PN-4462

Telecommunications

Telephone Terminal Equipment

Performance and Interoperability Requirements for Voice-over-IP [VoIP] Telephone Terminals

Formulated under the cognizance of TIA Engineering Committee TR-41,
User Premises Telecom Requirements

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FOREWORD

(This foreword is not part of this standard.)

This document is a TIA/EIA Telecommunications standard produced by Working Group TR-41.3 of Committee TR-41. This standard was developed in accordance with TIA/EIA procedural guidelines, and represents the consensus position of the formulating Working Group. This standard is based on TIA/EIA-579-A.

Suggestions for improvement of this standard are welcome. They should be sent to:

Telecommunications Industry Association
Engineering Department
Suite 300
250 Wilson Boulevard
Arlington, VA 22201

1 Introduction

1.1 This standard fills a recognized need in the telephone industry brought about by the use of equipment supplied by many different manufacturers. It will be useful to anyone engaged in the manufacture of terminal equipment and auxiliary devices, and to those purchasing, operating or using such equipment or devices.

1.2 In the Internet world, there are recognized standards that set out requirements for telephony-capable multimedia devices; however, none of these standards define a telephone terminal as is commonly used today. To define an interoperable IP telephone, physical connection requirements, a profile of network protocols, and speech transmission performance requirements must be specified.

Because of the dominance of Ethernet, this telephone is based on Ethernet.

Speech transmission..

1.4 Two categories of criteria are specified; mandatory and advisory. Mandatory requirements are designated by the word "shall". Advisory requirements are designated by the word "should," or "may," or "desirable" (which are used interchangeably in this standard). The mandatory criteria generally apply to safety and protection, signaling, and compatibility; they specify the absolute minimum acceptable requirements in areas such as transmission and equipment parameters and durability.

Advisory criteria represent product goals. In some instances, the advisory criteria are included in an effort to ensure universal product compatibility with equipment and facilities operating in statistically small quantities. In other cases, advisory criteria are presented when their attainment will enhance the general performance of the product in all its contemplated applications.

Where both a mandatory and an advisory level are specified for the same criterion, the advisory level represents a goal currently identifiable as having distinct compatibility or performance advantages, or both, toward which future designs should strive.

1.4 The intent of this standard is to conform with existing ITU-T, ANSI, and other standards; however, it is not the intent of this working group to duplicate the efforts of these other organizations.

1.5 This standard is intended to be a living document. The criteria contained in this standard are subject to revision and updating as warranted by advances in network or terminal equipment technology.

2 Scope

1.1 This standard specifies requirements for the connection method, physical layer connectivity, network protocols, and speech transmission performance for Voice over IP Telephone Terminals that are connected to Ethernet networks. The requirements should ensure interoperability between the equipment of different manufacturers and should ensure satisfactory voice grade service to the user.

1.2 This standard applies specifically to VoIP Telephone Terminals that are connected to Ethernet networks.

1.3 This standard harmonizes, wherever possible, with ITU-T recommendation H.323 [16].

3 Normative References

The following standards contain provisions, which, through reference in this text, constitute provisions of this Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below. ANSI and TIA maintain registers of currently valid national standards published by them.

[1] ETSI TS 101 319 v.1.6.4 (1998-12), *Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Signalling for basic calls from an H.323 terminal to a terminal in a Switched-Circuit Network (SCN).*

[2] IEEE 802.3 (1998), *Carrier Sense Multiple Access with Collision Detection (CSMA/CD) Access Method & Physical Layer Specifications*

[3] IEEE 802.3u (1995), *Local and Metropolitan Area Networks-Supplement - Media Access Control (MAC) Parameters, Physical Layer, Medium Attachment Units and Repeater for 100Mb/s Operation, Type 100BASE-T (Clauses 21-30)*

[4] IEEE 802.3i (1990), *Supplement to 802.3 - System Considerations for Multisegment 10 M/S Baseband Networks (Section 13) and Twisted-Pair Medium Attachmen Unit and Baseband Med Spec, Type 10BASE-T (Section 14)*

[5] IETF, *Internet-Draft RTP MIB "draft-ietf-avt-rtp-mib03.txt"* (1998)

[6] IETF, *RFC1213, McCloghrie, K. and M. Rose, "Management Information Base for Network Management of TCP/IP-based Internets – MIB-II"*

[7] IMTC Voice over IP Forum, *Service Interoperability Implementation Agreement 1.0* (1997)

[8] ISO/IEC 10646-1:1993, *Information technology – Universal Multiple-Octet Coded Character Set (USC) – Part 1: Architecture and Basic Multilingual Plane*.

