Signalling Protocols for NGN

Riccardo Passerini, ITU-BDT
Subscriber Growth

Fixed

Mobile

Minute migration

Fixed Internet

Mobile Internet

(Millions)

0 150 300 450 600 750 900 1050 1200 1350


ITU/ITC Regional Seminar on Network Evolution to NGN and Fixed Mobile Convergence, Moscow, 27-30 April 2004 - Riccardo Passerini ITU/BTD
Calling opportunities worldwide

Source: ITU Fixed-Mobile Interconnect website: http://www.itu.int/interconnect
Convergence

Computer & Data

Entertainment & Publishing

Consumer Electronics

Telecom

Business

Home

Information & Work Support

Multimedia

Information & Gadgets
Convergence

• The coming together of telecommunications, computing and broadcasting into information and communications technologies (ICT)
• Within telecom the convergence of voice and data and fixed and mobile services
• ICT uses same:
  – Technology used to code voice, data and video
  – Carrier for voice, data and video
• Expands the range and quality of services
• Requires broadband technologies
• Encourages the use of a single communications regulator
The emergence of the ‘networked’ society’

We are at the outset of a truly remarkable revolution where

- Anything that can be connected will be!
- Anything that can be digital will be!
- Anything that can become mobile will become!

Dismantling of traditional industry structures, disaggregation of traditional business models, a wealth of opportunities and considerable threats
Digital divide = Telecoms divide
User distribution, by income group, Jan 2000

Source: ITU World Telecommunication Indicators Database.
Digital divide reflected in the Telecoms divide
User distribution, by income group, Jan 2003

Source: ITU World Telecommunication Indicators Database.
The digital divide is shrinking, but also shifting

Share of low and lower-middle income countries in:

- **Telephone main lines**
- **Mobile subscribers**
- **Estimated Internet Users**

**Source:** ITU World Telecommunication Indicators Database.
The digital divide in January 2003

**Telephone lines**
- Upper-mid income + High income: 58%
- Low income + Lower-mid income: 42%

**Mobile users**
- Upper-mid income + High income: 64%
- Low income + Lower-mid income: 36%

**Internet users**
- Upper-mid income + High income: 76%
- Low income + Lower-mid income: 24%

*Source: ITU World Telecommunication Indicators Database.*

The growing importance of the telecommunication sector

Telecommunications are a crucial factor of efficiency for the administrations, public utilities and private companies.

- Synergy with computers for data processing,
- Faster information and better dialogue

Telecommunications play a crucial role for increasing the competitiveness of enterprises:

- Better productivity and better services
- More jobs with added values with new services
- Less intermediary positions without added values
Results of convergence of ICT
(Information and Communication Technology)

The telecommunications sector evolves in a broader « ICT » sector

Till 1980
Telecommunications
Informatics
Electronic media

After 1980
Telematics
Electronic media

After 1995
ICT or Mediamatics
Evolution Architecture

- IP can be considered as a “Data NGN”
Example: PSTN/SS7 signalling in IP Network

SS7 Links

SCP

STP

STP

MGC

MGC

MGC

IP Network

Media Gateway

Media Gateway

Media Gateway

IP terminal

Signalling Gateway

Signalling Gateway

Signalling Gateway

Phone (E.164)

Voice Trunks

SP

SP

IP terminal

Signalling Requirements to Support "IP telephony"

Network configuration A (phone to phone communication)
Terms and definitions (1)

*Media Gateway (MG)*:

Terminates voice calls on inter-switch trunks from the PSTN, compresses and packetizes the voice data, and delivers compressed voice packets to the IP network. For voice calls originating in an IP network, the media gateway performs these functions in reverse order. A Media Gateway converts the media provided by one type of network to the format required by another type of network. For example, an MG could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network). This gateway could be capable of processing audio, video and T.120 alone or in any combination, and would be capable of full-duplex media translations. The MG may also play audio/video messages and perform other IVR functions, or provide media conferencing. In the context of this document, the term “Media Gateway” refers to a voice gateway.
Terms and definitions (2)

**Media Gateway Controller (MGC):**

Controller that controls the parts of the call state that pertain to connection control for the media channels within a Media Gateway. A Media Gateway Controller handles the registration and management of resources at the Media Gateway(s). A Media Gateway Controller exchanges ISUP messages with central-office switches via a signaling gateway. Because vendors of media gateway controllers often use off-the-shelf computer platforms, a media gateway controller is sometimes called a softswitch.

