Trillium Digital Systems, An Intel Company.
A Case Study.
&
An Analysis of the Protocols in the Voice Over IP Arena

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Executive Summary:

This paper is an effort to showcase the pioneer work being done by Trillium Digital Systems in the field of telecommunications protocol software development. In this essay we have tried to summarize the various networking technologies that are being worked on at Trillium Digital Systems.

We have treaded a case study approach and have dealt with the protocols in the Voice Over IP protocols in great depth. This case study is aimed at providing an overview of the pioneer work being done at Trillium Digital Systems and also have tried to compare and contrast the three most important protocols in the Voice Over IP arena namely **Media Gateway Control protocol (MGCP),** The **H.323 standard** and the **Session Initiation Protocol** (SIP) in significant detail. Each of the protocols is organized into three different categories. The three being

- The Background of the protocol.
- The Basics of the protocol.
- The Architecture of the protocol.

After the detailed description of each of the three protocols, a comparison of each with respect to the other is given and a comparison of all three protocols is tabulated.

1. Introduction:

Trillium Digital Systems, Inc., an Intel company, is headquartered in Los Angeles, California and is a leading provider of communications software solutions for the converged network infrastructure. Trillium's source code solutions are used in more than 500 projects by industry-leading suppliers of wireless, Internet, broadband and telephony products. Trillium's high-performance, high-availability software and services reduce time, risk and cost of implementing SS7, IP, ATM, Wireless and other standards-based communications protocols. Trillium, an ISO 9001-certified software company, is the Networking Software Division of Intel's Access & Switching Group which focuses on the Intel® Internet Exchange Architecture.

2. About Trillium:

Trillium Digital Systems provides software for virtually every standard and protocol used in telephony today. SS7 technology is their self-professed "bread-and-butter", but has their hands in products from H.323 software used by AskJeeves, to RTP and RTCP, to the ever-expanding protocols in the wireless industry and the SIP technology released in February.

Intel bought trillium last August.

Since, the two companies have been exploring the new possibilities opened by merging the chip manufacturer with one of the leading behind-the-scenes communications technology companies around. Trillium comprises the Networking Software Division of Intel's Network Processing Group, focusing on the Intel Internet Exchange Architecture (IXA).

Trillium works solely in source code; they focus their energy on covering a broad range of standards and working quickly to get products to market. Two years ago Trillium focused primarily on carrier-grade technology, but the proliferation of smaller gateway start-ups have begun to garner a more significant portion of their time.

Given their flexibility with standards, including most of the varied global telephony protocols, start-ups find their all-important time-to-market needs can oft be handled quickest by Trillium. Implementations can be as quick as a couple of weeks with the training included from Trillium and the source code CD.

Trillium strives to be more than just a software library. Their Professional Services group located in Vancouver, B.C., handles contract implementations of all varieties. Purposely set up as a separate entity, the group handles the consulting and customization aspects of the business.

Trillium’s software is a proven, market-tested foundation on which you can build new solutions, and Trillium's Professional Services can helps the customer’s move forward even faster, on a price-competitive basis that applies Trillium's operational excellence and Intel's vast experience to solving problems quickly and flexibly.

The list of clients and partners is vast. Portable protocol software allows integration into proprietary environments with relative ease. The software has been licensed to over 250 companies for over 400 different projects.

From carrier-class solutions right down to end-user phones, from ATM to SS7 to SIP, Trillium supplies the source code.

3. Company History and Customers:

Jeff Lawrence and Larisa Chistyakov founded trillium in 1988. It has won numerous awards and recognitions and been first in market with many leading software products.

Since 1988, leading computer and communications equipment manufacturers such as has used Trillium software products and services

- 3Com
- Alcatel (including DSC Communications and Xylan)
- Apple Computer
- AT&T
- Cisco Systems
- ECI Telecom
- Ericsson
- Fujitsu
- General DataComm
- Hewlett-Packard
- Hitachi
- Hyundai

- IBM
- Lucent Technologies (including Ascend Communications and Cascade Communications)
- Motorola
- NEC
- Newbridge
- Nokia,
- Nortel Networks (including Bay Networks)
- NTT
- Qualcomm
- RAD Group
- Rockwell
- Samsung
- Siemens
- Toshiba.

Trillium also works with national labs, such as the Electronics and Telecommunications Research Institute (ETRI), and has donated the use of its software to educational institutions, such as Carnegie Mellon University, Dublin City University and the University of Hawaii..

Trillium has direct sales representation with offices in Los Angeles, Boston, Dallas, San Jose, Virginia, Seoul, Korea, the United Kingdom and Vancouver, British Columbia, and independent sales representatives in Germany, Japan, Korea and Switzerland. Trillium also has marketing partnerships with leading semiconductor, board, platform and operating system companies.

4. THE TRILLIUM ADVANTAGE:

Trillium products and services serve as integrated solutions in next generation applications such as:

- Telephony
- Network access
- Switching and routing
- Service platform and servers
- Gateway, gatekeeper and call agent

- Test equipment
- Appliances, devices and terminals
- Operating systems

Trillium Digital Systems makes software products for

- Network access
- Switching and routing
- Service platform and servers
- Gateway, gatekeeper and call agent
- Test equipment
- Appliances, devices and terminals
- Operating systems

That is used in

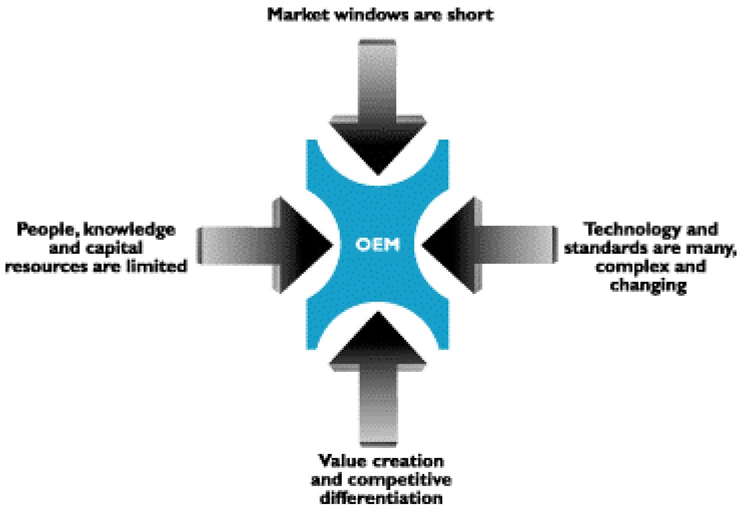
- Telephony
- Internet
- Wireless
- Cable television
- Interworking
- Unified and voice messaging
- Multimedia
- Remote access
- Security

Trillium has a vision: communications to the core powered by the confluence of wireless, IP telephony, broadband and SS7 signaling protocols. Driving convergence. Fulfilling the unlimited promise of the next generation network.

Trillium products work at the foundation of this historic convergence of hardware, software, companies and customer expectations.

Explosive demand for ubiquitous, mobile and instant access to data, voice and video content has fueled the market for innovative telecommunications products and services. To compete, OEMs must develop and deploy high-performance products with ever-greater speed and economy.

Success often depends on the ability to shorten product development cycles, reduce costs, and anticipate new applications and services.



To meet these challenges, telecom equipment manufacturers rely on Trillium to help them with software solutions and professional services. With Trillium as their partner, companies can focus on core competencies and value-added differentiation to build innovative products--streamlining the development process, saving time and winning a time-to-market advantage.
The convergent network brings not only new opportunities, but new pressures as well: short market windows, tight financing, limited knowledge resources and changing standards. The possibilities of this marketplace are greater than ever, but so are the challenges of time, money, and risk.

Since 1988, Trillium Digital Systems has been a trusted leader in communications software. Trillium has helped companies around the world--from the *Fortune* 500 to startups--with mission-critical products for wireless, IP telephony, broadband, and SS7 that form the strong foundation for the next generation network. Their acquisition by Intel in 2000 gives them additional strength to serve our customers with a depth of experience and resources that's unmatched in the industry.

What is source code?

Source code is software delivered in a high-level programming language, such as C, that can be modified as needed and compiled into binary or object-code form. In contrast, binary, or object code, is compiled into a machine-executable form only, and cannot be modified. Source-code solutions have other distinct advantages over binary solutions. Source-code software solutions enable computer and communications equipment manufacturers to use a wide variety of hardware platforms and add their own proprietary value. Trillium delivers its software in source-code form.

Trillium has the broadest range of protocol solutions and communications technology expertise in the industry. Because Trillium designs its software using a common architecture across its entire product line, you can easily integrate additional Trillium software to their products. Other benefits include: the assurance that it conforms completely to existing standards and is interoperable; the knowledge that upgrades will be available for future changes to the standards; extensive documentation; a group of professionals already trained in its design and use; and a warranty that it will work.

How Do Companies Use Trillium Software Solutions?

Trillium's customers build communications equipment for use in a wide variety of mission-critical applications including:

- Base stations
- Digital loop carrier systems
- Distributed network intelligence
- Intelligent networks
- Internet
- Operating system kernels
- Service platforms
- Signaling transfer points
- Switches
- Telephony
- Terminal adaptors and network interface cards
- Test equipment
- VLRs and HLRs
- Voice messaging systems
- Wireless and broadband networks
- Wireless local loop systems

This equipment supports services such as cellular communications, Internet access (Web, e-mail, e-commerce), multimedia, paging, remote access, traditional telephony, Internet telephony, voice messaging, and wireless applications.

5. NETWORK AVAILABILITY AND SCALABILITY PRODUCTS:

Today's telecommunication platforms must provide "five nines" availability, meaning practically no loss of service due to hardware or software errors, nor any downtime for software upgrades or hardware maintenance. This expectation places an unprecedented burden on service providers: to ensure that all the network elements needed to support a service are functioning whenever a user requests that particular service. Supporting five nines service availability depends on the near-flawless interaction of software (application, OS and management middleware), hardware and network design, as well as on environmental and operational factors. Hardware fault tolerance typically relies on redundant processors, memory, buses, power supplies and databases. Software fault tolerance uses a combination of software redundancy and simple hardware redundancy to provide the necessary availability in the case of failure. Network fault tolerance uses redundant data link cross connects (T1s, DS3s, OC3s, etc.).

