ITU-BDT Regional Seminar on Fixed Mobile Convergence and new network architectures for the Arab Region

Tunis, Tunisia, 21-24 November 2005

1.3.12/Part 1: Signalling Protocols and Evolving Architectures for NGN

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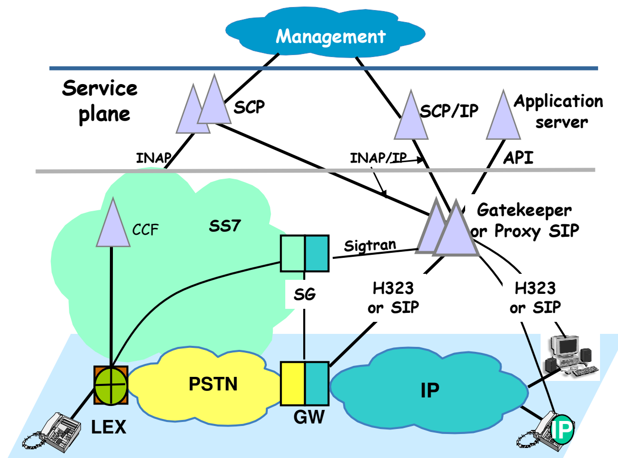
Evolution of PSTN/ISDN to NGN

NGN (Next Generation Network) is believed to provide new opportunities for and capabilities to the network and service providers. Considering that existing networks have different life span and vast amount of capital has been spent on them, complete replacement of their components is not considered to be either advisable or possible. So, a phased approach should be considered for evolution of existing networks to NGN.

PSTN/ISDN (Public Switched Telephone Network/Integrated Services Digital Network) being one of the first networks, is considered to be prime candidate for evolution. For PSTN/ISDN evolution to NGN a phased approach is considered

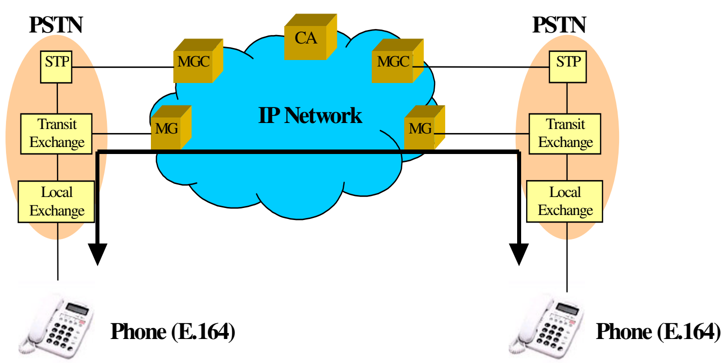
Different evolutionary Scenarios with PSTN/ISDN emulation (adaptation to IP infrastructure) and with PSDN/ISDN simulation (session control over IP interfaces and infrastructure) are presently under consideration in ITU in order to providereference for the evolution to NGN

Evolution Architecture



➔ IP can be considered as a " Data NGN "

Signalling Requirements to Support "IP telephony"

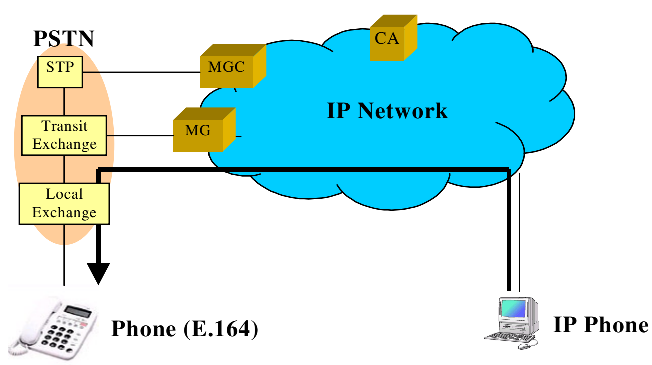


Network configuration A (phone to phone communication)

Configuration A: Phone to Phone Communication (with IP Transit Network)

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.

