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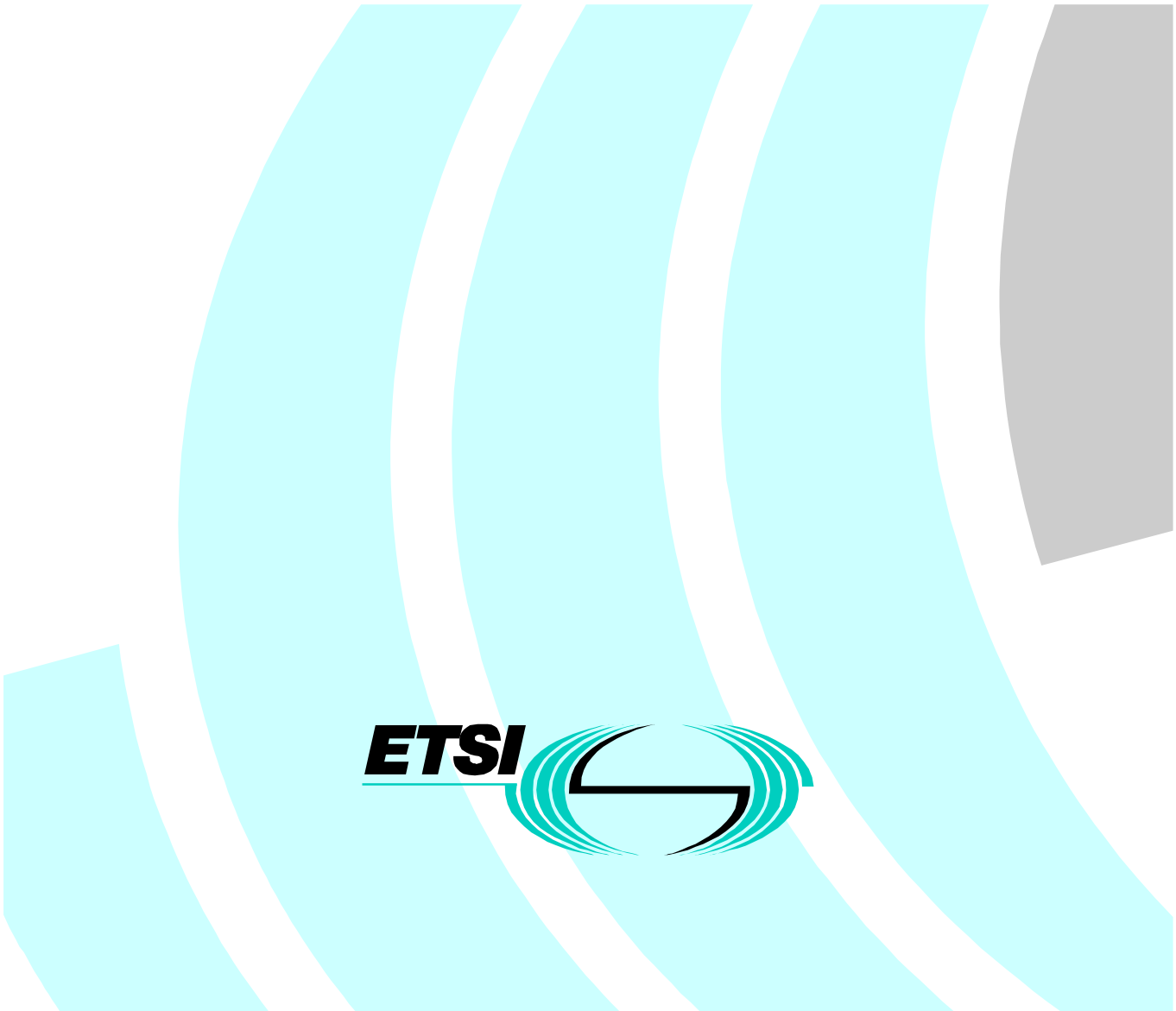
Technical Report

Access and Terminals (AT);
Study of the implications of standardization of IP terminals utilizing Ethernet, PSTN and ISDN connection methods



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Contents

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Access and Terminals (AT).

