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Technical Report

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); The procedure for determining IP addresses for routeing packets on interconnected IP networks that support public telephony

Reference

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650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

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Foreword

This Technical Report (TR) has been produced by ETSI Project Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON).

Introduction

The present document explains the procedures for *routeing* of public telephony calls to an IP network. Starting point are the existing requirements in TS 101 324 [2] on numbering, and the numbering options for users on IP terminals as identified in TR 101 327 [3]. Additional general requirements for E.164/IP resolution are identified. These requirements may form the basis for a **service capability description** for call routeing.

The present document is based on the architecture developed in Tiphon WG2.

1 Scope

The present document provides a collection of information and guidance relating to:

- the choice of naming schemes;
- the relationship of names to services;
- the role of the proposed ENUM system; and
- the resolution of names in the process of routing

for the routing of public telephone calls (i.e. calls where the called party is identified by an E.164 number) to a terminating IP network or an IP network that supports a gateway back to an SCN. The calls may originate from or transit public IP based or SCN based networks.

NOTE: This is intended to be approximately equivalent to the public telephone service defined in ITU-T Recommendation E.105.

The present document is applicable to all networks that support the public telephony service and is therefore written on the basis that the E.164 numbering scheme is used for calling and called party identification. Nevertheless the underlying principles could also be applied with minor adaptation to private network numbering schemes.

The present document applies to calls to most types of number structures within E.164 [13], and includes the support of carrier selection and number portability. It does not specifically address the support of mobility or roaming, although it would apply to the routeing of a call to the home mobile network.

The types of IP network considered include but are not limited to TIPHON Release 3. Because the routing aspects of the present document focus mainly on routing between networks for the support of a common service (public telephony), the report has a different emphasis from the main emphasis of TIPHON Release 3, which is focused on the provision of customized services to the customers of a single service provider.

The present document covers only the routeing between networks. It does not include the routeing inside a terminating network.

2 References

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

[1] ETSI TS 101 314: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Network architecture and reference configurations; TIPHON Release 2".

[2] ETSI TS 101 324: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Numbering; Scenarios 1, 2, 3 and 4".

[3] ETSI TR 101 327: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Guide to numbering options for public networks based on VoIP technology".

[4] ETSI TR 101 287: "Services and Protocols for Advanced Networks (SPAN); Terms and Definitions".

[5] ETSI TR 102 081: "Network Aspects (NA); Number Portability Task Force (NPTF); Signalling requirements to support number portability".

[6] ETSI TR 101 697: "Number Portability Task Force (NPTF); Guidance on choice of network solutions for service provider portability for geographic and non-geographic numbers".

[7] ETSI TR 101 119: "Network Aspects (NA); High level description of number portability".

[8] ETSI TR 101 118: "Network Aspects (NA); High Level Network Architecture and Solutions to support Number Portability".

[9] ETSI TR 101 122: "Network Aspects (NA); Numbering and addressing for Number Portability".

[10] ETSI EG 201 367: "Intelligent Network (IN); Number Portability Task Force (NPTF); IN and Intelligence Support for Service Provider Number Portability".

[11] ITU-T Recommendation H.225.0: "Call signalling protocols and media stream packetization for packet-based multimedia communication systems".

NOTE: See annex G: "Communication between Administrative Domains".

[12] ITU-T Recommendation Q.769.1: "Signalling system No. 7 - ISDN user part enhancements for the support of number portability".

[13] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".

[14] ITU-T Recommendation E.105: "International Telephone Service".

[15] ISO 3166: "Codes for the representation of names of countries and their subdivisions".

[16] ITU-T Recommendation E.191: "B-ISDN addressing".

[17] ETSI ETR 316: "Broadband Integrated Services Digital Network (B-ISDN); Numbering and addressing in B-ISDN".

[18] IETF RFC 2543: "SIP: Session Initiation Protocol".

[19] IETF RFC 2131: "Dynamic Host Configuration Protocol".

[20] IETF RFC 1715: "The H Ratio for Address Assignment Efficiency".

[21] IETF RFC 1035: "Domain names - implementation and specification".

[22] ITU-T Recommendation H.323: "Framework and wire-protocol for multiplexed call signalling transport".

[23] ITU-T Recommendation H.248: "Gateway control protocol".

[24] IETF RFC 2871: "A Framework for Telephony Routing over IP".

[25] IETF RFC 2327: "SDP: Session Description Protocol".

[26] ITU-T Recommendation Q.931: "ISDN user-network interface layer 3 specification for basic call control".

[27] ETSI TS 101 878: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Service Capability Definition; Service Capabilities for a simple call".

3 Definitions and abbreviations

3.1 Definitions

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

address: string or combination of digits and symbols which identifies the specific termination points of a connection/session and is used for routeing

called number: normally, name written as a numerical string identifying the called party or called terminal

contact ID: intermediate identifier for the destination of the next point of resolution, i.e. the destination of the next hop for the signalling messages

NOTE: The form of the Contact ID may vary and may or may not depend on the protocol and the technology used in the transport plane.

destination network: network to which a call is currently being routed

NOTE: For service resolutions that take place before the home network is reached, the destination network is the home network. For service resolutions performed by the home network (e.g. call forwarding or the support of roaming) this is the visited network.

E.164 number: number conforming to the numbering plan and structure specified in ITU-T Recommendation E.164

NOTE: See ITU-T Recommendation E.164 [13].

ENUM: telephone number mapping

NOTE: IETF working group.

home network name: network on which the customer's service application is provided whether by the network operator or a separate service provider, e.g. the network on which the customer has a subscription

NOTE: This is in most cases the network through which the customer is assigned its E.164 number.

internet named telephony: service that supports conversational voice and uses Internet names for the identification of the called party

name: combination of alpha, numeric or symbols that is used to identify end-users

NOTE: A name may be portable between Service Providers.

public telephony: service that conforms to ITU-T Recommendation E.105, i.e. it supports conversational voice and uses E.164 numbers for the identification of the called party

NOTE: From the perspective of the present document, the only point of significance is the use of E.164 numbers. The issue of whether any quality requirements should be applied to public telephony or whether E.164 numbers should be allocated only to services that achieve a certain threshold of quality is outside the scope of the present document. See ITU-T Recommendation E.105 [14].

Routeing Number (RN): within TIPHON, specific number that is used by the networks to route the call

NOTE: The Routeing Number conveys information in a form more readily usable by the network (e.g. to route calls to a ported number).

routeing: set of instructions on how to reach a destination

Second Level Domain name (SLD): part of the names in the DNS below the TLD

NOTE: Under the country code TLDs, there is a wide variation in the structure, in some countries the structure is very flat, in others there is substantial structural organization. In some country domains the second levels are generic categories (such as, AC, CO, GO, and RE), in others they are based on political geography, and in still others, organization names are listed directly under the country code.

Top Level Domain name (TLD): part of name structure in the Domain Name System (DNS) under the control of the Internet Corporation for Assigned Names and Number (ICANN)

NOTE: In the DNS naming of hosts (computers) there is a hierarchy of names. The root of system is unnamed. Below the root, there is a set of what are called "top-level domain names" (TLDs). They include the generic TLDs (EDU, COM, NET, ORG, GOV, MIL, and INT and new ones that are under creation), and the two letter country codes such as .UK, .DE and .JP from ISO-3166 [15].

transit network: network between two networks, e.g. between the originating network and the terminating network

NOTE: A transit network is not always present in a call, but in some calls there may be more than one transit network present.

3.2 Abbreviations

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

ACK	ACKnowledge
ALG	Application Layer Gateway
CR	Call Routing
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Server
ICANN	Internet Corporation for Assigned Names and Number
ID	IDentifier
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ISUP	ISDN User Part
ITU	International Telecommunication Union
LAN	Local Area Network
NAT	Network Address Translators
NOA	Nature Of Address
PSTN	Public Switched Telephone Number
RN	Routeing Number
RTP	Real Time Protocol
SC	Service Control
SCN	Switched Circuit Network
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SLD	Second Level Domain
SMS	Short Message Service
TCP	Transmission Control Protocol
TLD	Top Level Domain
TRIP	Telephony Routing over IP Protocol
TSAP	Transport layer Service Access Point
UAC	User Agent Client
UAS	User Agent Server
UCI	Universal Communications Identifier
UDP	User Datagram Protocol
UPT	Universal Personal Telephony
URL	Uniform Resource Locator
VoIP	Voice over the Internet Protocol

4 The choice of naming system

4.1 Introduction to naming and addressing

4.1.1 Naming

A name is a "combination of characters and is used to identify end users (character may include numbers, letters and symbols)".

NOTE: According to ITU-T Recommendation E.191 [16].

An end user is "a logical concept which may refer to a person, a persona (e.g. work, home etc.), a piece of equipment (e.g. NTE, phone etc.), an interface, a service (e.g. freephone), an application (e.g. video on demand), or a location".

A name is distinct in function from an address, which " identifies the specific termination points of a connection and is used for routeing". Addresses are essential for communication as the end points always have to be identified in a way that can be used for routing, but names are not essential. Names are added for some services to make it easier for users to identify the distant end-point or to provide an identification system that is independent of the structure of the networks or the current location of the entity to be communicated with.