[9] ITU-T Recommendation E.164 (1997), *The international public telecommunication numbering plan*.

[10] CCITT Recommendation G.711 (1988), *Pulse Code Modulation (PCM) of voice frequencies*.

[11] CCITT Recommendation G.722 (1988), *7 kHz audio-coding within 64 kbit/s*.

[12] ITU-T Recommendation G.723.1 (1996), *Speech coders: Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*.

[13] CCITT Recommendation G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction*.

[14] ITU-T Recommendation G.729 (1996), *Coding of speech at 8 kbit/s using Conjugate Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP).*

[15] ITU-T Recommendation T.120 (1996), *Data protocols for multimedia conferencing*.

[16] ITU-T Recommendation H.225.0 (1998), *Call signalling protocols and media stream packetization for packet based multimedia communication systems*.

[17] ITU-T Recommendation H.235 (1998), *Security and encryption for H-Series (H.323 and other H.245 based) multimedia terminals*.

[18] ITU-T Recommendation H.245 (1998), *Control protocol for multimedia communication*.

[19] ITU-T Recommendation H.323 (1996), *Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service*.

[20] ITU-T Recommendation H.323 (1998), *Packet-based multimedia communications systems.*

[21] ITU-T Recommendation H.332*, *H.323 extended for loosely-coupled conferences*.

[22] ITU-T Recommendation H.341* (1999), *Multimedia Management Information Base*

[23] ITU-T Recommendation H.450.1 (1998), *Generic functional protocol for the support of supplementary services in H.323*.

[24] ITU-T Recommendation H.450.2 (1998), *Call transfer supplementary service for H.323*.

[25] ITU-T Recommendation H.450.3 (1998) *Call diversion supplementary service for H.323*.

[26] ITU-T Recommendation H.450.4* (1998), *Call Hold Supplementary Service for H.323*.

[27] ITU-T Recommendation H.450.5* (1998), *Call Park and Call Pickup Supplementary Services for H.323*.

[28] ITU-T Recommendation H.450.6* (1998), *Call Waiting Supplementary Service for H.323*.

[29] ITU-T Recommendation H.450.7* (1998), *Message Waiting Indication Supplementary Service for H.323*.

* Presently at the stage of draft.
* Presently in the draft stage.

[30] ITU-T Recommendation H.450.8* (1999), *Conference out of Consultation Supplementary Service for H.323*.

[31] ITU-T Recommendation H.450.9* (1999), *Call Completion on Busy*.

[32] ITU-T Recommendation H.323 Annex F* (1999), *Single Use Audio Device*.

[33] ITU-T Recommendation I.250 (1988), *Definition of Supplementary Services*.

[34] ITU-T Recommendation I.251.5 (1995), *Connected Line Identification Presentation (COLP)*.

[35] ITU-T Recommendation I.251.6 (1995), *Connected Line Identification Restriction (COLR)*.

[36] ITU-T Recommendation I.251.7 (1992), *Malicious Call Identification*.

[37] ITU-T Recommendation I.257.1 (1995), *User-To-User Signalling (UUS)*.

[38] ITU-T Recommendation Q.931 (1993), *ISDN user-network interface layer 3 specification for basic call control*.

4 Definitions, Abbreviations and Acronyms

For the purposes of this Standard, the following definitions apply.

4.1 Abbreviations and Acronyms

Abbreviations and acronyms, other than in common usage, which appear in this standard are defined below.

ARJ	Admission Reject
ARQ	Admission Request
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Services Digital Network
BRJ	Bandwidth Change Reject
BRQ	Bandwidth Change Request
BTC	Broadband Transfer Capability
CID	Conference Identifier
CIF	Common Intermediate Format
D/A	Digital-to-Analog
DBR	Deterministic Bit Rate
DCF	Disengage Confirmation
DHCP	Dynamic Host Configuration Protocol
DRQ	Disengage Request
DTMF	Dual-Tone MultiFrequency
GCC	Generic Conference Control
GCF	Gatekeeper Confirmation
GK	Gatekeeper
GQOS	Guaranteed Quality of Service
GRJ	Gatekeeper Reject
GRQ	Gatekeeper Request
GSTN	General Switched Telephone Network
GW	Gateway
IACK	Information Acknowledgment
IANA	Internet Assigned Number Authority
IE	Information Element
IETF	Internet Engineering Task Force (http://www.ietf.org)
IP	Internet Protocol
IPX	Internetwork Protocol Exchange
IRQ	Information Request