**Call Agent (CA):**

Function that controls the provision of services to users.
Terms and definitions (3)

**Signalling Gateway (SG):**

A Signalling Gateway provides transparent interworking of signalling between switched-circuit and IP networks. The Signalling Gateway may terminate PSTN/SS7 signalling or relay messages over an IP network to a media gateway controller or another signalling gateway. Because of its critical role in integrated voice networks, signalling gateways are often deployed in groups of two or more to ensure high availability.

**Telephone Number Mapping (ENUM):**

Protocols for mapping telephone numbers to IP phone identifiers (i.e. E.164 numbers to URIs).
Terms and definitions (4)

**IP Network:**

An IP network is a network that uses IP technologies to transport information. It may be a Private IP network, or a Carrier’s network.

**Phone:**

Phone refers to a PSTN terminal.

**IP Phone:**

IP phone refers to a terminal (e.g. dedicated voice terminal or multipurpose personal computer) that is connected directly (e.g. Through an Ethernet interface or an xDSL line) to an IP network.
Terms and definitions (5)

**IP telephony:**

“IP telephony” is a service that enables the exchange of voice information, primarily in the form of packets, using IP protocols.

**Internet Telephony:**

The combination of the term ‘Internet’ with the term ‘telephony’ is regarded as a specific use of the Internet, rather than a service. The Internet offers many capabilities to users, including the ability to carry bi-directional speech in real time or near real time. This is considered to be an intrinsic capability of the Internet and not a telecommunications service.
Network configuration A (phone to phone communication)

This configuration uses the PSTN to originate and terminate a call (using the switching function of an existing PSTN), and converts speech into IP packets in the transit network.
In the IWF (such as MG, MGC, SG function) between the PSTN and IP network at the originating and terminating sides, control signalling (ISUP - H.323 / SIP conversion) and information signalling (64-kbps bearer - IP packet conversion) are converted. In the IP network, a call is controlled by the H.323 / SIP protocol. A user dials a phone number to identify the terminating phone terminal and also, in some cases, additional information (e.g. through the use of prefix dialling) to select an IP transit network.
Network configuration B1 (IP phone to phone communication)
Network configuration B1 (IP phone to phone communication)

In this configuration, the originating network is an IP network and the terminating network is a PSTN. In the IWF (such as MGC, MG, SG functions) between a PSTN and an IP network at the terminating side, the signalling protocol (ISUP - H.323 / SIP conversion) and the user information (64-kbps bearer - IP packet conversion) are converted.

In the IP network, a call is controlled by the H.323 / SIP protocol.

The originating IP phone user dials a phone number to identify the terminating phone terminal.
Network configuration B2 (phone to IP phone communication)
Network configuration B2 (phone to IP phone communication)

In this configuration, the originating network is a PSTN and the terminating network is an IP network.

In the IWF (such as MGC, MG, SG functions) between a PSTN and IP network at the originating side, the signalling protocol (ISUP - H.323 / SIP conversion) and the information (64-kbps bearer - IP packet conversion) are converted. In the IP network, a call is controlled by the H.323 / SIP protocol. The originating phone user dials a phone number to identify the terminating IP phone terminal.
Configuration C: IP Phone to IP Phone communication
Configuration C: IP Phone to IP Phone communication