4.1.2 Addressing

An address is defined as "a string or combination of digits and symbols that identifies the specific termination points of a connection and is used for routeing". An address is a specification of the location of the entity in terms of network structure. It includes information about the location within the network and may also include the identity of the network itself and its location in the topology of interconnected networks. An address identifies the interface at which the connection is to be delivered without regard to whether the connection continues beyond that interface. It contains location information and in telecommunications this is expressed in terms of the network structure in order to achieve as high as possible a degree of aggregation that reduces the complexity of routing tables in switches or routers.

NOTE 1: According to ETR 316 [17].

Addresses differ from names in that addresses contain explicit network information and this information is what makes them usable for routing. In order to route a call or a packet, the called name must be translated into an address that identifies the location in network terms and so can be used in the routing process. When a name is ported from one location or one service provider to another, the address associated with the name changes.

Unfortunately the distinction between name and address is not followed consistently and entities that are names, or closer to names than addresses, are often spoken of as addresses. A Uniform Resource Locator (URL) pointing to a company's web page is often called an Internet address, but is actually based on a domain name.

NOTE 2: Often the word "address" is used to mean "containing location information" but this is not sufficient for the purposes of the distinction between names and address in telecommunications. Here the critical issue is whether the location information is specified in terms of network structure. For example, An E.164 number may contain location information if numbering is related to geographical areas, but such a number may be a name rather than an address if the structure that provides the location information does not relate explicitly to network structure. This would be the case for example if there is number portability between competing networks.

NOTE 3: Where a communications system is structured in terms of layers with each layer offering a service to the layer immediately above and using the services of the layer immediately below, the identities offered to the layers tend to have the properties of names. Yet when viewed from the layer above, the same identifiers have the properties of addresses. This difference in perspective may explain why the term "address" is used for email and SIP (see IETF RFC 2543, [18]) identifiers e.g. "email address" and SIP "address".

4.1.3 IP addresses

IP addresses are allocated to interfaces, but different communication streams using different protocols may share the same interface. These streams are differentiated using port numbers which are carried in the protocol (e.g. TCP, UDP or RTP) that runs on top of IP. The combination of an IP address and a port number uniquely identifies the source or destination of a stream of packets flowing between two end points. Each application protocol has a "well known" fixed port number assigned to it plus a range of port numbers for dynamic assignment to communication streams.

IP addresses are divided, in principle, into two parts:

- the identity of the network (the network part);

- the identity of the interface attached to the network (the host address, which is the destination of the IP packet).

IP address allocations are normally made through ISPs to end networks. The allocations to ISPs are made in blocks and are organized as far as practicable to be aggregatable so that traffic on a particular route is likely to have addresses in contiguous blocks. This is important to reduce the size of the routeing tables in routers where several contiguous blocks that share the same route require only one entry. The size of these routeing tables is a potential bottleneck in the growth of the Internet as router technology is only just keeping ahead of the traffic growth.

ISPs normally allocate blocks of addresses to end networks. Where the end networks have permanently connected terminals e.g. PCs connected to a LAN, the addresses may be allocated permanently to the terminals.

Conversely where terminals are likely to be disconnected frequently and where dial-up access is used, IP addresses are normally allocated dynamically, e.g. using the Dynamic Host Configuration Protocol (DHCP) (see RFC 2131 [19]). Addresses are allocated from a pool only while the customer is logged-on. After logging-off the same address will be allocated to another user.

There are two versions of IP protocols, whose address formats differ significantly:

- IPv4, a 32-bit address, which is used throughout the Internet but which is considered to be in increasingly short supply and whose allocations are being controlled carefully.

- IPv6, a 128-bit address, which is just starting to be used and should provide more than adequate capacity for the future if it is administered effectively.

IPv4 is the version of the IP protocol in general use. Use of IPv6 is only just beginning. Because the address lengths are different, the two addresses are not compatible and a long process of migration is beginning.

There are two main drivers for moving to IPv6:

- Avoiding problems when IPv4 addresses reach exhaustion.

- Obtaining benefits from features that IPv6 offers that are not available in IPv4.

There is however a disadvantage. The IPv4 header has a variable length with the minimum being 192 bits. The IPv6 header has a fixed length of 320 bits, with the possibility of additional extension headers that are normally used only by the end nodes. The fixed header length simplifies the packet handling in routers but the increased length reduces the efficiency of transmission unless header compression is applied.

UDP has a 64 bit header and TCP a 224 bit header. Therefore the maximum reduction in efficiency is 33 % (100 x (1 - ((192 + 64) / (320 + 64)))) for a zero length packet. However for speech for a 4 kbit/s speech codec with a packetization delay of 40 ms the speech packet would have a length of 4 000 x 0,04 = 160 bits and the efficiency reduction would be 24 %. For data using TCP the minimum reduction for a zero length packet would be 21 %. Thus the reduction in efficiency is greater for speech than data.

A significant uncertainty is the speed with which IPv6 will be introduced generally in the Internet world. Here there are two extremes and a continuum of possibilities between them.

- The first extreme is that ISPs will perceive some real operational advantage in using IPv6 and will introduce it as soon as possible in order to capitalize on these advantages.

- The other extreme is that ISPs will regard the introduction of IPv6 as an avoidable expense and will delay its introduction as long as possible, i.e. until the shortage in IPv4 addresses begins to be felt.

Although IPv4 has a theoretical capacity of some 4 billion (4×10^9) addresses, in practice a realistic maximum is probably some 200 million hosts. The lower practical limit is the result of the structuring of the address space and is a prediction based on observations of the points at which other numbering schemes reach saturation (see RFC 1715 [20] by Christian Huitema).

It is very difficult to obtain a well founded estimate of the current world-wide situation on allocations or when the effects of exhaustion will first be experienced. According to a paper on the IANA part of the web site (see http://www.iana.org/assignments/ipv4-address-space) of the Information Sciences Institute, there were in October 2000 some 102 unallocated/8 IP addresses out of the maximum total of 256. There were 23 allocations to the Regional Internet Registries who currently handle the allocations to ISPs and large users. The demand for allocations from these RIRs is doubling every year according to RIPE, suggesting that a further 2-3 years' growth can be accommodated without making other changes. However the remaining 131 values are allocated to organizations and large corporate and eventually some of this space could be released if necessary.

There are many "variables" that make estimation of the remaining life of IPv4 difficult to quantify including:

- the use of Network Address Translators to increase the utilization of IPv4 address space,
- WAP proxy server deployment (similar to NATs in terms of saving IP address space),
- the impact of dynamic address assignment in an "always on" environment,
- the possible impact of Windows 2000™ which includes IPSec and may lead to an increase demand for secure end to end communication, (currently the use of NAT inhibits end to end IPSec),
- new demands from the "plug and play" (auto configuration) market.

In conclusion it does appear likely that some effects of IPv4 address exhaustion will begin to be felt in the 2004-2006 timeframe.

During the migration period, various techniques including dual stack will be used to provide interworking between IPv4 and IPv6. Some of these techniques require interworking equipment that itself needs IPv4 addresses. When eventually IPv4 becomes seriously exhausted, new allocations will be possible only from IPv6. This will mean that equipment with only IPv6 addresses will be able to communicate only with other equipment that have IPv6 addresses and therefore communications with the unmodified IPv4 world will not be possible. This will be a significant commercial issue and therefore there is a body of opinion that the introduction of IPv6 should be encouraged in order that it can become as widespread as possible before IPv4 exhausts so that the loss of compatibility will be minimized.

The TIPHON standards do not specify the choice of version of IP protocol and are compatible with either version because the TIPHON standards generally apply above the network layer. Thus the choice of Internet Protocol version and any interworking between versions is outside the scope of TIPHON.

4.2 Naming schemes

There are two common naming schemes:

- E.164 names (numerical strings) defined by ITU-T Recommendation E.164 [13]. This scheme is a mixture of names and addresses. It started primarily as an addressing system but has migrated to become more of a naming system because location and operator portability are functions of names rather than addresses.
- Internet names of the form "user@domain" defined by RFC 1035 [21].

NOTE: The prefixes "http:", "sip:" etc denote the protocol and are not parts of the domain name.

The choice of identification scheme is related to the nature of the service because a service description needs to specify which type of name is used. This is important because:

- users need to know how to identify their correspondents;
- the choice of identification system determines the set of potential correspondents that can be reached;
- interconnected networks need to have a common method of identifying communicating users.

For many services, names are used as the identification system, but some services allow addresses to be used as an alternative to names (e.g. http allows users to identify web sites by IP addresses or domain names), and some services use only addresses.

In the past services and hence name types were related to technology. For example telephony could be provided only on circuit switched technology and Telex had its own naming scheme and own technology. However, third generation mobile technology is designed to support multiple services and there is therefore the possibility of supporting more than one type of name. For Tiphon Release 3 compliant systems, only E.164 names and related national numbers (e.g. 0800, 112) are supported.

4.2.1 E.164

The international public telecommunication numbering plan is defined in ITU-T recommendation E.164 [13], and numbers which comply with it are referred to as E.164 numbers. It includes PSTN, ISDN and mobile networks and supports various services including public telephony, some special telephony services such as international freephone, fax, some data services and the GSM Short Message Service (SMS). According to the ITU-T recommendations, some telephony services such as national freephone in a strict sense do not use E.164 numbers although their numbering is compatible with E.164 and thought by most people to be E.164 numbers. The E.164 number is not necessarily identical to the dialled number as dialling prefixes and arrangements for local dialling are not part of ITU-T Recommendation E.164 [13].