Introduction

With new telephone infrastructures coming up as telephony over CATV networks, Intranets, the Internet we also see IP-Telephone terminals, connected via an Ethernet interface, replacing the legacy analogue and digital ISDN phones. To secure the overall quality of the communication system and to maintain certain level of functionality, as e.g. emergency functions, it will be necessary to agree on standards for these terminals.

For the standardization of IP terminals of course not all work has to be redone. A lot can be learned from what has been established for the analogue and ISDN telephony terminals, see clause 5 for an overview. Further, also other standardization organizations are active in the area of standards for IP terminals, see clause 6 for an overview.

The purpose of the present document is to identify possible open gaps in the standardization process for IP terminals and to make proposals for new work items if needed. The present documents aims also to facilitate to the experts information on the presently standardization work related with terminals.

1 Scope

The present document studies the implications of standardization of IP Terminals utilizing Ethernet, PSTN and ISDN connection methods.

2 References

For the purposes of this Technical Report (TR) the following references apply:

[1] ETSI EN 301 234 (V1.2.1): "Digital Audio Broadcasting (DAB); Multimedia Object Transfer (MOT) protocol".

[2] ETSI TBR 21: "Terminal Equipment (TE); Attachment requirements for pan-European approval for connection to the analogue Public Switched Telephone Networks (PSTNs) of TE (excluding TE supporting the voice telephony service) in which network addressing, if provided, is by means of Dual Tone Multi Frequency (DTMF) signalling".

[3] ETSI TBR 38: "Public Switched Telephone Network (PSTN); Attachment requirements for a terminal equipment incorporating an analogue handset function capable of supporting the justified case service when connected to the analogue interface of the PSTN in Europe".

[4] ETSI TBR 003: "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic access".

[5] ETSI TBR 008: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Attachment requirements for handset terminals".

[6] ITU-T Recommendation G.177 (1999): "Transmission planning for voiceband services over hybrid Internet/PSTN connections".

[7] ITU-T Recommendation H.248 (2000): "Gateway control protocol".

[8] ITU-T Recommendation G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

[9] ITU-T Recommendation G.723 (1988): "Extensions of Recommendation G.721 adaptive differential pulse code modulation to 24 and 40 kbit/s for digital circuit multiplication equipment application".

[10] ITU-T Recommendation G.729 (1996): "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)".

[11] ITU-T Recommendation G.108 (1999): "Application of the E-model: A planning guide".

[12] ITU-T Recommendation H.323 (2000): "Packet-based multimedia communications systems".

[13] IEEE 802.3: "IEEE Standard for Information technology--Local and metropolitan area networks--Part 3: Carrier sense multiple access with collision detection (CSMA/CD) access method and physical layer specifications.

[14] ANSI/TIA/EIA 464-B: "Business TeleCommunications (BTC) - Comparison of PBX transmission requirements in standards".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

Public telecommunications network: telecommunications network used to provide publicly available telecommunications services

Copied from:

Brussels, 12 July 2000, COM(2000)393, Proposal for a "DIRECTIVE OF THE EUROPEAN PARLIAMENT AND OF THE COUNCIL on a common regulatory framework for electronic communications networks and services".

Article 2:

Definitions:

For the purposes of this Directive:

a) 'electronic communications network' means transmission systems and, where applicable, switching or routing equipment and other resources which permit the conveyance of signals by wire, by radio, by optical or by other electromagnetic means, including satellite networks, fixed (circuit- and packet-switched, including Internet) and mobile terrestrial networks, networks used for radio and television broadcasting, and cable TV networks, irrespective of the type of information conveyed;

b) 'electronic communications service' means services provided for remuneration which consist wholly or mainly in the transmission and routing of signals on electronic communications networks, including telecommunications services and transmission services in networks used for broadcasting, but excluding services providing, or exercising editorial control over, content transmitted using electronic communications networks and services;

