently, regulators and administrations have to keep a close eye on the conditions under which access to such services is being provided by comparison with access to the Internet. For example, in the access there may be a certain percentage of the bandwidth or of the capacity reserved for the on-net services which is then not available for the Internet access. Similarly, the network operator packet-based backbone may serve both for the provision of its proprietary services, from which it receives the bulk of its revenues, and for the carriage of open Internet traffic, which generates lower revenues and is therefore liable to be accorded a lower priority.

- **Consumers** are challenged in the course of their personal use of telecommunication services (i.e. with regard to the discrepancy between the advertised and actual delivery speeds of the network). In communications between the European Commission and the Body of European Regulators for Electronic Communications (BEREC), the need for clear and transparent communication of QoS parameters and network management practices has been a recurrent theme.

  For example, BEREC states in its response of 30 September 2010 to the EU Commission consultation on the open Internet and net neutrality in Europe: “Consumers may not be able to detect the actual applications of discriminating traffic management techniques and find it difficult to distinguish between the effects of traffic management techniques on QoS from the effects of other quality degrading factors. For instance, a consumer who is observing that traffic is routinely throttled may not know whether this is done by intention, or is caused by other factors such as network congestion, which is leading to the degradation of service. Even if [network] operators or ISPs are required to declare which traffic management techniques and policies are being used, consumers may find it difficult to act upon such information if it is presented in a highly technical way which does not explain the ‘real world’ effects. Thus, it will be important to monitor the effectiveness of transparency and QoS.”

- **In technical terms**, the global challenges can be summarized as follows:

  - Owing to the dramatic increase in mobile communication, in terms both of the number of registered devices and volume of requested resources, it is quite likely that migration scenarios and hybrid connections with existing wire-bound and traditional networks and terminals will be neglected, and that appropriate QoS standards will not be established or enforced.

  - Service differentiation in modern packet-based networks is facilitated by means, for example, of the Internet Multimedia Subsystem (IMS), which, in its QoS part, is basically a resource allocation tool. Again, the exact services are not defined or standardized, which makes IMS less flexible for services to be offered across multiple packet networks. IMS is under the sole control of the 3rd Generation Partnership Project (3GPP), which is not an SDO in the classical sense. Opportunities for ITU members to influence the further development of IMS are thus very limited.

The main technical parameters to consider will therefore be:

- speed (data throughput) of the access network;
- congestion in the backbone;
- end-to-end delay (latency);
- delay-variation (jitter);
- packet loss (loss of information).

There are multiple facets here, depending on which kinds of gateway are used to interconnect IP networks: jitter is the variation in delay between different packets, and its compensation (by de-jitter buffers) converts it into additional delay which may build up and increase to unacceptable values, while packet loss may be concealed to such an extent that essential information is lost.

Bad terminal implementations may destroy reasonable performance delivered from the network(s), and users will be unable to judge the difference in end-to-end QoS.

G.1011

Recommendation ITU-T G.1011 provides a reference guide to existing standards for QoE assessment methodologies. It specifies QoE assessment approaches and classifications for different applications, and identifies the taxonomy of QoE assessment standards with different technical categorizations.

6.2 Obligatory Module OM2: Subjective assessment of voice quality

Module objectives

This module will provide an understanding of different types of subjective assessment of test purposes, of the design of subjective experiments and of test procedures and processing of the results. Consolidated subjective test results are a vital basis for standards and the setting of target values in telecommunications.

Target audience

Since properly designed and conducted subjective assessment forms the basis for instrumental methods for assessing QoS and QoE in the field, the target audience should include, but not be limited to, those involved in QoS/QoE matters within regulatory bodies and ICT/telecom providers and in test houses. In addition, students in their final year of engineering or ICT-related courses are targeted.

Learning outcomes

Those following this module will acquire knowledge which is essential for professional involvement in the conduct of subjective assessment experiments or evaluation of the results of such experiments from different sources.

There is an ongoing need for standard subjective test methodologies for the effective assessment of the transmission performance of new digital systems such as speech/music coders (telephony-band and higher quality bandwidths) or other devices and equipment designed for carrying voice and audiovisual signals, in order to follow the opinions and perceptions of real-world users. This module provides information on test purposes, experimental designs, test procedures, methods for handling the results, etc.

This module is related to Question 7/12 (Methods, tools and test plans for the subjective assessment of speech, audio and audiovisual quality interactions). Related Recommendations include P.85, P.800, P.805, P.810, P.830, P.835, P.840, P.851, P.880, P Suppl. 24 and P Suppl. 25.

P.85

Various services providing vocal responses related to telephone directory inquiries, weather forecasts, mail order, etc., are now available to public switched telephone network (PSTN) users using vocal servers. As the spoken messages are produced by machines, they may suffer from some impairment. In this Recommendation, a method is defined for subjective performance assessment of the quality of speech of voice output devices. This method allows for the intercomparison of several systems. It will be useful to system designers and service providers for checking the quality of their products. This method is of the listening-test type, whereby messages are presented aurally to subjects, who then express their opinion through one or more rating scales after having answered specific questions on the information contained in the messages. The results constitute a measure of several aspects of the perceived quality, making it possible to compare the effectiveness of different speech synthesis systems.

P.800

This Recommendation describes methods and procedures for conducting subjective evaluations of transmission quality. The main revision encompassed by this version of the Recommendation is the addition of an annex describing the comparison category rating (CCR) procedure. Other modifications have been made to align this Recommendation with recent revisions of Recommendation ITU-T P.830.

P.805

Recommendation ITU-T P.805 describes methods and procedures for conducting conversation tests to evaluate communication quality. The methodology uses examples of scenarios, rating scales and analysis procedures to estimate the subjective quality of telecommunication services. Conversation tests allow the simulation of more realistic situations close to the actual service conditions experienced by telephone customers. In addition, conversation tests are designed to assess the effects of impairments that can cause difficulty while conversing (such as delay, packet loss, echo, interruptions, noise, clipping, etc.), and may be used to study overall system effects as well as specific degradations.

P.806

This Recommendation describes a methodology for evaluating the subjective quality of speech samples using multiple rating scales. In addition to scores for overall quality and loudness, the methodology yields scores for six perceptual quality (PQ) attributes of the speech sample. Each of these PQ scores is based on ratings of the amount or degree of degradation present in the sample for an attribute that underlies the listener's judgment of speech quality. Four of the PQ scores represent degradation associated with the speech signal, while two of them represent degradation associated with the background noise. The methodology is designed to be used with naive subjects and yields scores for overall quality and loudness plus scores for the six PQ attributes of the speech sample. These PQ scores can be used to provide diagnostic information on the underlying causes of speech quality degradation.

In most standard ITU-T subjective test methodologies for evaluating speech quality, subjects are passive participants in the exercise. Typically, subjects listen to the test sample and provide a judgment of the overall quality of recorded passages of speech materials. These listening-quality test methodologies involve a single rating scale, and the quality estimate is an average of the ratings for multiple subjects where each subject typically rates samples from multiple talkers. Recommendation ITU-T P.800 describes a number of such methodologies, including the absolute category rating (ACR) method, which produces the mean opinion score (MOS).

More than three decades ago, the diagnostic acceptability measure (DAM) was developed to evaluate the underlying causes of degradation in speech quality. The DAM used 21 rating scales, nine associated with degradation in the speech signal alone, eight associated with degradation in the background noise alone, and four associated with overall quality. The MOS rating scale was one of those four. The DAM required expert subjects who were screened, trained, and calibrated to provide reliable and consistent responses on the large number of rating scales utilized by the DAM. While the DAM enjoyed considerable success for evaluating speech quality in government and industry in the United States, it was a proprietary method not suited to routine testing with naive subjects. The ITU-T P.806 test methodology extends the multiple rating scale approach of P.835 to the more general case of speech in most types of degradation.

P.810

This Recommendation describes the modulated noise reference unit (MNRU), a standalone unit for introducing controlled degradations to speech signals. As such, the MNRU has been used extensively in subjective performance evaluations of digital processes, both in conventional telephone bandwidth and in wideband (e.g. 70-7000 Hz) applications. Historically, the MNRU has been implemented in analogue hardware. The revisions encompassed in this version of the Recommendation are the inclusion of descriptions of digital implementations of the MNRU. These descriptions are suitable for implementation in software or on digital hardware. One further revision takes account of the need for a high-pass filter (removal of any DC component of the input speech material) for all implementations. Existing analogue hardware implementations of the MNRU will continue to meet the specifications in this Recommendation, provided such filtering is applied externally.

P.830

This revised Recommendation describes methods and procedures for conducting subjective performance evaluations of digital speech codecs. Revisions encompassed by this version of the Recommendation take account of new information that reflects current practices in subjective evaluation of digital codecs, including an expanded section on creating source recordings and the addition of two annexes. One annex describes an implementation of a PCM codec (A-/µ-law) that generates one quantization distortion unit (qdu) of distortion to input signals. The other new annex describes the modified IRS transmit and receive characteristics. These characteristics are recommended as the transmitting and receiving responses to be used in situations where the codec being tested is intended for use in fully digital circuits.

P.835

This Recommendation describes a methodology for evaluating the subjective quality of speech in noise, and is particularly appropriate for the evaluation of noise suppression algorithms. Naive subjects evaluate each sample in two dimensions, using rating scales to estimate the amount of distortion in the speech signal and degree of intrusiveness of the background noise, before making their rating of overall quality. The process of evaluating the sample separately with respect to speech distortion and background intrusiveness conditions the subjects to integrate the effects of both sources of degradation in making their ratings of overall quality. Routine use of the P.835 test methodology has shown that naive subjects can effectively and reliably use multiple rating scales to evaluate the quality of speech in background noise.

P.840

This Recommendation describes a subjective listening test method which can be used to evaluate the speech quality of circuit multiplication equipment (CME). It is intended for use with CME systems such as those described in Recommendations ITU-T G.763, G.767, G.768 (DCME), G.765 (PCME) and G.769/Y.1242 (IP-CME), which use digital speech interpolation (DSI) techniques. In this version, the scope is expanded to encompass more recent speech coders implemented in CME. Updating the Recommendation includes tandeming situations and comfort-noise test configuration. The new Appendix I gives guidance for conversation tests.

