Product Family

SS7 Pocket Guide

WHAT IS SS7?

Signaling System No.7 (SS7) is a global standard for telecommunications defined by the International Telecommunication Union (ITU) Telecommunication Standardization Sector (ITU-T).

SS7 defines the procedures and protocol by which network elements in the public switched telephone network (PSTN) exchange information over a digital signaling network to effect wireless (cellular) and wireline call setup, routing and control, as well as, network management and maintenance.

POCKET GUIDE OVERVIEW

This tutorial provides an introduction to basic SS7 network concepts. SS8 is also sponsoring the ss7.com web site where you can find more information on SS7, industry standards and other useful resources.

Don't forget to visit

www.ss7.com

TOPICS

Why SS7?

For the first fifty years of telephone communications, things moved at a fairly even pace. Demand for phones increased steadily, peaking at the time of the Stock Market collapse.

The Great Depression applied the brakes to the demand for phone services while the technology continued to increase, albeit more slowly. With the advent of World War II, the demand for phone services began, once again, to rise sharply. Initially sparked by military requirements, this demand was further fueled by the needs of a multitude of industries gearing up for the war effort.

The problems of meeting this demand were harrowing. For one thing, not all nations were party to any standards agreements which would facilitate the handling of international telephone calls. In many nations trying to make a telephone call was a lesson in handling frustration. That lesson was only compounded when a call originating in one nation had to be connected to a phone in another nation.

Telephone companies found it difficult to meet the demand in wartime. After the war, meeting the demand would become impossible. During the two decades following World War II, demand for telephone service reached astonishing proportions. New businesses popped up overnight like mushrooms. Existing businesses experienced growth spurts that would double or triple their demand for phones in a single year.

Comfortably employed workers gained the confidence to have second and even third phones installed in their homes. Areas where there had been few phones before the war now pressed to be able to become a part of the emerging world of modern communications.

To answer this demand, telephone companies could do little more than add more wires. A thousand new telephones might result in ten thousand new conversations every day. Those ten thousand conversations would require new wires to carry them. Worse still, telephone traffic doesn't occur at an evenly paced rate. There are peaks and valleys in telephone usage. Those thousand new telephones might well result in four or five hundred new conversations occurring at the same time.

By that time telephone industries around the world had attracted many of the best and brightest among those who had chosen to make their careers pursuing technology. Many of these now turned to thoughts of how telephone wires could be used more efficiently. The concept was both simple and obvious. If you could make wires ten percent more efficient, each wire would carry ten percent more conversations. The need for new wiring would then decrease by ten percent.

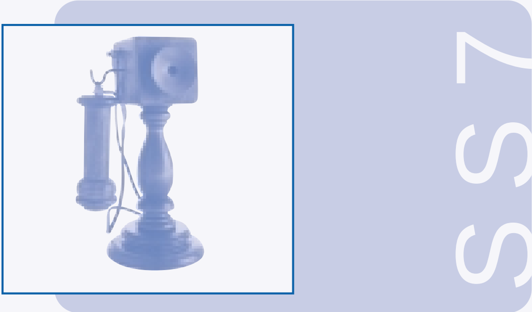
One way to make wires more efficient for conversations was to stop using the wire for anything other than conversations. The wire that was used for conversation was also used to carry all the information that was necessary to connect and manage that conversion. Such information was referred to as “signaling”. At the time, signaling consisted of analog electrical representations of sound. In exactly the same way that the voice was converted to an electrical current to be sent over the wire, these signals were sent over the wire in the form of an analog current which would be converted back to sound at the receiving end. In fact, both these signals and the electrical current representation of the voice were converted back to sound at the receiving end.

Most of us are familiar with at least some of these signals. Lift the phone off the hook and you hear what we refer to as “dial tone”. That sound tells the caller that the telephone line is connected to the local switch and that he/she may proceed to dial. At the telephone company end of your line, the completed circuit that allowed the telephone company to send you the dial tone tells them your phone is “off-hook”. If someone calls you now, the call won't be connected. Today if you have call waiting, they will put some sound on your line to tell you that someone is trying to reach you. But in the post war era, someone trying to reach you would simply get a busy signal.

Now you dial the phone. In most cases each digit you dial places two tones on the line. We call this “touch tone” dialing. Technical people at the phone company call it “Dual Tone Multi Frequency” (DTMF). In the 40s and 50s, this same information was conveyed by interrupting the line connection. The number of interruptions corresponded to the number dialed. A rotary dial on the phone accomplished these interruptions. But, if you timed it just right with pauses between the digits, you could actually dial just by clicking the button in the cradle of the phone.

Once you have finished dialing, the telephone company compares your dialed digits with a routing table that provides the switch with the information allowing it to choose another switch in the network to which it makes a voice circuit connection. That next switch also receives the dialed digit information so that it can consult its own routing table to determine where the next connection will be made. In the end, the switch that is connected to the line of the phone you are calling is connected into the circuit.
This switch now determines whether the call can be connected. If your party is talking, their line indicates an “off hook” condition. In the days before call waiting, this always meant that you would be sent another signal that we call the “busy” signal. This signal was not the only problem associated with signaling in the voice circuitry; but it was a major problem that we can examine to help understand the reasons for wanting to eliminate voice circuit signaling.

With all of the circuit connections in place, the busy signal was sent from the local switch serving the party being called. No



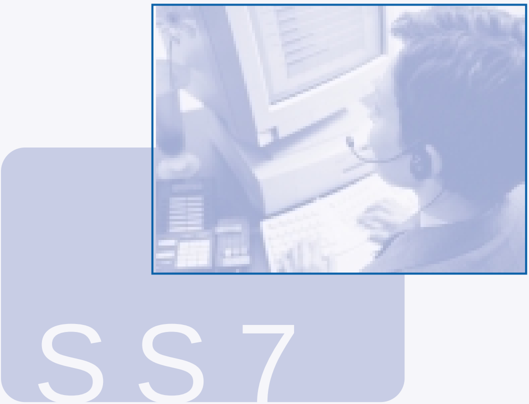
matter how far away you were, all of the connections had to remain in place just to carry the busy signal back to the caller. This same circuitry could not be used for any other phone call. The circuitry was lost for as long as the caller stayed on the phone. Very often, the caller would hang up and place the call again immediately. The result would usually be another busy signal.