c) 'public communications network' means an electronic communications network used wholly or mainly for the provision of publicly available electronic communications services.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ANSI	American National Standards Institute
ATM Forum	ATM Forum
CATV	Community Access Television
CEN	European Committee for Standardization
CENELEC	Committee European de Normalization Electrotechnique
DTMF	Dual Tone Multi Frequency
ECMA	Standardizing Information and Communication Systems
ECTF	Enterprise Computer Telephony Forum
ETSI	European Telecommunication Standardization Institute
ETSI-SPAN	ETSI-Services and Protocols for Advanced Networks
ETSI-TIPHON	ETSI-Telecommunication and Internet Protocol Harmonization Over Networks
ICTSB	Information and Communications Technologies (ICT) Standards Board
IEC	International Electrotechnical Commission
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IMTC	International Multimedia Teleconferencing Consortium
IP	Internet Protocol
IPv6 Forum	IPv6 Forum
ISDN	Integrated Services Digital Network
ISO	International Organization for Standardization
ISOC	Internet Society
ITU-T	International Telecommunication Union - Telecommunication standardization sector

NNI	Nederland's Normalization Institute
PAR	Project Authorization request
PISN	Private Integrated Services Network
POTS	Plain Ordinary Telephony Service
Softswitch	Softswitch consortium
TE	Terminal Equipment
TIA	Telecommunications Industry Association

4 The need for standards for IP terminals

With new telephone infrastructures coming up as telephony over CATV networks, Intranets, the Internet we also see IP-Telephone terminals, connected via an Ethernet interface, replacing the legacy analogue and digital ISDN phones. To secure the overall quality of the communication system and to maintain certain level of functionality, as e.g. emergency functions, it will be necessary to agree on standards for these terminals.

For the standardization of IP terminals of course not all work has to be redone. A lot can be learned from what has been established for the analogue and ISDN telephony terminals, see clause 5 for an overview. Further, also other standardization organizations are active in the area of standards for IP terminals, see clause 6 for an overview.

The purpose of the present document is to identify possible open gaps in the standardization process for IP terminals and to make proposals for new work items if needed. The present documents aims also to facilitate to the experts information on the presently standardization work related with terminals.

5 Available standards for most common terminals

In the legacy telecommunication network, the most common public offered interfaces are for the Public Switched Telephone Network:

- The analogue presented, commonly designated PSTN or POTS interfaces; and
- the digital presented, using ISDN techniques.

Typical basic standards for terminals to be connected to the above mentioned interfaces are:

- For analogue presented interfaces, defining basic harmonized terminal conditions for them to:
  - be connected to the network: TBR 21 [2],
  - inter-work via the network, if they support a simple voice service with a handset: TBR 38 [3].
- For digital presented interfaces, defining basic harmonized terminal conditions for them to:
  - be connected to the network TBR 003 [4],
  - inter-work via the network, if they support a simple voice service with a handset, TBR 008 [5].

Above referred standards are also very commonly used for private interfaces, e.g. to connect a terminal to a PABX branch interface.

Other standards and publications exist with more detailed information in these and related areas.

Standardized analogue and ISDN telephone terminals are used in public and private telecommunication systems. Besides these standardized terminals, also proprietary analogue and digital terminals are used.

The standards for analogue terminals take amongst others care of the following subjects:

- Acoustical terminal parameters as:
  - Loudness ratings;
  - Echo suppression.

- Line parameters;
- Impedance;
- Ring voltage;
- Ringing frequency;
- Open line voltage;
- Line signalling, DTMF.

The standards for ISDN terminals take amongst others care of the following subjects:

- Acoustical terminal parameters as:
  - Loudness ratings;
  - Echo suppression.
- Line protocol;
- ISDN line;
- Line power;
- Terminal power consumption:
  - Normal mode;
  - Restrictive mode operational/non operational.

The standards referred secure that the (voice) communication from endpoint to endpoint over the public network have a minimum acceptable QoS level, and that the terminals can interoperate with each other. By extension it is commonly assumed that simple private networks and installations do not significantly disturb the QoS offered by PSTN.