Trillium offers Fault-Tolerant/High Availability (FT/HA) and patent-pending Distributed Fault-Tolerant/High-Availability (DFT/HA) architectures and implementations. Trillium's FT/HA source code software solution provides a flexible, platform-independent, cost-effective framework that maintains active calls during software and hardware failures. Trillium's widely deployed FT/HA solution gives systems designers and engineers total freedom when choosing their hardware platform and OS. Developed primarily for telecommunications products, it is also applicable to other types of products.

Trillium's DFT/HA core software functionality allows the creation and management of Distributed Fault-Tolerant applications; Pure Distributed applications; and Pure Fault-Tolerant applications. The DFT/HA Core software distributes the protocol load onto available physical processors, dynamically re-distributes load on processor failure and new processor introduction, retains active calls and recovers from processor failure, and allows maintenance operations to be performed without bringing the system down.

6. Protocols In the Voice Over IP Arena:

- **Media Gateway Control Protocol: MGCP/Megaco**

Background:

MGCP/Megaco was developed by the telco community to address the issue of SS7/VoIP integration. The H.323 initiative had grown out of the LAN and had trouble scaling to public network proportions. The architecture that it defined was incompatible with the world of public telephony services, struggling with multiple gateways and the SS7.

To address this problem, the new initiative exploded the gatekeeper model and removed the signalling control from the gateway, putting it in a "media gateway controller" or "softswitch". This device would control multiple "media gateways". This is effectively a decomposition of the gatekeeper to its SS7 equivalents. MGCP/Megaco is the protocol used to communicate between the softswitch and the media gateways.

This initiative has come in a number of different guises. Firstly, IPDC (proposed by Level 3, 3Com, Alcatel, Cisco and others) and SGCP (Telcordia) were brought together by the IETF to form MGCP (Media Gateway Control Protocol), under the responsibility of the Megaco (Media Gateway Control) working group. MGCP is, at this stage, a working document and not a standard. The IETF and the ITU have decided to jointly mandate a single standard, endorsed by both communities and known as Megaco (IETF) and H.248 (ITU).

The existing implementations of this protocol are based on various stages of this initiative; IPDC, SGCP and MGCP form the vast majority of implementations.

Megaco does represent an enhancement of MGCP: it can support thousands of ports on a gateway, multiple gateways and accommodation for connection-oriented media like TDM and ATM.

The Basics of MGCP:

When a gateway detects an off hook condition, it tells the gateway controller, which might respond with a command to instruct the gateway to put dial tone on the line and listen for DTMF tones indicating the dialed number. After detecting the number, the gateway controller determines how to route the call and, using an inter-gateway signalling protocol such as SIP, H.323, or Q.BICC, contacts the terminating controller. The terminating controller could instruct the appropriate gateway to ring the dialled line. When the gateway detects the dialled line is off hook, both gateways could be instructed by their respective gateway controllers to establish two-way voice across the data network. Thus, these protocols have ways to detect conditions on endpoints and notify the gateway controller of their occurrence; place signals (such as dial tone) on the line; and create media streams between endpoints on the gateway and the data network, such as RTP streams.

There are two basic constructs in MGCP/Megaco: terminations and contexts. Terminations represent streams entering or leaving the gateway (for example, analogue telephone lines, RTP streams, or MP3 streams). Terminations have properties, such as the maximum size of a jitter buffer, which can be inspected and modified by the gateway controller. A termination is given a name, or TerminationID, by the gateway. Some terminations, which typically represent ports on the gateway, such as analogue loops or DS0s, are instantiated by the gateway when it boots and remain active all the time. Other terminations are created when they are needed, get used, and then are released. Such terminations are called "ephemerals" and are used to represent flows on the packet network, such as an RTP stream.

Terminations may be placed into contexts, which are defined as when two or more termination streams are mixed and connected together. The normal, "active" context might have a physical termination (say, one DS0 in an E3) and one ephemeral one (the RTP stream connecting the gateway to the network). Contexts are created and released by the gateway under command of the gateway controller. Once created, a context is given a name (ContextID), and can have terminations added and removed from it. A context is created by adding the first termination, and it is released by removing the last termination.

MGCP/Megaco uses a series of commands to manipulate terminations, contexts, events, and signals:

- Add - adds a termination to a context and may be used to create a new context at the same time.
- Subtract - removes a termination from a context and may result in the context being released if no terminations remain.
- Move - moves a termination from one context to another
- Modify - changes the state of the termination.
- Audit Value and AuditCapabilities - return information about the terminations, contexts, and general gateway state and capabilities.
- ServiceChange - creates a control association between a gateway and a gateway controller and also deals with some fail over situations.

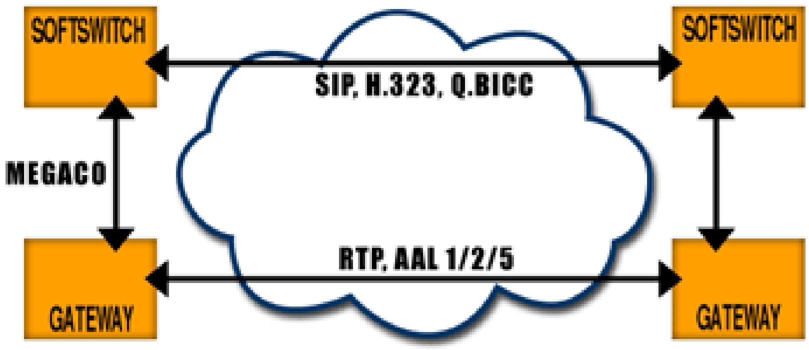
Architecture of MGCP/Megaco:

MGCP/Megaco exploded H.323 s gatekeeper model and removed the signalling control from the gateway, putting it in a "media gateway controller" or "softswitch". This device would control multiple "media gateways". Effectively, this was a decomposition of the H.323 architecture into SS7 equivalents, creating signalling intelligence that could act as a peer to the SS7 entities.

In the MGCP/Megaco architecture, the intelligence (control) is unbundled from the media (data). It is a master-slave protocol where the master has absolute control and the slave simply executes commands. The master is the media gateway controller, or softswitch (or call agent) and the slave is the media gateway (this can be a VoIP gateway, a DSLAM, MPLS router, IP phone etc.). This is a contrast to the peer-to-peer nature of SIP and other Internet protocols where a client can establish a session with another client.

MGCP/Megaco instructs the media gateway to connect streams coming from outside a packet network on to a packet stream such as RTP. The softswitch issues commands to send and receive media from addresses, to generate tones, and to modify configuration.

As can be seen from the diagram below, MGCP/Megaco is used for communication downward, to the media gateways. The architecture, however, requires a session initiation protocol for communication between gateway controllers.



In mirroring the SS7 architecture, MGCP/Megaco "soaks up" the complexity of the central offices. In this way, it can be used as a control protocol delivering services across the network via media gateways. By gathering intelligence (and service delivery) at the interconnect points of the network, MGCP/Megaco creates the "IP Central Office". This approach is seen as recreating the IN world or "PSTN over IP" thus precluding the potential and benefits of a new architecture. This is a contrast to the distributed service model of SIP and the world of the Internet.

- **The H.323 standard:**

Background of the H.323:

H.323 is part of a family of real-time communication protocols developed under the auspices of the ITU, the H.32x family. Each protocol in the family addresses a different underlying network architecture e.g. a circuit switched network, B-ISDN, LAN with QoS, and LAN without QoS (H.323). All borrow heavily from the original H.320's structure and modularity.

H.323 is not an individual protocol, but rather a complete, vertically integrated suite of protocols that defines every component of a conferencing network: terminals, gateways, gatekeepers, MCUs and other feature servers. Amongst others, H.323 uses:

- Q.931 - call setup
- H.225 call signalling
- H.245 - exchanging terminal capabilities and creation of media channels
- RAS - registration and admission control
- RTP/RTCP - sequencing audio and video packets
- G.711/712 - codec specification
- T.120 - data conferencing

All these protocols - dozens of back-and-forth messages - must be negotiated to set up a simple point-to-point voice call.

Architecture of the H.323:

The H.323 architecture includes the following elements:

- Gateways to link LAN-based H.323 endpoints and endpoints in the PSTN and other networks. These translate protocols, convert media formats and transfer information.

- Gatekeepers to translate addresses, allocate LAN bandwidth and provide other control and management functions. Gatekeepers are the brains of an H.323 network and act like SIP servers.
- Multipoint control units, which mix and distribute conference media streams for three or more H.323 terminals

The gatekeepers, gateways and MCUs are logically separate components of the H.323 architecture but can be implemented as a single physical device.

H.323 uses a number of protocols to set up a call. First of all, a supported client queries an H.323 gatekeeper for the address of a new user. The gatekeeper retrieves the address and forwards it to the client, which then establishes a session with the new client using H.225. Once the session is established, another H.323 protocol, H.245, negotiates the available features of each client. Because H.323 must first establish a session before it negotiates the features and functions of that session, call setup can take a long time. The amount of delay will depend upon the type of network.

- **Session Initiation Protocol (SIP):**

Background of the SIP:

The meteoric ascent of the Internet as a rival to the circuit-switched telephone network has given rise to strong economic and technological reasons for converged services and architectures. In order to assimilate telephony services with the ubiquitous technology of IP, a signalling protocol is required to set up and tear down connections.

A number of different communities put forward solutions, each colored by their own priorities and interests. The Internet community wanted to introduce innovative services based on enhanced web-authoring tools like XML and more open, peer-to-peer protocols and call models. The IETF offered SIP.

SIP was originally intended to create a mechanism for inviting people to large-scale multipoint conferences on the Internet Multicast Backbone (Mbone). At this stage, IP telephony didn't really exist. It was soon realized that SIP could be used to set up point-to-point conferences - phone calls.

The SIP approach exemplifies classic Internet-style innovation: build only what you need, to address only what is lacking in existing mechanisms. Because the SIP approach is modular and free from underlying protocol or architectural constraints, and because the protocols themselves are simple, SIP has caught on as an alternative to H.323 and to vendor-proprietary mechanisms for transporting SS7 protocols over IP.