IRR	Information Request Response
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
LAN	Local Area Network
LCF	Location Confirmation
LCN	Logical Channel Number
LRJ	Location Reject
LRQ	Location Request
MC	Multipoint Controller
MCS	Multipoint Communications System
MCU	Multipoint Control Unit
MP	Multipoint Processor
NACK	Negative Acknowledge
N-ISDN	Narrow-band Integrated Services Digital Network
NSAP	Network Layer Service Access Point
PABX	Private Automatic Branch Exchange
PBN	Packet Based Network
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
PDU	Packet Data Unit
PPP	Point-to-Point Protocol
QOS	Quality of Service
RAS	Registration, Admission and Status
RCF	Registration Confirmation
RIP	Request in Progress
RRJ	Registration Reject
RRQ	Registration Request
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SCN	Switched Circuit Network
SG16	ITU-T Study Group 16
SPX	Sequential Protocol Exchange
SQCIF	Sub QCIF
SSRC	Synchronization Source Identifier
TCP	Transport Control Protocol
TSAP	Transport layer Service Access Point
UCF	Unregister Confirmation
UDP	User Datagram Protocol
URJ	Unregister Reject
URQ	Unregister Request

5 Technical Requirements

5.1 Acoustic Performance Requirements

Note: TIA/EIA 579A (Telecommunications Telephone Terminal Equipment Transmission Requirements for Digital Wireline Telephones) specifies the acoustic-to-digital and digital-to-acoustic transmission performance for Digital Wireline Telephones, which applies to terminals that encode and decode analog signals in conformance with PCM 64kbit/s as described in ITU-T recommendation G.711 [10]. This standard mandates G.711 only, so it is acceptable to use TIA/EIA 579A; however, the option to support other codecs is also provided. A more effective standard, therefore, is PN-4352 (Telecommunications Terminal Equipment Transmission Requirements for Voice Over IP and Voice over PCM Digital Wireline Telephones). Because PN-4352 is still under development, we will, by default, reference TIA/EIA 579A.

5.1.1 Handset Technical Requirements

Note: This standard will reference the requirements from PN-4352.

5.1.1.1 Handset Frequency Response

5.1.1.2 Handset Loudness Ratings

5.1.1.3 Handset Noise

5.1.1.4 Distortion

5.1.1.5 Weighted Terminal Coupling Loss (TCLw)

5.1.1.6 Stability Loss

5.1.1.7 Maximum Acoustic Pressure

5.1.1.8 Handset Volume Control and Magnetic Output Requirements

Note: For the purposes of this standard, we reference TIA/EIA 504-B.

5.1.2 Hands-Free Technical Requirements

Note: This standard will reference the requirements from PN-4352.

5.1.2.1 Headset Frequency Response

5.1.2.2 Headset Loudness Ratings

5.1.2.3 Headset Noise

5.1.2.4 Distortion

5.1.2.5 Weighted Terminal Coupling Loss (TCLw)

5.1.2.6 Stability Loss

5.1.2.7 Maximum Acoustic Pressure

5.1.2.8 Handsfree Technical Requirements

5.1.2.9 Handsfree Frequency Response

5.1.2.10 Handsfree Loudness Ratings

5.1.2.11 Handsfree Noise

5.1.2.12 Distortion

5.1.2.13 Temporally Weighted Echo Return Loss

5.1.2.14 Stability Loss

5.1.2.15 Convergence Time

5.2 Ethernet Requirements

To ensure the interoperability of VoIP Telephone Terminals in common local area networks, this standard mandates that VoIP Telephone Terminals shall be compatible with IEEE 802.3u [3] (otherwise known as "Fast Ethernet") 100BASE-TX "Ethernet." Because many Ethernet networks currently in use do not provide 100 Mbit/s service, VoIP Telephone Terminals should also be compatible with the slower IEEE 802.3 [2] 10BASE-T standard. Terminals providing both 100 BASE-TX and 10 BASE-T through the same physical connector shall be able to automatically detect whether the network is configured for 100 BASE-TX or 10 BASE-T operation.

5.3 TCP/IP Requirements

VoIP Telephone Terminals shall contain a TCP/IP protocol stack. The TCP/IP protocol stack provides the transport layer services required by H.323.