An other point in the standardization and in some purchasing requirements established by distribution houses or network operators or even in some national legislation's is the capability for an analogue/ISDN voice terminal to stay operational under power fail conditions, to be able to call emergency services.

6 Related standards and work in progress

This clause gives an overview of standardization bodies that are active in areas related to IP terminals and identifies active work items and standards and for IP terminals that are already in place. The information provided by this clause can be used to identify the open gaps in the standardization process for IP terminals, leading to proposals for future work in clause 7.

6.1 ETSI-TIPHON

ETSI's Project TIPHON defines standards and specifications to ensure the interoperability of services built on traditional voice networks with those employing the emerging next generation networks based on Internet Protocol (IP) technology. EP TIPHON specifications also assist in the transparent migration of those services to next generation networks.

The project is organized into eight separate working groups, each focused on a specific development phase or specialized topic:

- WG1 Service Capabilities;
- WG2 Network Architecture and Information Flows;
- WG3 Protocols and Profiles;

- WG4 Naming and Addressing;
- WG5 Quality of Service;
- WG6 Testing and Interoperability;
- WG7 Mobility;
- WG8 Security.

There is no particular work on IP terminals and the TB is willing to collaborate with TC AT in this area.

6.2 ETSI-STQ

The objective of STQ is to ensure the co-ordination, production (where appropriate) and maintenance of end-to-end speech quality related deliverables, for the timely and economic development of equipment for use with existing and future fixed/mobile network telecommunications service offerings from network operators. TC/STQ should form the 'horizontal' technical nucleus for all speech performance issues including the implications of all new coding techniques. TC/STQ will work in close co-operation with the allied EPs ATA and DTA, with TC/SMG and other radio related EPs. Other such Technical Bodies may be identified in the course of restructuring ETSI. TC/STQ will contribute to work relating to speech performance within the ITU-T and other standards bodies. Specifically, STQ will monitor, contribute to and give guidance on:

- the speech transmission characteristics of networks and terminals including acoustic interfaces;
- the definitions for speech processing techniques (speech recognition, echo cancellation, coding, etc);
- the impact on the transmission quality of speech processing devices, as defined above, implemented in networks and terminals; Impacts and consequences of those equipment on the transmission quality of other types of signals;
- the specification of **end-to-end** speech transmission quality in fixed, mobile networks and interworking situations;
- the definition of appropriate test methods and/or test equipment (e.g. INMD);
- the validation of existing transmission rating models, maintaining existing or creating new deliverables as appropriate;
- the impact and limits of the individual transmission parameters (delay, echo, signals and noise levels, impairments - including coding impairments, etc);
- the relationships between signalling data and transmission quality aspects.

TC/STQ will not produce deliverables, where work is within the Terms of Reference of another Technical Body.

6.3 ETSI-SPAN

ETSI SPAN (Services and Protocol for Advanced Networks) is ETSI's core competence centre for fixed networks standardization including IP based networks. It is responsible for all aspects of standardization for present and future converged networks including mobility aspects within fixed networks, using existing and emerging technologies, in line with, and driven by, the commercial objectives of the ETSI membership. This will be accomplished in close co-operation with other ETSI TB's and external standardization activities.

SPAN's core competence is the definition of general network and service aspects for all existing and new wireline access and core networks. Definition of information transport capabilities, signalling requirements, protocol design and associated test specifications. Standardization may also be based on requirements from other Technical Bodies or external bodies. Matters of consistency between public and corporate networks and between fixed and mobile networks including ensuring that standards take account of security and regulatory requirements.

NB: these activities cover both circuit- and packet-switched networks including IP and ATM technologies.