This wasn't stupidity on the part of the caller. It was simply that they knew they may have dialed incorrectly and that it might not really be their party that was busy. Sometimes it was because the party who was calling felt an urgent need to contact the other party. Sometimes these dialing compulsions led to the call being placed again and again and again. The resultant inefficient use of circuitry was one of the reasons why the phone companies could not keep up to the demand for new wiring.

Digital concepts were already well enough advanced that telephone company thinkers could envision turning the analog data into digital packets and sending them through the network using existing wiring set up for digital use. A single "channel" or individual circuit would only handle conversation and signaling for one phone call at a time. A digital packet could share a common channel with hundreds or thousands of other digital packets. Each packet could contain a signal.

Thus, thousands of signals could share a single channel and only one voice circuit was lost to remove the signaling from thousands of voice circuits. Because this was so, the approach became known as "Common Channel Signaling" (CCS).

The results of this Common Channel Signaling approach were almost immediately apparent. If the local switch could get the information back from the remote switch that the called party's line was busy, then the local switch could send the busy signal back to the caller. None of the circuitry between the local and remote switches would be required to carry the busy signal back. The only wiring being tied up would be the wire to the caller's phone.



Having a digital interface with the telephone network would evolve to a point where removing the signaling from the voice network would seem to be a minor advantage. Common Channel Signaling would pave the way for 800 numbers, 900 numbers, telephone credit cards, calling cards, the delivery of numerous services (such as short text messages) to cell phones, caller identification and a host of other intelligent (programmable) services available in the Common Channel network.

Nevertheless, having the concept fifty years ago was a long way from experiencing the reality. Everyone in the industry understood that such a system would be almost useless, unless a telephone call could be connected from any phone in the world to any other phone in the world. It was time to develop a standard that would set the guidelines for all the details of how the new system would handle every situation.

The standards organization that would do the work was the CCITT (Consultative Committee on International Telephone and Telegraph).
Telecommunications standards go all the way back to May of 1865 when the International Telegraph Convention was signed by 20 countries. Once the agreement was signed, the organization known as the International Telegraph Union was formed to perform the ongoing work of recommending changes to the first agreement because all parties recognized that time and technology would likely result in the need to make changes.

A mere ten years later, the invention and rapid deployment of telephone services led the Telegraph Union to begin recommending legislation for international use of telephony.

Wireless telegraphy joined the communications mix only twenty years later. The need for yet another set of standards prompted the calling of an International Radio Conference in 1906.The result was the signing of the first International Radiotelegraph Convention.

By 1927, there was a Consultative Committee for International Radio (CCIR), a Consultative Committee for International Telephone (CCIF), and a Consultative Committee for International Telegraph (CCIT). In 1932, the ITU decided to combine the Telegraph and Radiotelegraph Conventions and form the International Telecommunication Convention. In 1934, the ITU renamed itself as the International Telecommunication Union.

After World War II, the ITU became a United Nations Treaty Organization. Finally (or almost) in 1956, the CCIF and CCIT were combined and became the CCITT (Consultative Committee for International Telephone and Telegraph). To this group fell the task of making the recommendations that would collectively become known as Signaling System No.7. In subsequent years, the subcommittees were reorganized and CCITT was replaced with today's ITU-TS.

One question often asked is “Were there six signaling systems before SS7?” The answer is that there were, but the earliest versions existed no where except on paper. The immediate predecessor to SS7 actually saw some limited deployment. It was not called “SS6” (though some call it that in hindsight) but, rather, Common Channel Interoffice Signaling Systems #6 (CCIOS6). It would be 1980 before a fully deployable version would be completed. Every four years beginning in 1976, the standards were grouped into collections which became identified with the colors used for the bound covers. In 1976, it was the Orange Book followed by the Yellow Book (1980), the Red Book (1984), the Blue Book (1988) and the White Book (1992).

Why SS7?

The answer is simply that the time had come for the world to begin its move into the high-tech, highly communicative world of the latter part of the Twentieth Century. A new network signaling architecture was needed.

SS7 was developed to satisfy the telephone operating companies' requirements for an improvement to the existing signaling systems.

SS7

2. SS7 Network Architecture

SS7 Architecture

A telecommunications network consists of a number of switches and application processors interconnected by transmission circuits. The SS7 network exists within the telecommunications network and controls it. SS7 achieves this control by creating and transferring call processing, network management, and maintenance to the network's various components. An SS7 network has three distinct components: Service Switching Points, Signal Transfer Points, and Service Control Points. These components may be generically referred to as “nodes” or “signaling points” and are connected to each other via “data links”. The SS7 Architecture is illustrated in Figure 1 and components are explained in the following sections.