For various historical reasons, E.164 numbers are a mixture of names and addresses, but the trend is to reduce the degree of address information and make them more names than addresses (i.e. to reduce the network specific information). Various parts of E.164 [13] include structures related for example to geography. This structure may in the past have been related precisely to network architecture but the relationship to network architecture has reduced or been removed for example by operator and location portability.

For routeing in switched circuit networks (i.e. when network information is needed), routeing numbers are used to add at least some location information. A routeing number may be:

- a separate E.164 number or E.164-like number (i.e. a number similar in format to E.164 numbers and compatible with the E.164 [13] plan but not formally part of the plan) that contains the necessary network information;
- a non-E.164 number that contains the necessary routeing information;
- or a routeing prefix added to the front of the E.164 number.

4.2.2 Internet "names"

Within IP based networks, there are separate naming and addressing schemes. Names normally have the form:

User@domain

IP addresses are completely separate binary strings and there are different forms depending on whether IPv4 or IPv6 is being used.

Table 1 shows some examples of the relationship between names and addresses for telephony and current web applications. It includes the differences between the Tiphon and Internet telephony.

Table 1: Examples of names and addresses

	Telephony on switched circuit network	Email	Tiphon Release 3 solution for telephony on IP	Internet telephony
Name	E.164 number	user@domain where domain may be a host name	E.164 number	user@domain, possibly with an E.164 alias for incoming calls from the switched circuit networks
Address	Routeing E.164 number, or (routeing prefix +E.164 number) (see note)	IP address	IP address	IP address
NOTE: There is also additional lower level addressing information in equipment numbers.				

4.2.3 Coding of names

Table 1 refers to the naming scheme used, i.e. allocated to the users of a service, and not to the carriage of the names by a signalling protocol which would depend on the specification for the protocol. For example, an E.164 name could be coded in the form of <E.164 number>@domain.

4.3 The relationship of naming to services

Normally each service that uses names specifies a single type of name that is used. However a type of name may be used by several different services. Sometimes these services are distinguished by different ranges as is the case of ITU-T Recommendation E.164 [13]. Where there is a separate method to identify different services, the same value of a name of the same type may be used for different services. For example a given E.164 number may be shared for both telephony and fax if the terminal can distinguish the services; equally under Internet naming the same value of user@domain may be used for both email and various SIP (see RFC 2543 [18]) based services.

A user may therefore have several names for different services, e.g. GSM users have three different E.164 names for voice fax and data services on GSM. This is not very user friendly because business cards become cluttered up with the different names. A reasonable long term objective could be to work towards having one name per person for private use and perhaps a separate name for business use, however this goal is constrained by the need for compatibility with existing systems. Work within ETSI on the Universal Communications Identifier (UCI) is considering these possibilities.

Figure 1 shows an example of the relationship of named entities to name types and values for different services.

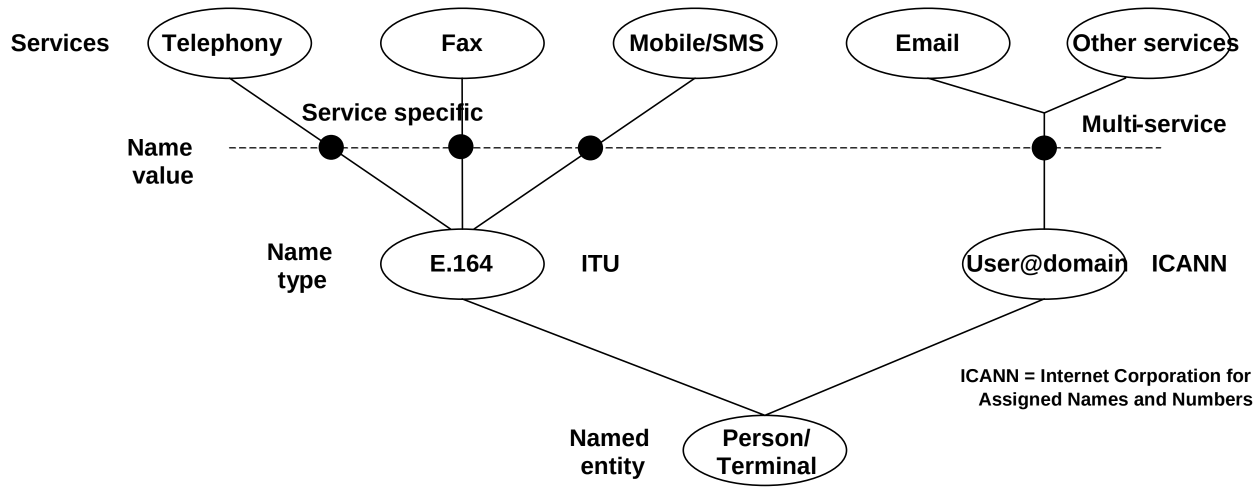

Figure 1: Naming relationships from a caller's perspective

A single name can also support several different users. Examples are an E.164 name for a telephone service in a house shared by several occupants, or an E.164 name used by a call centre, or an email name used by several people who fulfil the same function in an organizations (e.g. sales@company.com).

Although one name of the form "user@domain" may be used for several services, a given name may be used only for services supported on a single host because the name will be mapped by DNS to an IP address of an interface on the host. In most cases the host is operated or connected to a single service provider. Therefore a user that wishes to use one service provider for one set of services and another for another set will need different names.

The IETF has distinguished names and addresses and introduced the public DNS to support the resolution of names into addresses, however the IETF does not define services or service capabilities in the way that ETSI and ITU-T do, because it assumes that users assemble their own services using application protocols. In other words, services are created at the edge of networks and are not embedded in them. This means that there is a lack of clarity in relating services defined in ITU-T/ETSI to work in IETF. This lack of clarity in turn leads to some confusion over the choice of naming schemes for voice in IETF. From the perspective of ETSI, voice communications that use Internet names (e.g. what the press calls PC-PC communications) are a different service from public telephony, we call this "Internet named telephony". Internet named telephony will require interworking with the public telephony service and this interworking will have to enable callers on the public telephony service to identify called parties on the Internet telephony service. This is service interworking and there are two alternative methods of handling its numbering:

- to allocate hitherto unused E.164 numbers for this interworking (each customer of Internet telephony who requires interworking would need to be allocated an E.164 number to use in parallel with their existing Internet name);

- to use a database of associations between E.164 numbers already allocated for the public telephony service and the Internet name used for Internet telephony (this is ENUM).

Figure 2 shows the differences between the approaches of Tiphon and the IETF.

The aim of Tiphon is to support the existing public telephony services, which use different parts of E.164, on IP technology. This technology could include Internet at the transport layer but does not necessarily do so.

In contrast, the IETF is supporting a different "service" with interworking using ENUM as a method to translate E.164 numbers to Internet names. More information about ENUM is given in a later clause.

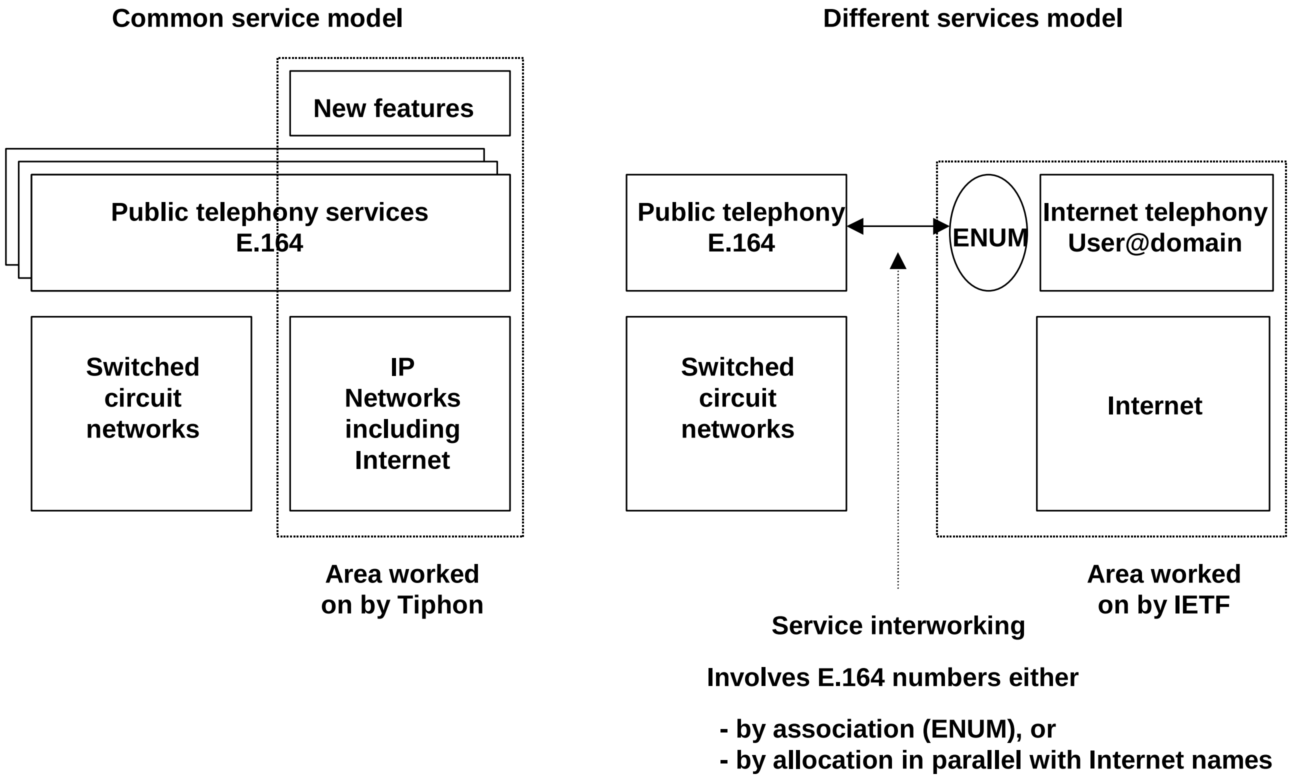

Figure 2: The different approaches to telephony

There are several different forms of public telephony service that use different ranges of numbers within ITU-T Recommendation E.164 [13]. For example in addition to the basic geographic telephony service there are mobile services, freephone, shared cost and premium rate services. These different forms are shown by the stacked boxes in the diagram.