Activities of SPAN include:

- Network architecture and its evolution;
- Network dimensioning;
- Service descriptions from the user perspective (excluding details of the human interface);
- Relations between services and network; network capabilities for service provision and interoperability of services (e.g. for Service Provider Access);
- Functional capabilities and information flows needed to support services;
- Service interworking;
- Technical requirements on terminals and network components to support the implementation of services, including the support of mobile services;
- Switching functions and switching systems for public networks;
- Common channel signalling systems and signalling networks;
- Access Networks and protocols;
- Numbering, naming, addressing and routeing;
- Interworking between different network types;
- Interworking of protocols and signalling systems;
- Charging capabilities of circuit and packet switched networks;
- Quality of Service and Network Performance;
- Resource management e.g. congestion control in IP based networks;
- Network intelligence;
- Universal Mobility, including global mobility aspects in fixed networks;
- Specification of protocols and the means of testing those protocols;
- Operational, maintenance and technical performance requirements of switching and signalling systems including testing requirements.

6.4 ECMA TC32-TG17

ECMA TC32-TG17 develops Standards and Technical Reports for IP-based multimedia communications in a business environment. Emphasis of their work is on:

- the interconnection of PISN components via IP networks
- the interworking of PISNs and IP networks via a gateway

No particular work on IP terminals is being done in ECMA TC32 TG17.

6.5 ITU-T

6.5.1 SG12

The terminal standard P.31x is in an early state. For the new Study Period a special question (B/12) was created. Therefore a new wider approach is expected in February 2001.

For the time being the most important terminal parameters are captured in ITU-T Recommendation G.177 [6]. Actually it is a network standard but there was a special agreement between the rapporteur of Question 23/12 and rapporteur of Question 9/12 for this approach.

6.5.2 SG16

H series

SG16 is the home of all ITU-T Multimedia standards. The H-Series standards with the protocol definition. In the moment there is a big discussion on what protocol family should used (H-Series vs. IETF-Protocols).

NOTE 1: There is a heavy debate.

NOTE 2: In one case there was a common standardization approach MEGACO/ ITU-T Recommendation H.248 [7].

3GPP already decided to use SIP, now there is a debate on the constraints on this decision. A migration towards All-IP is discussed in 3GPP2.

CODECs

ITU-T SG16 CODECs ITU-T Recommendation G.711 [8], ITU-T Recommendation G.723 [9], ITU-T Recommendation G.729 [10] are mostly used in IP-Phones of today.

Since ITU-T Recommendation G.723 [9] and ITU-T Recommendation G.729 [10] have got quality disadvantages (caused by delay) but low bit-rate codecs are needed for some services, in the IP-community the AMR-Codec defined by GSM/3GPP has attracted attention. In the moment the Equipment Impairment Factor is not available. Therefore the E-Model cannot be used (ITU-T Recommendation G.108 [11]) for Voice Quality Discussions. On the other hand there is no negative voice concerning the AMR quality. Ii is assumed that the Ie-factor will be determined on short notice in the SG12 meeting.

Beside this traditional approach of 3,1kHz voice telephony in TIPHON there are discussions on new approaches concerning voice quality aspects based on the idea that in the internet a 3,1kHz restriction is not necessary any more.

6.6 TIA

As the communications industry rushes to provide homes and businesses with reliable two-way voice, video and data transmission, TIA's User Premises Equipment Division (UPED) - and more specifically, **Engineering Committee TR-41** User Premises Telecommunications Equipment Requirements - has kept pace by launching projects to standardize system functionality of Internet protocol (IP)-telephony infrastructures.

VOIP Standards Development

In October 1998, TR-41 announced the initiation of a project (PN-4352) in Engineering Subcommittee TR-41.3, Analog and Digital Wireline Telephones, to create a voice-transmission performance standard for narrowband digital telephones that includes products used for Internet telephony. The scope of the standard includes handset, headset and handsfree audio performance, and encompasses IP-based interfaces, as well as pulse code modulation (PCM)-based interfaces, with codecs of 64 kbit/second and lower. PN-4352 (soon to be published as TIA/EIA/IS-810) also includes quality of service aspects of voice-over-IP (VOIP). This is the first all-encompassing standard in the world for audio performance of narrowband digital telephones.