The Basics of the SIP:

The Session Initiation Protocol (SIP) is a signalling protocol used for establishing sessions in an IP network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. The ability to establish these sessions means that a host of innovative services become possible, such as voice-enriched e-commerce, web page click-to-dial, Instant Messaging with buddy lists, and IP Centrex services.

Over the last couple of years, the Voice over IP community has adopted SIP as its protocol of choice for signalling and the industry has focused a great deal of attention on this emerging standard. Currently, SIP is a draft from the Internet Engineering Task Force (IETF), the body responsible for administering and developing the mechanisms that comprise the Internet. SIP is still evolving and being extended as technology matures and SIP products are socialised in the marketplace.

The IETF's philosophy is one of simplicity: specify only what you need to specify. SIP is very much of this mould; having been developed purely as a mechanism to establish sessions, it does not know about the details of a session, it just initiates, terminates and modifies sessions. This simplicity means that SIP scales, it is extensible, and it sits comfortably in different architectures and deployment scenarios.

SIP is a request-response protocol that closely resembles two other Internet protocols, HTTP and SMTP (the protocols that power the world wide web and email); consequently, SIP sits comfortably alongside Internet applications. Using SIP, telephony becomes another web application and integrates easily into other Internet services. SIP is a simple toolkit that service providers can use to build converged voice and multimedia services.

In order to provide telephony services there is a need for a number of different standards and protocols to come together - specifically to ensure transport (RTP), to authenticate users (RADIUS, DIAMETER), to provide directories (LDAP), to be able to guarantee voice quality (RSVP, YESSIR) and to inter-work with today's telephone network.

Architecture of the SIP:

There are two basic components within SIP: the SIP user agent and the SIP network server. The user agent is the end system component for the call and the SIP server is the network device that handles the signalling associated with multiple calls.

The user agent itself has a client element, the User Agent Client (UAC) and a server element, the User Agent Server (UAS). The client element initiates the calls and the server element answers the calls. This allows peer-to-peer calls to be made using a client-server protocol.

SIP user agents can be lightweight clients suitable for embedding in end-user devices such as mobile handsets or PDAs. Alternatively, they can be desktop applications that bind with other software applications such as contact managers.

The main function of the SIP servers is to provide name resolution and user location, since the caller is unlikely to know the IP address or host name of the called party, and to pass on messages to other servers using next hop routing protocols.

SIP servers can operate in two different modes: stateful and stateless. The difference between these modes is that a server in a stateful mode remembers the incoming requests it receives, along with the responses it sends back and the outgoing requests it sends on. A server acting in a stateless mode forgets all information once it has sent a request. These stateless servers are likely to be the backbone of the SIP infrastructure while stateful-mode servers are likely to be the local devices close to the user agents, controlling domains of users.

Other functions fulfilled by the SIP servers are re-direct and forking. A re-direct server receives requests but, rather than passing these onto the next server, it sends a response to the caller indicating the address for the called user. Forking is the ability to split or "fork" an incoming call so that several locations can ring at once. The first location to answer takes the call.

Together these components make up a basic SIP infrastructure. Application servers can sit above these components delivering SIP services to end-users. Application server's host service modules such as IM and presence, third party call control and user profiling. They also interact with other media servers and can be responsible for load balancing across a distributed architecture. These servers will also typically contain the management interface.

Custom services can be created by accessing subroutines in the application servers using APIs (Application Program Interfaces). When service modules are used in combination the service possibilities are vast. For further information on service creation and APIs see the Programming SIP section.

SIP follows the client/server model that has proved so successful with the Internet. Backbone service providers can offer SIP infrastructure as part of their IP service offering to other service providers. These can, in turn, offer their own SIP services over this infrastructure in a ISP/ASP model. It is even possible for applications to be written by end-users in the same way that web applications are today.

Characteristics of the SIP:

SIP is described as a control protocol for creating, modifying and terminating sessions with one or more participants. These sessions include Internet multimedia conferences, Internet (or any IP Network) telephone calls and multimedia distribution. Members in a session can communicate via multicast or via a mesh of unicast relations, or via a combination of these. SIP supports session descriptions that allow participants to agree on a set of compatible media types. It also supports user mobility by proxying and redirecting requests to the user's current location. SIP is not tied to any particular conference control protocol. In essence, SIP has to provide or enable the following functions:

- Name translation and user location

  Ensuring that the call reaches the called party wherever they are located. Carrying out any mapping of descriptive information to location information. Ensuring that details of the nature of the call (Session) are supported.

- Feature negotiation

  This allows the group involved in a call (this may be a multi-party call) to agree on the features supported - recognizing that not all the parties can support the same level of features. For example, video may or may not be supported.

- Call participant management

  During a call a participant can bring other users onto the call or cancel connections to other users. In addition, users could be transferred or placed on hold.

- Call feature changes

  A user should be able to change the call characteristics during the course of the call. For example, a call may have been set up as 'voice-only', but in the course of the call, the users may need to enable a video function. A third party joining a call may require different features to be enabled in order to participate in the call.

  SIP fulfils these functions and re-uses other web elements to make it flexible and scalable.

Rather than defining a new type of addressing system, SIP addresses users by an email-like address. Each user is identified through a hierarchical URL that is built around elements such as a user's phone number or host name. This means that it is just as easy to redirect someone to another phone, as it is to redirect someone to a web page.

SIP uses MIME, the de facto standard for describing content on the Internet, to convey information about the protocol used to describe the session. As a result, SIP messages can contain Java applets, images, audio files, authorization tokens or billing data.

SIP borrows from the email model, using the Domain Name System to deliver requests to the server that can appropriately handle them. This simplifies the integration of voice and email. Servers along the call path can easily create and forward email messages, and vice versa, enabling various combined services.

SIP provides its own reliability mechanism and is therefore independent of the packet layer and only requires an unreliable datagram service. SIP is typically used over UDP or TCP.

SIP provides the necessary protocol mechanisms so that end systems and proxy servers can provide services:

- User location
- User capabilities
- User availability
- Call set-up
- Call handling
- Call forwarding, including:
    - The equivalent of 700-, 800- and 900- type calls
    - Call-forwarding no answer
    - Call-forwarding busy
    - Call-forwarding unconditional
    - Other address-translation services
- Callee and calling "number" delivery, where numbers can be any (preferably unique) naming scheme
- Personal mobility, i.e., the ability to reach a called party under a single, location-independent address even when the user changes terminals
- Terminal-type negotiation and selection: a caller can be given a choice how to reach the party, e.g. via Internet telephony, mobile phone, an answering service, etc.;
- Terminal capability negotiation
- Caller and callee authentication
- Blind and supervised call transfer

- Invitations to multicast conference.

Comparisons:

SIP vs. H.323:

SIP is, more or less, equivalent to the Q.931 and H.225 components of H.323. These protocols are responsible for call setup and call signalling. Consequently, both SIP and H.323 can be used as signalling protocols in IP networks. The main advantage of SIP is its full integration with other Internet protocols and functions, such as email, web and instant messaging. For example, it is very easy to "forward" calls to web pages or email; web pages can be included in responses to call attempts. SIP is also codec-neutral and has been used to set up anything from audio and video to Doom distributed games and screen sharing. There a number of open-standard programming interfaces, SIP servlets, sip-cgi and CPL, that are particularly suited to SIP-based devices. These programming interfaces make creating SIP-based services very similar to writing web scripts or web pages.
It also supports a number of services, such as ACD and follow-me, which are much harder to implement in H.323.

Philosophy

SIP	**H.323**
New World" - a relative of Internet protocols - simple, open and horizontal.	"Old World" - complex, deterministic and vertical.
IETF	ITU
Carrier-class solution addressing the wide area.	Borne of the LAN - focusing on enterprise conferencing priorities.

Characteristic Differences:

SIP	**H.323**
A simple toolkit upon which smart clients	H.323 specifies everything including the

and applications can be built. It re-uses Net elements (URLs, MIME and DNS).	codec for the media and how you carry the packets in RTP
Leaves issues of reliability to underlying network.	Assumes fallibility of network - an unnecessary overhead.
SIP messages are formatted as text. (Text processing lies behind the web and email).	Binary format doesn't sit well with the internet - this adds complexity.
SIP allows for standards-based extensions to perform specific functions.	Extensions are added by using vendor-specific non-standard elements.
Hierarchical URL style addressing scheme that scales.	Addressing scheme doesn't scale well.
Minimal delay - simplified signalling scheme makes it faster.	Possibilities of delay (up to 7 or 8 seconds!).
Slim and Pragmatic.	The suite is too cumbersome to deploy easily.

Services:

SIP	H.323
Ability to 'fork' calls.	Not possible in the existing standard.
User profiling.	
Unified messaging.	
Presence management.	
Unique ability to mix media.	Cannot mix media within a session.
URLs can be embedded in web browsers and email tools	H.323 has no URL format.
Works smoothly with media gateway controllers controlling multiple gateways - crucial in a multi-operator environment.	"Shoehorn" interworking with SS7 is problematic - H.323 has trouble connecting calls to and from PSTN endpoints.
Gateways - crucial in a multi-operator environment.	Services are nailed-down and constricted - voice only ceiling.

Seamless interactions with other media - services are only limited by the developers' imagination.	
Standard IP Centrex services.	Standard IP Centrex services.

Status:

Industry endorsed.	Popularity due to the fact that it was the first set of agreed-upon standards.
Many vendors are developing products.	The majority of existing IP telephony products rely on the H.323 suite.

SIP vs MGCP/Megaco:

SIP and MGCP/Megaco are not peers; they can and will coexist in converged networks. There are, however, a number of issues surrounding implementations that will influence future directions and capabilities.

As discussed in the Architecture section, MGCP/Megaco does not constitute a complete system: a session initiation protocol is required between gateway controllers. SIP is eminently suitable and is a requisite where there is more than one softswitch.

A more contentious area of discussion is the use of MGCP/Megaco to control end-points. A media gateway could be an IP phone but, due to the service limitations this imposes, this is likely to be unpopular. MGCP/Megaco would only be able to support basic IN-type services in a dumb black phone.