In this configuration, all networks are IP. Calls are controlled by H.323/SIP signalling in the IP network. The terminating IP phone user is identified by an ID (e.g. a sequence of alphanumeric characters). The network operator assigns an ID to each user as they are registered. In addition to IDs, the IP phones can also have phone numbers which can be used to dialing (in the call control level IDs are used).
Network capabilities to support “IP telephony” interworking between the PSTN and the IP network

Network configuration connecting carriers (NNI between IP-IP network)
In the case of NNI between IP-IP networks, when the protocol (H.323, SIP, and so on,) being used by the IP network differs between carriers, a study of the protocol conversion feature is necessary. Also, in the future, it will be necessary to study the agreement on the quality assurance and quality evaluation between the different IP network carriers.
(Example: configuration B1 and B2)

Network configuration connecting carriers (NNI between PSTN-IP network)

Note: A study of the signalling requirement for the interworking between the PSTN and the IP network must be done in ITU-T
Control Protocols for support of “IP telephony”

Example of protocol adoption

(Example: configuration B1 and B2)
Control Protocols for support of “IP telephony”

- Call Control protocols: SIP (IETF), H.323 (ITU-T), BICC (ITU-T)
- Media Gateway Control Protocols: H.248 (ITU-T) /Megaco (IETF)
- Signalling transport protocols: UDP (IETF), TCP (IETF) and SCTP (IETF) including the specified adaptation layers.
- Media Transport Protocols: RTP/RTCP (IETF) over UDP (IETF)
Protocol Stacks (1)

Protocol stack (example for configurations B-1 and B-2)
Protocol Stacks (2)

Protocol stack (example for configurations B-1 and B-2)
General Framework for migrating Telephony networks towards Next Generation Networks (NGN)

In markets with a high growth in traditional voice services (which is the case for most developing countries), substantial extensions will be required to the existing telephony network in order to cover the huge need for new lines. Established Service Providers will have to decide on how to extend their networks: using more traditional circuit-switched solutions or implementing a distributed network architecture, with a common, packet-based transport layer for voice and data.
For this to occur many aspects like network consolidation, expansion and migration need to be taken into account in a way that is specific for each operator. However one can devise the following generic step-wise approach. Such steps are generic in the sense that they are not mandatory for each specific operator case. Still, they offer interest by highlighting major evolutionary steps of networks that might occur in the following years.

*Step 1*: use of today’s TDM based network for voice telephony and Internet access

*Step 2*: consolidation of switching and access equipment;

*Step 3*: introduction of Voice-over-Packet technology for trunking;

*Step 4*: introduction of Voice-over-Packet technology in access and CPE

*Step 5*: multimedia services and new applications;

*Step 6*: end-of-life replacement of legacy infrastructure and migration to all-IP signalling.
Step 1: PSTN for Voice and Internet
Step 1: PSTN for Voice and Internet

**TDM and SS7 [A]**
In this network, all voice traffic is transported over TDM, and controlled by a hierarchy of local (LEX or Class 5) and transit (TEX/Class 4) circuit switches. All the voice related signalling network (ISUP and INAP) is handled by the SS7 signalling network.

**Intelligent Network Services [B]**
Value Added Services are provided inside the switches, or through the Intelligent Network (IN). Widely spread IN services include Calling Card services, Number Translation and routing services (such as Freephone, Premium Rate and Universal Access Number), and Enterprise Network services (such as Virtual Private Networks and Wide Area Centrex).

**Internet Access [C]**
With the growing number of Internet users, carriers are providing connectivity to Internet Service Providers (ISP) either through narrow-band (PSTN or ISDN) dial-up services, or through introduction of broadband ADSL (with voice split off as a separate service).
Step 2: PSTN Consolidation
Step 2: PSTN Consolidation

Established Carriers in growing markets will face major capital expenses (CAPEX) to extend the capacity of their network in line with the growth in subscribers. These investments will be needed at all levels of the network, local level as well as in the long distance network.