NOTE: The issue of whether any quality requirements should be applied to public telephony or whether E.164 numbers should be allocated only to services that achieve a certain threshold of quality is outside the scope of the present document.

4.4 The choice of naming for Tiphon

Tiphon is aiming to produce standards that meet the needs primarily of new and old "telcos" who wish to offer new services on IP-based technology or migrate existing services from circuit switched technology to IP-based technology. (The commercial framework for TIPHON is assumed to be networks where service providers create and control services and the services and networks can be accessed only by authorized users. This is the traditional telco approach to communications and differs fundamentally from the Internet model where "services" are created at the network edges and the network provides universal interconnectivity). This aim makes it essential that Tiphon supports the existing services that use E.164 so that users of these services can be migrated transparently from switched circuit to IP technology without any number or name changes. In other words, Tiphon has specified E.164 as the numbering scheme for its public telephony because it is supporting the same public telephony services that already use E.164. This approach is wholly consistent with the principle that services should be defined in a way that is technology-independent.

Tiphon is developing an architecture and meta-protocol (abstract message descriptions and information flows) for the support of a range of service capabilities. This work is protocol independent. It is also developing "profiles and deltas" that will specify how existing protocols can be used/adapted for these information flows and service capabilities. The protocols covered include H.323 [22], SIP [18] and H.248/Megaco [23].

A service capability in TIPHON is a unit of technical functionality that can be mixed-and matched to implement part of a (non-standardized) service application. It is important to understand that a service capability is defined only as a technical function from the perspective of network implementation and is not an element of service defined from the perspective of the user. Also the same service capability may be part of more than one service application (e.g. user registration or user profile) in one or more service offerings.

This approach to services in TIPHON means that services are defined initially for use only between the customers of the same service provider. Access to customers of a different service provider may be achieved only if a service level interconnection is established. Thus TIPHON views a service such as public telephony, much of whose value lies in the ability of any customer to call any customer of any other service provider in the world as an exception rather than the norm. The common standard that TIPHON provides for service capabilities is, however, intended to help the establishment of such service level interconnection agreements.

The service capability definitions in TIPHON do not specify a naming scheme. The implication is that the choice of naming scheme should be specified as a separate part of the service description of a particular service and as part of any relevant service level interconnection agreement.

The following three naming schemes may be used in TIPHON:

- public telephony numbers E.164 defined by ITU;
- internet names defined by ICANN; and
- unspecified private naming schemes.

Although the service capabilities defined in TIPHON Release 3 do not specify a particular naming scheme, they have been written to ensure that they are, inter alia, suitable for use in supporting public telephony based on ITU-T Recommendation E.164 [13].

4.5 The relationship of the present document to ENUM

ENUM is the name of a chartered working group in IETF. It is attracting a great deal of attention in relation to numbering and is supported by several large manufacturers. ENUM is the name given to a set of standards that define a protocol for Telephone Number Resolution.

The function of the ENUM protocol is to map telephone numbers, defined in ITU-T Recommendation E.164 [13], to one or more Internet resources using the existing DNS system. Here a "resource" is an Internet destination that has an associated application protocol such as an email address or a SIP address. ENUM can also support mapping to resources outside the Internet such as fax and mobile numbers. The system is based on telephone numbers because these numbers are widely known and can be input from any telephone keypad.

The name "ENUM" is used to describe both:

a) a protocol for interrogating and receiving a response from DNS that could be used for public or private numbering applications;

b) a global public service for resolving already allocated E.164 numbers, including the administrative methods for populating the part of DNS used by ENUM.

The main functions of the global public ENUM service are intended to:

- enable calling users or entities to make a selection from the range of services that are available for communicating with a particular person or entity when the calling user knows only their telephone number;
- enable users to access Internet based services and resources from ordinary telephones where they are only able to input digits;
- enable users to specify their preferences for receiving incoming communications (e.g. specifying a preference for voicemail messages over live calls or indicating a destination for call forwarding). ENUM will give much improved user control over communications.

Figure 3 shows an example of ENUM used from an IP terminal.

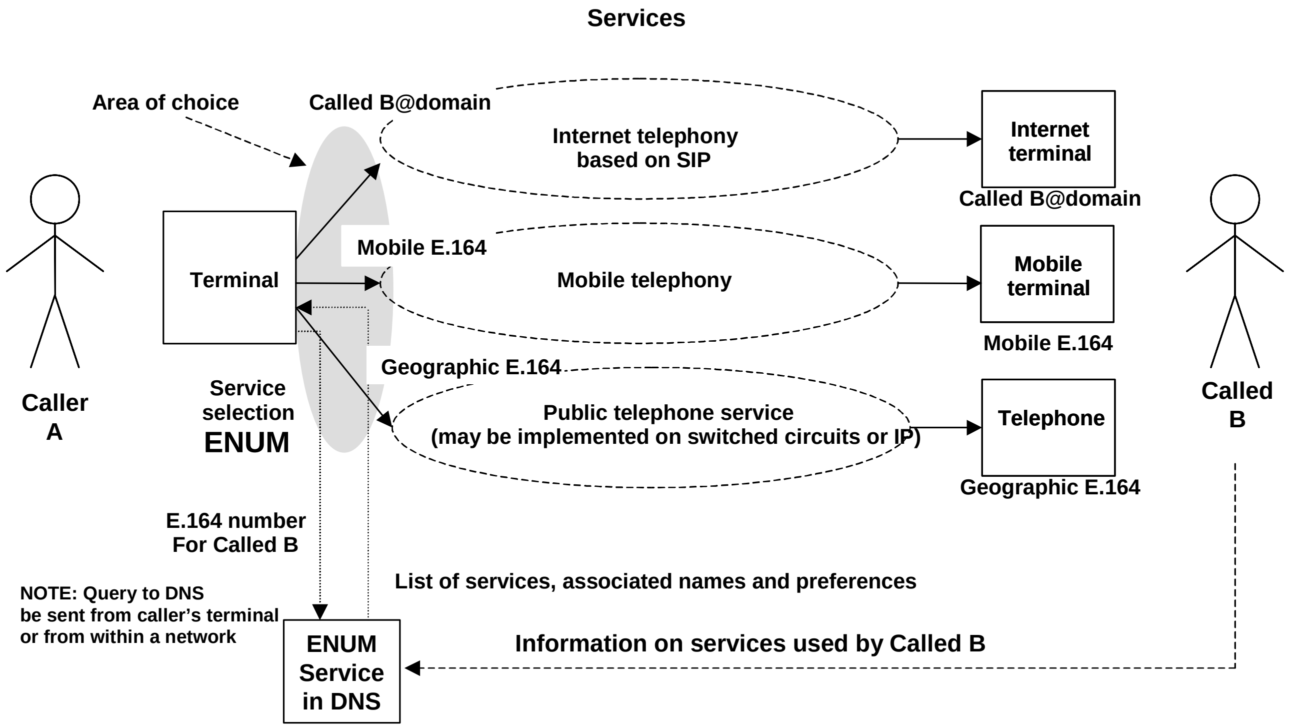

Figure 3: Example of ENUM used from a terminal

Figure 4 shows an example of ENUM used from a Switched Circuit Network (SCN). In this case, the functionality to provide access from a simple telephone and the mechanism to allow the caller to select the required option is assumed to be built into a network that is part of the PSTN.