For advanced services (i.e. anything more sophisticated than IN services), SIP is required to reside both in the endpoints and above the signalling network, acting as the service intelligence. The issue that then arises is where should services reside.

Softswitch vendors would prefer the service intelligence to reside in the IP Central Office, tied in to the softswitch architecture. This perspective holds firm in the short-term where emphasis on convergence means that the interconnect point between a circuit-switched environment and an IP network will be a major focus. In this scenario, SIP application servers reside with the softswitches in the IP Central Office with MGCP/Megaco controlling multiple media gateways across the network, delivering services to all endpoints. As the legacy circuit-switched network diminishes in importance, and focus shifts squarely onto the IP infrastructure, then this model will become increasingly imbalanced and irrelevant. The softswitch function will need to evolve away from the interconnect point

In a pure IP environment, service creation would be distributed across the network. This is the model that has produced the startling innovations that we have seen on the Internet: anyone with a few dollars and a good idea has the opportunity to give it a try. The Application Service Provider model can be extended to offer voice-type services. ASPs, ISPs, or even the end-users themselves can create their own SIP services; after all, SIP's similarity to other Internet protocols makes it a familiar programming language to web developers. A SIP-centric implementation would use MGCP/Megaco only for internally controlling an IP telephony gateway. SIP application servers would distribute services throughout the network via SIP proxy servers.

SIP	MGCP/Megaco
Peer-to-peer signalling protocol.	Can be used as a control protocol for delivering services across the network.
A session initiation protocol required between separate softswitches.	Used for internally controlling an IP telephony gateway.
Client-server architecture.	Master-slave architecture.
"Pure" IP solution.	An interim solution for co-existing.
	networks - "PSTN over IP".

Horizontal architecture that re-uses Internet elements.	Mirrors the signalling and control architecture of IN.
Intelligent clients.	Assumes dumb end-points.
Abstracts the signalling layer from the network.	Pre-supposes the existence of hardware.
"New world" approach simple open and horizontal.	"Old world" Decentralized, controlled and vertical.

Comparison between SIP, H.323 and MGCP/Megaco:

	SIP	**H.323**	**MGCP/Megaco**
Philosophy	Horizontal	Vertical	Vertical
Complexity	Low	High	High
Scope	Simple	Full	Partial
Scalability	Good	Poor	Moderate
New Service Revenues	Yes	No	No
Internet Fit	Yes	No	No
SS7 Compatibility	Poor	Poor	Good
Cost	Low	High	Moderate

7. Products and Services:

➢ IP Telephony

Internet Telephony, or IP Telephony, refers to a new class of applications that merge Internet capabilities with PSTN functions. Internet telephony applications enable the transmission of real-time voice traffic over the ubiquitous Internet infrastructure and the seamless integration with the existing PSTN infrastructure. While IP Telephony primarily focuses on voice calls, it can also be used to carry other voice-band and multimedia applications, such as fax, video, and modem data. Convergence of the Internet and PSTN provides more effective network usage, substantial cost savings, and new revenue opportunities.

Trillium Digital Systems' IP Telephony software source code solutions enable the convergence of the IP and PSTN networks. Trillium's IP Telephony products helps network equipment vendors build complete end-to-end solutions ranging from Softswitches to IP phones.

IP Telephony Solutions from Trillium are mainly aimed at reducing risk, costs and accelerate time to market. The primary features of the IP telephony products form Trillium is given below. Fully portable, hardware-independent products enable integration into any processor or Operating System (OS).

- Simple, fully featured, and flexible interfaces
- Common mapping layer (TUCL) to socket interfaces of the IP stacks in operating systems such as Windows, Solaris, VxWorks and LINUX
- Highly efficient integrated ASN.1 PER library eliminates the need for third-party software packages and enables developing enhanced proprietary customer applications
- Accurate implementation of standards and assures interoperability
- High performance, key building blocks, combined with expert Professional Services, and unparalleled technical support
- Small footprint and small dynamic memory enable optimized solutions.
- Consistent architecture enables the efficient use of resources and eases integration
- Unparalleled service for standards updates, together with complementary solutions, enable customers to meet market demands.
- Consistent fault-tolerant and distributed architecture enables:
    - Developing scalable and reliable products
    - Integrating both fault-tolerant and non-fault-tolerant components
- Enables customers to focus on core competency to rapidly develop and deploy high-performance products, to compete successfully in the dynamic marketplace

Trillium offers the following IP Telephony network elements:

- Signaling Gateway (SG)
- Media Gateway Controller (MGC)
- Media Gateway (MG)
- Gatekeeper (GK)
- SoftSwitch
- IP Phones/Appliances
- Multi-service Switch
- 3G Serving GPRS Support Node (SGSN)
- 3G Radio Network Controller (RNC)
- Cable modem
- Multipoint Conference Unit (MCU)

Trillium's IP Telephony solutions include:

- H.323
- TCP/UDP Convergence Layer (TUCL)
- RTP/RTCP
- MGCP and H.248/MEGACO
- SIP and SIP-BCP-T
- SS7 over IP

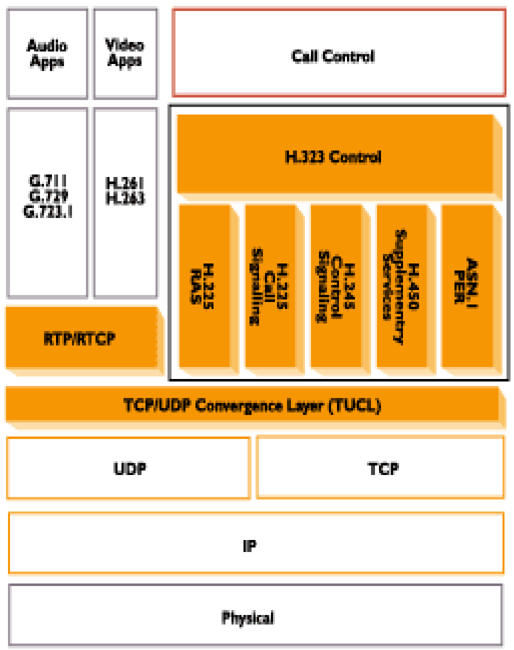
Trillium's Services can be further used to provide complete IP Telephony solutions including:

- Interworking Call Control applications
- Audio/Video Codec integration
- Enhanced application-specific development
- Proprietary value-added supplementary services

H.323 Standard:

H.323 is a standard from the International Telecommunications Union (ITU-T) that specifies the components, protocols, and procedures for multimedia communication over packet-based networks such as IP-based Local Area Networks (LANs) and Wide Area Networks (WANs), including the Internet. Trillium's H.323 source code solutions enable the implementation of Internet telephony devices

H.323 STACK DIAGRAM



H.323 Stack Components

H.245 Control Signaling:

- Control signaling between endpoints (client, Gateway) or between endpoints and Gatekeeper/MCU to determine the capabilities of the communicating endpoints.
- Opening and closing logical channels for media streams.
- Conference control commands

H.225 RAS:

- Registration, Admission, and Status (RAS) control signaling between an endpoint (client, Gateway) and Gatekeeper.

RTP:

- Real-time Transport Protocol (RTP) for carrying real-time audio/video data.

RTCP:

- Real-time Transport Control Protocol (RTCP) for providing feedback on the transmission and reception quality of data carried by the RTP.

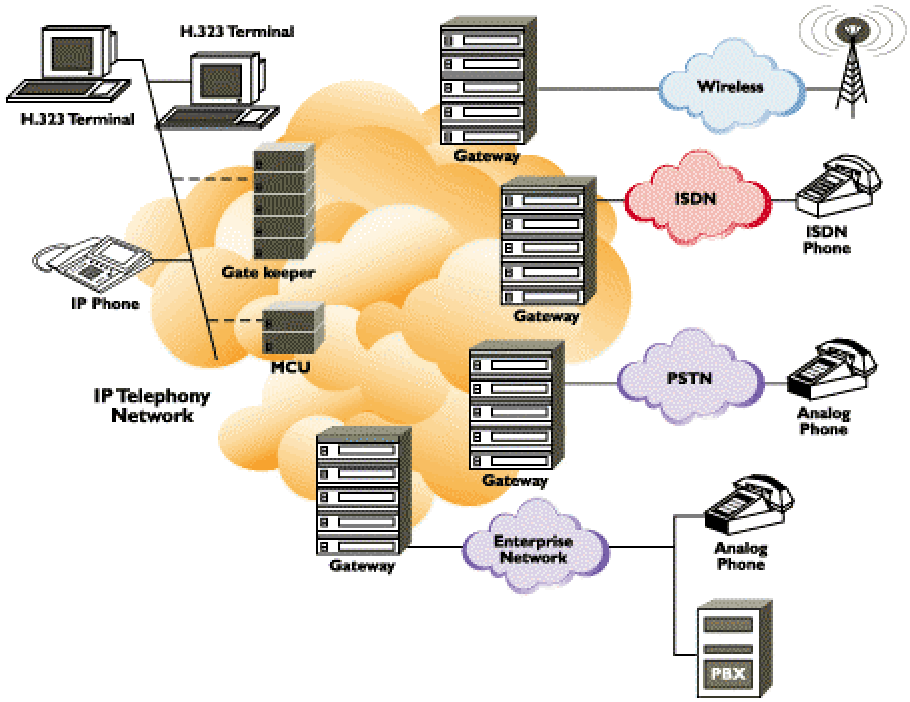
Audio/Video Codecs:

- Encoding/Decoding audio/video signals from and to the H.323 Terminal.

H.450 Supplementary Services:

- Adds basic call management features to a Terminal, such as Call Transfer, Caller ID, Call Forwarding (RTCP) for providing feedback on the transmission and reception quality of data carried by the RTP.