A safe approach (from a technical as well as from an economical viewpoint) is to start with consolidation of the existing PSTN infrastructure while selecting “NGN-ready” products for expansion, and introduction of new services for additional revenue generation.
Step 2: PSTN Consolidation (1)

Switch Consolidation [D]
Deployment of a small number of large exchanges (local and transit) with increased switching capacity, and high-speed interfaces (e.g. SDH, ATM) reduce the Operator’s operational expenses (OPEX) and enable faster deployment of new services. “Redundant” switches may be converted to additional remote access concentrators.

Introduction of new technology with e.g. smaller footprint, or packet fabrics inside the exchanges, allows the carrier to reduce expenses and reuse the switching equipment for new data services.

Access Consolidation [E] and VoDSL [F]
Adding new Access Nodes and upgrading the existing ones lets the carrier capitalise on his PSTN, while extending the coverage area and the bandwidth offered to individual subscribers (fibre closer to the end user). New access technology provide seamless multi-service access to voice (POTS, ISDN) and data (ADSL, ATM, IP, …) services and pave the way to Next Generation Networks.

Optimisation of the ADSL access infrastructure is realised through introduction of Voice-over-DSL (VoDSL) loop-emulation services (inverse gateway, with a V5.2/GR303 connection to the LEX).
Step 2 : PSTN Consolidation (2)

**IN-Internet Convergence Services [G]**

Providing an external server to the PSTN and the Internet, the IN Service Control Point (SCP) may be used as a means to integrate voice and data into common applications. Example IN-Internet convergence applications are Click-to-Dial, Internet Call Waiting, Web Augmented Calling, Unified Messaging, etc. In order to communicate with the Internet servers, the SCP has to adopt some IETF protocol suites (e.g. PINT and SPIRITS).

**Open Service Access [H]**

To prepare for the NGN and to get extra revenues from new services, the network operator may deploy Application Gateways (ApGW) with open interfaces (e.g. OSA/Parlay, JAIN, SIP) towards (3rd party) Application Servers (AS).
Step 3: Voice over Packet for Trunking
Step 3: Voice over Packet for Trunking (1)

As one of the basic goals of NGN introduction is to move to a unique, packet-based infrastructure (presumed with lower OPEX and CAPEX), voice transport will smoothly migrate to IP or ATM technology. Initially, carriers will focus on trunking scenarios to offload long-distance voice from their TDM network.
Step 3: Voice over Packet for Trunking (2)

**Trunking through Integrated Gateways [I]**
The first step towards VoP migration is extending the existing (local) exchanges with integrated Trunking Gateways (TGW) for converting TDM voice into packets (ATM or IP). This approach guarantees full protection of TDM investments, while providing the operator with a full fledged trunking-over-packet solution, as well as continued access to switch based and IN based Value Added Services.

**GW [J] with Class 4 Softswitch [K]**
In order to address existing switches without integration of a gateway, external trunking gateways (TGW) controlled by a Class 4 Softswitch (through the H.248 or Megaco[1] protocol may be added. From a functional point-of view, the Softswitch performs like a Class 4 (Toll/Transit) Exchange, with similar features (e.g. screening and routing), signalling interfaces (ISUP, INAP) and access to Value Added Services (IN).

[1] For early deployment the MGCP protocol – which is a precursor to H.248/Megaco protocol – might be used due to the availability of gateways supporting it.
Step 4: Voice over Packets in access and CPE
Step 4: Voice over Packets in access and CPE (1)

In fast growing markets or in markets with aggressive deployment of broadband access (ADSL, LMDS, cable) operators may introduce voice-over-packet technology to capture growth in the access network, or as a means to offload the Local Exchanges from DSL.
Step 4: Voice over Packets in access and CPE (2)

Class 5 Softswitch [L]
The Class 5 Softswitch with local features (e.g. CLASS, custom calling) will be a shared control element, but several alternatives for voice gateways (depending on end-users topology, density, service requirements, etc.) may be deployed [figure 4]. Just as in the Class 4 case, the softswitch will address the gateways using the H.248/Megaco protocol (or MGCP).