VoIP Telephone Terminals shall support IPv4. VoIP Telephone Terminals may choose to support IPv6 but should implement the mechanisms specified in RFC1933 in order to maintain compatibility with IPv4.

The Dynamic Host Configuration Protocol (RFC2131) shall be used to determine the terminal's IP address and DNS servers.

5.4 H.323 Requirements

The basic procedures for multimedia communications over packet based networks are defined by ITU-T Recommendation H.323 and the recommendations which it references, H.225 and H.245.

Two versions of H.323 products are currently recognized. Version 1 products are compliant with the 1996 versions of H.323, H.225 and H.245. Version 2 products are compliant with the 1998 versions of H.323, H.225 and H.245.

There is currently a White Paper submission for an Annex F to the 1998 version of H.323 defining a Single Use Audio Device (SUD-Audio), which operates using a well-defined subset of H.323 protocols intended to be well-suited for IP Telephony applications while retaining interoperability with regular H.323 Version 2 devices.

The VoIP Telephone Terminal defined by this standard is based upon this SUD-Audio definition. The intent of this standard is to follow the basic procedures defined by Annex F wherever possible. Additional limitations and extensions to Annex F that the TR-41.3.4 Engineering Committee has felt it necessary to impose are detailed in this section. This section and it's subsections mirror 7/H.323 Annex F.

5.4.1 RAS Signaling (H.225.0 RAS)

VoIP Telephone Terminals shall comply with the RAS procedures defined in 7.1/H.323 Annex F.

Note: The RAS procedures defined in 7.1/H.323 Annex F specify that SUD-Audio terminals shall comply with the RAS procedures defined in H.323 version 2 and H.225.0 version 2.

5.4.2 Call Signaling (H.225.0 Call Control)

Whereas section 7.2 of H.323 Annex F states that all supplementary services are optional, certain of them shall be made mandatory in this standard. Section 5.4.5 defines these mandatory supplementary services. VoIP phones shall be able to safely ignore H.225.0 Facility messages not associated with the mandatory supplementary services.

5.4.3 Multimedia System Control Signaling (H.245)

5.4.3.1 H.245 Control Channel

The procedures of 7.3.1 of H.323 Annex F shall define the H.245 Control Channel procedures for VoIP telephone terminals.

5.4.3.2 H.245 Messages Supported

The procedures of 7.3.2 of H.323 Annex F shall apply unmodified.

5.4.3.3 Master-Slave Determination

The procedures of 7.3.3 of H.323 Annex F shall apply unmodified.

5.4.3.4 Terminal Capability Exchange

5.4.3.4.1 Audio Capability

Section 7.3.4.1 of H.323 Annex F defines G.711 [2] alternatives that shall be implemented, and G.723.1 [12], G.729 [14] and GSM alternatives that may be implemented by SUD-Audio devices.

The IMTC VoIP Service Interoperability Implementation Agreement 1.0 [38] states that endpoints claiming compliance with the agreement shall implement G.723.1 in addition to G.711. It is therefore desirable that VoIP phones compliant with this standard implement the G.723.1 variants specified by H.323 Annex F.Video Capability

5.4.3.4.2 Video Capability

As per 7.3.4.2 of H.323 Annex F, video capability shall not be used.

5.4.3.4.3 Data Capability

As per 7.3.4.3 of H.323 Annex F, data capability shall not be used.

5.4.3.4.4 Conference Capability

As per 7.3.4.4 of H.323 Annex F, conference capability shall not be used.

5.4.3.4.5 User Input Capability

The procedures of 7.3.4.5 of H.323 Annex F shall apply unmodified.

5.4.3.4.6 Security Capability

Section 7.3.4.6 states that security capabilities shall not be used and that Secure SUD-Audio devices are for further study.

The TR-41.3.4 Engineering Committee may decide to seek an interim solution, as per section 5.5, in which case the VoIP Telephone Terminals defined by this standard may incorporate the security capabilities so specified.

5.4.3.4.7 maxPendingReplacementFor

The procedures of 7.3.4.7 of H.323 Annex F shall apply unmodified.

5.4.3.4.8 nonStandardCapability

The procedures of 7.3.4.8 of H.323 Annex F shall apply unmodified.

5.4.3.4.9 Additional Rules for the Use of Capabilities

The procedures of 7.3.4.9 of H.323 Annex F shall apply unmodified.