An exciting aspect of IP telephony is the possibility of wideband (150 to 6 800 Hz) telephony. Building on the narrowband PN-4352 project is a new wideband project, which will be starting in November 1999.

A new working group, TR-41.1.2, Voice Quality Over IP, was started in December 1998 to address end-to-end voice quality over an IP/switched-circuit network (IP/SCN). This project will build on the European Telecommunications Standards Institute Telecommunications and Internet Protocol Harmonization Over Networks (ETSI TIPHON) and IP/SCN network connection scenarios. The end-to-end voice quality of the various scenarios will be analysed using the "E-Model," a computer tool widely used by the industry for transmission planning, and will build on private network transmission planning guidelines in TIA/EIA/TSB [Telecommunications Systems Bulletin] 32-A. Quality-of-service categories will be defined, taking into account the impairments introduced by IP networks. This project includes collaboration with ETSI Speech Transmission Quality (ETSI STQ).

Also in December 1998, the parent subcommittee, TR-41.1, Multiline Terminal Systems, began work to revise the private branch exchange (PBX)-loss plan to include IP telephones and gateways. It is part of a revision of the PBX performance standard, TIA/EIA-464-B [14], which is being addressed in PN-3673.

In January of this year, TIA started developing a standard for IP telephone interoperability in TR-41.3.4, VOIP Terminals. TR-41 initiated this project to address the convergence of voice- and IP-based networks and the resulting need for standards-based telephones for use on IP networks, so that telephones from different suppliers can be interchanged.

The project, (PN-4462) (soon to be published as TIA/EIA/IS-811), defines the "whole device" standard for IP telephones and includes voice-transmission requirements, ensure consistent speech performance physical connectors, Ethernet interface, safety considerations, the basic set of supplementary services for IP business telephones and the gateway-to-telephone options-control protocols. so that telephones from different suppliers can be interchanged. The new standard will reference several existing standards. Voice-quality aspects are governed by PN-4352, a voice transmission standard for Internet telephony. Control options include ongoing efforts such as the Internet Engineering Task Force (IETF) Megaco Protocol (Media Gateway Control), the IETF Session Initiation Protocol (SIP), and the International Telecommunication Union (ITU) H.323 [12]; protocol suite.

In an effort to promote global acceptance of the standard, Subcommittee TR-41.3 is working with IP network standards developing organizations (SDOs) such as IETF and ETSI STQ.

New IP-Centred Subcommittee:

To date, little has been done to standardize system functionality of IP-telephony infrastructures, while there has been extensive development of peripheral device standards for IP telephony systems in other standards developing organizations and industry fora such as ETSI, the IETF and the ITU.

As a result, TIA formed a new subcommittee in July dedicated to developing standards for IP telephony gateways, infrastructure and interworking. The new subcommittee, TR-41.4, IP Telephony Infrastructure and Interworking Standards, is part of TR-41.

TR-41.4 will deal with VOIP standards aspects of enhanced-911, enterprise-to-public-switched-telephone-network-gateway equipment standards, interworking between dissimilar islands of control, and interworking between peer and master/slave-controlled telephone stimulus-to-peer networking proxy equipment standards.

The subcommittee specifically will focus on VOIP connectivity and interoperability between endpoints within a single enterprise and its service provider networks. The area of interest does not extend to the external service provider networks.

- TIA engineering Committee TR-41;
- User Premises Telecommunication Requirements;
- TIA Subcommittee TR-41.1;
- Multiline Terminal Systems;
- TIA-4689 V2.0 TIA TSB;
- Telecommunications;
- Terminal equipment;
- Voice Quality Recommendations for IP Telephony.

Discusses effects of delay, coders ITU-T Recommendation G.711 [8], ITU-T Recommendation G.729 [10], ITU-T Recommendation G.723 [9], Echo and echo cancellation compared to the user satisfaction. R rating model is used.

- Standard set up for North America IP telephony;
- TIA engineering Committee TR-41.

User Premises Telecommunication Requirements:

- PN-4462:
  - Telecommunications;
  - Telephone terminal equipment;
  - Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Phones;
  - Reference to Acoustic performance requirements set out by TIA PN-4352.