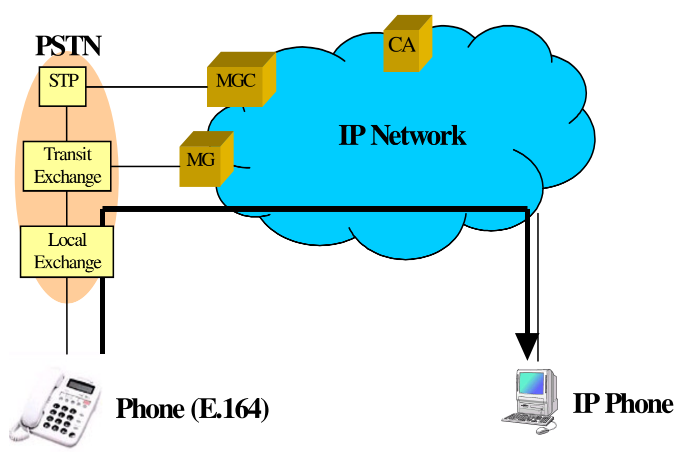


Network configuration B1 (IP phone to phone communication)

Configuration B-1: 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.

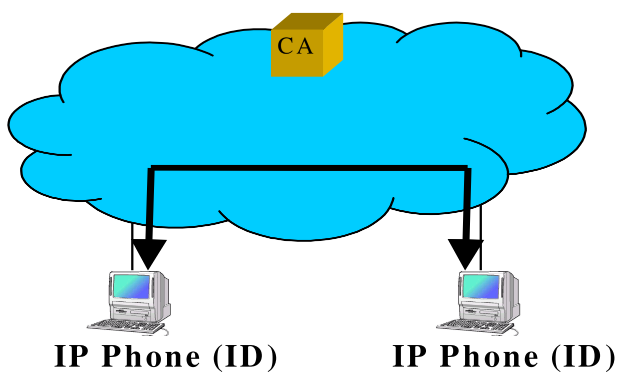


Network configuration B2 (phone to IP phone communication)

Configuration B-2: 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

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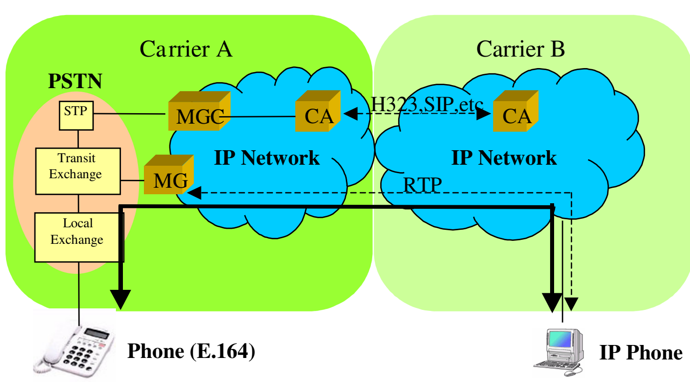
Configuration C: IP Phone to IP Phone communication

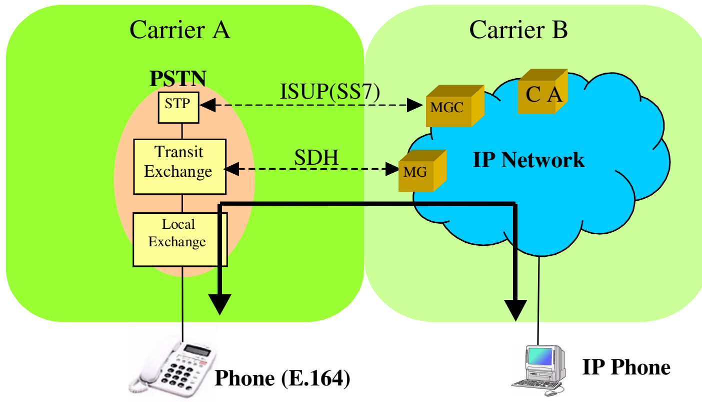
In this configuration, all networks are IP.