Residential Gateway [M]
ADSL subscribers may install a Residential Gateway (RGW) or Integrated Access Device (IAD) with VoP coding capability. Contrary to the ADSL with split-off voice [B] or VoDSL loop emulation [E] solutions the RGW provides the broadband user with end-to-end voice-over-packet.
Step 4: Voice over Packets in access and CPE (3)

**Access Gateway in the DSLAM [N]**
As an alternative to upgrading the CPE of its subscribers, an ADSL operator may choose to extend the DSLAMs with VoP gateway functionality.

**Distributed Access Gateways [O, P]**
Another solution for connecting the voice subscribers directly to the data network is to introduce new Access Gateways [AGW] or to upgrade the existing access nodes with AGW functionality.

**IP Phones [Q]**
In order to address new generation voice terminals (IP Phones), the Class 5 Softswitch can also terminate emerging user-to-network signalling protocols such as H.323 and SIP.
Step 5: Multimedia
It is beyond question that in the near (and even midterm) future voice will be the predominant service, even in Next Generation Networks. The introduction of broadband access in the network, however, is enabling the deployment of a new range of data and multimedia services. These new services will allow carriers to differentiate and compete with new entrants.
Step 5 : Introduction of Multimedia (2)

**IP Clients [R] with MM Softswitch [S]**

A prerequisite for the deployment of multimedia services is the general availability of appropriate terminals. Today's Personal Computers are a good starting point, but it is expected that the convergence of computer, consumer and communications technology will result in a number of new multimedia devices. These new terminals will communicate with the softswitch through emerging multimedia signalling protocols such as H.323 and SIP. In order to fully support the new network and terminal capabilities, the Softswitch is extended with mixed-media session and QoS control.
Retailer Portal and Open Interfaces [T]

With the introduction of new business models and new players (e.g. Virtual Network Operators, 3rd party application providers, content providers), there is a need for application access (for authentication, authorisation, accounting, roaming, subscriber profiles, etc.) and service brokering platforms (terminal capabilities negotiation, bandwidth brokering, content aggregation, etc.). Such portals do not only provider the Network Operator with new business opportunities as a Service Retailer, but also clearly separate the network control from the services functionality.

In a full-fledged NGN architecture, applications and network will interface through standardised protocols (e.g. SIP) and APIs (e.g. JAIN, OSA/Parlay).
New Applications [U]

From an applications (and thus a revenue) viewpoint ‘plain vanilla’ Voice-over-Packet is not considered as a differentiator. It is even assumed that voice services offered on VoP networks will have fewer features than the ones on circuit networks (especially in a H.323 environment). Therefore evolution of the applications portfolio towards data and multimedia is considered an absolute prerequisite for telecom service providers to differentiate, grow and generate new revenues. Typical examples of multimedia applications include:
- Mixed-media calls/conferences
- Real-time data streaming
- Instant Messaging, Presence and Location services
Massive deployment of new applications will be enabled by the availability of application servers and terminals, with easy-to-use service creation tools.
Step 6: The full NGN
Step 6 : Migration to the full NGN (1)

As a final migration step toward the full NGN, the remaining legacy PSTN equipment is transformed to or replaced by NGN ‘compliant’ network components. Aim of this ultimate (though optional) transformation, is to capitalise on existing CAPEX (e.g. access concentrators connected to local exchanges) while further reducing the OPEX (packet-only network for transport and signalling).
Step 6: Migration to the full NGN (2)

**Replacement of Legacy Equipment [V]**

At the end of their life, remaining TDM exchanges and access nodes are gracefully transformed to or replaced by Trunking Gateways, Access Gateways and Softswitches as outlined in the previous sections.