H.323 NETWORK ARCHITECTURE



Trillium's H.323 Solutions

The H.323 Control portable software product supports:

- H.323v3, H.225v3 Call Signaling, H.245v5 Control Signaling, H.225v3 RAS
- All H.450 Supplementary Services
- Annex E-based call signaling over UDP
- Annex G-based Gatekeeper-to-Gatekeeper (inter-domain) communication
- Flexible RAS API that allows minimal or full control for device implementers
- Fast connection procedures
- Tunneling of H.245 massages in the H.255.0 Call Signaling PDU's
- Interfaces for developing and deploying Gateway, Gatekeeper, multipoint controller, and Terminal services
- Static and round-robin distribution of incoming calls
- Transparent Gatekeeper-routed call mode
- API's for H.235 security protocol services
- Coexistence of multiple H.323 entities such as Gatekeeper and Gateway, using the same instance of the code
- Integrated ASN.1 Encoder/Decoder-PER library

The ASN.1 Encoder/Decoder-PER portable software product supports:

- Basic PER aligned variant encoding of ASN.1 data types to generate the transfer syntax, as specified in ITU-T Recommendation X.691

TCP/UDP Convergence layer (TUCL):

- TUCL portable software product is a generic protocol software layer that can be used as a transport layer with the H.323 stack.

MGCP/MEGACO:

Trillium Digital Systems' Gateway Control Protocol software code solutions include Media Gateway Control Protocol (MGCP) and H.248/MEGACO--products that accelerate the convergence of the Internet and PSTN.

Media Gateway Control Protocol (MGCP) is an IETF informational draft that has gained wide acceptance in the IP telephony market segment. MGCP has also been adopted by CableLabs® PacketCable™ initiative. An enhanced version of MGCP, known as the H.248/MEGACO protocol, has evolved into a new joint ITU-IETF standard.

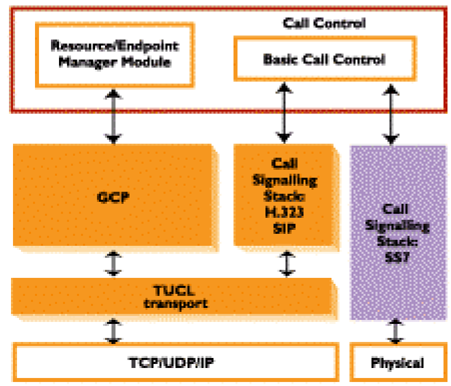
The Gateway Control Protocol (GCP) is an essential technology for IP Telephony Gateway solutions. The decomposed Gateway model safeguards investments in legacy telephone networks, while enabling the migration to a distributed and flexible IP network architecture.

The GCP solution represents a simplified messaging path between the Media Gateway Controllers (MGCs), also referred to as Call Agents (CAs), and Media Gateways (MGs). GCP functions in a call control architecture consisting of signaling gateways that are responsible for PSTN control and intelligent MGCs, controlling reduced intelligence MGs.

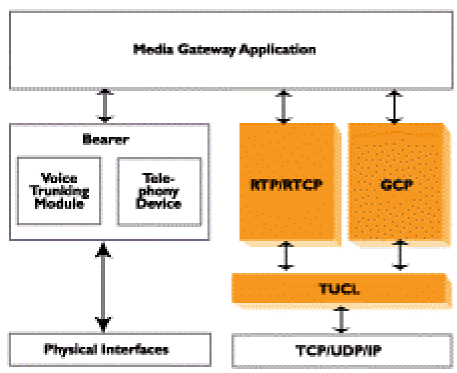
The MGCP and H.248 MEGACO allow for the "decomposition" of telephony gateways by separating the signaling and service capability (provided by the MGC), SS7 termination (provided by the SG), and the media transmission capability (provided by the MG). This model of a decomposed Gateway allows for the convergence of legacy telephony networks with an IP-based, next generation network infrastructure.

MGCP STACK

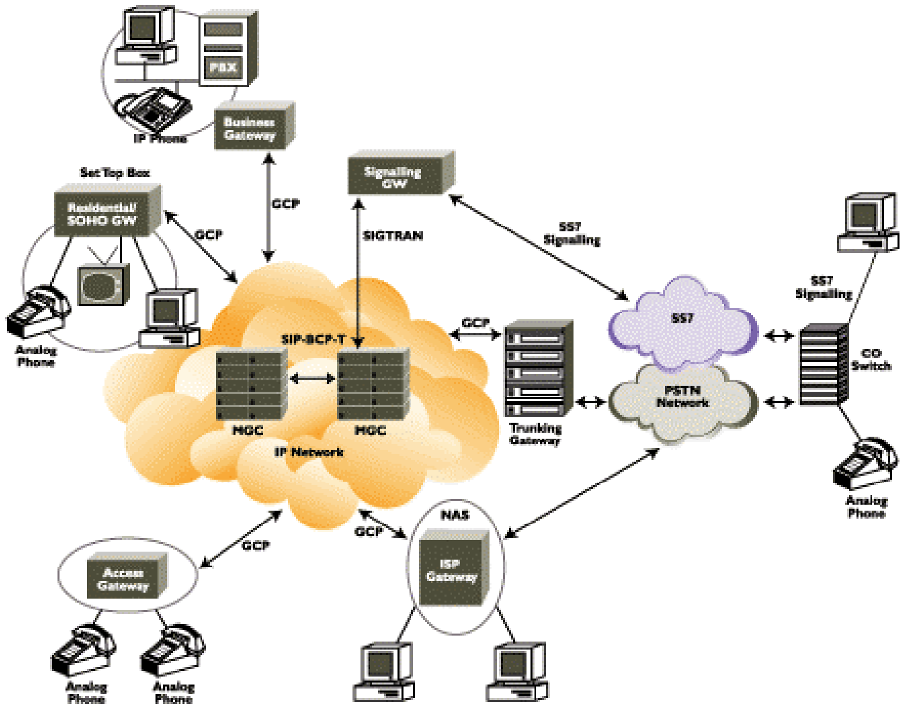
MGC stack



MG stack



MGCP/MEGACO NETWORK ARCHITECTURE



Trillium's MGCP/MEGACO Solutions

Trillium's GCP portable software product conforms to:

- IETF-established RFC 2705 MGCP standard
- CableLabs® PacketCable™ Network Call-based Signaling Protocol Specification (NCS)
- Joint ITU-IETF RFC 3015 H.248/MEGACO standard.

Common MGC and MG features:

- APIs to build MGCs and MGs
- Encode/Decode library engine for transmission and reception of all messages using text encode/decode
- Encodes/Decodes SDP descriptions in messages
- Manages the MGCP transactions over the UDP
- Supports H.248/MEGACO over the TCP
- Supports configured and discovered MGs
- Supports virtual and physical MGs in one instance
- Provides failover/service change support

MG features:

- Support for all defined packages
- Supports interaction with the configured MGC
- Supports protocol operation on the default port (protocol-specific) or on any other configured port
- Provides failover support
- RTP/RTCP for the media layer to transfer media

MGC features:

- Supports management of multiple MGs within a single instance
- Supports a distributed call control application for managing multiple MGs

- Supports communication on the default port or any other user-selected port
- Support for managing the transaction load on the MGC

The TCP/UDP Convergence Layer (TUCL) portable software product provides:

- Common transport functions for applications operating over the TCP/UDP/IP stacks
- Provides transparent mapping to the underlying socket interface
- Direct mapping to the socket interface for OSs such as Solaris, Windows, VxWorks, and pSOS

Gateway Control Protocol (GCP)
Description:

Gateway Control Protocol (GCP) is an implementation of the MGCP and MEGACO (H.248) protocol stack for use in a Media Gateway Controller and a Media Gateway.

Product deliverables consist of C source software, documentation, training, a warranty, and technical support. IP Telephony gateway equipment and PacketCable equipment manufacturers to speed the time to market can use the GCP product and lower the cost of developing the MGCP and MEGACO (H.248) protocol stacks for their Media Gateway Controllers (MGCs), Call Agents (CAs), and Media Gateways (MGs).

The following figures illustrate the MGC and MG protocol stack architectures:

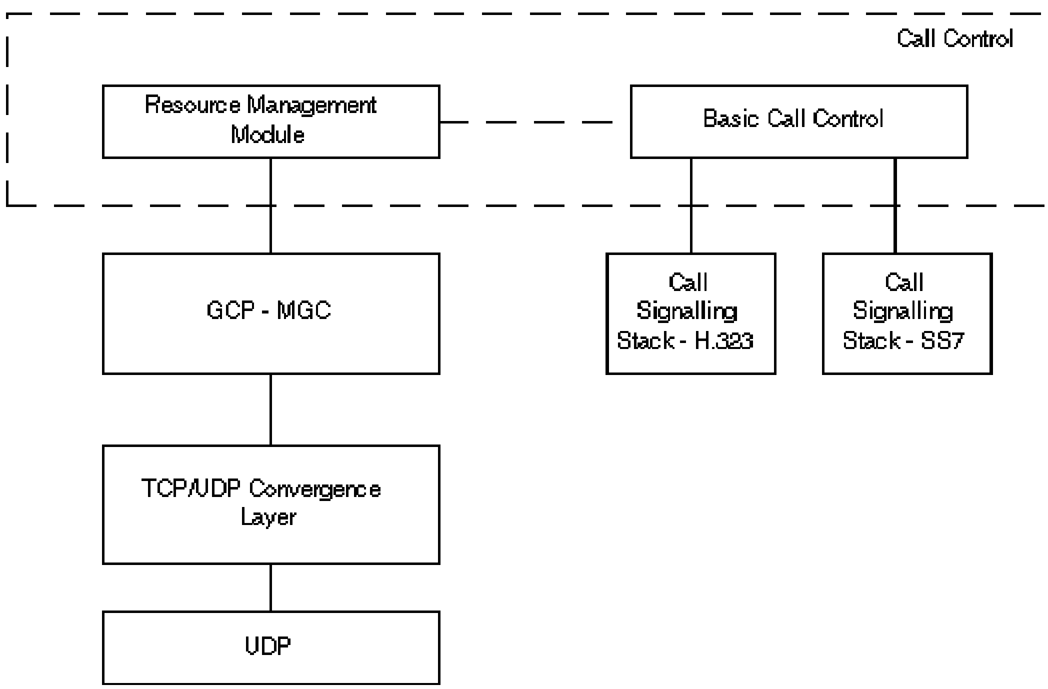


Figure 1: MGC protocol stack architecture

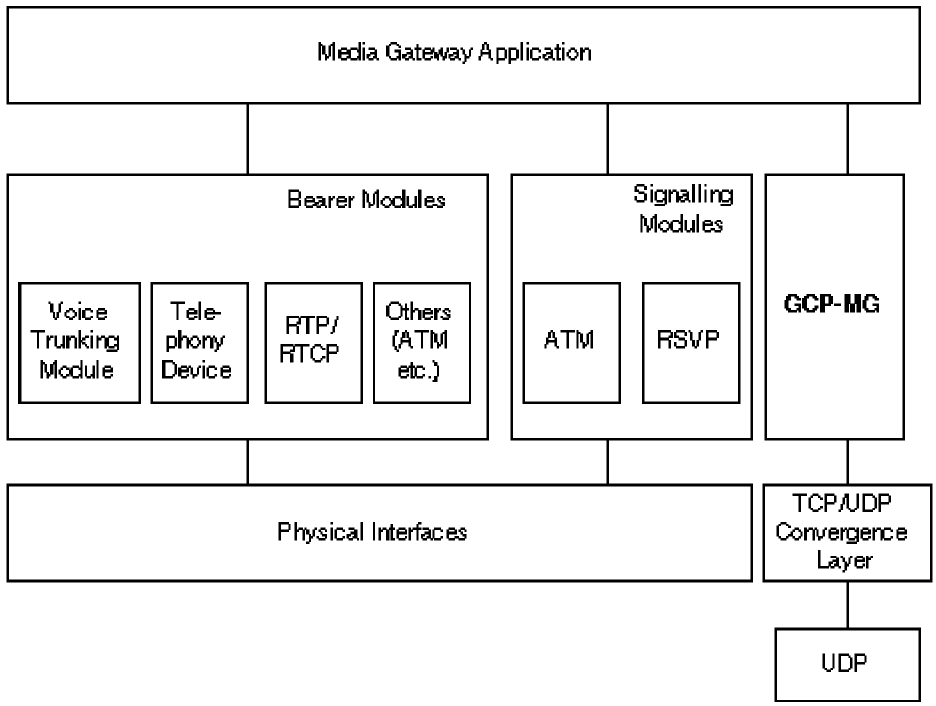


Figure 2: MG protocol stack architecture

Features and Benefits:

Trillium's GCP supports the following features:

Common Features

- Provides interfaces to build MGC and MG components
- Provides management interfaces for configuration, control, and retrieval of status and statistics. It also provides protocol state and alarm information at the management interface.
- Provides extensive run-time error checking support
- Provides extensive debugging support for easy system integration and testing
- Provides support for function call traces and PDU traces. The trace information is provided at the management interface to support remote logging and analysis.
- Provides an encode/decode library engine for transmission and reception of all protocol messages using text encode/decode
- Encodes/decodes SDP session descriptions contained in the messages
- Encodes/decodes ATM SDP session descriptions contained in the messages
- Supports transaction management as specified in the specification
- Receives notification on the responses received for transactions
- Prevents restart avalanche
- Performs version negotiation
- Conforms to Trillium Advanced Portability Architecture (TAPA)®

MGCP-Specific Features

- Supports transmission and delivery of the following MGCP commands and notifications:
  - EPCF — Endpoint configuration
  - CRCX — Create connection
  - MDCX — Modify connection
  - DLCX — Delete connection
  - RQNT— Request to notify
  - NTFY — Notify
  - AUEP — Audit endpoint
  - AUCX — Audit connection
  - RSIP— Restart in progress
- Supports the following basic MGCP packages:
  - Generic media package
  - DTMF package
  - MF package
  - Trunk package
  - Line package
  - Handset emulation package
  - RTP package
  - Network access server package
  - Announcement server package
  - Script package
- Supports transparent handling of unsupported/non-standard packages
- Supports communication between MGC and MG over UDP

MEGACO-Specific Features

- Supports transmission and delivery of the following MEGACO commands and notifications:
  - Add
  - Modify
  - Subtract
  - Move
  - AuditValue
  - AuditCapabilities
  - Notify
  - ServiceChange
  - Redundant MG support
  - Failure and handoff procedures
- Supports the following basic MEGACO packages:
  - Generic package
  - Base root package
  - Tone generator package
  - Tone detection package
  - Basic DTMF generator package
  - DTMF detection package
  - Call progress tone generator package
  - Call progress tone detection package

    - Analog line supervision package
    - Basic continuity package
    - Network package
    - RTP package
    - TDM circuit package
    - NAS package
- Supports transparent handling of unsupported/non-standard packages
- Supports communication between MGC and MG over UDP or TCP

MGC-Specific Features

- Supports the management of multiple MGs within a single instance of the product
- Supports a distributed call control application for managing the multiple media gateways
- Supports communication on the default port or on any other user selected port when communicating with MGs. This allows support for managing the transaction load on the MGC.
- Supports configured and discovered MGs

MG-Specific Features

- Supports interaction with the configured MGC
- Can support protocol operation on the default port (protocol-specified) or any other port configured

Conformance:

The Gateway Control Protocol software conforms to the following standards:

- Media Gateway Control Protocol Version 1.0, Internet RFC 2705
- PacketCable Network-Based Call Signaling (NCS) Protocol Specification, PKT-SP-EC-MGCP-I02-991201
- PacketCable PSTN Gateway Call Signaling (TGCP) Protocol Specification, PKT-SP-TGCP-I01-991201
- SDP: Session Description Protocol, Internet RFC2327
- Augmented BNF for Syntax Specifications: ABNF, Internet RFC 2324
- Megaco Protocol Version 1.0, Internet RFC3015, November 2000
- ITU-T recommendation H.248, June 2000

SIP: Session Initiation Protocol
The Session Initiation Protocol (SIP) is an IETF standard that specifies the basic and supplementary services to create, modify, and delete multimedia sessions or calls. These multimedia sessions include, but are not limited to, multimedia conferences, distance learning, and Internet telephony. The SIP-BCP-T is an IETF-defined extension of SIP, enabling inter-MGC communication.

Trillium Digital Systems' comprehensive Session Initiation Protocol (SIP) software source code solutions support SIP and SIP-BCP-T Protocols.

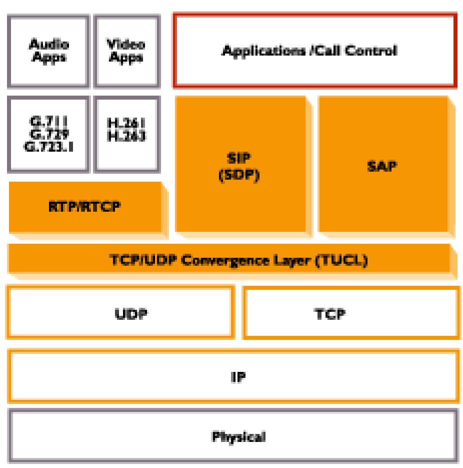
SIP is a lightweight, transport-independent, text-based protocol used as a signaling protocol for Internet conferencing and telephony:

- Lightweight in that SIP has only six different types of methods, reducing the level of complexity for the user.
- Transport layer-independent because SIP can be used with any datagram or stream protocol, such as UDP, TCP, ATM, and so on, thereby providing flexibility of use
- Text-based allowing for low overhead.

These factors allow for seamless call routing, based on the currently specified call flow instructions, and contribute to making SIP highly scalable.

SIP also provides ease in adding enhanced services. This extensible paradigm helps expedite customer adoption of SIP. Another driver of SIP is SIP-BCP-T—a set of extensions to the SIP protocol—to enable inter-Media Gateway Controller (MGC) communications. SIP has also been extended to support third-party call control.

SIP STACK DIAGRAM



SIP Stack Components

User Agent Client (UAC)

- Supports the caller application that initiates and sends the SIP requests

User Agent Server (UAS)

- Receives and responds to SIP the requests on behalf of the clients.
- Accepts, redirects, or refuses the calls

Proxy Server

- Receives requests from clients and forwards them to next-hop servers.
- Contains the UAC and UAS.

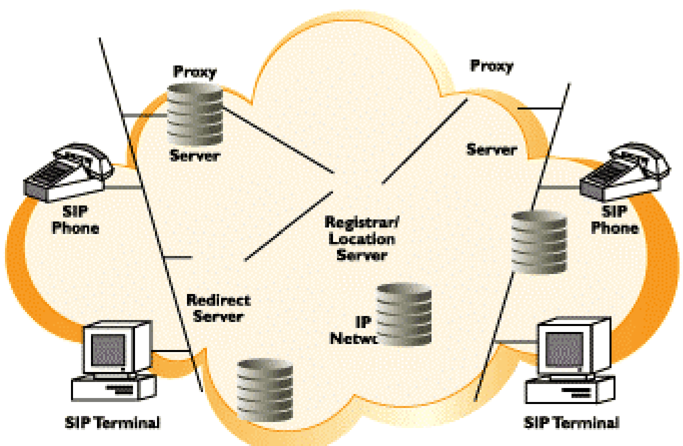
Redirect Server

- Accepts the SIP requests, maps the address into zero or more new addresses, and returns those addresses to the client.
- Does not initiate SIP requests or accept calls

Registrar/Location Server

- Provides information about a caller's possible locations to redirect and proxy servers
- Capable of being co-located with a SIP server

SIP Network Architecture



Session Initiation Protocol (SIP)

Description:

The Session Initiation Protocol (SIP) (RFC 2543) defined by IETF, is an application layer control protocol that can establish, modify, and terminate multimedia sessions or calls.

SIP is a text-based client-server protocol. It is modeled after the Hypertext Transfer Protocol (RFC 2616). SIP can invite parties to both unicast and multicast sessions and it is independent of the type of session being established.