SS7 Architecture

A Access Links

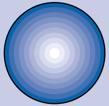
B Bridge Links

C Cross Links

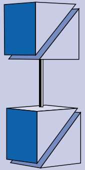
D Diagonal Links

E Extended Links

F Fully Associated Links

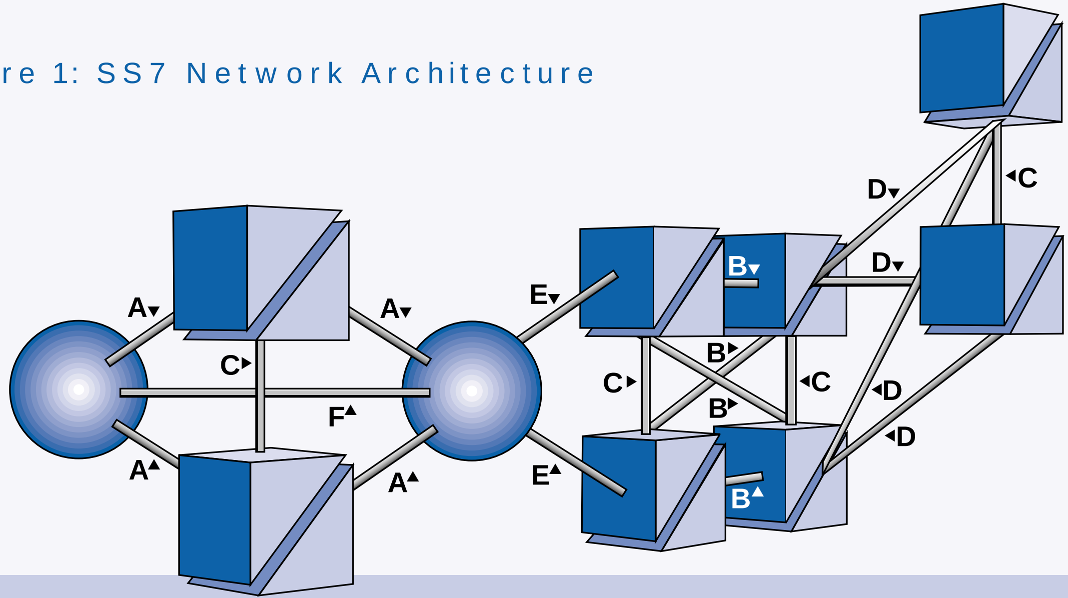


Signaling Point



STP Pairs

Figure 1: SS7 Network Architecture



2–1. Signaling Transfer Point (STP)

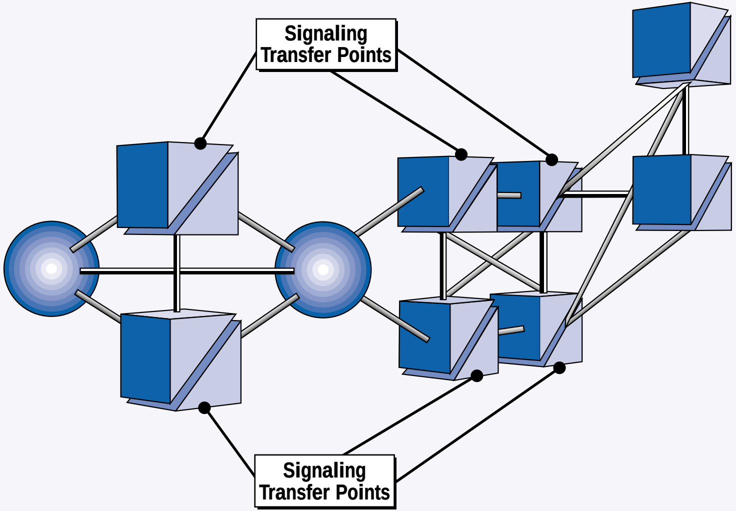
Signaling Transfer Point (STP)

The STP is to the SS7 Network what the switch is to the Public Switched Telephone Network. While a switch routes calls by making actual voice connections, the STP simply directs the digital traffic by selecting links on which to place the outgoing traffic. STPs are paired for redundancy with consideration being given that both members of the pair are not subject to the same hazards. For example both members of the same pair would not be placed on the same earthquake fault.

Signaling Transfer Point (STP) — (Local)

The STPs indicated here are at the lowest level of the SS7 network hierarchy. What makes them local STPs is the fact that the sphere which represents a network location (or node) providing and/or using network services is directly connected to these STPs.

Just as a local telephone office is the direct connection point for the phone lines of telephone users, the local STP pair provides the direct connection for users of the SS7 network.

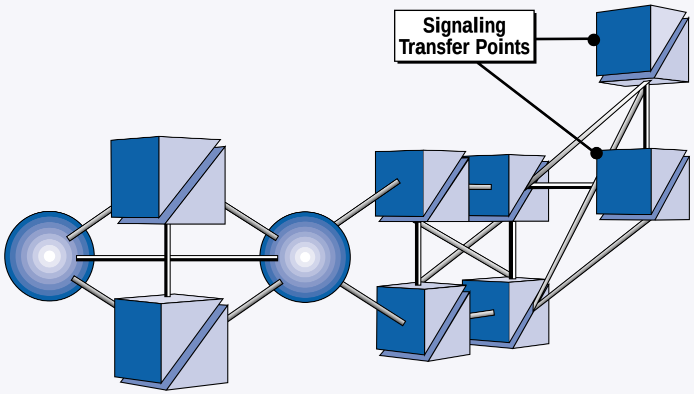


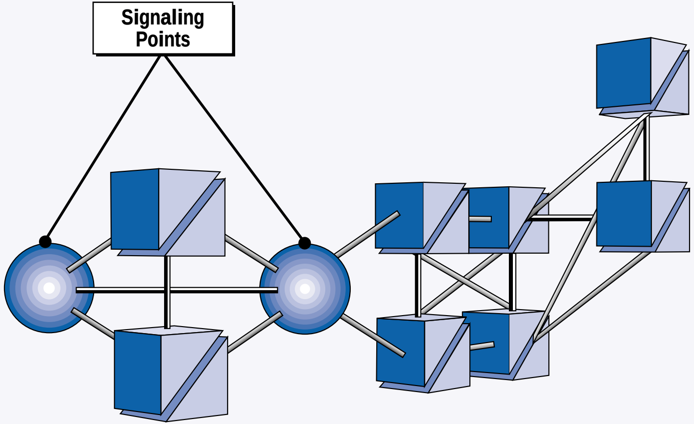
2–1. Signaling Transfer Point (STP)

Signaling Transfer Point (STP) — (Regional)

The STP pair indicated here is at a higher level of the SS7 network hierarchy. The drawing indicates this by showing no direct node connections, and also by placing this STP at a higher position on the page.

The two pairs of STPs shown at the center of the network are, therefore, local pairs whose job is to provide network access to the services nodes. The higher level pair is the regional pair used to connect local STP pairs from different areas together.





Signaling Point (SP)

When a telecommunications service is to be connected to the SS7 network, it is given a Signaling Point Code (SPC) identity much the same way a new telephone location is given a telephone number. A service with such a code is known as a Signaling Point. SP, however, is a broad generic term which does not identify the type of service being offered. Other terms, such as Service Switching Point (SSP), Service Control Point (SCP), Mobile Switching Center (MSC) and others, define the services offered in more narrow categories.

For example, an SSP is a location offering voice circuit connections (in the telephone network) and SS7 connections for the exchange of circuit information and for call routing and maintenance requests. SCPs provide services such as database information, while MSCs control Mobile networks and provide voice connections for subscribers.