5.4.3.5 Logical Channel Signaling Messages

The procedures of 7.3.5 of H.323 Annex F shall apply unmodified.

5.4.3.6 Request Mode Messages

The procedures of 7.3.6 of H.323 Annex F shall apply unmodified.

5.4.3.7 Non-Standard H.245 Messages

The procedures of 7.3.7 of H.323 Annex F shall apply unmodified.

5.4.4 Media Exchange

The procedures of 7.4 of H.323 Annex F shall apply unmodified.

5.4.5 Supplementary Services (H.450.x)

At present, Annex F of H.323 specifies that all supplementary services are optional, and states that a baseline for supplementary services to be provided by SUD-Audio devices is for further study.

PN-4462 will establish it's own baseline for supplementary services as specified in this section. Wherever possible this baseline will be aligned with the existing H.450 series of ITU-T Recommendations for H.323 devices, and where none exists, with any anticipated direction that the ITU may take in defining the service.

As there is currently no complete listing of supplementary services for H.323 networks, the list of supplementary services considered was taken from those defined by ITU for ISDN networks. Recommendation I.250 [33] defines most of these supplementary services.

The following table summarizes the supplementary services to be considered by PN-4462:

Table 1 Supplementary Services

Legend

Service is the acronym by which the supplementary service is commonly known.

Description is the full name assigned to the supplementary service.

Reference is the reference document where the implementation of the service is defined:

#.#… references a section within this document

H.323 or #.#…/H.323 references H.323 generally or specifically respectively

x.#.#… references another ITU recommendation

[#] indicates the element in the Normative References list in which complete details for the reference are provided.

Requirement indicates the requirement on VoIP phones to implement the service:

M – the requirement is mandatory

O – the requirement is optional

I – the service is intrinsically provided in H.323

FFS – the requirement is for further study

(#) – indicates a note at the bottom of the table regarding the requirement.

Service	Description	Reference	Requirement
DDI	Direct-Dialing-In	H.323	I(1)
MSN	Multiple Subscriber Number	7.1.3 / H.323	I(2)
CLIP	Calling Line Identification Presentation	5.4.5.1	M
CLIR	Calling Line Identification Restriction	5.4.5.2	M
COLP	Connected Line Identification Presentation	5.4.5.3	FFS
COLR	Connected Line Identification Restriction	5.4.5.4	FFS
MCI	Malicious Call Identification	5.4.5.5	FFS
SUB	Sub-addressing	H.323	I(1)
CT	Call Transfer	H.450.2 [24]	M
CFB	Call Forwarding Busy	H.450.3 [25]	M
CFNR	Call Forwarding No Reply	H.450.3 [25]	M
CFU	Call Forwarding Unconditional	H.450.3 [25]	M
CD	Call Deflection	H.450.3 [25]	M
CW	Call Waiting	H.450.6 [28]	M
HOLD	Call Hold	H.450.4 [26]	M
CCBS	Completion of Calls to Busy Subscriber	[31]	O
CONF	Conference Calling	5.4.7.2	O
PNP	Private Numbering Plan	7.1.3 / H.323	I(2)
CRED	Credit Card Calling	5.6	FFS
AOC	Advice of Charge	5.6	FFS
REV	Reverse Charging	5.6	FFS
UUS	User-to-User Signaling	5.4.5.6	FFS
PARK	Call Park	H.450.5 [27]	M
PICKUP	Call Pickup	H.450.5 [27]	M
MWI	Message Waiting Indication	H.450.7 [29]	M

Notes

1. Addressing features of H.323 intrinsically provide this capability.
2. Alias addressing features of 7.1.3/H.323 intrinsically provide this capability.

Note: the following subsections are not present in 7.5/H.323 Annex F because it does not deal with the supplementary services to the level of detail attempted by this standard.

5.4.5.1 Calling Line Identification Presentation (CLIP)

There are multiple means for specifying the source address in the Setup message defined by H.225. Procedures are therefor required to define the order in which they are selected for presentation where more than one is present.

Addresses shall be selected for presentation according to the following decreasing priority sequence:

1. Calling Party Number
2. sourceAddress(es)
3. sourceCallSignalAddress

5.4.5.2 Calling Line Identification Restriction (CLIR)

Note 7.2.2.6/H.225 states that octet 3a of the Calling Party Number information element, the octet containing the fields which control presentation, shall not be used. However, since all source addressing information is optional the CLIR supplementary service can be achieved by not including the information. VoIP phones shall not send the source addressing information if the user chooses to not have it revealed.