P.851

This Recommendation describes methods and procedures for conducting subjective evaluation experiments for telephone services which are based on spoken dialogue systems. The respective systems enable a natural interaction via spoken language and possess speech recognition and interpretation, dialogue management and speech output capabilities. The setup and running of appropriate interaction experiments is described, and questionnaires for quantifying the relevant quality dimensions perceived by the user are given.

P.880

This Recommendation describes a methodology called continuous evaluation of time varying speech quality (CETVSQ) that can be used for evaluating the impact of the time fluctuations of speech quality on the instantaneous perceived quality (that is perceived at any instant of a speech sequence) and on the overall perceived quality (at the end of the speech sequence). The method uses a two-part task: first, an instantaneous judgment on a continuous scale with a slider during the speech sequence, and second, an overall judgment on a standard five-category scale at the end of the speech sequence.

P Suppl. 24

This supplement provides definitions for a set of parameters which can be extracted from services which rely on spoken dialogue systems. The parameters can be extracted from logged (test) user interactions with the service under consideration. They quantify the flow of the interaction, the behaviour of the user and the system, and the performance of the speech technology devices involved in the interaction. They provide useful information for system development, optimization and maintenance, and are complementary to subjective quality judgments collected according to Recommendation ITU-T P.851.

P Suppl. 25

Supplement 25 to the ITU-T P-series Recommendations provides definitions for a set of parameters which can be extracted from services which rely on multimodal dialogue systems. The parameters can be extracted from logged (test) user interactions with the service under consideration. They quantify the flow of the interaction, the behaviour of the user and the system, and the performance of the speech technology devices involved in the interaction. They provide useful information for system development, optimization and maintenance, and are complementary to subjective quality judgments. The list is an amendment and extension of the respective list of parameters for speech-based services which is given in Supplement 24 to the ITU-T P-series Recommendations.

6.3 Obligatory Module OM3: Objective assessment of voice quality

Module objectives

This module will provide an understanding of different types of objective assessment of test purposes, of the design of objective tests and test campaigns and of processing of the results. Consolidated objective assessment results are a common basis for benchmarking systems and networks.

Target audience

Since objective assessment results are a common basis for benchmarking systems and networks, the target audience should include regulators and ICT/telecom providers, test equipment manufacturers, as well as students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire knowledge which is essential for professional involvement in the carrying out of objective testing and test campaigns or evaluation of the results derived from such tests from different sources.

This module is related to Question 9/12 (Perceptual-based objective methods for voice, audio and visual quality measurements in telecommunication services). Related Recommendations include P.862, P.862.1, P.862.2, P.862.3, P.863 and P.863.1.

Objective speech quality models analyse speech samples and predict the mean opinion score (MOS) that would be achieved for that sample in a well-designed, well-balanced subjective experiment. By comparison with subjective experiments, objective measurements are fast, inexpensive to perform and highly repeatable. However, since objective models may not fully consider the cognitive aspects of human quality assessment or the physiological aspects of the senses, caution must be exercised when using an objective model, and subjective experiments should still be used to verify their accuracy. This module provides information on objective speech quality assessment methods such as requirements regarding the speech material to be used, algorithms for objective assessment, etc., based on Recommendation ITU-T P.863 and related documents.

P.862

This Recommendation describes an objective method for predicting the subjective quality of 3.1 kHz (narrowband) handset telephony and narrowband speech codecs. It presents a high-level description of the method, advice on how to use it, and part of the results from a Study Group 12 benchmarking exercise carried out in the period 1999-2000. An ANSI-C reference implementation, described in Annex A, is provided in separate files and forms an integral part of this Recommendation. A conformance testing procedure is also specified in Annex A to allow a user to validate the correctness of an alternative implementation of the model. This ANSI-C reference implementation takes precedence in case of conflicts between the high-level description as given in this Recommendation and the ANSI-C reference implementation. This Recommendation includes an electronic attachment containing an ANSI-C reference implementation of PESQ and conformance testing data.

P.862.1

Recommendation ITU-T P.862 provides raw scores in the range -0.5 to 4.5. The aim is to provide an MOS-LQO (P.800.1) score from P.862 to allow a linear comparison with MOS. This Recommendation presents the mapping function and its performance for a single mapping from raw P.862 scores to the MOS-LQO (P.800.1).This will allow MOS-LQO scores from Recommendation ITU-T P.862 to be comparable independently of the implementation of Recommendation ITU-T P.862. The given function for transformation presented in this Recommendation has been optimized on a large corpus of subjective data representing different applications and languages.

P.862.2

Recommendation ITU-T P.862.2 describes a simple extension to the perceptual evaluation of listening speech quality (PESQ) algorithm defined in Recommendation ITU-T P.862. It allows Recommendation ITU-T P.862 to be applied to the evaluation of conditions such as speech codecs, where the listener uses wideband headphones. (By contrast, Recommendation ITU-T P.862 assumes a standard IRS-type narrowband telephone handset which attenuates strongly below 300 Hz and above 3100 Hz.)

This Recommendation is mainly intended for use with wideband audio systems (50-7000 Hz), although it may also be applied to systems with a narrower bandwidth.

P.862.3

Recommendation ITU-T P.862.3 provides some important remarks that should be taken into account in the objective quality evaluation of speech conforming to Recommendations ITU-T P.862, P.862.1 and P.862.2. Users of Recommendation ITU-T P.862 should understand and follow the guidance given in this Recommendation. This Recommendation forms a supplementary guide for users of Recommendation ITU-T P.862, which recommends a means of estimating listening speech quality by using reference and degraded speech samples. The scope of Recommendation ITU-T P.862 is clearly defined in itself. This Recommendation does not extend or narrow the scope, but provides necessary and important information for obtaining stable, reliable and meaningful objective measurement results in practice.

P.863

Recommendation ITU-T P.863 describes an objective method for predicting overall listening speech quality from narrowband (300 to 3 400 Hz) to superwideband (50 to 14 000 Hz) telecommunication scenarios as perceived by the user in an ITU-T P.800 or ITU-T P.830 ACR listening-only test. It supports two operational modes, one for narrowband and one for superwideband. It presents a high-level description of the method, advice on how to use it, and part of the results from a benchmarking exercise carried out in the period

2006-2010. All essential parts of the model are given as source code specification in C++ notation in Annex B, and are provided in separate files which form an integral part of this Recommendation. A conformance testing procedure is also specified in Annex A to allow a user to validate the correctness of an alternative implementation of the model. The source code in Annex B takes precedence in case of conflicts between the high-level descriptions as given in the main body of this Recommendation and the corresponding source code parts. The source code does not contain basic libraries and structure definitions and cannot be compiled to an executable programme.

This Recommendation includes an electronic attachment containing source code in C++ notation (see Annex B) and conformance testing data (see Annex A).

P.863.1

Recommendation ITU-T P.863.1 provides some important remarks that should be taken into account in the objective quality evaluation of speech conforming to [ITU-T P.863]. Users of [ITU-T P.863] should understand and follow the guidance given in this Recommendation.

This Recommendation forms a supplementary guide for users of [ITU-T P.863], which recommends a means of estimating listening speech quality by using reference and degraded speech samples. The scope of [ITU-T P.863] is clearly defined in itself. This Recommendation does not extend or narrow the scope, but provides necessary and important information for obtaining stable, reliable and meaningful objective measurement results in practice.

6.4 Obligatory Module OM4: QoS and QoE for Multimedia and assessment methods

Module objectives

This module will provide an understanding of QoS and QoE of multimedia services and the related assessment methods. This is more complex and is lacking the long history of standards which are available for voice QoS/QoE. The focus is on multimedia services in packet-based networks, definition of specific QoS and QoE parameters and their assessment at different technical levels. The development of new standards in this field by ITU-T is progressing very rapidly.

Target audience

Those involved in new services within ICT/telecom providers, test equipment manufacturers and test houses, as well as students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire knowledge which is essential for professional involvement in the assessment of the QoS and QoE of new services or in evaluating the results derived from such tests from different sources.

A major challenge for emerging IP-based networks is to provide adequate QoE and QoS for new multimedia services and applications. An example is web-based applications, including so-called Cloud-based applications. In such applications, QoE is affected not only by network QoS, but also by server and terminal performance. These services are inherently multimedia, incorporating audio, video, text, graphics and interactive control functions, and performance requirements and associated measurement methodologies for each of these aspects need to be defined. This module provides relevant information in that regard, such as end-user performance expectations and associated metrics for audio, video, text, graphics quality and control functionality, the key performance parameters and the values required to satisfy end-user expectations as well as methods to assess quality.

This module is related to Question 13/12 (QoE, QoS and performance requirements and assessment methods for multimedia) and Question 14/12 (Development of parametric models and tools for multimedia quality assessment). Related Recommendations include G.1010, G.1011, G.1030, G.1040, G.1050, G.1070, G.1080, G.1081, G.1082, P.1010, Y.1562, P.1201, P.1201.1, P.1201.2, P.1202, P.1202.1 and P.1202.2.

G.1010

This Recommendation defines a model for multimedia QoS categories from an end-user viewpoint. By considering user expectations for a range of multimedia applications, eight distinct categories are identified, based on tolerance to information loss and delay. These categories are intended to form the basis for defining realistic QoS classes for underlying transport networks and associated QoS control mechanisms.

G.1011

Recommendation ITU-T G.1011 provides a reference guide to existing standards for QoE assessment methodologies. It specifies QoE assessment approaches and classifications of different applications, and identifies the taxonomy of QoE assessment standards with different technical categorizations.

G.1030

This Recommendation provides a framework of tools to obtain IP network performance, estimate the performance of user applications and apply perceptual models to gauge user satisfaction with the end-to-end performance.

The user-perceived performance of data applications on packet networks is dependent on many factors, including the end-to-end performance of the packet network, the application dependency on the communications network, the performance of the terminals and other devices beyond the purview of the network operator(s), and the user's task and the extent of user interaction with the application. Network designers take these factors into account to assure user satisfaction. Once the application performance has been estimated, then perceptual models can be applied to interpret the level of end-to-end performance attained.

This Recommendation assumes that the reader will be able to provide at least some level of detail about each of the key factors above, and then use the framework of tools to estimate end-to-end performance.

G.1040

This Recommendation defines a new performance metric for networks that transport short data transactions, such as those associated with credit cards and other point-of-sale transactions. The main factors contributing to transaction time are packet network performance and host processing time.