Trillium's SIP product is a complete implementation of the SIP protocol stack for use in a SIP-enabled device. SIP can run on the top of either the Transmission Control Protocol (TCP) or the

User Datagram Protocol (UDP). SIP provides its own reliability, and UDP enables SIP messages to be multicast. The SIP network has two primary components:

- User Agent (UA)
- Network Server (NS)

Features and Benefits:

- Procedures to establish, change, and tear down multimedia sessions or calls
- Two sets of APIs to implement the SIP user agent and network server applications. The network server may include the proxy, redirect, or registrar server functions.
- Fast and robust ABNF parses for SIP and SDP
- Extensive run-time error-checking support
- Extensive debugging support to ease system integration and testing
- Detailed call statistics and accounting information
- Procedures for error detection and recovery
- Conforms to Trillium Advanced Portability Architecture (TAPA)®

Features Unique to Trillium's SIP Implementation

Trillium's SIP can:

- Act as a standalone registrar, proxy, or redirect server with minimal interaction with the service user applications
- Optionally pass SDP as a string to the user. This allows efficient implementations at the call agents.
- Seamlessly handle unknown methods and unknown headers
- Implement the cache and registry databases maintained using Trillium's multiway radix tree implementation, optimized for memory usage
- Simultaneously perform different logical functions - such as user agent, proxy, redirect, and registrar server - in the same instance of the code
- Support inherent call distribution to cater scalable, carrier-grade applications. Incoming calls are distributed on a round-robin basis to the service users. All incoming call-independent messages can be conveyed to service users dedicated to handling these messages; outgoing calls can be distributed across different transport servers or even different transport providers, if required.
- Maintain a database of users registered at a registrar server inside the SIP layer
- Maintain various caches, such as, caches for DNS (SRV and A records), for location results returned from an external location service, for registrations learned by snooping on multicast addresses, and so on. These caches help expedite lookups and reduce the effort of customers - enabling them to concentrate on developing the call control application.
- Support APIs to contact the external/third-party registrar server and location server for address lookups
- Support automatic switchover from a stateful proxy to a stateless proxy upon depletion of available resources
- Support the automatic generation of a number of SIP headers from inside the stack

- Support a rich set of APIs allowing support for different possible network configurations
- The network server-proxy with a colocated registrar can take care of directed pickup cases
- As a user agent, SIP can take care of recursing on redirect responses received from a downstream server with minimal interaction with the call control application
- As a network server, SIP can handle call forking and can recurse on redirect responses received from a downstream server with minimal or no interaction with the network server application
- Act as a standalone registrar server (without an application)

Product Interworking:

SIP runs on top of Trillium's TCP/UDP Convergence Layer (TUCL) and is also fully compatible with Trillium's Interworking Call Control (ICC) product for interworking between SIP and other protocols.

Conformance:
The SIP software supports the following standards:

- IETF RFC 2543 (bis-02), *SIP*: *Session Initiation Protocol*, Handley, Schulzrinne, Schooler, and Rosenberg, November 24, 2000.
- Internet Draft, *The SIP INFO Method*, Donovan, January 2001.
- *IETF RFC 2327, SDP: Session Description Protocol*, Handley, Jacobson, April 1998.
- Internet Draft, *MIME Media Types for ISUP and QSIG Objects*, Zimmerer, Vemuri, Ong, Zonoun, and Watson, June 2000.
- Internet Draft, *Integration of Resource Management and SIP*, Marshall et all, June 2000.
- Internet Draft, *Reliability of Provisional Responses in SIP*, Rosenberg and Schlzrinne, June 2000.
- Internet Draft, *Establishing QoS and Security Preconditions for SDP Sessions*, Rosenberg, Schulzrinne, and Donovan, June 1999.

➢ **SS7 TELEPHONY:**

Signaling System 7 (SS7) is a global telecommunications, protocol standards suite that defines the procedures by which network elements, within the Public-Switched Telephone Network (PSTN), exchange control information over digital signaling links for setting up, managing, and tearing down wireline and wireless (cellular) calls.

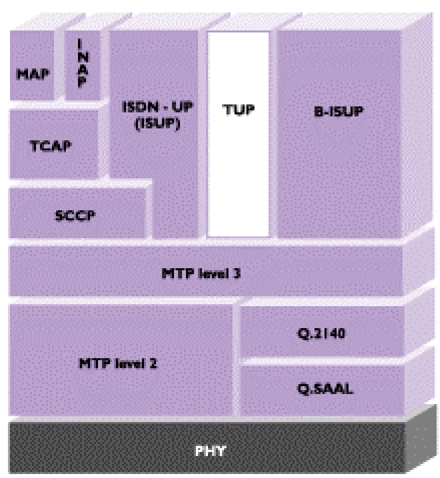
The standard has been extended for multiple country-specific variations, such as the American National Standards Institute (ANSI) and Telcordia Technologies (formerly Bell Communications Research, "Bellcore") standards within North America, and the European Telecommunications Standards Institute (ETSI) standard in Europe.

The SS7 network has emerged as a common, low-delay, highly secure, and reliable infrastructure designed to support wireline and wireless (cellular) calls over the circuit-switched network.

Trillium's SS7 software source code solutions extend the signaling capabilities of telecommunications networks. The solutions offer a complete standards-based implementation, and are ideal for equipment manufacturers seeking to reduce their time-to-market by using proven components. Trillium’s carrier-grade SS7 products provide scalability, security, quality, reliability, speed and increased network utilization for the converged network.

SS7 Stack Diagram

Signaling System 7 (SS7) is a global telecommunications, protocol standards suite that defines the procedures by which network elements, within the Public-Switched Telephone Network (PSTN), exchange control information over digital signaling links for setting up, managing, and tearing down wireline and wireless (cellular) calls.



SS7 stack architecture (narrowband and broadband)

SS7 Stack Components

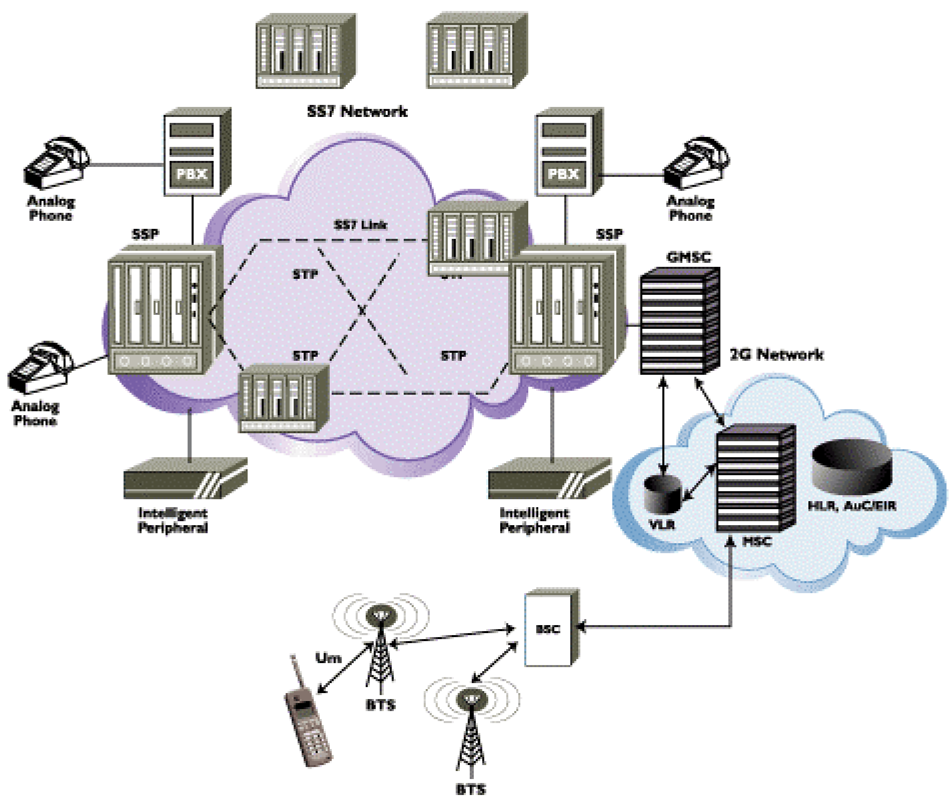
SS7 comprises the following protocol layers/components:

- MTP1 defines the physical and functional characteristics of the digital signaling link.
- MTP2 supports reliable transfer of signaling messages.
- MTP3 supports signaling traffic management.

- SCCP supports connection-oriented and/or connectionless services for transferring data across an SS7 network.
- ISUP supports signaling to establish, maintain, and release circuit-switched network connections.
- TCAP supports the exchange of non-circuit related information between signaling points using the SCCP connectionless service. The Mobile Switching Center (MSC), Serving GPRS Support Node (SGSN), and Gateway GPRS Support Node (GGSN) use MAP in wireless networks to query the Home Location Register (HLR) or Visitor Location Register (VLR) to determine and/or verify subscriber services.
- B-ISUP supports signaling to establish, maintain, and release broadband-switched network connections across an SS7/ATM network. It can directly interwork with Trillium's ATM and SS7 products.
- TUP supports signaling to establish, maintain, and release circuit-switched network connections.

SS7 NETWORK ARCHITECTURE

Signaling System 7 (SS7) is a global telecommunications, protocol standards suite that defines the procedures by which network elements, within the Public-Switched Telephone Network (PSTN), exchange control information over digital signaling links for setting up, managing, and tearing down wireline and wireless (cellular) calls.



Trillium's SS7 Solutions:

Trillium’s comprises highly experienced and dedicated experts available to provide system integration, customization, application development, and consulting services.