6.7 IEEE 802,3af

The IEEE 802,3af [13] committee is very active on the powering issue of IP equipment over Ethernet, see the parts of the Project Authorization Request below, and the web site http://grouper.ieee.org/groups/802/3/af/.

IEEE-SA Standards Board Project Authorization Request (PAR) (1999-Rev 1)

TITLE: [Information technology—

Telecommunications and information exchange between systems—

Local and metropolitan area networks—

Specific requirements—

Part 3: Carrier sense multiple access with collision detection (CSMA/CD)

access method and physical layer specifications

Data Terminal Equipment (DTE) Power via

Media Dependent Interface (MDI)]

Scope of Proposed Project:

Define methodology for the provision of power via balanced cabling to connected Data Terminal Equipment with 802.3 interfaces. The amount of power will be limited by cabling physics and regulatory considerations. Compatibility with existing equipment will be considered.

Purpose of Proposed Project:

To provide power for a new class of devices with 802.3 interfaces enabled by progress in silicon technology. These devices are characterized by low power requirements and LAN connectivity.

Analysis of IEEE 802,3af [13**]:**

The Work in this workgroup Power via Media Dependent Interface (MDI). Focuses on the maximum amount of power send to a terminal. The detection whether a terminal should receiver power or not, the detection of a power requiring terminal. Maximum current through the transformers, noise effects.

In conclusion a very thorough piece of work.

However if we compare the work of IEEE 802,3af [13] with the powering standards for an ISDN terminal:

An ISDN terminal has special provisions for power outage emergency situations.

In case of a power outage the polarity on the line will reverse and one of the terminals on the bus is allowed to use power from the line, restricted mode.

In normal power mode a terminal can use up-to 1 W if active, deactivated 100 mW.

In restricted mode activated 400 mW and deactivated 25 mW.

Other terminals connected to the bus are allowed to draw 3 mW.

In several countries terminals connected to the public network need to be capable of calling emergency number in case of power outage.

With the upcoming use of CATV systems with IP-Telephony as public telephony infrastructure also the emergency powering for the envisaged use of IP Telephony terminals will have to be standardized.

7 Proposals for further standardization work

7.1 General

For the standardization of IP-Telephone a lot of work is on going in different standardization bodies.

Protocol and interoperability for IP-Telephony in general is handled by ETSI-Tiphon this of-course is also valid foe IP-Telephony terminals.

Acoustical aspects of terminals are handled by ETSI-STQ and collaboration is open to TC AT members. The acoustical specification of the end point should also specifically laid down for an IP_Telephony terminal, which is not yet the case.

The acoustical quality is handled by TIA. TC STQ is liasing with TIA in this area.

Remote powering, without restricted mode, by the IEEE.

7.2 ACTIONS

7.2.1 Acoustics

ETSI, preferably ETSI-STQ can take actions to define the acoustical specifications for an IP-Telephony terminal, taking the ISDN terminal specifications as a staring point.

7.2.2 Powering over Ethernet

To be able to use IP telephones also in emergency situations the powering concept has to be capable of handling a normal mode and restricted mode, in analogy with the ISDN situation.

ETSI-TC AT, based on earlier well accepted standardization of terminals and public offered interfaces, is in the best position to take the lead in the completion of the remote powering of IP-Telephony terminals in co-operation with IEEE 802,3af [13].

In Meeting AT#2 a new work item was proposed, document 02_72, to produce a standard detailing the line powering requirements for IP terminals. It was decided that this should be based upon work already carried out in IEEE 802af. It was agreed that a liaison should be sent to IEEE informing them of our work and requesting access to their work in this area.

- Cooperate, if possible, with IEEE in the basic standardization of IP Terminal powering over the Ethernet.
- Propose to IEEE 802,3af [13] to study the possibility of amendments to implement the restricted mode capabilities and make the remote powered IP-Telephony terminals comply to the most common requirements, e.g. national rules for public Telephones.

History

Document history		
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