The new metric, called network contribution to transaction time (NCTT), uses packet-transfer performance levels (such as round-trip time and packet-loss ratio) as inputs in order to estimate the portion of transaction time attributable to the network alone. Since this is the portion under the control of the network operator, knowledge of this value is useful for operators and users alike. The scope of this Recommendation is limited to the performance of the path between user-network interfaces (UNI-UNI) and uses limited configuration information from transaction client and host systems.

G.1050

Recommendation ITU-T G.1050 describes an IP network model that can be used for evaluating the performance of IP streams. The focus is on packet delay, delay variation and loss. IP streams from any type of network device can be evaluated using this model. The following are possible uses for this Recommendation:

- Simulation of real-world IP network impairments (packet-delay variation and packet-loss characteristics).
- Testing of any type of IP stream(s) under simulated network conditions using pcap files. The IP stream(s) can be evaluated using standard test cases or user-defined simulated network conditions.
- Testing of any type of IP stream using hardware emulation of simulated network models using standard test cases or user-defined simulated network conditions.

This revision of Recommendation ITU-T G.1050 replaces Recommendation ITU-T G.1050 (2007) in its entirety. Technical changes from Recommendation ITU-T G.1050 (2007) include:

1) Revised Recommendation ITU-T G.1050 models the mechanisms that contribute to packet delay, jitter and loss: interfering streams, queue delays in network elements, and the characteristics of specific access technologies. The intention is to provide more realism than the earlier version.
2) Recommendation ITU-T G.1050 (2007) defined a Gilbert-Elliott mathematical model that fits a certain observed network behavior but that was not easily extendible to other scenarios. The new approach is based on discrete event simulation.
3) The "likelihood of occurrence" concept is no longer applied to IP networks.
4) Revised Recommendation ITU-T G.1050 is a true bidirectional model.
5) Impairment levels are updated to maintain currency with evolving IP networks.
6) The number of standard test cases is greatly reduced.
7) Users can customize test cases to fit their specific needs.

This Recommendation includes an electronic attachment containing the discrete event simulator source code, input packet capture files of interfering traffic, standard test cases and the simulator output.

G.1070

Recommendation ITU-T G.1070 proposes an algorithm that estimates videophone quality for QoE/ QoS planners. This model can be used by QoE/QoS planners to help ensure that users will be satisfied with end-to-end service quality. The model provides estimates of multimedia quality that take interactivity into account and allows planners to avoid under-engineering.

The application of this Recommendation is limited to QoE/QoS planning. Other applications such as quality benchmarking and monitoring are outside its scope.

G.1080

Recommendation ITU-T G.1080 defines user requirements for QoE for Internet protocol television (IPTV) services. The QoE requirements are defined from an end user perspective and are agnostic to network deployment architectures and transport protocols. The QoE requirements are specified as end to end and information is provided on how they influence network transport and application layer behaviour. QoE requirements for video, audio, text, graphics, control functions and metadata are provided. Compression coding schemes addressed in this Recommendation are examples, and detailed numeric values given as performance targets, e.g. bit rate, packet loss rate, are also examples. The readers may choose or replace these parameter values as appropriate in order to be consistent with the requirements of each IPTV service context to which they are to be applied.

G.1081

Successful deployment of IPTV services requires performance parameters to be monitored at a number of different points in the complete end-to-end chain, including the customer premises, key aggregation points and interconnect points between disparate and service provider network domains. Recommendation ITU-T G.1081 defines five monitoring points where such performance measurements can be made.

G.1082

Recommendation ITU-T G.1082 provides a framework for improving the robustness of IPTV performance based on the results of real-time measurements. The primary application of this framework is to control the media and network resources based on the measurement information and according to policy rules to support high quality of experience of IPTV services. For IPTV services, service providers and network providers may have separate monitoring systems. Measurement information is provided by the monitoring system. This Recommendation first describes the possible measurement information used in different monitoring domains and the information exchanged between providers. It then gives guidance on how to take these factors into account to adjust media and network resources in order to maintain the quality of experience for IPTV services.

P.1010

This Recommendation provides 3.1 kHz telephony speech transmission performance requirements for the whole range of packet-based gateways and terminals, including wireless and softphones. While measurement methodologies are not covered by this Recommendation, work on this topic is under way in Study Group 12 and is slated for incorporation in a future revision or, alternatively, a separate new Recommendation. Also, requirements for wideband telephony may be added in a future version of this Recommendation.

Y.1562

Recommendation ITU-T Y.1542 considers various approaches towards achieving UNI-UNI IP network performance objectives. Detailed examples are provided as to how some approaches might work in practice, including how service providers might handle cases where the aggregated impairments exceed those specified in a requested QoS class (such as those of Recommendation ITU-T Y.1541). The advantages and disadvantages of each approach are summarized.

P.1201

Recommendation ITU-T P.1201 provides an overview of algorithmic models for non-intrusive monitoring of the audio, video and audiovisual quality of IP-based video services based on packet header information. The ITU-T P.1201 series of Recommendations addresses two application areas:

- Recommendation ITU-T P.1201.1 specifies the model algorithm for the lower resolution (LR) application area, including services such as mobile TV.
- Recommendation ITU-T P.1201.2 specifies the model algorithm for the higher resolution (HR) application area, including services such as IPTV.

The two ITU-T P.1201 model algorithms are no-reference (i.e. non-intrusive) models which operate by analysing packet header information, as available from the respective packet trace data, provided to the model algorithms in the packet capture format (PCAP). Further input information on more general aspects of the stream, such as the video resolution, which may not be available from packet header information, is provided to the model algorithm out of band, for example in the form of stream-specific side information.

As output, the model algorithms provide individual estimates of audio, video and audiovisual quality in terms of the five-point ACR MOS scale. Further diagnostic information on causes of quality degradation can also be made available.

Complementary to the ITU-T P.1201 models, there are further models described in the ITU-T P.1202 series of Recommendations. These so-called ITU-T P.1202 models are bitstream-based video quality models. The main differences with ITU-T P.1201 can be summarized as follows:

- The ITU-T P.1201 models provide audio, video and audiovisual quality estimates, while the ITU-T P.1202-only models provide video quality estimates.
- The ITU-T P.1201 models use packet header information, while the ITU-T P.1202 models exploit further bitstream information, such as coding-related information. As a consequence, the ITU-T P.1202 models can be more accurate in their quality predictions. In turn, they require non-encrypted streams to enable access to payload information. Since the ITU-T P.1202 models are more complex, they also require more computational power to estimate the video quality.

P.1201.1

Recommendation ITU-T P.1201.1 specifies the algorithmic model for the LR application area of Recommendation ITU-T P.1201. The ITU-T P.1201 series of Recommendations specifies models for monitoring the audio, video and audiovisual quality of IP-based video services based on packet-header information. The lower resolution application area of the ITU-T P.1201.1 part of ITU-T P.1201 can be applied to the monitoring of performance and QoE of video services such as mobile TV. The algorithm for the HR case is specified in Recommendation ITU-T P.1201.2. See Recommendation ITU-T P.1201 for details and respective application ranges and limitations of use.

P.1201.2

Recommendation ITU-T P.1201.2 specifies the algorithmic model for the HR application area of Recommendation ITU-T P.1201.

The ITU-T P.1201 series of Recommendations specifies models for monitoring the audio, video and audiovisual quality of IP-based video services based on packet-header information. The higher resolution application area of the ITU-T P.1201.2 part of ITU-T P.1201 can be applied to the monitoring of performance and QoE of video services such as Internet Protocol television (IPTV). The algorithm for the LR case is specified in Recommendation ITU-T P.1201.1. See Recommendation ITU-T P.1201 for details and respective application ranges.

P.1202

Recommendation ITU-T P.1202 provides an overview of algorithmic models for non-intrusive monitoring of the video quality of IP-based video services based on packet-header and bitstream information. The ITU-T P.1202-series of Recommendations addresses two application areas:

- Recommendation ITU-T P.1202.1 specifies the model algorithm for the LR application area, including services such as mobile TV;
- Recommendation ITU-T P.1202.2 specifies the model algorithm for the HR application area, which includes services such as IPTV.

The ITU-T P.1202 model algorithms are no-reference (i.e. non-intrusive) models which operate by analysing packet header and bitstream information as available from the respective packet-trace data provided to the model algorithms in the packet capture format (PCAP). Further input information on more general aspects of the stream, which may not be available from packet-header and bitstream information, is provided to the model algorithm out of band, for example in the form of stream-specific side information.

Recommendation ITU-T P.1202.1 describes one model, namely the model for the LR application area. Recommendation ITU-T P.1202.2 describes two models for the HR application area corresponding to two modes: mode 1 and mode 2, which are both no-reference (i.e. non-intrusive) models. Mode 1 refers to a parsing mode; the model operates by analysing information in the video bitstream without fully decoding the bitstream (i.e. no pixel information is used) for MOS estimation. Mode 2 refers to a full decoding mode. In addition to the bitstream information which mode 1 uses, the model can also decode parts or all of the video bitstream (i.e. pixel information is used) for MOS estimation. Further client-specific information, such as concealment type, is provided to the algorithm out of band, for example in the form of stream-specific side information. As output, the model algorithms provide individual estimates of video quality in terms of the five-point ACR MOS. Further, diagnostic information on causes of quality degradation can be made available too, since different types of performance parameters are derived during model calculations.

Complementary to the ITU-T P.1202 models, two further models are described in Recommendations ITU-T P.1201.1 and ITU-T P.1201.2. The respective entry Recommendation for these models is ITU-T P.1201. It describes packet-header-only-based video, and audio and audiovisual quality models. The main differences with Recommendation ITU-T P.1202 can be summarized as follows:

- The ITU-T P.1201 models provide audio, video and audiovisual quality estimates, while the ITU-T P.1202-only models provide video quality estimates.

The ITU-T P.1201 models use packet header information, while the ITU-T P.1202 models exploit further bitstream information, such as coding-related information. As a consequence, the ITU-T P.1202 models can be more accurate in their quality predictions. In turn, they require non-encrypted streams to enable access to payload information. Since the ITU-T P.1202 models are more complex, they also require more computational power to estimate the video quality.

P.1202.1

Recommendation ITU-T P.1202.1 specifies the algorithmic model for the LR application area of Recommendation ITU-T P.1202. The ITU-T P.1202 series of Recommendations specifies models for monitoring the video quality of IP-based video services based on packet-header and bitstream information. The LR application area of the ITU-T P.1202.1 part of ITU-T P.1202 can be applied to the monitoring of the performance and QoE of video services such as mobile TV.