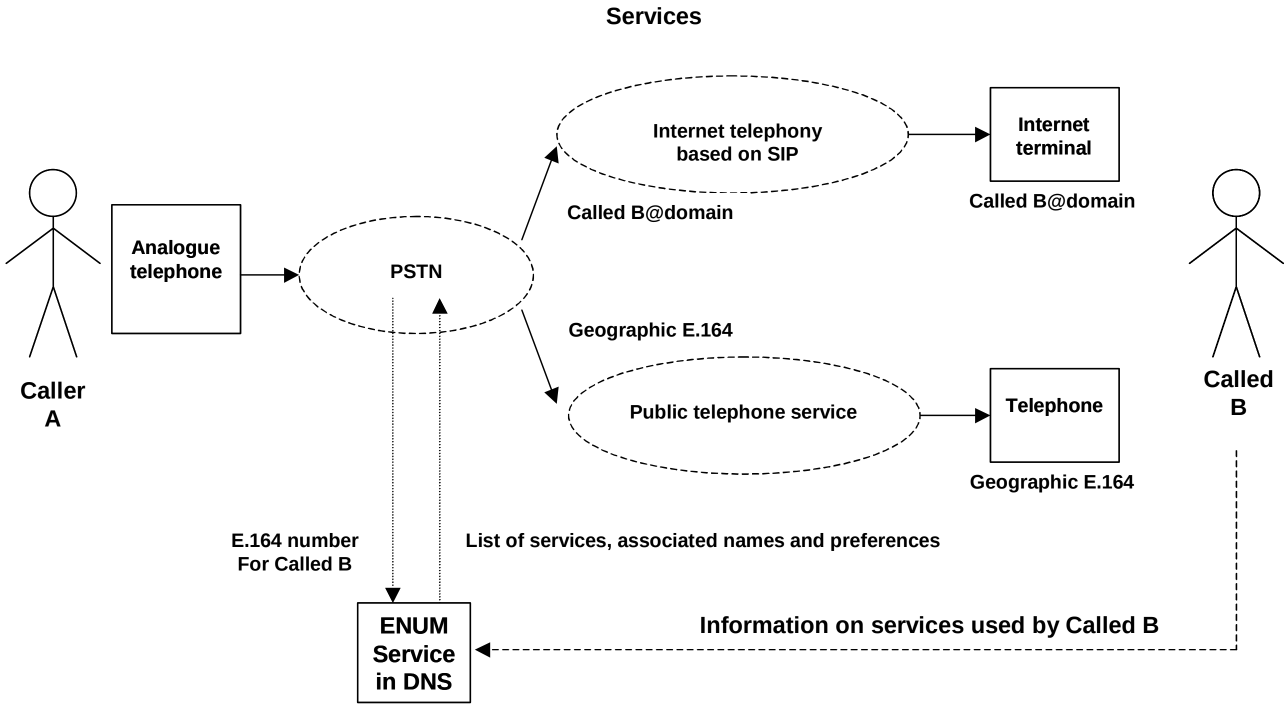

Figure 4: Example of ENUM used from a network

The ENUM service can hold E.164 numbers that are supported on IP as well as on SCN. An example would be when the access network that provides the public telephone service to the number concerned is IP based.

In practice, a prime role of ENUM may be to facilitate the migration of traffic from circuit switched networks to the Internet.

When presenting a telephone number to the DNS, the digits are reversed and dots are inserted between each digit (this will be done by software and not by the human user). ENUM is designed for users that have exclusive use of an E.164 number. It does not specify how shared E.164 numbers (e.g. a household with one line and E.164 number but several occupants each with their own SIP addresses for Internet telephony) are supported.

The issues currently under discussion are:

- what exactly will be the form of the top level part of the name to be resolved by DNS. IETF has proposed .e164.arpa and this proposal is being discussed with ITU-T;
- which organizations will be registries and run the DNS servers for E.164 numbers. The intention is that each government will select the registry for numbers under its own country code but selections had not been made at the time the present document was written;
- what control will be exercised over the submission of E.164 numbers and associated information for storage in DNS. The aim is to ensure that only the legitimate user of an E.164 number can submit and alter information and that records are updated when E.164 numbers are ported or cease. This is difficult to achieve especially in countries where there is no national database of customer information. The issues are also related to privacy laws that differ from country to country.

The resolutions considered in the present document could use a database that uses the ENUM protocol. However the fundamental difference between the global public ENUM service and the resolutions discussed here is that ENUM can use E.164 numbers that have already been allocated for service on one network (the circuit switched network) for directing calls that will be delivered on a different network (e.g. the Internet). In contrast the present document considered the use of E.164 numbers for the delivery of calls on the same networks for which or through which the numbers have been allocated.

4.6 The use of aliases

An alias is an alternative name. Aliases are commonly used in email where there may be multiple alternative values of "user" in "user@domain". This type of alias is local to the end system since other networks only use the value of "domain". Aliases of this type are supported in H.323 [22].

Users may have more than one name type and more than one name value, where these names are not just local aliases but are recognized by the networks. An example would be an E.164 number and an Internet name. In this case in the terminology of the present document, the user is using two services, public telephony and Internet named telephony, and the choice of name used to call the user is in effect selecting the service by which to call him.

4.7 Master IDs and personal numbering

There is no "master ID" and no ultimate personal number. There are various personal numbering services and TIPHON is supporting global personal numbering under the code +87810, which is a subset of the numbering range for Universal Personal Telephony. There is always a translation between a personal number and some other form of identification such as a name or an address.

4.8 Relationship to back end services

Figure 5 shows the relationship of the customer ID used in support systems such as service management systems and billing to:

- the user name;
- the end IP addresses and port numbers for the signalling and media paths.

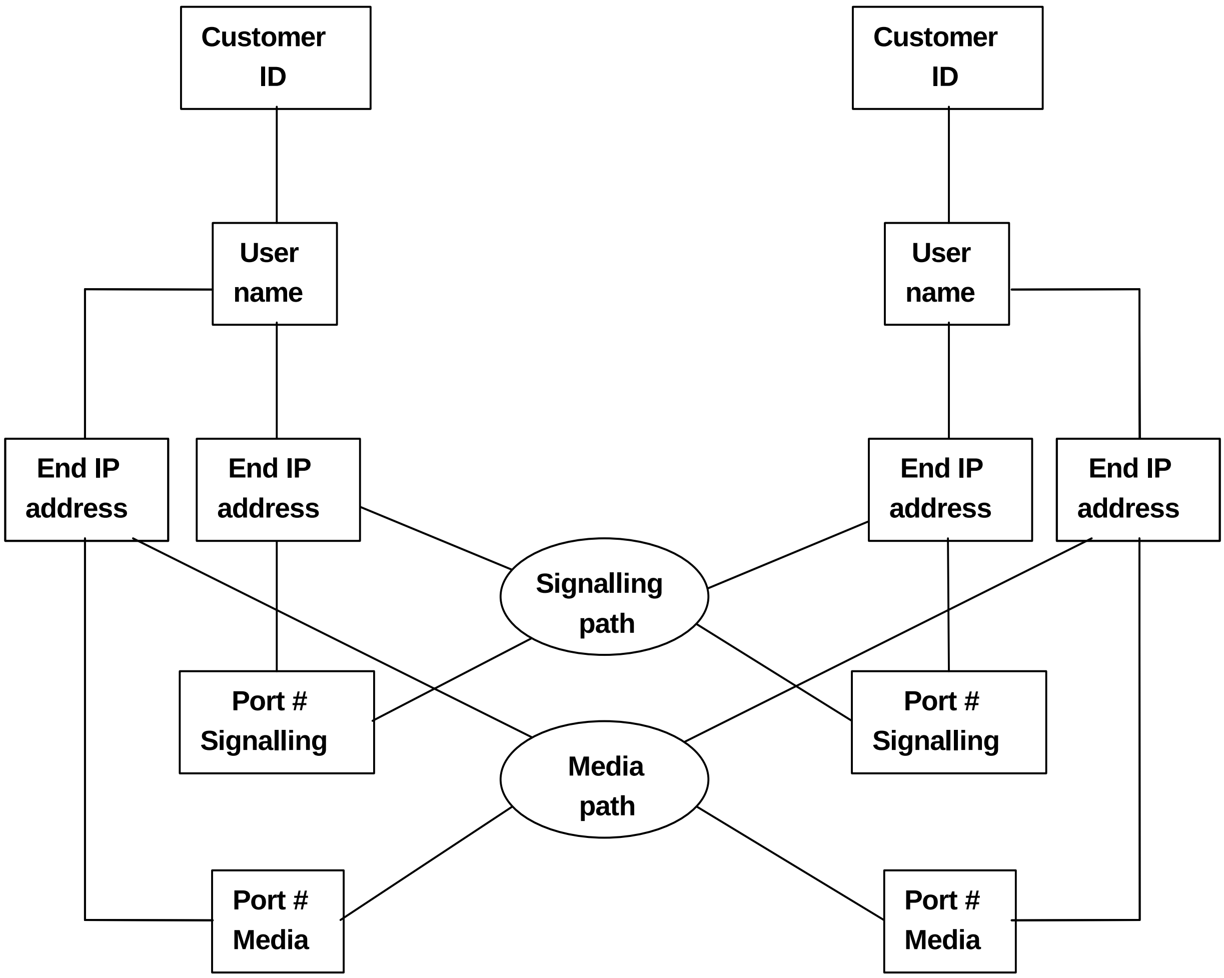

Figure 5: Relationships at the ends of the call

The figure shows the relationships for both ends of a call. Additional IP addresses and port numbers may be used at different points in the signalling and media paths, especially if network address translators are used.

In the back end system, the customer ID is related to the account ID. This relationship is outside the scope of the present document.

5 Types of resolution and their order

5.1 Introduction

The purpose of this clause is to distinguish the various different resolutions that may be encountered in an SCN to IP call. It is written for the public telephony service and so assumes that the E.164 number is the principal identifier for a telephony correspondent whatever the technology used (SCN or IP), however the principles that would apply in a private domain would be similar.

The term "E.164 number" is used loosely to mean the number that would be dialled or built up immediately from the dialled number (i.e. by expanding a local number).

There are three types of resolution regardless of the underlying technology of the network:

- search resolution to determine the E.164 number (name) from any information about the called party;
- service resolution to resolve any options for how the call should be carried and to determine ultimately the name or other identity of the destination network from the called E.164 number (examples are freephone numbers and number portability). There are three different types of service resolution and they are explained further in clause 5.3;
- routing resolution to determine the information needed for routing the signalling and media streams.