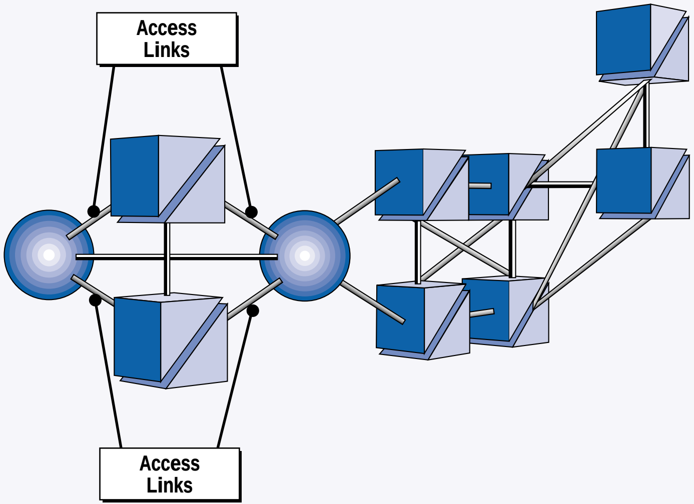
Data Links

What allows messages to travel around the network is the existence of connections between the nodes (Signaling Points) which are called links.

The SS7 network is unconcerned as to the type of transmission being employed except as it may impact the considerations of the physical layer of the protocol. Therefore, links are categorized by what they connect rather than how the data is transmitted.

The names given to link types can be represented by the letters of the alphabet “A” through “F”.

Access Links	**Diagonal Links**
Bridge Links	**Extended Links**
Cross Links	**Fully Associated Links**



Access Links

To provide (or acquire) services from the SS7 a node needs first to gain access. This is normally done through connections to a Signaling Transfer Point (STP). STPs exist throughout the network on a hierarchical basis. That is, some exist for the prime purpose for providing access on a local basis to service providers (or service users). Other STPs may exist solely to expand the network by connecting local STPs. At a still higher level STPs can provide for international communication.

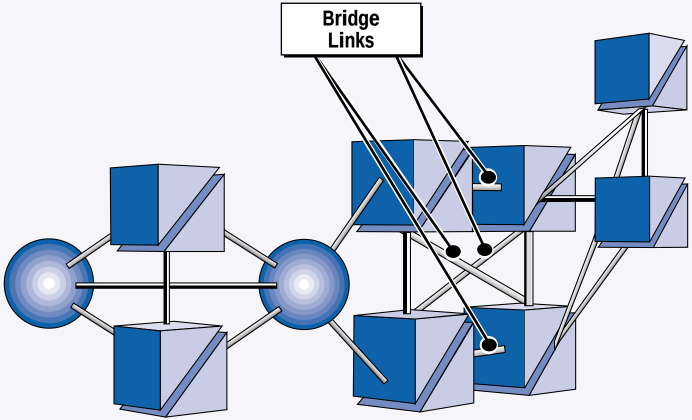
The links that connect a node to a local STP pair provide access to the network, and are therefore called Access Links.

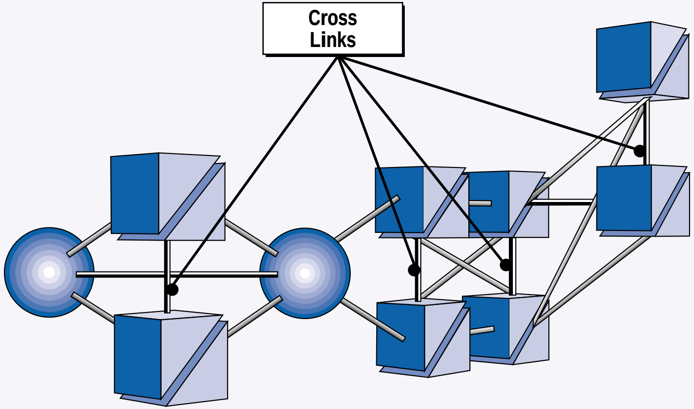
2–5. Bridge Links

Bridge Links

The more links available to an STP for connection into the network, the greater will be that STP's routing flexibility. To gain such flexibility an STP will link to a second STP at the same hierarchical level (e.g. Local to Local). The linking arrangement employed connects each of the STPs in one area with each of the STPs in the other area. To do so requires four links.

Since these links form a bridge from one area to the other, they are referred to as Bridge Links.





Cross Links

For the sake of redundancy, STPs are paired. In a redundant pair, it is generally assumed that both members of the pair perform exactly the same functions. Both members of a pair of STPs can be considered to be the same logical location.

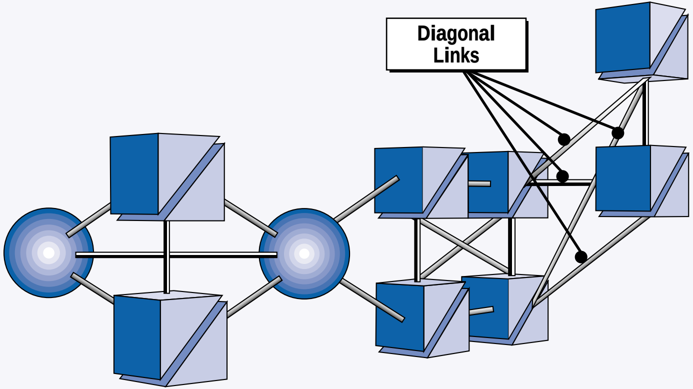
Since these links allow messages to cross over from either STP to its mate, they are called Cross Links.

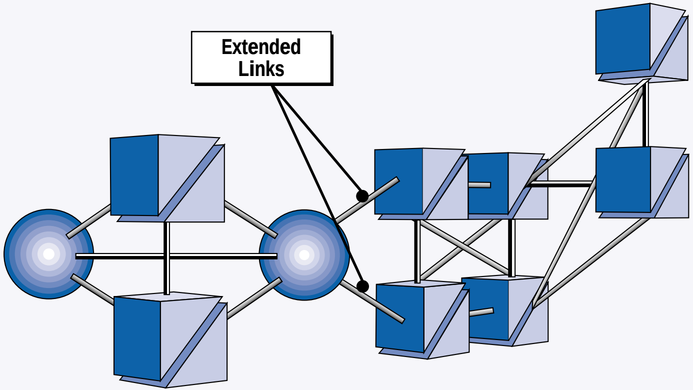
Diagonal Links

Even an STP linked to another STP at the same network level can gain additional routing strength by connection to an STP at a higher level (e.g. Local to Regional). The linking arrangement employed connects each of the STPs in one area with each of the STPs in the other area just as Bridge links do. To do so likewise requires four links.