P.1202.2

Recommendation ITU-T P.1202.2 specifies the algorithmic model for the HR application area of Recommendation ITU-T P.1202. The ITU-T P.1202 series of documents specifies models for monitoring the video quality of IP-based video services based on packet-header and bitstream information. The HR application area of the ITU-T P.1202.2 part of ITU-T P.1202 can be applied to the monitoring of performance and QoE of video services such as IPTV, and has two modes: mode 1, where the video bitstreams are parsed and not decoded into pixels, and mode 2, where the video bitstreams are fully decoded into pixels for analysing.

P.1501

This Recommendation describes the method and procedures for subjective testing of the user-perceived quality of web browsing. The method generally applies to degradations and characteristics that can be introduced on the network level (e.g. round-trip time, downlink and uplink bandwidth, packet losses etc.), as well on the application level (page load times, etc.). Combinations of two or more of such factors have to be catered for. Impacting factors due to webpage design are not in the scope of this Recommendation.

6.5 Elective Module EM1.1: Telephonometry

Module objectives

This module will provide an understanding of the details of telephonometry electroacoustic terminal measurements. With the continued development of new types of terminal and new signal processing techniques inside the terminal, the development of new standards in this field by ITU-T is progressing very rapidly.

Target audience

Terminal manufacturers, test equipment manufacturers, test houses, students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire detailed knowledge in the field of electroacoustics and state-of-the-art terminal testing.

HANDBOOK ON TELEPHONOMETRY

Obviously, the handbook is closely related to most of the P-Series Recommendations, to which it may be considered as forming a commentary. The preparation of this edition involved close cooperation with the CCITT/Study Group XII/Speech-Quality Experts Group, who harmonized the P.70 and P.80 Series of

Recommendations. It will be noticed, for example, in Recommendation ITU-T P.80 “Methods for the subjective determination of transmission quality” that there is frequent reference to this handbook.

Although, basically speaking, the handbook is intended to serve as a practical manual, with the methods described within it being seen as a guide to sound engineering practice, it has been considered useful to include passages of a textbook nature to help in understanding the underlying principles. The inclusion of too many obsolete Recommendations in the handbook has been avoided.

It is hoped that the handbook will continue to meet the needs of the developing countries as initially intended, but it may also serve as a guide for technical personnel at different educational levels in any country. In addition, it may pave the way for further international standardization of all kinds of telephonometric measurements.

The methods described in the handbook represent only what is considered as sound practice. ITU-T Recommendations should be given priority wherever they exist. The reader is invited to refer to the most recent amendments of the Recommendations. Furthermore, the length of the sections should not be considered as reflecting the importance of the subjects treated.

At the 1988 Plenary Assembly in Melbourne the decision was taken to close the CCITT laboratory.

P.32

The purpose of this Recommendation is to define the methods of measurement to evaluate the efficiency of acoustic hoods or telephone booths intended to improve the quality of telephone transmission in noisy environments. In addition to the improvement of the transmission quality during a conversation between two users, this Recommendation takes into consideration the need to guarantee speech privacy for the user speaking beneath the acoustic hood or inside the booth with respect to a listener located nearby.

P.48

This purpose of this Recommendation is to specify the intermediate reference system (IRS) to be used for defining loudness ratings. The description should be sufficient to enable equipment having the required characteristics to be reproduced in different laboratories and maintained to standardized performance.

P.50

The signal described here reproduces the characteristics of human speech for the purposes of characterizing linear and non-linear telecommunication systems and devices, which are intended for the transduction or transmission of speech. It is known that for some purposes, such as objective loudness-rating measurements, simpler signals can also be used. Examples of such signals are pink noise or spectrum-shaped Gaussian noise, which nevertheless cannot be referred to as "artificial voice" for the purposes of this Recommendation.

The artificial voice is a signal that is mathematically defined and that reproduces the time and spectral characteristics of speech which significantly affect the performances of telecommunication systems. Two kinds of artificial voice are defined, reproducing respectively the spectral characteristics of female and male speech.

The following time and spectral characteristics of real speech are reproduced by the artificial voice:

a) long-term average spectrum;

b) short-term spectrum;

c) instantaneous amplitude distribution;

d) voiced and unvoiced structure of speech waveform;

e) syllabic envelope.

Appendix I/P.50 includes a CD-ROM containing useful test signals. The signals on this CD-ROM include the signal described in Recommendation ITU-T P.50 as well as other signals that have been found useful by some administrations. Additionally, the full-speech database that was used to develop Recommendation ITU-T P.50 is also on this CD-ROM. Appendix I/P.50 is published separately.

P.51

This Recommendation specifies the artificial mouth for telephonometric use. The methods of use of the artificial mouth are outside the scope of this Recommendation.

P.52

ITU-T considers that, in order to ensure continuity with previous practice, it is not desirable to modify the specification of the ARAEN volume meter.

The Recommendation gives the principal characteristics of various measuring devices used for monitoring the volume or peak values during telephone conversations or sound-programme transmissions.

The measurement of active speech level is defined in Recommendation ITU-T P.56. Comparison of results using the active speech level meter and some meters described in this Recommendation can be found in Supplement No. 18 to the P-series Recommendations.

NOTE – Descriptions of the following devices are contained in the Supplements to *White Book*, Volume V:

– ARAEN volume meter or speech voltmeter: Supplement No. 10.

– Volume meter standardized in the United States of America, known as the "VU meter": Supplement No. 11.

– Peak indicator used by the British Broadcasting Corporation: Supplement No. 12.

– Maximum amplitude indicator Types U 21 and U 71 used in the Federal Republic of Germany: Supplement No. 13.

P.57

This Recommendation specifies the artificial ears for telephonometric use. Three types are recommended, covering the different transducers, types, sizes and technologies.

The methods for use of the artificial ears are outside the scope of this Recommendation; however, some general rules are provided about the application force and the positioning of transducers.

Three types of artificial ears are defined:

1) telephone-band type for measurements on traditional telephone sets;
2) type for measuring insert earphones;
3) type which faithfully reproduces the characteristics of the median human ear.

P.58

This Recommendation specifies the electroacoustic characteristics of the head and torso simulator (HATS) for telephonometric use. Both sound emissions and sound pick-up characteristics are specified; the free-field acoustic diffraction is specified as well.

The device is intended for airborne acoustic measurements, and is not suitable for measurements that depend on vibration conduction paths such as bone conduction. The HATS is intended to provide acoustic diffraction similar to that encountered around the median human head and torso, and to generate an acoustic field similar to that generated by the human mouth, both in proximity and in the far field.

The methods for use of the HATS in telephonometry are outside the scope of this Recommendation. However, the sound pick-up and diffraction characteristics specified by this Recommendation resemble those recommended by the International Electrotechnical Commission (IEC) for the measurement of hearing aids. The electroacoustic measurement methodologies for assessing the performance of hearing aids in their telecommunication applications are then, to the extent applicable, specified by the relevant IEC publications.

The characteristics of the device are fully specified for narrowband and wideband speech measurements. Some of its characteristics are described for an enlarged frequency range enabling measurements to be performed for super-wideband speech. Regarding sound pickup, some characteristics are specified up to 20 kHz, thus enabling measurements to be performed for full-band speech as well. Regarding ear simulator acoustic impedance, characteristics are described in IEC 60318-4 and Recommendation ITU-T P.57.

P.59

The signal described here reproduces the on-off temporal characteristics of human conversational speech for characterizing speech processing systems which have speech detectors, such as loudspeaker telephones, echo control devices, digital circuit multiplication equipment (DCME), packet systems, and asynchronous transfer mode (ATM) systems. This signal reflects parameters of human conversation such as the length of the talk-spurt, pause, double talk and mutual silence. The following chapters describe these characteristics and a method of generating artificial conversational speech.

P.61

Primary and secondary calibrations of condenser microphones can be carried out using the methods described in this Recommendation.

P.64

The sending, receiving or sidetone sensitivity/frequency characteristic of a local telephone system (LTS) is usually measured directly.

NOTE 1 – The sending, receiving or sidetone sensitivity/frequency characteristic can also be calculated provided that the relevant information of the telephone line and feeding bridge is known. Some of the information required for sidetone is outside the scope of existing Recommendations.

NOTE 2 – The same principles also apply to the measurement of microphones and earphones.

Since electroacoustical measurements of the type being considered may be required for different purposes, it is important to distinguish the following:

a) Supplying the designer of a transducer with information concerning the success he has achieved in aiming at a given sensitivity/frequency response;

b) Checking that the manufactured product meets the specified requirements;

c) Supplying sensitivity/frequency characteristics suitable for use in calculating loudness ratings or estimating other subjectivity-determined quantities.

This Recommendation is mainly concerned with c), but the principle is also applicable to a) and b). For these purposes, especially for c), measurements under real conditions must form the basis. Artificial mouths and artificial ears must be used with due regard for obtaining good agreement between these measurements and those from real mouth and ear determinations. Measurements under real conditions are complicated, time-consuming and not reproducible with great precision. This Recommendation describes measurement methods using recommended forms of artificial mouths and artificial ears (see Recommendations ITU-T P.51 and P.57).

This Recommendation applies mainly to LTSs with handset telephones. However, the principles also apply to other types of telephone. Specific considerations for headsets are described in Recommendation ITU-T P.380 and, for loudspeaker telephones, ITU-T P.340.

See Recommendation ITU-T P.76 for general principles concerning the determination of loudness ratings.

P.75

Since the characteristics of carbon microphones are strongly dependent on conditioning techniques, it is necessary to follow a consistent procedure prior to measuring sensitivity/frequency characteristics in order to obtain reproducible results. ITU-T recommends that, for best reproducibility, automatic mechanical conditioning be used.

P.76

This Recommendation is one of a set of closely related Recommendations concerned with determination of loudness ratings, and deals with fundamental principles. The others deal with certain additional matters.

P.78

This Recommendation contains the essential particulars for defining the method for determining loudness ratings in accordance with Recommendation ITU-T P.76, when use is made of subjects performing equal loudness balances. Details are included concerning the balancing method, choice of subjects, speech material, design of experiment, method of analysis and presentation of results.