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

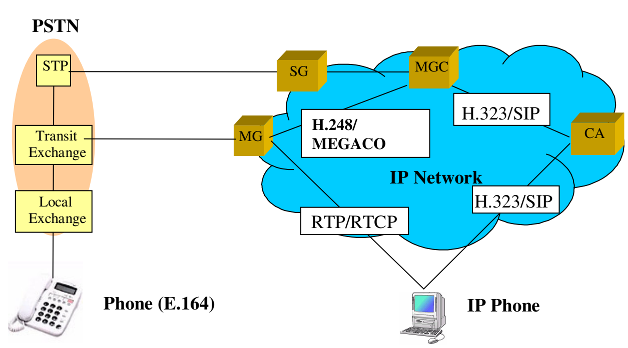


Network configuration connecting carriers (NNI between IP-IP network)



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

Control Protocols for support of "IP telephony"



Example of protocol adoption

TERMS AND DEFINITIONS

Media Gateway (MG): **A media gateway converts the media provided by one type of network to the format required by another type of network. 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.**

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 (described below). Because vendors of media gateway controllers often use off-the-shelf computer platforms, a media gateway controller is sometimes called a softswitch.**

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**

Call Agent (CA): **Function that controls the provision of services to users.**

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TERMS AND DEFINITIONS

Telephone Number Mapping (ENUM): Protocols for mapping telephone numbers to IP phone identifiers (i.e. E.164 numbers to URIs).

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.

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

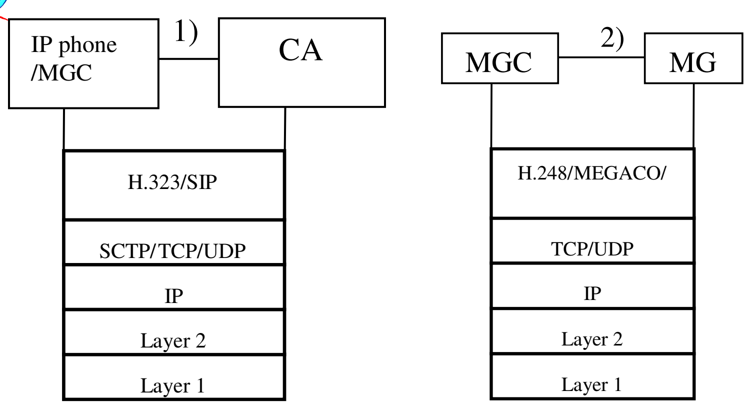
Internet Telephony: The combination of the term 'Internet' with the term 'telephony' is regarded as a specific use of the Internet, rather than a service. The Internet offers many capabilities to users, including the ability to carry bi-directional speech in real time or near real time. This is considered to be an Intrinsic capability of the Internet and not a telecommunications service.
(Note) Internet telephony is a particular application of the Internet and therefore falls outside of the scope of this document.

CONTROL PROTOCOLS FOR SUPPORT OF "IP TELEPHONY"

This section describes the protocol stacks used for the call control and media transport in "IP telephony".

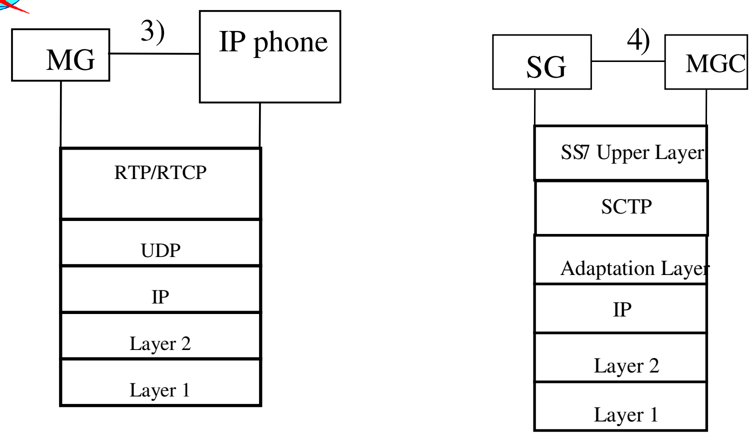
- **Call Control protocols: SIP (IETF), H.323 (ITU-T), BICC (ITU-T)**
- **Media Gateway Control Protocols: H.248(ITU-T) /Megaco(IETF)**
- **Signalling transport protocols: UDP(IETF), TCP(IETF) and SCTP(IETF) including the specified adaptation layers.**
- **Media Transport Protocols: RTP/RTCP(IETF) over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