The search resolution always takes place first, if it is needed, and is performed at the calling end only.

Each network whose call control function handles the signalling may perform:

- a service resolution followed by a routing resolution; or
- a routing resolution only.

These resolutions may be carried out repeatedly by successive networks, but normally a service resolution for a particular service capability will be carried out only once.

In SCNs, the network is generally unaware whether it is doing a service resolution or a routeing resolution.

The range of numbers that contains the called number determines whether a network needs to perform a service resolution. Normally a network will perform a service resolution if it can before performing a routing resolution. For example, a network in one country will perform a service resolution on its own national freephone numbers but only perform a routing resolution for calls to a national freephone number in another country. Service resolutions for different services may be undertaken successively by different networks, e.g. a call to an international freephone number terminated on a ported geographical number.

Service and routing resolutions may be combined in practice.

A network that is traversed by the call, but where the call control function does not handle the signalling, will not perform any resolution. An example of this is where signalling packets are routed across a network using an IP address that has already been established for an interface in a network closer to the destination.

The resolutions are summarized in table 2. The "From" column shows the information that is converted in the resolution process and the "To" column shows what it is converted to.

Table 2: Types of resolution

Resolution type	From	To	Carried out by	Public or internal	Comment
Search Resolution	Any information	Called E.164	Calling terminal or calling party	Either	Address book, or Directory, or Search engine
Service Resolution	Called E.164 name	Destination network name or other identity, which may be the home network of the called E.164 number or their visited network	Service control in any network	Either	Specific to service or geographical area (NOTE: a service may be specific to a network) Not specific to IP technology
Routeing Resolution	Called E.164 name or Destination network name or other identity, which may be the home network of the called E.164 number or their visited network	Routeing information for next hop	Call control in any network	Normally internal	Local or national or global Not specific to IP technology Process may be repeated as signalling progresses hop-by-hop

A service resolution is objective or absolute in that it always returns the same result irrespective of the location from which the resolution request is made. In contrast, a routing resolution is subjective or relative in that the result depends on the location from which the resolution request is made. Figure 6 shows an example of the sequence of resolutions for a call to a ported freephone number. The caller first uses a directory service (search resolution). Then the call is passed from the originating network, which determines that, because the called number is in a number range of another country, only a routing resolution is needed to reach a network in the terminating country. This network detects that the called number is in a range that supports number portability and so performs an originating network number portability service resolution followed by a routing resolution. The call then passes via a national transit network to the terminating network. The terminating network detects that the number range is used for non-geographic freephone numbers and therefore that a service resolution is needed to determine the destination followed by a routing resolution to reach it.

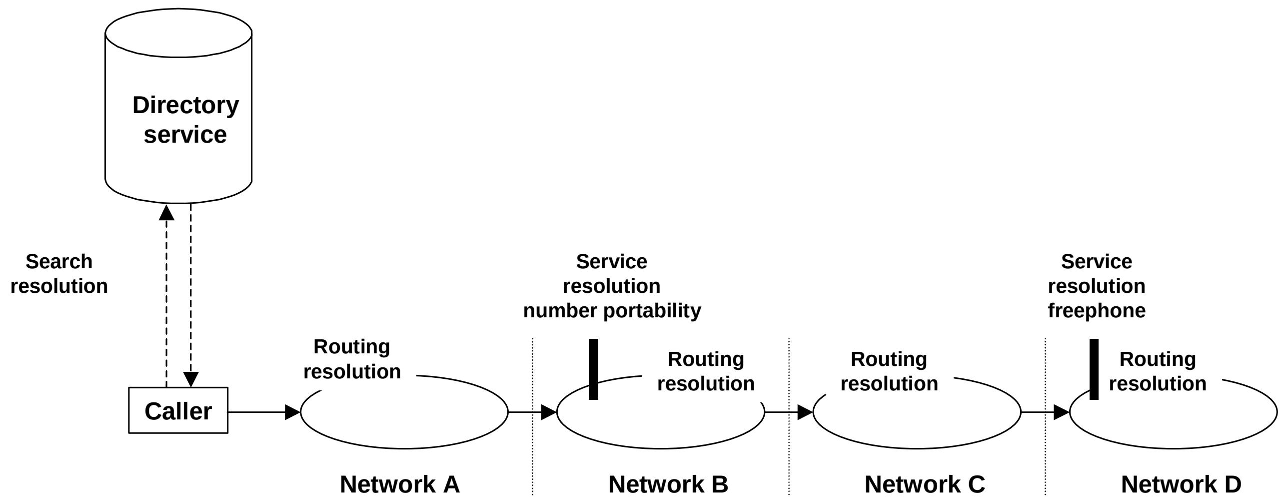

Figure 6: Example of a sequence of resolutions

5.2 Search resolution

The function of the first resolution is to find the E.164 number (sometimes called the directory number) for the called party. This step may be skipped if this number is already known, or it may be carried out by:

- an address book or directory in the terminal;
- an address book or directory in the calling network;
- a public directory function;
- a search engine.

A search engine may be a successor to the current directory enquiry service to provide a user friendly public service interface for finding correspondents. There is likely also to be a user friendly customized terminal specific systems like an address book for commonly called numbers.

The search resolution will take some or any information and provide the E.164 number. The query is made from the calling terminal and the response is returned to the calling terminal or calling party.

NOTE: The details of this resolution are out of scope of the present document.

5.3 Service resolution

This resolution supports the particular service that is using a range of E.164 numbers. It resolves the called E.164 number to a home network name and optional location information.

NOTE 1: Practices vary in different countries. In general the provision of location information enables the calling network to make more optimal routing decisions.

The main function of the home network name is to give information on the identity of the call control entity for the called party. This resolution may be needed:

- to support number portability;
- to support personal numbering;
- to support non-geographic services such as national or international freephone;
- to support call redirection services.

The resolution may be local, regional or national and is not normally related to IP technology specifically because services are not normally technology specific. The resolution capability would normally be available to all operators that need to route calls into the domain that requires the resolution. The information needed for the resolution is may be public and in some cases may be provided through a public reference database. Real-time resolutions may also be public services, but are more commonly provided by the operators for themselves using information downloaded from the public reference database. Any IP network that supports a service that requires a service resolution will need access to the resolution system.

NOTE 2: There is as yet no standardization for queries to a real-time resolution service, although there is a work item for standardizing a protocol for the resolution service.

For Tiphon compliant systems, the form of the destination network identity should be a URI. There are two reasons for using a URI:

- The URI may be used in a SIP address with the form <E.164 number>@<home network>.
- The URI enables cheap standardized hardware to be used for resolving the home network name to an address.

There are two options for the form of the URI below the sled:

<home network id>.<service capability>.SLD.TLD

<service capability>.<home network id>.SLD.TLD

Both options may be implemented and some operators may prefer one and some the other.

For different service capabilities, different forms of <home network id> could be used including a numerical form with a standard letter prefix if necessary. The home network ID can also include identities within the home network e.g. "Bracknell.CW" or "London.BT".

The values of SLD and TLD would need to be chosen carefully because fast responses are needed to minimize the call set-up time, there would be advantage in using the same values for all service capabilities because this would enable the domain name services and the networks around them to be organized to produce fast responses.

In terms of the use of TLDs, the most appropriate value of .TLD could be .arpa, which is set aside by the IAB for infrastructural support.

5.4 Routing resolution

The function of routing is to determine the direction in which a network should send a signalling packet. The routing resolution resolves from a called E.164 name or home network name to an IP address for routing the packets. Routing is discussed in more detail in the following clauses.

Routing is subjective because, unless the routing is capable of reaching the far end point, the destination to be reached depends on where the routing resolution is taking place. Routing destination therefore differ from service resolutions which are normally objective, i.e. they give the same result wherever you are starting from. Because routing is subjective, routing resolutions can normally only be internal to a network.

6 Routing in SCNs

6.1 Introduction

Within the SCN, **numbers** are used to identify (or to name) destinations. In many cases today, the address of the destination and the name are identical. The number is used to route the call to the terminating switch. It is therefore convenient for routeing, to attach numbering blocks (43 1 979 xxxx) or numbering ranges to switches (a numbering range is consecutive blocks of numbers which follow the same routeing instructions e.g. 43 1 979 2xxx to 43 1 979 5xxx). Block sizes depend on national policy and differ from country to country and also within countries. The most common block size is 10,000 numbers. A disadvantage of this approach is the uneconomic use of the available numbering space, if the demand is less than the block size. Consequently in areas where there is a shortage of numbers and in rural areas smaller block sizes such as 1,000 are increasingly being used.

The mechanism that performs the analysis of numbers looks to see if a given number fits into a numbering block or numbering range and extracts the given set of routeing instructions for this numbering block, to be executed in call set-up. The information stored in switches that relates number blocks to routeing instructions is referred to as "routeing tables".

Because the analysis needed for the application of routeing tables is costly in terms of switch processor power and because of the problem of maintaining up-to-date routeing tables, the route to the final destination is evaluated in most cases step by step. The process of number analysis is distributed amongst switches, with each switch normally knowing only the route to the next switch (next hop). The next switch may repeat and refine the analysis process. Eventually the call is routed to the terminating switch that serves the called party. The terminating switch translates from the number (name) attached to the called party and the hardware address of the line card.