The method described in this Recommendation requires both the “unknown” telephone system whose loudness rating is to be determined, and the IRS, to be balanced against the fundamental reference system NOSFER. An alternative method, in which the “unknown” is directly balanced against the IRS, is described in the annex to this Recommendation (P.78).

P.79

This Recommendation describes the preferred method for calculating loudness ratings in the following cases:

- Narrowband local telephone systems (which transmit a band of frequencies not exceeding about 180-4500 Hz), and
- "Dual-mode" narrowband/wideband end-to-end transmissions, including terminals (300-3400 Hz and 100-7000 Hz, respectively).
- Wide-band only end-to-end transmission (100 Hz to 7 kHz) between wide-band terminals.

The purpose of using loudness ratings for telephone sets is twofold: first, to provide the transmission planner with an adequate measure of how the sets perform in the network; second, to enable valid and unambiguous comparison between sets. Therefore, to avoid confusion, this version of the Recommendation contains only those telephone set loudness ratings which are of interest for these purposes.

Annex A sets out the fundamental principles of loudness-rating calculations and explains the relationships between Recommendations ITU-T P.76, ITU-T P.78 and this Recommendation, as well as the physical basis of this Recommendation.

Annex B explains the fundamental concept of the sidetone-masking rating (STMR) used for evaluation of the talker's sidetone.

Annex C gives an alternative form of the loudness-rating algorithm, which is useful for estimating the relative importance of how the sensitivity in different frequency bands influences the loudness-rating value.

Annex D provides, as a reference only, W_i weights for OLR, SLR and RLR over the wider band 100-8000 Hz.

Annex E describes how the listener's sidetone factor D can be determined.

Annex F shows how the sidetone sensitivity S_{meST} can be computed from the send and receive sensitivity and impedance data.

Annex G gives a set of W weights suitable for the calculation of sending and receiving loudness ratings of wideband (100 to 7000 Hz) only terminals.

P.300

This Recommendation provides the audio performance requirements and test methods for group audio terminals (GATs), i.e. terminals which are specifically designed for use by several users at the same time.

GATs cover a wide range of products, ranging from hands-free telephone sets, when used by several users at the same time, to teleconference audio terminals incorporating sophisticated echo control mechanisms. GATs can be designed to operate on both analogue POTS networks and digital ISDN links. While the former can only offer telephone bandwidth performances, the latter can be designed to offer either narrowband or wideband audio communication facilities.

The Recommendation addresses this whole range of devices by identifying, as far as possible, the already existing Recommendations applicable to each terminal type.

P.310

This Recommendation deals with sending and receiving frequency response, loudness rating, noise, distortion, out-of-band signals, linearity, sidetone, echo and delay of narrowband (300-3400 Hz) digital handset and headset telephones using "waveform" encoding according to Recommendations ITU-T G.711 (PCM at both 64 and 56 kbit/s) and G.726 (ADPCM, 32 kbit/s). IP terminals and wireless headsets are not covered in this Recommendation.

The use of digital telephones using Recommendation ITU-T G.728 (LD-CELP, 16 kbit/s) and of mobile/cordless telephones are under study.

Requirements applicable to low acoustic impedance transducers and digital telephone sets using non-linear techniques are likewise under study.

The requirements listed in this Recommendation should also be used as the basis of requirements for other "waveform" encoding schemes.

The values given in this Recommendation should be used for developing specifications which will include assigning tolerances, etc.

P.311

This Recommendation provides audio performance requirements and test methods for digital handset and headset telephones capable of transmitting an audio bandwidth extending beyond the conventional telephony bandwidth of 300 Hz to 3400 Hz, to a bandwidth of approximately 100 Hz to 8000 Hz. Such telephones are known as wideband audio telephones, and will make use of digital encoding schemes such as in Recommendation ITU-T G.722. IP terminals may support other coding algorithms. Wideband audio telephones are expected to be used in new services such as high-quality audioconferencing, videoconferencing and multimedia applications.

The requirements listed in this Recommendation are primarily applicable to telephones using ITU-T G.722 encoding at 64 kbit/s, but should also be used as the basis of requirements for other wideband audio encoding schemes. This is still under study in ITU-T.

Conventional telephone band (300 Hz-3400 Hz) digital handset telephones using encoding according to Recommendation ITU-T G.711 and [ITU-T G.726] are covered by Recommendation ITU-T P.310.

P.313

This Recommendation deals with electroacoustic performance parameters of portable digital cordless and mobile terminals. The requirements given below should ensure satisfactory voice service in a high percentage of installations under normal conditions, but other performance-impacting factors, such as the radio link, are not included.

This Recommendation includes specifications for the handset, headset and speakerphone set of mobile terminals. These specifications may be applicable to digital cordless handset, headset and speakerphone sets. The requirements contained in this Recommendation apply only to narrowband systems (3.1 kHz), regardless of the coding algorithm used.

Specifications for speakerphone sets are recommended for handheld terminals and desktops.

Specifications for car-mounted hands-free terminals will be included in a separate Recommendation.

Requirements are given for handset, headset and speakerphone set in conjunction with a 0 dBr 4-wire reference.

P.330

This Recommendation applies to the generic transmission characteristics, performance and testing principles of speech-processing devices for acoustic enhancement (SPDA) intended for use in terminals, whatever the applications.

An SPDA is defined as any signal-processing function integrated in terminals that performs voice enhancement. Voice enhancement functions include the control of acoustic echo and noise reduction. Dereverberation and any advanced signal processing for multi-channel pick-up and restitution are for further study.

The purpose of this Recommendation is to define a framework for specifying performance constraints for terminals which include SPDA and, where appropriate, to define tests that may be performed on such terminals to verify that these constraints are met. This Recommendation covers generic characteristics that are applicable to both analogue and digital terminals. Requirements that are applicable strictly to hands-free terminals can be found in Recommendation ITU-T P.340.

Test methods appropriate for parameters defined in this Recommendation may be found in Recommendation ITU-T P.502.

For the use of HATS for testing, Recommendation ITU-T P.581 applies.

P.340

The object of this Recommendation is to obtain for hands-free terminals transmission performance equivalent with handset telephones, at least with respect to send and receive loudness.

Other important features contributing to the quality of telephone calls made from hands-free terminals, as switching characteristics, duplex capability, are defined in this Recommendation.

This Recommendation covers generic characteristics and requirements that are applicable to both analogue and digital hands-free terminals. Additional requirements that are applicable strictly to digital terminals can be found in Recommendations ITU-T P.342 and P.341.

For loud-speaking telephones (see Recommendation ITU-T P.10) which do not provide full hands-free operation, the relevant parts of this Recommendation may be referred to.

Test methods appropriate for parameters defined in this Recommendation may be found in Recommendation ITU-T P.502.

For the use of HATS for testing, Recommendation ITU-T P.581 applies.

P.341

This Recommendation provides audio performance requirements and test methods for loud-speaking and hands-free telephones capable of transmitting an audio bandwidth extending beyond the conventional telephony bandwidth of 300 to 3400 Hz, to a bandwidth of approximately 100 Hz to 8000 Hz. Such telephones are known as wideband audio telephones, and will make use of digital encoding schemes such as in [ITU-T G.722]. IP terminals may support other coding algorithms. Wideband audio telephones are expected to be used in new services such as high-quality audio conferencing, videoconferencing and multimedia applications.

The requirements listed in this Recommendation are primarily applicable to telephones using ITU-T G.722 encoding at 64 kbit/s, but should also be used as the basis of requirements for other wideband audio encoding schemes. This is still under study in ITU-T.

General information on hands-free terminals, which includes switching characteristics, can be found in [ITU-T P.340], and information on acoustic echo controllers can be found in Recommendation ITU-T G.167.

For loud-speaking telephones which do not provide full hands-free operation, the relevant parts of this Recommendation may be used.

Conventional telephone band (300-3400 Hz) digital hands-free telephones using encoding according to Recommendations ITU-T G.711 and G.726 are covered by Recommendation ITU-T P.342. Audio performance requirements for wideband headset terminals are included in Recommendation ITU-T P.311. Specifications for car-mounted wideband hands-free terminals are included in Recommendation ITU-T P.1110. Transmission characteristics for narrowband cordless and mobile digital terminals are included in Recommendation ITU-T P.313.

P.342

This Recommendation provides audio performance requirements for hands-free and loud-speaking telephones using, in the telephone narrowband (300-3400 Hz), the waveform encoding according to Recommendations ITU-T G.711 (PCM at both 64 kbit/s and 56 kbit/s) and G.726 (ADPCM 32 kbit/s). Audio performance requirements for headset terminals are included in Recommendation ITU-T P.310. IP terminals are not included in this Recommendation.

Audio performance requirements for digital telephones using coding schemes other than waveform encoding and at bit rates lower than 32 kbit/s are under study.

P.350

Handset dimensions for traditional corded analogue and digital telephones are recommended. For very short designs, information about the influence on the D-factor is given.

P.360

The use of devices for preventing the occurrence of excessive acoustic pressure by telephone receivers is recommended in Recommendation ITU-T K.7. Methods for checking the efficiency of such devices in response to short-duration impulses, longer-duration disturbances, such as tones, and daily noise exposure are given in this Recommendation. A method is also given for checking that such devices do not have adverse effects on normal speech signals.

P.370

Clause 4 applies to telephone handsets having supra-aural earphones that can be connected to the PSTN or ISDN and are intended for direct application to the ear (e.g. traditional handsets, operators’ headsets) and which provide a magnetic field for coupling to hearing aids. It specifies the level, linearity and frequency dependence of the magnetic field strength produced by the handset and characteristics for the calibrated probe coil.

Clause 5 specifies the electroacoustic performance characteristics of telephony terminals that are intended for direct application to the ear (e.g. traditional handsets, operators’ headsets) and which provide, at the earphone, additional amplification in the receiving direction compared with the receiving loudness rating (RLR) specified in the requirements of the national system.

Clause 6 specifies the electrical characteristics for the electrical coupling of the telephony function, implemented telecommunication terminal equipment, to hearing aids. It specifies the level and frequency response relative to the acoustic output at the earphone, as well as the noise and maximum level.

Annex A specifies the measuring method for an acousto-magnetic adapter that converts the acoustic output of an associated telephone receiver to a magnetic field, in accordance with 4.2.1 and 4.2.2, that can be received by the magnetic pick-up coil in a hearing aid.