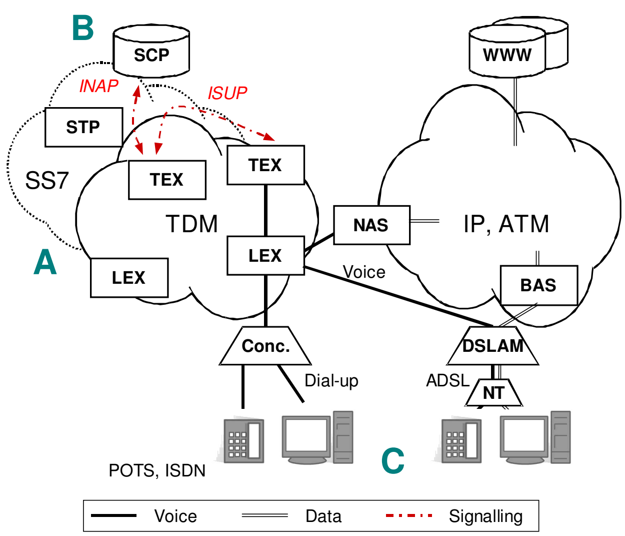
General Framework for migrating Telephony networks towards Next Generation Networks (NGN)

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.

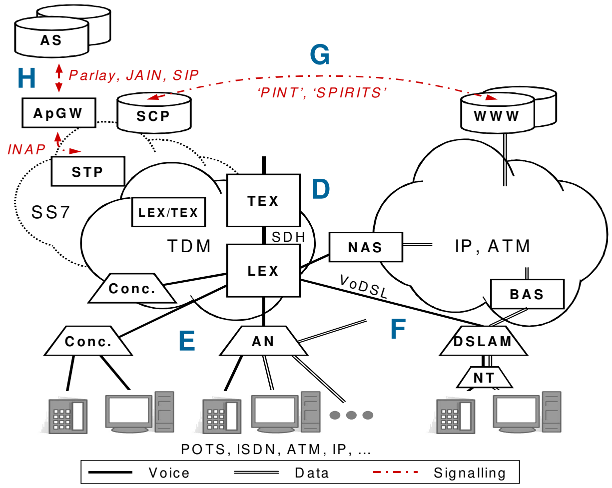
General Framework for migration to NGN (The Essential Report on IP telephony – ITU-D

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.

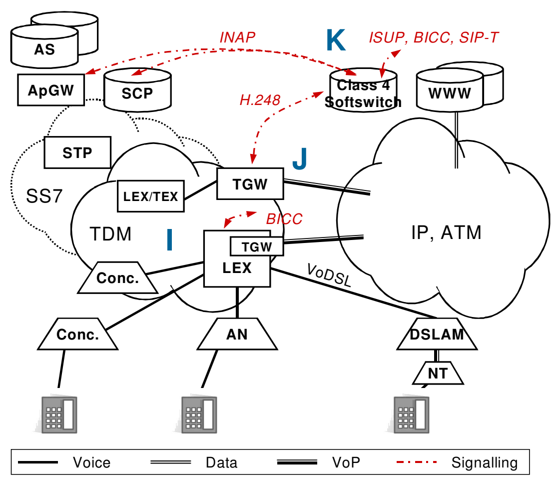
Step 1: use of today's TDM based network for voice telephony and Internet access
Step 2: consolidation of switching and access equipment;
Step 3: introduction of Voice-over-Packet technology for trunking;
Step 4: introduction of Voice-over-Packet technology in access and CPE
Step 5: multimedia services and new applications;
Step 6: end-of-life replacement of legacy infrastructure and migration to all-IP signalling.