Then how do these links differ from Bridge links? Look at network drawings and you will notice that network hierarchy is usually indicated by placing STPs of a higher level at a higher position on the page. When the four connections are drawn, the lines must be drawn on the Diagonal.

In more abstract terms, Diagonal simply implies the connection of two levels of network hierarchy.





Extended Links

While STPs are often connected to other pairs at the same level of network hierarchy, these links are commonly made to the closest pair on that level. Further routing flexibility can be acquired by connecting to still another pair of STPs on that same level. To do so requires adding links to some more distant pair. Such links would be made in the same quad-linking arrangement as B links.

Since these links form a connection to a more distant pair of STPs, they are considered to be extended further than other links and are, therefore, called Extended Links.

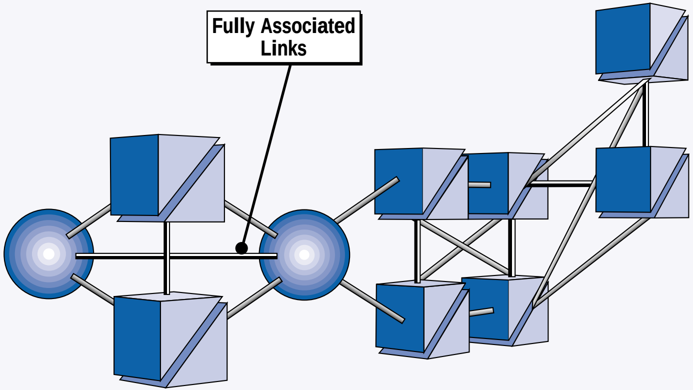
One might also think of these links as extending the routing capabilities to the STPs.

2–9. Fully Associated Links

Fully Associated Links

From time to time, particularly in a proprietary network, users find it desirable to share data directly between nodes and to bypass intervening STPs. This is only done for nodes that are directly and completely associated such as those owned and operated by the same company.

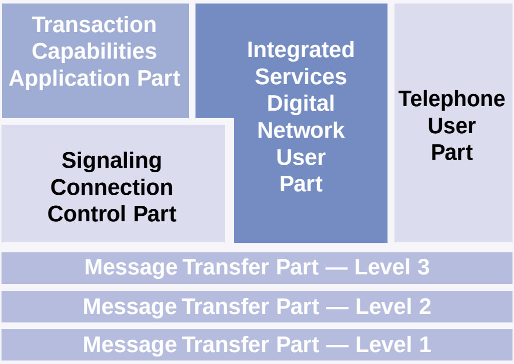
Since such linking occurs only between nodes with this complete association, the links are referred to as Fully Associated Links.



SS7 Protocol Layers

SS7 standard was developed in a modular approach. This approach leads to the creation of what is referred to as a "layered" protocol. Protocol means nothing more than a rigid set of rules which determine how communication should be handled. It covers everything from what should occur to when and how it should occur. It also prescribes exactly what the message consists of when it is sent over the links. "Layered" means each module performs its function in sequence and then hands the message off to the next module (which is "above" for incoming messages and "below" for outgoing messages).

Each of the functional program modules is termed as a "user part." The rules (protocol) dictate the sequence in which things must be done. To show this graphically, a convention has been adopted for the drawing. In this drawing, the functional modules that deal with a message just about to be transmitted over the links (or one just received from the links) are shown at the bottom. Other modules are shown "stacked" above in the sequence in which their functions are performed. The resulting picture is commonly called a "stack." A typical SS7 stack is shown below.



Message Transfer Part — Level 1

The Message Transfer Part Level 1 (MTP L1) is called the "physical layer". It deals with hardware and electrical configuration.

Bear in mind that a protocol is only a set of rules. Those rules extend to what occurs in the equipment to control the links. For example, one rule for MTP L1 is that a link must consist of two data channels operating in opposite directions at the same bit rate. In other words, the links must be bi-directional.

The standard also refers to the need to disable certain attachments to the link that would interfere with Full Duplex operation and might challenge bit integrity. In other words, MTP level 1 is a user part that deals with physical issues at the level of links, interface cards, multiplexors etc. It does not, therefore, concern software providers except that they need to understand these requirements in order to interface the software module layers with the physical layer.

Message Transfer Part — Level 2

This is a busy user part. It is the last to handle messages being transmitted and the first to handle messages being received. It monitors the links and reports on their status. It checks messages to ensure their integrity (both incoming and outgoing). It discards bad messages and requests copies of discarded messages. It acknowledges good messages, so the transmitting side can get rid of superfluous copies. It places links in service, and restores the service links that have been taken out of service. It tests links before allowing their use. It provides sequence numbering for outgoing messages. And finally it reports much of the information it gathers to MTP Level 3.

Message Transfer Part — Level 3

The MTP Level 3 provides the functions and procedures related to Message Routing (or Signaling Message Handling) and Signaling Network Management. MTP L3 handles these functions assuming that signaling points are connected with signaling links. The message routing provides message discrimination and distribution. Signaling Network Management provides traffic, link and routing management, as well as, congestion (flow) control.

Signaling Connection Control Part (SCCP)

SCCP provides connectionless (class 0) and connection-oriented (class 1) network services and extended functions including specialized routing (GTT-global title translation) and subsystem management capabilities above MTP Level 3.

Many of the benefits of the use of the SCCP lie in the specialized routing functions. The addressing capabilities are what allow the locating of database information or the invoking of features at a switch.

A global title is an address (e.g., a dialed 800 number, calling card number, or mobile subscriber identification number) which is translated by SCCP into a destination point code and subsystem number. A subsystem number uniquely identifies an application at the destination signaling point. SCCP is used as the transport layer for TCAP-based services.