P.380

This Recommendation is the result of a study held within ITU-T for defining the electroacoustic testing methodologies for headsets, which provide the best correlation with the performance of headsets in real use, when using the couplers currently recommended in Recommendation ITU-T P.57.

The results of this Round Robin test, aiming to compare the acoustic behaviour of headsets placed on humans and on HATS, can be found in the bibliography.

The recommended test methodology is based on the use of HATS, as this is the best approximation of acoustic conditions occurring in the real use of headsets.

This Recommendation focuses specifically on headsets and overrules Recommendation ITU-T P.57 regarding the applicability rules of artificial ears to specific receivers, as long as these devices belong to headsets.

This Recommendation is complementary to the relevant Recommendations (ITU-T P.64, P.79, etc.), which specify the electroacoustic and telephonometric testing methods applicable to telephone devices.

The recommendation of performance descriptors, such as masks or limit values, is left to the relevant performance standards.

P.581

This Recommendation specifies the use of the HATS for hands-free (including speakerphone, loud-speaking and headset functions) and handset terminal testing. It applies for subjective tests, e.g. as described in Recommendation ITU-T P.832, or for objective measurements as described in, for example, Recommendation ITU-T P.340. The test positions, equalizations and calibration are specified. The binaural test conditions are also considered. Test methods for both send and receive characteristics testing are specified.

For the applications described in this Recommendation, the HATS consists of a head mounted on a torso that extends to the waist. The head is equipped with two artificial ears according to Recommendation ITU-T P.57 and with a mouth simulator. The HATS is specified physically as well as acoustically, in Recommendation ITU-T P.58.

P.501

Recommendation ITU-T P.501 describes test signals which are applicable for several purposes in telephonometry. This Recommendation gives a wide variety of test signals, starting with low complexity test signals up to test signals with a high degree of complexity incorporating a large number of typical parameters of speech. Besides technical signals such as sine waves or noise, more speech-like signals are described.

This Recommendation describes the principles of the signal construction for each type of test signal. Characteristic properties such as power density spectra, probability density functions or shaping filter responses are shown.

The Recommendation gives an overview about the typical application of the test signals described. This overview is a guideline giving general application rules. The detailed description of the application, however, should be found in the individual Recommendations describing the measurement procedures for a certain application.

In order to avoid problems in creating the test signals described, all these test signals are freely available for download from the ITU-T test signals database.

P.502

The aim of this ITU-T Recommendation is the definition of test methods which can be used to evaluate specific artefacts influencing the speech quality transmission of terminals and speech transmission systems. The methods described in the Recommendation are based on test signals as defined in Recommendations ITU-T P.50, P.59 and P.501.

The Recommendation provides a collection of test methods which allow the investigation of various parameters which were found to be important for the assessment of speech communication systems. Each performance parameter is qualified by the speech degradation perceived subjectively and the related objective parameters. For the individual parameters, analysis methods are described.

P.505

This Recommendation provides a novel quality-representation methodology of parameters that determine the speech quality of telecommunication equipment as well as the end-to-end speech quality. This methodology is easy to use and also easy to understand for non-experts, and can serve as a basis for commercial decisions on a management or marketing level.

The Recommendation does not provide methods for the acquisition of speech quality measurement results, it being assumed that its user has readily at hand those test results which are needed as an input for the representation methodology recommended here; furthermore, it does not state any requirements with respect to the parameters mentioned herein.

P SUPPLEMENT 10

Clause 4 applies to telephone handsets having supra-aural earphones that can be connected to the PSTN or ISDN and are intended for direct application to the ear (e.g. traditional handsets, operators' headsets), and which provide a magnetic field for coupling to hearing aids. It specifies the level, linearity and frequency dependence of the magnetic field strength produced by the handset and characteristics for the calibrated probe coil.

Clause 5 specifies the electroacoustic performance characteristics of telephony terminals that are intended for direct application to the ear (e.g. traditional handsets, operators' headsets) and which provide, at the earphone, additional amplification in the receiving direction compared with the RLR specified in the requirements of the national system.

Clause 6 specifies the electrical characteristics for the electrical coupling of the telephony function, implemented telecommunication terminal equipment, to hearing aids. It specifies the level and frequency response relative to the acoustic output at the earphone, also the noise and maximum level.

Annex A specifies the measuring method for an acousto-magnetic adapter that converts the acoustic output of an associated telephone receiver to a magnetic field, in accordance with 4.2.1 and 4.2.2, that can be received by the magnetic pick-up coil in a hearing aid.

P SUPPLEMENT 16

This Recommendation provides basic rules for assessing the acoustics of telephone conference rooms and for installing group audio terminals consistent with maximum speech intelligibility and easy talker recognition.

P SUPPLEMENT 20

The subject of handset earcap leakage losses has been studied on and off for a quite considerable time by many organizations. The liberalization of the telecom market and emergence of "innovative" handset designs have now made the leakage problem more acute, i.e. in terms of customer complaints. The information provided in this supplement is intended to give some instructive examples as general information on the subject.

6.6 Elective Module EM1.2: Network performance and OAM for performance measurement

Module objectives

As critical communication services increase their reliance on new networking technologies such as MPLS and Ethernet over various network domains, network performance remains a key factor in the user's experience. When several network operators work together to provide end-to-end communications, each needs to understand how to achieve the end-to-end performance objectives. Such objectives must be both adequate for the service being offered and feasible based on the available networking technologies. This module provides information on this issue, looking at matters such as performance parameters, methods of measurement and/or numerical objectives.

The module is related to Question 17/12 (Performance of packet-based networks and other networking technologies). Related Recommendations include ITU-T Y.1540, Y.1541, Y.1543, Y.1544, Y.1560, Y.1561, Y.1563, Y.1564 and Y.1565 (also related to IETF bmwg, ippm).

This module is also related to Question 10/15 (Interfaces, interworking, OAM and equipment specifications for packet based transport networks). Related Recommendations include ITU-T G.8013/Y.1731, G.8113.1 and G.8113.2. (IEEE 802.1ag and IETF mpls wg are also related but focus more on fault management than on performance management).

OAM capabilities are essential for all network technologies when developing a carrier-class network. This is also particularly the case when developing a transport network, because a transport network is expected to support a wide variety of services in terms of reliability and performance, including highly reliable and high-quality services requiring effective network management. This module provides information on performance measurement OAM functions for various packet-based networks such as Ethernet and multi-protocol label switching transport profile (MPLS-TP).

Target audience

Network operators, students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire knowledge that is essential for any technical work or decision making related to end-to-end performance objectives where several network operators work together to provide end-to-end communications of ICT media quality.

Y.1540

Recommendation ITU-T Y.1540 defines parameters that may be used in specifying and assessing the speed, accuracy, dependability and availability performance of IP packet transfer in international Internet Protocol (IP) data communication services. The defined parameters apply to end-to-end, point-to-point IP service and to the network portions that provide, or contribute to the provision of, such service in accordance with the normative references specified in clause 2. Connectionless transport is a distinguishing aspect of the IP service that is considered in this Recommendation.

Y.1541

This Recommendation defines classes of network QoS with objectives for Internet Protocol network performance parameters. Two of the classes contain provisional performance objectives. These classes are intended to be the basis for agreements among network providers, and between end users and their network providers.

Appendix I provides information about how ATM might support IP layer performance. Appendix II discusses alternatives for defining IP delay variation. Appendix III presents the hypothetical reference paths (HRP) against which the ITU-T Y.1541 QoS objectives were tested for feasibility. Appendix IV gives example computations of packet-delay variation. Appendix V discusses issues that must be considered whenever IP measurements are made. Appendix VI describes the relationship between this Recommendation and the IETF-defined mechanisms for managing QoS. Appendix VII gives estimates of speech transmission quality for the hypothetical reference paths of Appendix III. Appendix VIII discusses digital television transport on IP networks. Appendix IX estimates transmission control protocol (TCP) file transfer performance on paths conforming to ITU-T Y.1541 objectives. Appendix X gives example calculations for combining delay variation measurements from multiple sections to estimate UNI-UNI performance; and Appendix XI estimates the packet loss requirement for digital circuit emulation.

Y.1543

Recommendation ITU-T Y.1543 specifies a set of IP performance parameters and methods of measurement applicable when assessing the quality of packet transfer on inter-domain paths. The methods anticipate that there will be multiple measurement systems, each conducting measurements of a segment of the customer-to-customer path, and recommend configurations that should produce useful results in this cooperative scenario. The methods rely on existing parameter definitions and encompass both active and passive measurement techniques.

Y.1544

Recommendation ITU-T Y.1544 extends the framework of Recommendation ITU-T Y.1540 to the point-to-multipoint, or multicast case. It also expands the key Y.1540 concepts with details needed to define parameters for the point-to-multipoint configuration. The performance of point-to-multipoint packet transfer to a set of destinations can first be considered a set of point-to-point packet transfers, and characterized using any or all of the point-to-point parameters found in Recommendation ITU-T Y.1540. This Recommendation defines parameters that are specific to the point-to-multipoint case. There are three general categories of point-to-multipoint parameters which focus on different entities in this network topology: parameters that describe the source performance, parameters that describe the performance at one or more destinations, and parameters that can be applied to describe the performance of subsections of the multicast tree. In its present version, this Recommendation primarily addresses destination performance. It also specifies a complete set of parameters for the access and disengagement phases of communication. This aspect goes beyond the scope of Recommendation ITU-T Y.1540, which covers only the information transfer phase.

Y.1560

This Recommendation defines the end-to-end TCP performance in terms of speed, accuracy and dependability in an IP-based network with middleboxes, which are nodes terminating TCP connections.

Y.1561

This Recommendation defines parameters that may be used in specifying and assessing the speed, accuracy, dependability and availability performance of packet transfer over a label switched path on an MPLS network. The defined parameters apply to end-to-end, point-to-point and multipoint-to-point LSP and to any MPLS domain that provides, or contributes to the provision of, packet transfer services. Two categories of MPLS networks are considered:

1) TE-LSP: traffic engineering label switched path, or configured LSP. These are point-point paths.
2) LDP-based LSP: This includes point-to-point and multipoint to point LSPs.