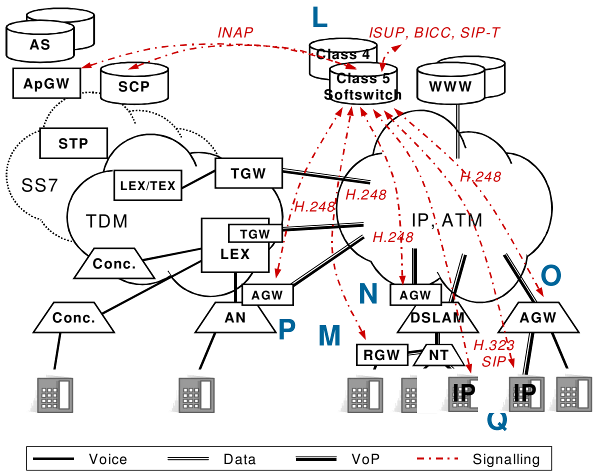
Step 1: PSTN for Voice and Internet



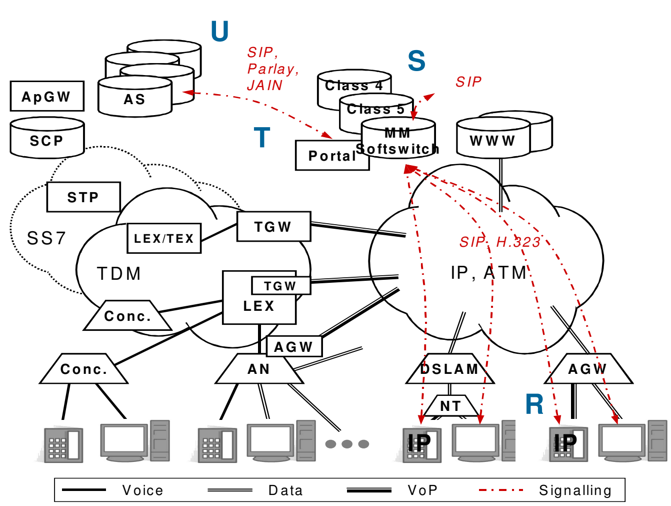
Step 2 : PSTN Consolidation



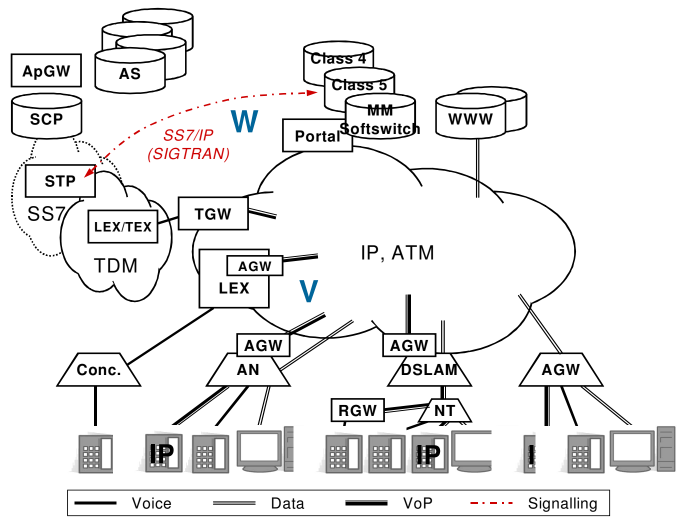
Step 3: Voice over Packet for Trunking



Step 4: Voice over Packets in access and CPE



Step 5 : Multimedia



Step 6 : The full NGN

Protocols: Status of studies being undertaken by international standards organizations

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

Note: http://www.ietf.org/html.charters/sipping-charter.html

ANNEX: Abbreviations

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