There are at least two benefits of global title translations. The first is that SPs can have access to data of all types without having to maintain cumbersome tables. New data can become universally available very quickly. The second is that companies can have better control over the data kept within their own networks.

Transaction Capabilities Application Part (TCAP)

The Transaction Capabilities Application Part offers its services to user designed applications, as well as, to OMAP (Operations, Maintenance and Administration Part) and to IS41-C (Interim Standard 41, revision C) and GSM MAP (Global Systems Mobile).

TCAP supports the exchange of non-circuit related data between applications across the SS7 network using the SCCP connectionless service. Queries and responses sent between SSPs and SCPs are carried in TCAP messages. TCAP is used largely by switching locations to obtain data from databases (e.g. an SSP querying into an 800 number database to get routing and personal identification numbers) or to invoke features at another switch (like Automatic Callback or Automatic Recall). In mobile networks (IS-41 and GSM), TCAP carries Mobile Application Part (MAP) messages sent between mobile switches and databases to support user authentication, equipment identification, and roaming.

Integrated Services Digital Network User Part (ISUP)

The ISDN User Part (ISUP) is used throughout the PSTN (Public Switched Telephone Network) to provide the messaging necessary for the set up and tear-down of all circuits, both voice and digital. Wireless networks also make use of ISUP to establish the necessary switch connections into the PSTN. In the telephone network, ISUP messages follow the path of the voice circuits. That is, ISUP messages are sent from one switch to the other where the next circuit connection is required.

ISUP offers two types of services, known as Basic and Supplementary. Basic Services consist of those services employed in the process of setting up and tearing down a call. Supplementary Services consist of those services employed in passing all messages that may be necessary to maintain and/or modify the call. ISUP functionality can be further broken down into 3 major procedural categories: Signaling Procedure Control, Circuit Supervision Control, and Call Processing Control.

Telephone User Part (TUP)

In some countries (e.g., China, Hong Kong, Brazil), the Telephone User Part (TUP) is used to support basic call setup and tear-down. TUP handles analog circuits only. In most regions of the world, ISUP is used instead of TUP for call management.

Operations, Maintenance and Administration Part (OMAP)

OMAP services are used to verify network routing databases and to diagnose link problems.

NewNet Product Family

SS8's NewNet product family is the leading supplier of global telecommunications software for the converging voice and data communications for wireless, wireline and IP networks. SS8's NewNet product family includes Signaling System No.7 (SS7) middleware (NewNet AccessMANAGER™, and NewNet Connect7™) commercially deployed in over 40 countries, network level applications for short messaging and over-the-air service provisioning (NewNet SMserver™ and NewNet OTAserver™), SS7-IP gateways (NewNet Internet Offload Gateway) and lawful intercept platform (NewNet CALEAserver™).

With NewNet products, wireless and wireline carriers can quickly define, develop, and deploy intelligent networking applications and services, and perhaps more significantly, differentiate their services while tightly linking investment with revenue potential. NewNet products provide operators with state of the art enhanced services platforms and software infrastructure such as SS7 connectivity for revenue generating applications.

NewNet AccessMANAGER™ — *High Performance Platform for Signaling System No.7 (SS7) Applications*

AccessMANAGER is an open-architecture, real-time, scalable, reliable, and high-performance telecommunications application development platform. It enables the rapid development and deployment of enhanced Intelligent Network (IN) services and features for global wireline, wireless and IP networks. AccessMANAGER provides value-added application components on open-architecture computer platforms and integrates industry standard boards into computers with standard backplanes. It enables the rapid development and deployment of enhanced Intelligent Network (IN) services and features for global wireline, wireless and IP networks.

AccessMANAGER middleware is a collection of telecommunications software building blocks such as SS7 (MTP, SCCP, TCAP, ISUP, TUP, OMAP), and IS-41, GSM MAP and A-Interfaces. The building blocks are implemented on industry-standard, open-architecture platforms and the UNIX operating system. The platform frequently takes advantage of UNIX STREAMS to provide a truly layered software architecture, modularity, and performance.

Using a fast-packet switch software backplane implemented in UNIX STREAMS, the AccessMANAGER software also provides Inter-Process Communications (IPC) and extended timer facilities essential for telecommunications applications. The services of AccessMANAGER are available to applications via dynamic binding and a series of Applications Programming Interface (API) library calls. Consistent with its object-oriented architecture and rapid, simple application development philosophy, AccessMANAGER supports protocol-related communications and IPC on the same application interface.

NewNet Connect7™ — *Board Level Based Signaling System No.7 (SS7) Solution*

Connect7 is a host-independent SS7 controller board embedded with full SS7 functionality. All the protocol layers necessary for communicating with the SS7 network reside on the controller board. The board plugs into any host, workstation or switch that supports common bus architectures. The embedded board design isolates the application developer from the complexity of the network protocols and allows the application to be developed on the most appropriate platform — whether it's an NT Server, a UNIX Server, a UNIX workstation, or any type of off-the-shelf PC. Connect7 works with them all.

Connect7 offers a redundant, high-available and high performance solution for network connectivity. The implementation of services using Connect7 is cost-effective, delivering complex global SS7 protocol stacks embedded on a board.

NewNet SMserver™ — *Short Message Service Center (SMSC)*

SMserver is a robust, flexible, open architecture short messaging platform ready to deploy with value added short messaging services. SMserver manages the transmission of alphanumeric messages between mobile subscribers and external systems such as paging, electronic mail and voice mail systems. Built around a client/server architecture, it supports connectivity to external systems via dedicated client modules. It accepts, stores and manages alphanumeric messages to be delivered to mobile subscribers.

SMserver manages all network interactions and provides sophisticated redelivery mechanisms to ensure reliable delivery of short messages. It supports performance monitoring and full billing capabilities. The open Short Message Client Interface (SMCI) facilitates prototyping and deployment of value added services.

NewNet OTAserver™ — *Over-the-Air Service Provisioning (OTASP) for CDMA and TDMA wireless networks*

Over-the-Air Service Provisioning (OTASP) holds the key to wireless growth and revenue generation. Call it a revolution or an evolution, over-the-air service provisioning is the vehicle to grow rapidly in the new generation wireless service offerings. SS8 provides NewNet OTAserver to help service providers activate wireless subscribers and provision wireless services over-the-air quickly, cost-effectively and securely.