**Migration to all-IP Signalling [W]**

While keeping the upper layers (SCCP, ISUP, TCAP, INAP) intact, the lower layers of the SS7 signalling network are replaced by a packet-based equivalent, as defined by the IETF SIGTRAN working groups.
### Protocols: Status of studies being undertaken by international standards organizations

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Call control</th>
<th>Media control</th>
<th>Interwork</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>H.323</td>
<td>ITU-T SG16</td>
<td>H.323</td>
</tr>
<tr>
<td></td>
<td>SIP</td>
<td>IETF SIP-WG</td>
<td>RFC3261</td>
</tr>
<tr>
<td></td>
<td>H.248/MEGACO</td>
<td>IETF MEGACO-WG</td>
<td>RFC3015</td>
</tr>
<tr>
<td></td>
<td>ITU-T SG16</td>
<td>H.248</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Protocol</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>RTP/RTCP</td>
<td>IETF MMUSIC-WG</td>
<td>RFC1889</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIP-ISUP inter-working</td>
<td>IETF SIPPING-WG</td>
<td>RFC3398</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Note: For the latest draft document, see URL of SIPPING-WG</td>
</tr>
<tr>
<td></td>
<td>ITU SG11</td>
<td></td>
<td>TRQ.2815 supplement 45</td>
</tr>
<tr>
<td></td>
<td>ITU SG11</td>
<td></td>
<td>Q.1912.5</td>
</tr>
</tbody>
</table>

**Note:** [http://www.ietf.org/html.charters/sipping-charter.html](http://www.ietf.org/html.charters/sipping-charter.html)
## ANNEX: Abbreviations