5.4.5.3 Connected Line Identification Presentation (COLP)

This service is for further study. The definition of this supplementary service for ISDN is provided by Recommendation I.251.5 [34], which may be used as a basis for defining it here.

5.4.5.4 Connected Line Identification Restriction (COLR)

This service is for further study. The definition of this supplementary service for ISDN is provided by Recommendation I.251.6 [35], which may be used as a basis for defining it here..

5.4.5.5 Malicious Call Identification (MCI)

This service is for further study. The definition of this supplementary service for ISDN is provided by Recommendation I.251.7 [37], which may be used as a basis for defining it here.

5.4.5.6 User-to-User Signaling (UUS)

This service is for further study. The definition of this supplementary service for ISDN is provided by Recommendation I.257.1 [37], which may be used as a basis for defining it here.

5.4.6 Third-party initiated pause and rerouting

The procedures of 7.6/H.323 Annex F shall apply unchanged.

5.4.7 Conference-Mode Operation

Annex F/H.323 provides two different ways that SUD-audio devices can participate in multipoint conferences. VoIP Telephone Terminals may be either aware or unaware of the conferencing procedures described in H.245. Terminals that are aware of the procedures shall follow the recommendations of section 5.4.7.2 of this standard.

The implementation of these messages and procedures in VoIP Telephone Terminals is optional; however, if these messages and procedures are implemented, they shall be implemented as specified by 7.7.2/H.323 Annex F.

5.4.7.1 Conference-unaware SUD-Audio Terminals

The procedures defined in section 7.7.1 of H.323 Annex F place no requirements on the SUD-Audio terminal in providing the conferencing functionality. The SUD-Audio terminal is proxied into the conference through a dedicated external device.

5.4.7.2 Conference-aware SUD-Audio Terminals

Section 7.7.2 of H.323 Annex F defines the H.245 messages and procedures that should be supported by SUD-Audio devices to be able to participate in multipoint conferences without the need for a proxy.

5.4.8 Support for Loosely-coupled Conferences (H.332)

The procedures of H.323 Annex F shall apply unchanged.

5.4.9 Management Information Bases (MIBs)

The Call signaling, Terminal entity and RAS MIBs are specified in the H.341 Recommendation, a white paper submission of which is targeted for decision at the upcoming SG 16 Meeting in May 1999.

The RTP MIB, as per 8.5/H.341, is specified in an Internet-Draft [5], expected to reach RFC status in early 1999.

As per 8.6/H.341, it is assumed that any device implementing H.341 will also support MIB-II [6] or its successors.

The H.245 MIB specified in H.341 does not apply to SUD-Audio devices since they are restricted to the use of FastConnect and H.245 tunneling procedures only.

5.5 Security Extensions

The stated intent is to produce specifications for Secure SUD-Audio devices as simple extensions to the plain SUD-Audio specifications by incorporating a subset of the mechanisms specified in H.235.

This should be the appropriate means for providing security functions in VoIP phones, but since the details of this extension are for further study on an unspecified timeframe the TR-41.3.4 Engineering Committee may seek to assist the process or to pursue an interim solution. The first order of business should be to try and ascertain an expected timeline for the Secure SUD-Audio device specification.

Pursuit of an interim solution, if required, should start with looking at the IMTC Security Profile 1 (SP1) [38] and the ETSI TIPHON project TS 101 319 Annex B [1]. The cooperation agreement between IMTC and ETSI, announced January 8 1999, should ensure a great deal of synergy between these security profiles.

5.6 Settlements and Billing

Settlement and billing is primarily a network concern, however, supplementary services for alternate billing and advise of charge require the telephone to interact with the network in order to provide the service.

There are currently no established standards addressing settlement and billing in H.323 network environments, but the IMTC has it on it's "to do list" for 1999.

This standard does not, at present, define any mandatory requirements for settlement and billing. The TR-41.3.4 Engineering Committee will monitor developments in settlement and billing standards and should any clear standards emerge shall look to reflect corresponding requirements for the telephone in this section of the document.

5.7 Conformance

TIA has traditionally provided test and measurement techniques to establish conformance with the standards it publishes.

The Internet community has traditionally enforced their standards through the use of benchmark implementations and interoperability forums.

The TR-41.3.4 Engineering Committee needs to consider which mechanism or combination of mechanisms is appropriate for ensuring conformance to the standards specified in this section.