OTAserver manages the transmission of service provisioning data between the mobile station and service provider's customer service center. OTAserver accepts and manages service provisioning commands. It provides complete encoding and decoding of service provisioning data to support both CDMA and TDMA air-interfaces. It handles all mobile network interactions and provides a reliable delivery.

NewNet CALEAserver™ — *Solution for Communications Assistance for Law Enforcement Act (CALEA)*

SS8 offers the solution that allows carriers to meet CALEA requirements today. Our NewNet CALEAserver product helps take the time, effort and hassle out of becoming CALEA-compliant. CALEAserver provides lawful intercept delivery services that ensure network-proven, telco-quality wiretap access support.

CALEAserver is an off-switch lawful intercept platform which is compatible with offerings from a variety of switching vendors, while also supporting both wireless and wireline networks. NewNet CALEAserver collects pertinent call information and allows carriers to provide secure access to that information by law enforcement agencies that obtain court-ordered wiretap directions.

NewNet Internet Offload Solution™

The explosive growth in dial-up internet traffic is choking carrier switches nation-wide. SS8's Internet Offload Solution (IOS) is an SS7 based solution that enables dial-up Internet traffic to be diverted from Competitive Local Exchange Carrier (CLEC) or Incumbent Local Exchange Carrier (ILEC) switches. The need for this product is driven by the rapid increase in Internet traffic, as well as long Internet hold times.

These two factors contribute to heavy congestion on switches, particularly switches that directly serve Internet Service Providers' (ISPs) Network Access Servers (NAS).

SS8's NewNet IOS offering is a switched solution that enables a CLEC or ILEC to service ISPs who choose to maintain ownership or control of their own Network Access Servers (NAS). The solution consists of an SS7 Signaling Gateway (including Controller functionality) and a Media Gateway/Switch.

The Signaling Gateway functionality converts SS7 ISUP messages to Q.931 messages, which are transported over IP. The Controller functionality is used to control the switch and ISPs' NAS devices via Q.931 signaling. The Media Gateway/Switch routes individual calls to the correct ISP NAS via an ISDN Primary Rate Interface (PRI). The Media Gateway can be controlled over MGCP or switch specific interfaces.

Glossary of Terms

Terms	Description
ABS	Alternate Billing Service
ACD	Automatic Call Distributor
ACG	Automatic Code Gapping
ACK	Acknowledgment
ACM	Address Complete Message
AFR	Automatic Flexible Routing
AHT	Average Handle Time
AIN	Advanced Intelligent Network
AIOC	Automatic Identified Outward Calling
AMA	Automatic Message Accounting
AMATPS	AMA Teleprocessing System
AMP	AIN Maintenance Parameter
AMPS	Advanced Mobile Phone System
ANI	Automatic Number Identification
ANM	Answer Message
ANSI	American National Standards Institute
API	Application Programming Interface
APPN	Advanced Peer-to-Peer Networking
ARP	Address Resolution Protocol
ASA	Average Speed of Answer
ASCII	American Standard Code for Information Interexchange
ASE	Application Service Element
ASN.1	Abstract Syntax Notation 1
ATB	All Trunks Busy
ATM	Asynchronous Transfer Mode
ATP	Acceptance Test Procedure
AUI	Attachment Unit Interface
AW	Admin Workstation
B-ISDN	Broadband Integrated Services Digital Network (ISDN)
BAF	Bellcore AMA Format
BBG	Basic Business Group
BCC	Bellcore Client Company
BCD	Binary Coded Decimal
BCI	Backward Call Indicators
BCLID	Bulk Calling Line Identification

Terms	Description
BCM	Basic Call Model
BER	Basic Encoding Rules
BG	Business Group
BGID	Business Group Identification
BGP	Border Gateway Protocol
BRI	Basic Rate Interface (ISDN)
BSN	Backward Sequence Number
CAC	Carrier Access Code
CAP	Competitive Access Provider
CC	Call Control
CCA	Call Control Adjunct
CCC	Clear Channel Capability
CCITT	Consultative Committee on International Telephone & Telegraph
CCS	Common Channel Signaling
CDAR	Customer Dialed Account Recording
CDMA	Code Division Multiple Access
CDP	Customized Dialing Plan
CDPD	Cellular Digit al Packet Data
CDSL	Customer Digital Subscriber Line
CED	Call Entered Digits
CGB	Circuit Group Blocking Message
CGU	Circuit Group Unblocking Message
CIC	Carrier Identification Code
CIDS	Calling Identity Delivery & Suppression
CL	Connectionless
CLID	Calling Line ID
CLLI	Common Language Location Identification
CMC	Cellular Mobile Carrier
CMS	(AT&T's) Call Management System
CNAB	Call Name Delivery Blocking
CO	Central Office or Connection Oriented
COT	Continuity Test Message
CPC	Call Processing Control
CPE	Customer Premises Equipment
CPG	Call Progress Message
CR	Conditional Requirement

Terms	Description
CRA	Circuit Reservation Acknowledgment Message
CRC	Cyclic Redundancy Check
CRM	Circuit Reservation Message
CRP	Customer Routing Point
CS-1	Capability Set 1
CSC	Circuit Supervision Control
CSU	Channel Service Unit
CT	Call Type
CVR	Circuit Validation Response Message
CVT	Circuit Validation Test Message
DACS	Digital Access Cross-Connect System
DCE	Data Circuit (terminating) Equipment
DLC	Digital Loop Carrier
DMP	Device Management Protocol
DMS	Digital Multiplex Switch
DMT	Discrete Multitone Technology
DN	Directory Number (SS7)
DN	Dialed Number
DNIS	Dialed Number Identification Service
DP	Dial Pulse
DPC	Destination Point Code
DSL	Digital Subscriber Line
DSU	Data Service Unit
DSVD	Digital Simultaneous Voice and Data
DTE	Data Terminal Equipment
DTMF	Dial Tone Multifrequency
DUP	Data User Part
DXI	Data Exchange Interface
EA	Equal Access
EADAS	Engineering & Administration Data Acquisition System
EADASNM	EADAS Network Administration
EAEO	Equal Access End Offic
EAMF	Equal Access Multifrequency
EBCDIC	Extended Binary Coded Decimal Interchange Code