6.2 Routeing numbers

Routeing numbers have been introduced to provide more flexibility and provide routeing control in cases where the called party is at a terminating switch that is not identified by the block that contains the called party number. This situation occurs either:

- where the number has been ported away from the switch identified by the number block; or
- where number blocks are not used (e.g. in mobile services or where numbers are allocated individually).

Routeing numbers may also be used to route calls to the correct termination point in the case where the same set of dialled digits is used for access to a particular service even though the service may be delivered from a number of different points in the network (e.g. 112 calls).

In these cases a routeing number is used instead of the called party number for the routeing of the call.

The routeing number is normally generated by either:

- a query to an IN database (e.g. a Home Location Register in mobile networks to obtain the Mobile Station Roaming Number, which is the routeing number, or a number portability database in some number portability solutions); or
- on-switch processing (e.g. in some onward routeing number portability solutions).

A routeing number is either:

a) added in front of the called party number in the field in the signalling system that carries the called party number; or

b) placed instead of the called party number in the field in the signalling system that carries the called party number, with the called party number being carried in a separate field; or

c) placed instead of the called party number in the field in the signalling system that carries the called party number, with the called party number no longer being carried.

With a), the routeing numbers are added to the routeing tables. With b) changes to the routeing tables in the switches may not be needed if the routeing number is chosen to match the number structure already stored in the routeing tables.

NOTE: Older signalling systems (e.g. early versions of ISUP) allow only one number to be transported, therefore the combined approach is used. New signalling systems (e.g. the latest version of ISUP) allow both numbers to be transported, but this opens another problem. Some networks use the called party number field for the routeing number, and put the called party number in the new field, others leave (for other compatibility reasons) the called party number in the old field and put the routeing number in the new field (both approaches are conforming to ITU-T Q.series). Compatibility is achieved by giving additional information indicating what changes have been made possibly by using the Nature Of Address (NOA) parameter.

A routeing number in an SCN network may be used to identify either:

- the home network (i.e. the equivalent of the home network name); or
- routing information for the next hop.

A routeing number may not need to be forwarded in signalling if a new routeing number is generated at the next switch.

7 Resolutions in Tiphon Release 3 networks at the meta-protocol level

The function of the service and routing resolutions are to provide the routing information for packets to be passed forwards on the correct path with sufficient information to reach their next destination. This means identifying where the packets should be routed to. This destination may be the end destination or an intermediate destination where a further routing resolution (or combination of service and routing resolutions) will take place.

The TIPHON architecture (see TS 101 314 [1]) defines five functional layers:

- Services;
- Service Control;
- Call Control;
- Bearer Control;
- Media Control.

The Services functional layer contains the Call Routing function (CR). This function provides the service and routing resolution functions (i.e. the translation functions) referred to in the present document. These functions may or may not be co-located with the functions in the Service Control functional layer, e.g. they may be in remote databases.

The Service Control function (SC) in the Service Control functional layer accesses the various resolution functions at the Services functional layer to obtain and assemble the information needed for call control.

The Call Control functional layer provides the signalling messages and maintains state information for the call.

Figure 7 lists the service capabilities specified in TS 101 878 [27] and identifies what resolutions they involve.

Service capability	Resolution
Simple call establishment	Routing
Calling user identity generation	-
Calling user identity conveyance	-
Calling user identity delivery	-
Call rejection	-
Number portability	Service
Emergency Calls	Service
QoS Bearer selection	
Alternative Media Path	

Figure 7: Resolutions involved in service capabilities

8 Other issues

8.1 Firewalls

Firewalls are devices that are placed at the boundary of networks to protect the networks from denial of service attacks and unwanted traffic. Firewalls are used mainly to protect company intranets and web sites, i.e. they are used on end networks. However the need for protection and access control to support charging in transit networks may lead to firewalls being used more widely on interconnected networks that provide VoIP services.

Firewalls work by examining the IP addresses and port numbers used within incoming and outgoing packets and allowing only certain ranges of addresses and port numbers through. This examination adds delay that degrades the quality of real-time communications, and firewall developers are being challenged by the need to keep this delay adequately low for conversational voice. It is quite difficult to formulate policies for firewalls that will provide adequate protection whilst not rejecting too much wanted traffic.

A group in IETF called MIDCOM is developing requirements for the control of "middle-boxes" including firewalls by the devices that handle the call signalling. This will enable the signalling to instruct firewalls to open "pinholes" (particular IP address: port number combinations) that relate to calls that are in progress. These pinholes are then closed when a call is terminated.

8.2 NATs

Network Address Translators (NATs) are devices that enable a small number of public IP addresses to be pooled and shared by a larger number of terminals. The terminals inside the area served by a NAT have private IP addresses. The NAT changes the values of the public address in the incoming packets to a private address, and changes the value of the private address in an outgoing packet to a public address. Because NATs hide the internal private addresses of a network, they provide some protection.

NATs are used widely at present both to hide internal addresses and to reduce the demand for public IP addresses.

Because NATs change the values of IP addresses in packets they interfere with the operation of applications that are aware of IP addresses. The SIP signalling messages may contain end IP addresses in the call-ids, and these addresses will need to be altered as the SIP messages cross a NAT. This is a messy situation and requires an Application Layer Gateway (ALG) to make the necessary changes.

9 Application to SIP and H.323

9.1 Application to SIP

In SIP the call control and service control functions are performed by proxy servers which interrogate:

- redirect servers; or
- location servers.

depending on the type of resolution needed. A SIP server resolves a SIP address to the URL for the next hop. This may be either a service resolution or a routing resolution. A redirect server would perform either service or routing resolutions, whereas a location server would normally perform a routing resolution to reach the location where the called terminal has registered.

It is recommended that a public SIP address for the support of public telephony would have the form:

<E.164 number>@<home network name>

where:

<E.164 number> is the called E.164 number. Although this is the form of SIP address, users would be likely not to put this address on their business cards but to put only their E.164 number.

SIP addresses may also have the form:

<E.164 number>@<server>

where

<server> is the domain name of the SIP server for the range of numbers that contains the called E.164 number. Different SIP servers will be used for different ranges of numbers and these ranges may equate to services or geographic regions or networks.

The URL for the next hop may be a domain name or an IP address. If it is a domain name, then DNS will have to be used to resolve it to an IP address.

EXAMPLE: Consider the routing in figure 8, which is the same as the example in clause 5.1.

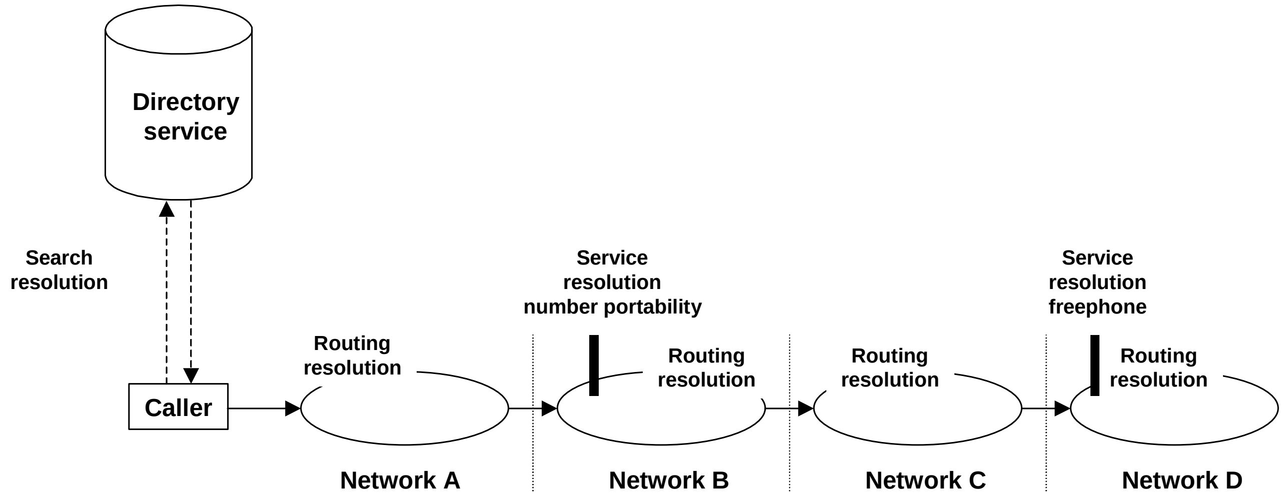

Figure 8: Example of a sequence of resolutions

Suppose all networks use SIP, although some could use H.323. After the search resolution, network A knows only the called number not the home network name.

Network A would interrogate its routing (redirect) server with:

SIP:<called E.164>@<Network A server>

and would obtain the URL for a public number portability server in the area served by network B. Suppose the URL is in the form of an IP address. This is network A's routing resolution.

Network B would interrogate the public number portability server with:

SIP:<called E.164>@<Public number portability server>

and receive the URL of the SIP server in network D. This URL is the home network name. Suppose this is in the domain name form rather than an IP address. This is the service resolution for number portability.

Network B would subsequently interrogate its internal DNS with the URL for network D and obtain an IP address for the server in network C. This is Network B's routing resolution.

Network C would interrogate its internal DNS with the URL for network D and obtain an IP address for the server in network D. This is network C's routing resolution.