Y.1563

Recommendation ITU-T Y.1563 defines parameters that may be used in specifying and assessing the speed, accuracy, dependability and availability performance of Ethernet frame transfer in an Ethernet communication service. The defined parameters apply to end-to-end, point-to-point connections and multipoint connectivity in the Ethernet layer, and to the network portions that provide, or contribute to the provision of, such service in accordance with the normative references specified in clause 2.

Y.1564

This Recommendation defines a test methodology that may be used in assessing the proper configuration and performance of an Ethernet network to deliver Ethernet-based services. This out-of-service test methodology was created to provide service providers with a standard way of measuring the performance of Ethernet-based services.

Y.1565

Recommendation ITU-T Y.1565 describes the performance model, reference events and performance parameters for generic home networks and their interfaces to operators' broadband access networks. It augments existing information on packet performance parameters in ITU-T Recommendations. IPv4 and IPv6 are both within its scope, as is the possibility to perform network address and port translation, allowing the use of private address space in the home network. These parameters are also applicable to non-IP networks, such as Ethernet VLANS and IEEE 802.11 wireless networks.

G.8013/Y.1731

OAM functions and mechanisms for Ethernet-based networks.

G.8113.1

Recommendation ITU-T G.8113.1/Y.1372.1 specifies mechanisms for user-plane operations, OAM in MPLS TP networks to meet the MPLS-TP OAM requirements defined in IETF RFC 5860. It also specifies the MPLS-TP OAM packet formats, syntax and semantics of MPLS-TP OAM packet fields.

The OAM mechanisms defined in this Recommendation assume common forwarding of the MPLS TP user packets and MPLS-TP OAM packets. In transport networks, the OAM return path is always in band.

The MPLS-TP OAM mechanisms described in this Recommendation apply to co-routed bidirectional point-to-point MPLS-TP connections. Unidirectional point-to-point and point to multipoint MPLS-TP connections will be addressed in a future version of this Recommendation.

This Recommendation is compliant with the transport profile of MPLS as defined by IETF. In the event of a misalignment in MPLS-TP-related architecture, framework and protocols between this ITU-T Recommendation and the normatively referenced IETF RFCs, the RFCs will take precedence.

G.8113.2

Recommendation ITU-T G.8113.2/Y.1372.2 specifies OAM mechanisms based on the tools defined for MPLS for data-plane OAM in MPLS-TP networks. It also specifies the MPLS-TP OAM packet formats, syntax and semantics of MPLS-TP OAM packet fields. The OAM mechanisms defined in this Recommendation assume common forwarding of the MPLS-TP user packets and MPLS-TP OAM packets.

6.7 Elective Module EM 2.1: Hands-free communication and user interfaces in vehicles

Module objectives

This module will provide an understanding of user scenarios, test methods and requirements for hands-free communication in motor vehicles. This includes the testing of complete systems and subsystems of hands-free microphones and of telephones with short range wireless transmission links used to transmit the speech signals from the hands-free system to the mobile network.

Target audience

Car manufacturers, terminal manufacturers, test houses, students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire knowledge which is essential for professional involvement in the assessment of QoS and QoE of hands-free communication in cars or in the evaluation the results derived from such tests from different sources.

P.1100

This Recommendation, the aim of which is to define usage cases and test methods for narrowband hands-free communication in vehicles, covers:

- built-in hands-free systems,
- after-market hands-free car kits,
- corded headsets, and
- wireless headsets

to be used in vehicles for communication while driving.

It addresses the testing of complete systems as well as the subsystems of hands-free microphone and telephones with short-range wireless transmission links used to interconnect the hands-free system to the mobile network.

For testing, the test setup and recommended environmental conditions are described.

The methods, analysis and performance parameters described in this Recommendation are based on test signals and test procedures as defined in Recommendations ITU-T P.50, P.501, P.502, P.340 and P.380.

P.1110

This Recommendation, the aim of which is to define user scenarios, test methods and requirements for wideband hands-free communication in motor vehicles, covers:

- built-in hands-free systems,
- after-market hands-free car kits,
- corded headsets and
- wireless headsets

to be used in motor vehicles for communication.

Compatibility between narrowband and wideband implementations is also addressed.

The Recommendation addresses the testing of complete systems as well as the subsystems of hands-free microphones and telephones with short-range wireless transmission links used to transmit the speech signals from the hands-free system to the mobile network.

For testing purposes, the test set-up and recommended environmental conditions are described.

The methods, analysis and performance parameters described in this Recommendation are based on test signals and test procedures, as defined in Recommendations ITU-T P.50, P.501, P.502, P.340 and P.380 and ETSI ES 202 739, and ETSI ES 202 740.

6.8 Elective Module EM2.2: Traffic management

Module objectives

This module provides information on end-to-end QoS, performance and resource management of packet-based networks. There is an ongoing need for guidance on general transmission planning and keeping it up to date with technological evolution. Especially in light of the migration of modern telecommunication networks towards packet-based technologies, replacing traditional circuit-switched systems, guidance is needed on transmission planning with respect to heterogeneous and interconnected networks.

With the increasing industry focus on packet-based networks, there is a need for guidance on the associated end-to-end QoS, performance and resource management issues for multimedia services (e.g. voice, video and data) carried by packet-based networks, in order to ensure customer satisfaction. This includes interworking aspects between different types of network (e.g. cellular, wireless, wireline) using various packet-based technologies (including IP, ATM, Ethernet and MPLS), as well as apportionment of performance objectives between different network segments.

This module is related to Question 11/12 (Performance interworking and traffic management for next-generation networks). Related Recommendations include ITU-T Y.1221, Y.1222, Y.1223, Y.1530, Y.1531 and Y.1542 (also related to IETF technologies such as DiffServ/Intserv/RSVP/RSVP-TE).

Target audience

Network operators, test-equipment manufacturers, test houses, regulators and students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire knowledge which is essential for any technical work or decision making related to traffic management in IP based networks.

Y.1221

Recommendation ITU-T Y.1221 provides a general description as well as objectives and procedures for traffic control and congestion control for IP-based networks. In particular, it describes the concepts of the traffic contract between a user and the network. It specifies the IP transfer capabilities (IPTCs) including, for each IPTC, the service model, the associated traffic patterns and conformance definition.

Y.1222

Recommendation ITU-T Y.1222 provides a general description of, and procedures for, traffic control and congestion control in Ethernet-based networks. It describes the concepts of the traffic contract between a user and the network. It specifies the Ethernet transfer capabilities (ETCs) including, for each ETC, the service model, the associated traffic patterns and conformance definition for an Ethernet flow that are observable at any point in the network.

Y.1223

In order to transport IP flows with assured end-to-end quality and reliability in a multi-provider environment, coordinated decisions must be made regarding the admission, policing and assignment of resources to particular offered flows. To do this, a uniform way of characterizing such IP flows is needed, together with some shared decision rules for handling them. Recommendation ITU-T Y.1223 defines a set of IP flow specifications that could offer a basis for such cooperation.

Y.1530

Recommendation ITU-T Y.1530 defines performance parameters and objectives for point-to-point call processing for voice service provided by hybrid IP networks. The parameter definitions are based on the principles and generic performance parameters defined in Recommendation ITU-T I.350, and make use of relevant definitions from ISDN call processing performance Recommendations where appropriate.

Y.1531

Recommendation ITU-T Y.1531 defines three performance parameters that may be used in specifying, measuring and comparing the speed, accuracy and dependability of call setup processing in networks that employ the session initiation protocol (SIP), together with other protocols, in establishing and terminating media sessions ("calls") between users. The parameters may also be used in call-processing performance apportionment or accumulation. This Recommendation does not specify numerical performance values.

Y.1542

Recommendation ITU-T Y.1542 considers various approaches for achieving end-to-end (UNI-UNI) IP network performance objectives. Detailed examples are provided as to how some approaches might work in practice, including how service providers might handle cases where the aggregated impairments exceed those specified in a requested QoS class (such as those of Recommendation ITU-T Y.1541). The advantages and disadvantages of each approach are summarized.

6.9 Elective Module EM3.1: QoS for mobile services

Module objectives

This module provides information on QoS for mobile services such as definition of QoS parameters and their computation, procedures and requirements for QoS measurement equipment, etc. It is based on draft Recommendation E.MQoS (QoS aspects for popular services in mobile networks), which is currently under development. QoS issues for mobile services are becoming more and more important for network operators as well as regulators.

Target audience

Mobile operators, test equipment manufacturers, test houses, regulators and students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire knowledge which is essential for any technical work or decision making related to mobile QoS as seen from an end-user perspective, by contrast to the mobile network QoS management in operation and maintenance centres.

E.804

This new Recommendation provides sets of QoS parameters from an end-user perspective for the operational aspects of mobile communication. And since services per se are not standardized, it focuses on popular services, which means commonly or widely-used services.

This does not preclude applying the definitions in this Recommendation for other (not widely-used) services, where feasible.

6.10 Elective Module EM3.2: Bit-rate measurement of Internet connections

Module objectives

Unified methodology of Internet speed quality measurement usable by end users on the fixed and mobile networks (currently available as a working draft under Question 15/11).

Target audience

Network operators, test equipment manufacturers, test houses, regulators and students in the final year of engineering or ICT-related courses.

Learning outcomes

Those following this module will acquire knowledge that is essential for any technical work or decision making related to assessments and optimization based on bit-rate measurements in IP-based networks.

ITU experts are discussing ways to meet user and operator demand for a standardized methodology for measuring Internet access speeds. The Bundesnetzagentur (Federal Network Agency), in Germany, the testing laboratory of the Central Research Telecommunication Institute (ZNIIS,) in Russia, and a vendor of telecom testing equipment, Arcatech, in the UK, are promoting a new work item on this issue.

Nowadays, most telecommunication networks are based on IP technologies, where QoS plays a key role. Carriers (operators) are focusing on IP technology, inasmuch as their networks provide access to Internet resources and Internet-based services predominate in their traffic. This obviously has a significant negative impact on operator telecommunication services.

From all these factors, it is clear that independent measurement of the quality of Internet resources has become a high-level task for all ICT players, including content providers, service providers, regulators, etc. The measurement approach consists of two parts: the assessment of Internet access speed and assessment of access to Internet resources.

Various organizations have been addressing this issue, and several methods of Internet speed testing have already been developed. At the same time, some operators provide their services based on third-party measurement results (for example, www.speedtest.net, www.speed.io, etc.). Also, some European regulators and a number of operators have launched their own projects aimed at resolving this issue (e.g. a project being pursued by the University of the Basque Country (Spain) and Telefónica, Germany speed test systems, VoIP Test, Hyperiontest).