<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADPCM</td>
<td>Adaptive Differential Pulse Code Modulation</td>
</tr>
<tr>
<td>ADM</td>
<td>Adaptive Delta Modulation</td>
</tr>
<tr>
<td>ADSL</td>
<td>Asymmetric Digital Subscriber Line</td>
</tr>
<tr>
<td>AGW</td>
<td>Access Gateway</td>
</tr>
<tr>
<td>AN</td>
<td>Access Node</td>
</tr>
<tr>
<td>API</td>
<td>Application Programming Interface</td>
</tr>
<tr>
<td>ATM</td>
<td>Asynchronous Transfer Mode</td>
</tr>
<tr>
<td>BAS</td>
<td>Broadband Access Server</td>
</tr>
<tr>
<td>BICC</td>
<td>Bearer Independent Call Control</td>
</tr>
<tr>
<td>CAPEX</td>
<td>Capital Expenditure</td>
</tr>
<tr>
<td>CCF</td>
<td>Call Control Function</td>
</tr>
<tr>
<td>CPE</td>
<td>Customer Premises Equipment</td>
</tr>
<tr>
<td>CME</td>
<td>Circuit Multiplication Equipment</td>
</tr>
<tr>
<td>DPCM</td>
<td>Differential Pulse Code Modulation</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name System</td>
</tr>
<tr>
<td>DSL</td>
<td>Digital Subscriber Line</td>
</tr>
<tr>
<td>DSLAM</td>
<td>Digital Subscriber Line Access Multiplexer</td>
</tr>
<tr>
<td>DFFSERV</td>
<td>Differentiated Services</td>
</tr>
<tr>
<td>FTP</td>
<td>File Transfer Protocol</td>
</tr>
<tr>
<td>HTTP</td>
<td>Hyper Text Transfer Protocol</td>
</tr>
<tr>
<td>IETF</td>
<td>Internet Engineering Task Force</td>
</tr>
<tr>
<td>INAP</td>
<td>Intelligent Network Application Part</td>
</tr>
<tr>
<td>INTSERV</td>
<td>Integrated Services</td>
</tr>
<tr>
<td>IP</td>
<td>Internetworking Protocol</td>
</tr>
<tr>
<td>IPTN</td>
<td>IP Telephony Network</td>
</tr>
<tr>
<td>ISUP</td>
<td>ISDN User Part</td>
</tr>
<tr>
<td>ITSP</td>
<td>Internet Telephony Service Provider</td>
</tr>
<tr>
<td>JAIN</td>
<td>Java API for Integrated Networks</td>
</tr>
<tr>
<td>LAN</td>
<td>Local Area Network</td>
</tr>
<tr>
<td>LEX</td>
<td>Local Exchange</td>
</tr>
<tr>
<td>LMDS</td>
<td>Local Multipoint Distribution System</td>
</tr>
<tr>
<td>MEGACO</td>
<td>Media Gateway Control (an IETF Workgroup)</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>MOS</td>
<td>Mean Opinion Score</td>
</tr>
<tr>
<td>MPEG</td>
<td>Motion Picture Expert Group</td>
</tr>
<tr>
<td>MPLS</td>
<td>Multi-Protocol Label Switching (an IETF W.G.)</td>
</tr>
<tr>
<td>NAS</td>
<td>Network Access Server</td>
</tr>
<tr>
<td>NAT</td>
<td>Network Address Translation</td>
</tr>
<tr>
<td>NGN</td>
<td>Next Generation Network</td>
</tr>
<tr>
<td>NT</td>
<td>Network Termination</td>
</tr>
<tr>
<td>OPEX</td>
<td>Operational Expenditure</td>
</tr>
<tr>
<td>OSA</td>
<td>Open Service Access</td>
</tr>
<tr>
<td>OSI</td>
<td>Open System Interconnection</td>
</tr>
<tr>
<td>PABX</td>
<td>Private Automatic Branch Exchange</td>
</tr>
<tr>
<td>PC</td>
<td>Personal Computer</td>
</tr>
<tr>
<td>PHT</td>
<td>Public Key Infrastructure</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>QoS</td>
<td>Quality of Service</td>
</tr>
<tr>
<td>RGW</td>
<td>Residential Gateway</td>
</tr>
<tr>
<td>RTCP</td>
<td>Real-time Transmission Control Protocol</td>
</tr>
<tr>
<td>RTP</td>
<td>Real-time Transport Protocol</td>
</tr>
<tr>
<td>SCP</td>
<td>Signalling Control Point</td>
</tr>
<tr>
<td>SCCP</td>
<td>Signalling Connection Control Part</td>
</tr>
<tr>
<td>SCN</td>
<td>Switched Circuits Network</td>
</tr>
<tr>
<td>SCTP</td>
<td>Signaling Connection Transfer Protocol</td>
</tr>
<tr>
<td>SIGTRAN</td>
<td>Signalling Transport (an IETF W.G, PSTN over IP)</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SLA</td>
<td>Service Level Agreement</td>
</tr>
<tr>
<td>SLS</td>
<td>Service Level Specification</td>
</tr>
<tr>
<td>SPIRIT</td>
<td>Service PSTN/IN Requesting Internet Services (IETF W.G)</td>
</tr>
<tr>
<td>SS7</td>
<td>Signalling System N°7</td>
</tr>
<tr>
<td>STP</td>
<td>Signalling Transfer Point</td>
</tr>
<tr>
<td>TEX</td>
<td>Transit Exchange</td>
</tr>
<tr>
<td>TCAP</td>
<td>Transaction Capabilities Application Part</td>
</tr>
<tr>
<td>TCP</td>
<td>Transmission Control Protocol</td>
</tr>
<tr>
<td>TDM</td>
<td>Time Division Multiplexing</td>
</tr>
<tr>
<td>TGW</td>
<td>Trunking Gateway</td>
</tr>
<tr>
<td>TLS</td>
<td>Transmission Level Security</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
</tr>
<tr>
<td>URI</td>
<td>Universal Resource Identification</td>
</tr>
<tr>
<td>VoDSL</td>
<td>Voice over Digital Subscriber Line</td>
</tr>
<tr>
<td>VoIP</td>
<td>Voice over IP</td>
</tr>
<tr>
<td>VPN</td>
<td>Virtual Private Network</td>
</tr>
<tr>
<td>WAN</td>
<td>Wide Area Network</td>
</tr>
</tbody>
</table>