Terms	Description
EDI	Electronic Data Interchange
EDP	Event Detection Point
EGP	Exterior Gateway Protocol (IETF)
EIA	Electronics Industry Association
EIR	Equipment Identification Register
EKTS	Electronic Key Telephone Service
EMS	Event Management Service
EO	End Office
ESME	External Short Message Entity
ESN	Electronic Serial Number
ETSI	European Telecommunications Standards Institute
EXM	Exit Message
FCS	Frame Check Sequence
FISU	Fill-in Signal Unit
FR	Frame Relay
FRAD	Frame Relay Access Device
FRL	Facility Restriction Level
FSD	Feature Specific Document
FSK	Frequency Key Shifting
FSN	Forward Sequence Number
FSS	Facility Selective Service
FTE	Full Time Equivalent
FTP	Fast Transfer Protocol (IETF)
FUNI	Frame User Network Interface
FX	Foreign Exchange
GN	Generic Name
GRS	Group Reset Message
GSC	Gateway Switching Center
GSM	Global System for Mobile Communication
GTT	Global Title Translations
GTV	Global Title Value
GUI	Graphical User Interface
HDLC	High Level Data Link Control
HFC	Hybrid Fiber Coaxial Cable
HLR	Home Location Register
IAM	Initial Address Message
IC	Interexchange Carrier

Terms	Description
ICP	Intelligent Call Processing
ICR	Intelligent Call Router
IDLC	Integrated Digital Loop Carrier
IDT	Integrated Digital Terminal
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IGP	Interior Gateway Protocol
INR	Information Request Message
IP	Intelligent Peripheral or Internet Protocol (IETF)
IPC	Interprocess Communication
IPI	Intelligent Peripheral Interface
ISDN	Integrated Services Digital Network
ISDNUP	ISDN User Part
ISO	International Organization for Standardization
ISP	Intermediate Service Part
ISPC	International Signaling Point Code
ISUP	ISDN User Part (circuit related)
ITU	International Telecommunication Union
ITU-T	Telecommunication Standardization Sector (of ITU)
ITU-TS	Telecommunication Standardization Sector (of ITU)
IWX	Interworking Function
IXC	Interexchange Carrier
LAA	Longest Available Agent
LAN	Local Area Network
LATA	Local Access & Transport Area
LCN	Logical Channel Number (x.25)
LEC	Local Exchange Carrier
LI	Length Indicator
LNP	Local Number portability
LOCREQ	Location Request
LSSGR	LATA Switching & Signaling Generic Requirements
LSSU	Link Status Signaling Unit

Terms	Description
MAP	Mobility Application Part
MBG	Multi-switch Business Group
MCC	Mobile Country Code
MGW	Mini-Gateway Prototype
MIB	Management Information Base
MIN	Mobile Identification Number
MLHG	Multi-Line Hunt Group
MMI	Man-Machine Interface
MSC	Mobile Switching Center
MSISDN	Mobile Station ISDN Number
MSO	Mobile Switching Office
MSU	Message Signaling Unit
MTP	Message Transfer Part
MTSO	Mobile Telephone Switching Office (Cellular)
MUX	Multiplexor
NA-TDMA	North American Time Division Multiple Access
NAA	Next Available Agent
NACN	North American Cellular Network
NANP	North American Numbering Plan
NCA	Non-Call Associated
NCP	Network Control Point
NDC	National Destination Code
NETID	Network Identifier
NIC	Network Interface Controller
NMS	Network Management System
NNI	Network-to-Network Interface or Network Node Interface
NPA	Numbering Plan Area
NSP	Network Services Part
NT	New Technology (Windows)
OAM&P	Operations, Administration, Maintenance and Provisioning
ODBC	Open Database Connectivity
OE	Office Equipment
OMAP	Operations & Maintenance Application Part
OPC	Origination Point Code

Terms	Description
OPI	Open Peripheral Interface
OS	Operations System
OSI	Open Systems Interconnection
OTA	Over The Air
OTGR	Operations Technology Generic Requirement
PAM	Pulse Amplitude Modulation
PANS	Pretty Amazing New Services (B-ISDN)
PBX	Private Branch Exchange
PCS	Personal Communications Services
PG	Peripheral Gateway
PIC	Point In Call
PIM	Peripheral Interface Manager
PPP	Point-to-Point Protocol
PRI	Primary Rate Interface
PROFREQ	Profile Request
PSN	Alternative to PSTN (Public Switched Telephone Network)
PSTN	Public Switched Telephone Network
RADIUS	Remote Authentication Dial-In User Service
RARP	Reverse Address Resolution Protocol
REGNOT	Registration Notification
RISC	Reduced Instruction Set Computing
ROUTREQ	Routing Request
SANC	Signaling Area Network Code
SAP	Service Access Point
SCCP	Signaling Connection Control Part
SCP	Service Control Point (SS7)
SCP	Service Control Point
SDLC	Synchronous Data Link Control
SDSL	Symmetric Digital Subscriber Line
SEP	Signaling Endpoint
SF	Status Field
SI	Service Indicator
SIB	Signaling Information Field
SIO	Signaling Information Octet
SLC	Signaling Link Code
SLIP	Serial Line Internet Protocol

Terms	Description
SLP	Service Logic Program
SLS	Signaling Link Selection
SMDS	Switched Multimegabit Digital Service
SMPP	Short Message Peer to Peer
SMS	Service Management System
SMTP	Simple Mail Transfer Protocol
SNMP	Simple Network Management Protocol (IETF)
SS7	Signaling System No.7
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TDMA	Time Division Multiple Access
TUP	Telephone Users Part
UDP	User Datagram Protocol
UDT	Unitdata
UDTS	Unitdata Service
VAD	Voice Activated Dialing
VANC	Voice Activated Network Control
VLR	Visitor Location Register
VPN	Virtual Private Network
VRU	Voice Response Unit
WAN	Wide Area Network
WATS	Wide Area Telephone Service
XUDT	Extended Unitdata
XUDTS	Extended Unitdata Service