However, these methods are not unified, and the framework and testing results cannot be used by other countries, especially developing countries. In addition the existing methods cannot be used to maintain the service level agreement (SLA) for the following reasons:

- Not all of the existing methods have been developed by international SDOs, resulting in a lack of transparency for all ICT players, especially regulators and operators.
- The lack of a unified measurement framework means that there is no guarantee that testing results include only the operator network and no other Internet segment.
- The lack of unified measurement principles.

A working draft is currently available under Question 15/11 as COM 11 – C 0096-E.

Annex 1: Possible laboratory/practical exercises and demonstrations on voice, audio and multimedia QoS measurements and QoE aspects

Laboratory exercises for OM1 introduction - QoS and QoE

Elementary subjective experiments – e.g. listening to different speech and music fragments degraded with elementary impairments and their combinations: background noise, quantizing and (multiple-) coding distortion, jitter delay and relevant PLC response, etc. Masking effects demonstration (background noise vs. coding distortion, listening level vs. background and room noise, etc.).

*Requirement*s: Sets of predefined speech samples, HiFi headphones, quiet listening environment, presentation HW and SW (e.g. laptop and media player).

Laboratory exercise for OM2 – Subjective assessment of voice quality

Advanced subjective experiments – basics of audiometry and principles of human hearing (human ear sensitivity and its dependency on age; frequency response and its dependency on age; temporal and frequency masking demonstrations; etc.), listening tests as per P.800 and P.835, speech sample preparation, randomization and presentation, data processing, conversational tests as per P.800 and P.805, scenario preparation, test handling and data processing, listening tests for higher-quality samples (BS.1116 and similar), differences in perception driven and influenced by location (e.g. environmental noise), situation (e.g. expectation), mood (positive/negative manipulation), listening level and terminal frequency response influence on listening test results, etc.

Requirements: Sets of clean speech samples and noise recordings, defined high-quality headphones for sample presentations, defined listening environment as per P.800, SW for sample edition and play-out.

Laboratory exercise for OM3 – Objective assessment of voice quality

Experiments with PESQ and POLQA demo SW on impaired samples as above. Analysis of impairments that are/are not detectable by the algorithm. Algorithm sensitivity to “listening” levels and proper frequency pre-filtering and its consequences for their correct application. Detailed (optional) algorithm output parameter analysis and understanding. Statistical processing of results, reporting.

Requirements: Sets of predefined speech samples, algorithm demo versions for offline sample processing, presentation HW and SW as an option (e.g. laptop and media player).

Laboratory exercise for OM4 – QoS and QoE for multimedia and assessment methods

QoS impact on final QoE, different aspects of QoE, QoE manipulation techniques, long-term and age-dependent QoE changes and developments.

Requirements: Sets of predefined speech samples and marketing scenarios/situations, environment for relevant testing techniques.

Laboratory exercise for EM1.1 – Telephonometry

Sensitivity, linearity and frequency response of terminals, their measurement on HATS and their influence on perceived QoS.

Requirements: Sets of predefined speech samples, samples of various terminals, headsets and headphones, HATS, SW for sample edition, playout and recording.

Laboratory exercise for EM2.1 – Hands-free communication and user interfaces in vehicles

Sensitivity, linearity and frequency response of hands-free terminals, their measurement on HATS and their influence on perceived QoS.

Requirements: Sets of predefined speech samples, samples of various hands-free terminals, HATS, SW for sample edition, playout and recording

Abbreviations

3GPP	3rd Generation Partnership Project
ACR	absolute category rating
ADPCM	adaptive differential pulse code modulation
ANSI-C	American National Standards Institute C
ARAEN	volume meter specified by CCITT
ATM	asynchronous transfer mode
BDT	Telecommunication Development Bureau
BEREC	Body of European Regulators for Electronic Communications
BR	Radiocommunication Bureau
CCITT	Comité Consultatif International Téléphonique et Télégraphique
CCR	comparison category rating
CD-ROM	Compact Disc – read-only memory
CETVSQ	continuous evaluation of time-varying speech quality
CME	circuit multiplication equipment
CoE	Centre of Excellence
ComSoc	Communication Society (of IEEE)
DAM	diagnostic acceptability measure
DCME	digital circuit multiplication equipment
DSI	digital speech interpolation
ECTS	European Credit Transfer and Accumulation System
EM	elective module
EPFL	École Polytechnique Fédérale de Lausanne
ETC	Ethernet transfer capabilities
ETSI	European Telecommunications Standards Institute
EU	European Union
FTW	Forschungszentrum Telekommunikation Wien - Telecommunications Research Center Vienna
G	G-series of ITU-T Recommendations
GAT	group audio terminals
GPS	Global Positioning System
HATS	head and torso simulator
HiFi	high fidelity
HR	higher rate
HW	hardware
ICT	information and communication technology
IEC	International Electrotechnical Commission
IEEE	Institute of Electrical and Electronics Engineers

IET	Institution of Engineering and Technology
IETF	Internet Engineering Task Force
IMS	Internet Multimedia Subsystem
IP	Internet Protocol
IP-CME	circuit multiplication equipment optimized for Internet Protocol
IPTV	television over Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
IRS	intermediate reference system
ISDN	integrated services digital network
ITR	International Telecommunication Regulations
ITU	International Telecommunication Union
ITU-D	International Telecommunication Union – Development Sector
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
KPI	key performance indicator
LD-CELP	low-delay code excited linear prediction
LR	lower resolution
LTS	local telephone system
MNRU	modulated noise reference unit
MOS	mean opinion score
MOS-LQO	mean opinion score - listening quality objective
mp3	MPEG-1 or MPEG-2 audio layer III
MPLS	multi-protocol label switching
MPLS TP	multi-protocol label switching transport profile
NCTT	network contribution to transaction time
NGN	next-generation network
NGO	non-governmental organization
NNI	network-to-network interface
NOSFER	new fundamental system for the determination of reference equivalents
NP	network performance
OAM	operation, administration and maintenance
OLR	overall loudness ratio
OM	obligatory module
P	P-series of ITU-T Recommendations
PCAP	packet capture format
PCM	pulse-code modulation
PCME	packet circuit multiplication equipment

PESQ	perceptual evaluation of speech quality
PLC	packet loss concealment
POLQA	perceptual objective listening quality assessment
PSTN	public switched telephone network
PQ	perceptual quality
Q	Question, or Q-series of ITU-T Recommendations
QoE	quality of experience
QoS	quality of service
QoSTP	quality of service training programme
RFC	Research Task Force
RLR	receiving loudness ratio
RSVP	Resource reSerVation Protocol
RSVP-TE	Resource reSerVation Protocol - Traffic Engineering
SDO	standards development organization
SG	study group
SIP	session initiation protocol
SLR	sending loudness ratio
SPDA	speech processing devices for acoustic enhancement
STQ	speech and multimedia transmission quality
SW	software
TIPHON	Telecommunications and Internet Protocol Harmonization Over Networks
TCL	terminal coupling loss
TCP	transmission control protocol
TU	technical university
UK	United Kingdom
UNI-UNI	path between user network interfaces
US	United States
VDE	Verband der Elektrotechnik, Elektronik und Informationstechnik
	German Association for Electrical, Electronic & Information Technologies
WCET	Wireless Communication Engineering Technologies Certification Programme
Y	Y-series of ITU-T Recommendations
ZNIIS	Russian Central Scientific and Research Institute of Communication (Центральный научно-исследовательский институт связи)

References

1. ITU-T Handbook on Quality of Service and Network Performance, 2004
2. ITU-T Handbook on Practical Procedures for Subjective Testing , 2011
3. ITU-T Handbook on Telephonometry, 1987
4. Raake, A.: Speech Quality of VoIP: Assessment and Prediction, Wiley, January 2007, ISBN: 978-0-470-03299-2
5. Marchese, M.: QoS Over Heterogeneous Networks, Wiley, 2007, ISBN 978-0-470-01752-4
6. Hardy, W.: VoIP Service Quality: Measuring and Evaluating Packet-Switched Voice, McGraw-Hill 2003, ISBN 978-0-071-41076-2
7. Wallace, K.: Implementing Cisco Unified Communications Voice over IP and QoS (CVOICE), CCIE 7945, ciscopress.com
8. Cisco Networking Academy, www.cisco.com/web/learning/netacad/index.html
9. www.wraycastle.com/course/telecoms-training/ip-networks-and-protocols-training/ip-quality-of-service-161
10. www.technology-training.co.uk/qosinipandmplsnetworks_28.php
11. www.telecomsacademy.com/
12. www.opticom.de/products/services.php
13. www.swissqual.com/index.php/services/training.html
14. www.mesaqin.com/training.html
15. www.theiet.org/membership/career/index.cfm
16. www.ieee.org/education_careers/education/accreditation/ctaa/ieee_role.html
17. ECTS Users' Guide. European Commission, Education and Culture DG, 2009, http://ec.europa.eu/education/lifelong-learning-policy/doc/ects/guide_en.pdf
18. European Erasmus Mundus Master Courses, http://eacea.ec.europa.eu/erasmus_mundus/results_compendia/selected_projects_action_1_master_courses_en.php
19. www.itu.int/en/ITU-T/Workshops-and-Seminars/qos/201309/Pages/default.aspx
20. www.itu.int/en/ITU-T/Workshops-and-Seminars/qos/201307/Pages/default.aspx
21. www.itu.int/en/ITU-T/Workshops-and-Seminars/qos/201207/Pages/default.aspx
22. www.itu.int/ITU-T/worksem/qos/201007/index.html
23. REPORT ON SPECTRUM MANAGEMENT TRAINING PROGRAMME (SMTP), ITU-R Human Capacity Building Division, Geneva, 2013
24. http://Mead.ch/mead
25. Ofcom approach to net neutrality, online at http://stakeholders.ofcom.org.uk/consultations/net-neutrality/statement/
26. ITU GSR10 Best Practice Guidelines for Enabling Open Access, www.itu/int/ITU/treg/bestpractices.html
27. www.itu.int/md/T09-CWG.WCIT12-INF-0005/en

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ISBN 978-92-61-15481-3

9 789261 154813

Printed in Switzerland
Geneva, 2014
Photo credits: Shutterstock

12/2014