Network D would interrogate the freephone server with:

SIP:<called E.164>@<Network D freephone server>

and receive the URL of the terminal. Suppose this is in the domain name form rather than an IP address. This is the service resolution for the freephone service. Network D would then perform a routing resolution using its internal DNS to obtain the IP address of the called terminal at its current location.

This example assumes step-by-step routing. Depending on the information available to the servers and the connectivity at the IP level some steps could be skipped.

9.2 Application to H.323

In H.323 the call control and service control functions are performed by the gatekeeper.

The name for a terminal (endpoint) is called the alias address because the name is considered to be an alias of the transport address. Various forms of alias address are defined in ITU-T Recommendation H.225.0 [11] including E.164 numbers and H.323 Ids that have the form of a email address.

Endpoints are identified by the Transport Address, which consists of an address relating to the network protocol being used (e.g. an IP address) and a TSAP (Transport layer Service Access Point) identifier.

The gatekeeper therefore provides the resolution from alias address to transport address. This resolution may be either a service resolution and routing resolution combined, or just a routing resolution.

Annex G of H.323 defines communications between different administrative domains that use H.323. It defines the exchange between border elements of templates that define:

- the range of alias address identifiers that can be reached;
- pricing information;
- protocols to be used.

Annex G is broadly equivalent to TRIP (see RFC 2871 [24]), which is used for the exchange of routing information to support SIP.

A border element may indicate the gatekeeper to be used for the next step of the routing process.

Annex A:
Overview of SIP

The main function of SIP is to enable a calling host to establish a media path defined by IP addresses and port numbers to a called host that is identified by a SIP address. The SIP address is the same form (user@domain) as an email address, except that the value of "user" could be either a name such as "john_smith" or a telephone number. Domain indicates the user's home network. Once the media path is defined, the media communications (session) are controlled by the Session Description Protocol (SDP).

NOTE 1: See RFC 2327 [25].

In SIP all communications are between clients and servers:

- User Agent Clients (UAC) send SIP messages;
- User Agent Servers (UAS) receive SIP messages;
- proxy servers in networks act as both clients and servers and pass requests and responses to and from other servers;
- redirect servers accept a SIP request, map the address into zero or more new addresses and return these addresses to the client;
- registrar servers accept registration requests.

Figure A.1 shows the operation of SIP in proxy mode.

NOTE 2: SIP can also operate in redirect mode without proxies, but proxies would normally be used in the provision of services.

The calling UAC sends a request (INVITE) message indicating the SIP address of the called party and the type of media communications to which they are being invited. The proxy server in the network sends this message to a redirect server that indicates the URL of the next server to which the message must be sent. The next proxy server accesses a location server to determine the current location of the called party in the form of a URL or IP address. The called party sends a response message (200 = success) to the calling party indicating whether or not they accept the session and giving a Call-id. If the session is accepted then the call identity is given as "call-id@host". "host" may be either a URL or an IP address and it indicates the destination for the requested media session. The calling party sends an ACK message and the media session is then established directly between the calling and called parties using SDP. Either the caller or the called parties may terminate the session using a BYE message.

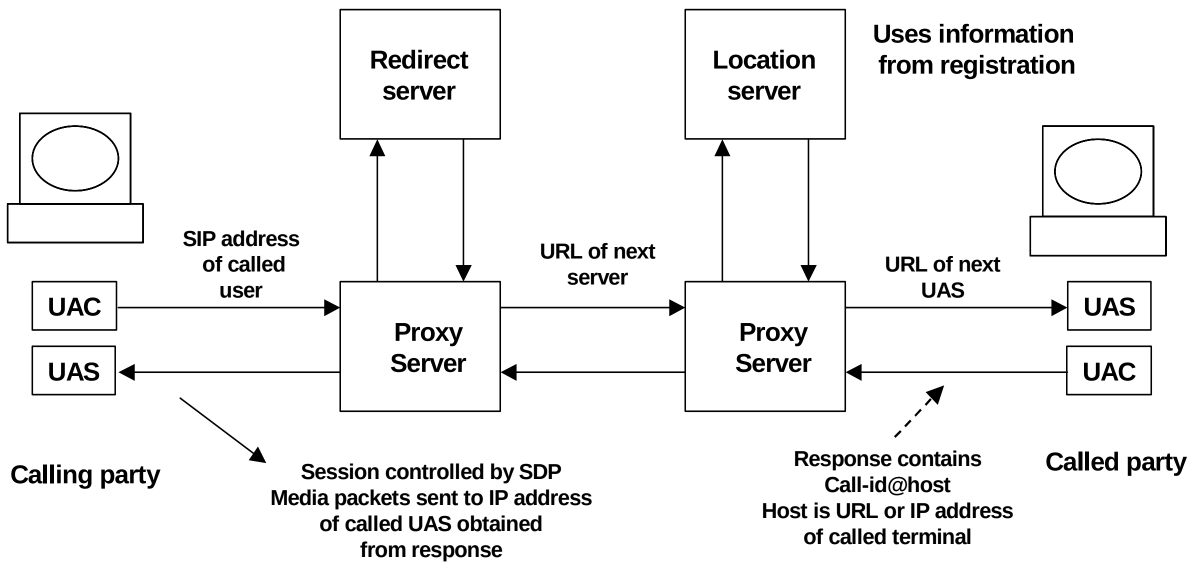

Figure A.1: SIP in proxy mode

The request and response messages contain a "record-route" header that enables proxy servers to add their identity in the form of a URL to a list of the proxy servers that the message has traversed. This list is then used to force all response messages to take the same route as the request messages so that the proxy servers can keep track of the calls.

The proxy servers may be either:

- stateless, in which case they finish their task and then forget what they have done; or
- stateful, in which case they keep a record of their action until a sequence of actions is completed, i.e. they know that a call is in progress until the call is terminated.

Where calls are charged by the minute, the proxy servers will have to be stateful to create a call record for billing and will therefore have to use the record-route function.

SIP provides no control over the routeing of the SDP session media packets, which may take a different route compared to the SIP packets.

Annex B: Overview of H.323

H.323 is the ITU-T's standard for "Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service". H.323 defines the signalling and the components of the system, but it does not define the LAN or transport layer, and therefore can be used for voice and multi-media provided over IP. The signalling concepts in H.323 are based on ISDN access signalling (see ITU-T Recommendation Q.931 [26]).

H.323 is a "system" standard that makes reference to:

- ITU-T Recommendation H.225.0 [11]: Call signalling protocols and media stream packetization for packet based multimedia communications systems;
- H.245 Control of communications between visual telephone systems and terminal equipment;
- various standards in the H-series on video codecs;
- various standards in the G-series on audio codecs.

H.323 defines the signalling between:

- endpoints, which are terminals or gateways, and
- gatekeepers, which may manage the communications of terminals.

Figure A.2 shows the general structure:

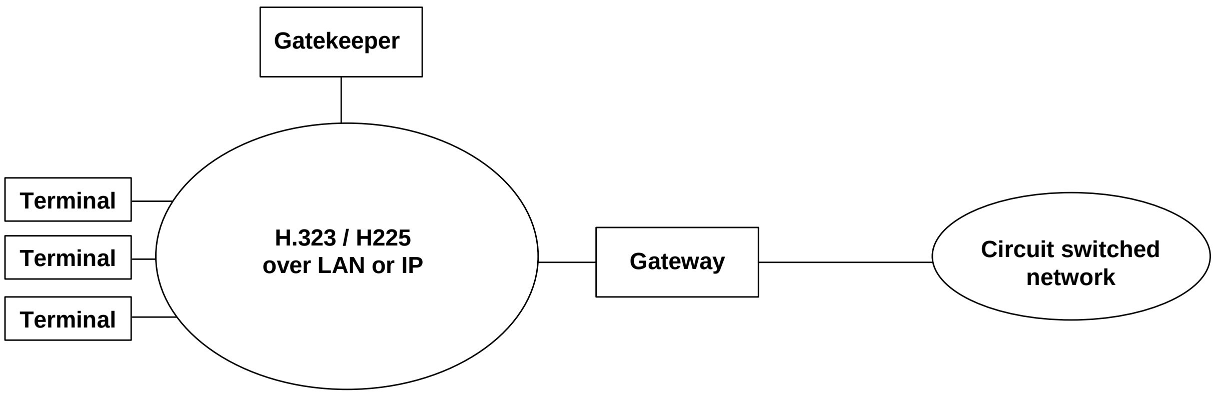

Figure A.2: H.323

There are four stages in a communication (see ITU-T Recommendation H.323 [22], section 7.3.1):

- signalling between the calling endpoint and the gatekeeper to obtain admission to the network;
- signalling between the calling and called endpoints to establish the call. This signalling may go either direct or via the gateway;
- establishment of the media control channel using the in-band H.245 protocol. This signalling may go either direct or via the gateway;
- the media communications themselves using the same transport addresses as the media control channel.

Each endpoint of an information flow is identified by a Transport Address, which consists of an address relating to the network protocol being used (e.g. an IP address) and a TSAP (Transport layer Service Access Point) identifier, which allows multiplexing of flows for a single terminal. Endpoints may have separate transport addresses for signalling and media.

Endpoints may also have alias addresses which include E.164 numbers, Internet names and other identifier strings. The gateways provide translation between aliases and transport addresses if incoming traffic is sent to an alias address.

History

Document history		
V1.1.1	September 2000	Publication (Historical)
V2.0.0	February 2002	Publication