Trillium’s services can be leveraged to provide complete SS7 solutions, including:

- Interworking Call Control applications.
- Driver development and integration.
- Element Management Application and integration

Fault-Tolerant/High Availability:

- Carrier-Grade Network Reliability Requirements of Five 9s Availability
- Flexible Architecture
- Software-Based Active-Standby Configuration to Achieve Redundancy
- Provides Run-Time State Update of Standby
- Controlled (Manual) Switchover and Forced (Automatic) Switchover

Distributed Fault-Tolerant/High-Availability:

- Carrier-grade Network Requirements of Six 9s Availability
- Controlled (Manual) Switchover and Forced (Automatic) Switchover
- Allows Co-Existence of Distributed and Non-Distributed Layers
- High Performance and Scalability Achieved by Distributing Processing Load Across Multiple Processors

Trillium’s SS7 product line includes the following protocol layers:

MTP2 supports:

- Reliable transfer of signaling messages over signaling links
- ITU-T, ANSI, TTC (Japan), NTT (Japan), China, and other variants

MTP3 supports:

- Broadband and narrowband signaling traffic management, signaling link and route management
- Functionality as an SSP or STP; Direct interworking with Trillium’s Q.2140 ATM product
- ITU-T, ANSI, TTC (Japan), NTT (Japan), Singapore, China, B-ICI, and other variants

SCCP supports:

- Connection-oriented and/or connection-less services (Class 0, 1, 2, and 3) for transferring data across an SS7 network
- ITU-T, ANSI, and China variants

ISUP supports:

- The signaling protocol to establish, maintain, and release circuit-switched network connections across an SS7 network. It can act as originating, destination, or intermediate exchange
- ITU-T, Telcordia (formally Bellcore), ANSI, Singapore, Q.767, ETSI, FTZ, Russia, Italy, NTT (Japan), and other variants

TCAP supports:

- End-to-end, connectionless network service protocol between transaction capability users, across an SS7 network
- An ASN.1 encoding/decoding engine to encode/decode all operation codes and dialog parameters
- ITU-T, ANSI, ETSI, and TTC (Japan) variants

INAP supports:

- Capability Set 1 (CS1), as defined by the ITU, ETSI, and the Generic Requirement (GR) standards of the Bellcore Advanced Intelligent Network (AIN)
- The interaction between the SSF, SCF, Specialized Resource Function (SRF), and the Service Data Function (SDF)

MAP supports:

- Refer to Wireless Section

B-ISUP supports:

- Establishing, maintaining, and releasing broadband-switched network connections across an SS7/ATM network. It can act as the originating, destination, or intermediate exchange. Also, it can directly interwork with Trillium's ATM and SS7 products
- ITU-T and ATM Forum variants

Q.2140 supports:

- Convergence functions necessary to map the SS7 MTP Level 3 protocol to the ATM Q.SAAL protocol
- ITU-T Q.2140, B-ISDN - ATM Adaptation Layer - SSCF at NNI and Q.2110, B-ISDN - ATM Adaptation Layer - SSCOP

TUP supports:

- Establishing, maintaining, and releasing circuit-switched network connections across an SS7 network. It can act as the originating, destination, or intermediate exchange
- ITU-T and China variants
- Available only through Professional Services

The Motorola MC68360 SCC Driver portable software product supports the HDLC and SS7 interfaces. It can be directly used with Trillium’s LAPB, LAPD, Frame Relay, and MTP2 products.

The Motorola MPC8260 Driver and MPC860 Driver portable software product supports the HDLC, AAL5, and SS7 interfaces. It can be directly used with the LAPB, LAPD, Frame Relay, ATM, and MTP2 products.

Trillium's SS7 is a global telecommunications protocol standards suite for functions including set-up, management, and teardown of basic calls; call forwarding; three-way calling, and caller ID; Local Number Portability (LNP); wireless services such as Personal Communications Services, (PCS), wireless roaming, and mobile subscriber authentication.

Trillium's SS7 software solution offers the following features:

- Fully portable, hardware-independent products enable integration into any processor and Operating System (OS).
- Simple, fully featured, and flexible interfaces.
- Mature product, integrated into a wide variety of solutions, ensures interoperability.
- Accurately implements standards and assures interoperability.
- High-performance, key building blocks, combined with expert Professional Services, and unparalleled technical support.
- Small footprint and small dynamic memory enable optimized solutions.
- Consistent architecture enables the efficient use of resources and eases integration.
- Unparalleled service for standards updates, together with complementary solutions, enable customers to meet market demands.
- Consistent fault-tolerant and distributed architecture enables, Developing scalable and reliable products
- Integrating both fault-tolerant and non-fault-tolerant components.
- Enables customer to focus on core competency to rapidly develop and deploy high-performance products, to compete successfully in the dynamic marketplace.

Trillium's SS7 enables:

- **PSTN/IP** Convergence
- **IP Telephony** (voice)
- **3G Wireless** infrastructure

Trillium's SS7 product line includes the following protocol layers:

MTP2 supports:

- Reliable transfer of signaling messages over signaling links
- ITU-T, ANSI, TTC (Japan), NTT (Japan), China, and other variants

MTP3 supports:

- Broadband and narrowband signaling traffic management, signaling link and route management
- Functionality as an SSP or STP
- Direct interworking with Trillium's Q.2140 ATM product
- ITU-T, ANSI, ETSI, Telcordia (formerly Bellcore), TTC (Japan), NTT (Japan), China, RFC2719 and other variants

SCCP supports:

- Connection-oriented and/or connection-less services (Class 0, 1, 2, and 3) for transferring data across an SS7 network
- ITU-T, ANSI, and China variants

ISUP supports:

- The signaling protocol to establish, maintain and release circuit-switched network connections across an SS7 network. It can act as originating, destination, or intermediate exchange ITU-T, Telcordia, ANSI, Singapore, Q.767, ETSI, Germany, Russia, Italy, NTT (Japan) and other variants

TCAP supports:

- End-to-end, connectionless network service protocol between transaction capability users, across an SS7 network
- An ASN.1 encoding/decoding engine to encode/decode all operation codes and dialog parameters
- ITU-T, ANSI, ETSI and TTC (Japan) variants

INAP supports:

- Capability Set 1 (CS1), as defined by the ITU, ETSI, and the Generic Requirement (GR) standards of the Telcordia Advanced Intelligent Network (AIN)
- The interaction between the SSF, SCF, Specialized Resource Function (SRF), and the Service Data Function (SDF)

B-ISUP supports:

- Establishing, maintaining, and releasing broadband-switched network connections across an SS7/ATM network. It can act as the originating, destination, or intermediate exchange. Also, it can directly interwork with Trillium's ATM and SS7 products
- ITU-T and ATM Forum variants

Q.2140supports:

- Convergence functions necessary to map the SS7 MTP Level 3 protocol to the ATM Q.SAAL protocol

- ITU-T Q.2140, B-ISDN - ATM Adaptation Layer - SSCF at NNI and Q.2110, B-ISDN - ATM Adaptation Layer - SSCOP

TUP supports:

- Establishing, maintaining, and releasing circuit-switched network connections across an SS7 network. It can act as the originating, destination, or intermediate exchange
- ITU-T and China variants
- Available only through Professional Services

SCTP provides:

- Persistent associations
- Reliable data transport to handle missing and duplicated datagrams
- Congestion control algorithms

M3UA provides:

- Address Translation and the Mapping of MTP Level 3 user primitives to SCTP associations/streams, for message routing
- MTP Level 3 User Part Signaling (ISUP, SCCP) transport
- Seamless MTP Level 3 network management interworking IUA supports transporting ISDN network layer signaling over the IP network.

SUA provides:

- Carries SCCP User Part signaling such as ISUP, TCAP and RANAP
- Allows seamless interworking between SCCP in both the SS7 and IP domains
- Supports operation between SCCP peers in an all-IP domain

IUA provides:

- Transport of Q.921/Q.931 boundary primitives
- Communication between protocol management modules on the SG and MGC
- Supports management of active associations between the SGs and MGCs

The Motorola MC68360 SCC Driver portable software product supports the HDLC and SS7 interfaces. It can be directly used with Trillium’s LAPB, LAPD, Frame Relay, and MTP2 products.

The Motorola MPC8260 Driver and MPC860 Driver portable software product supports the HDLC, AAL5, and SS7 interfaces. It can be directly used with the LAPB, LAPD, Frame Relay, ATM, and MTP2 products.

SS7/IP (SIGTRAN):
Signaling Transport (SIGTRAN) is an Internet Engineering Task Force (IETF) standard for transporting message-based Public-Switched Telephone Network (PSTN) Signaling System 7 (SS7) traffic over IP networks. The SIGTRAN framework defines a modular structure that uses a common reliable transport protocol and allows the definition of adaptation modules for different PSTN control protocols.

The transport protocol, Stream Control Transmission Protocol (SCTP), allows carriers to use the IP infrastructure to transport SS7 telephony traffic over an IP network. However, its generic design provides a reliable transport delivery mechanism for other multimedia and wireless frameworks such as H.323, MGCP, H.248/MEGACO, SIP and 3G.

SIP supports node-to-node transport of SS7/ISDN traffic between Signaling Gateways (SGs)/Media Gateways (MGs) and Media Gateway Controllers (MGCs). It works on the basic concept of associations and streams.

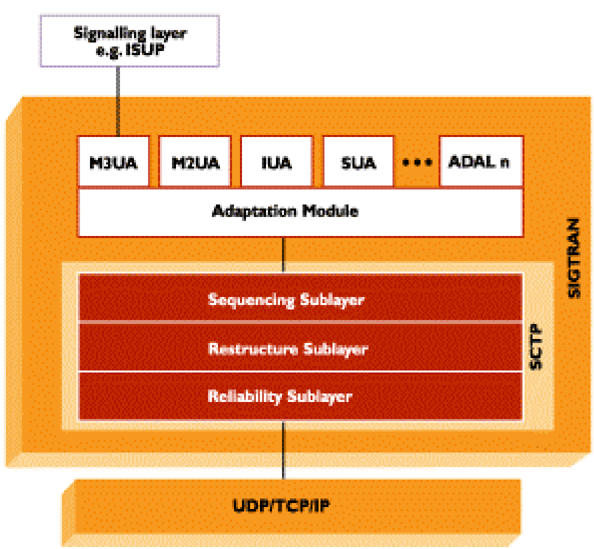
Trillium Digital Systems' SS7/IP (SIGTRAN) software source code solutions enable seamless convergence and interworking of PSTN and IP protocols under a unified architecture

An SCTP association:

- Is similar to a TCP connection
- Supports multiple IP addresses at either or both ends
- Supports multiple logical streams
- Provides sequenced delivery for user datagrams within a single stream

SCTP's design includes the appropriate congestion-avoidance behavior, message validation, and path management capabilities required by the PSTN.
Adaptation modules may be added as extensions to the transport protocol. Currently, an IETF SIGTRAN working group has defined adaptation modules for MTP Level 3, MTP Level 2, ISDN Q.921 and SUA.

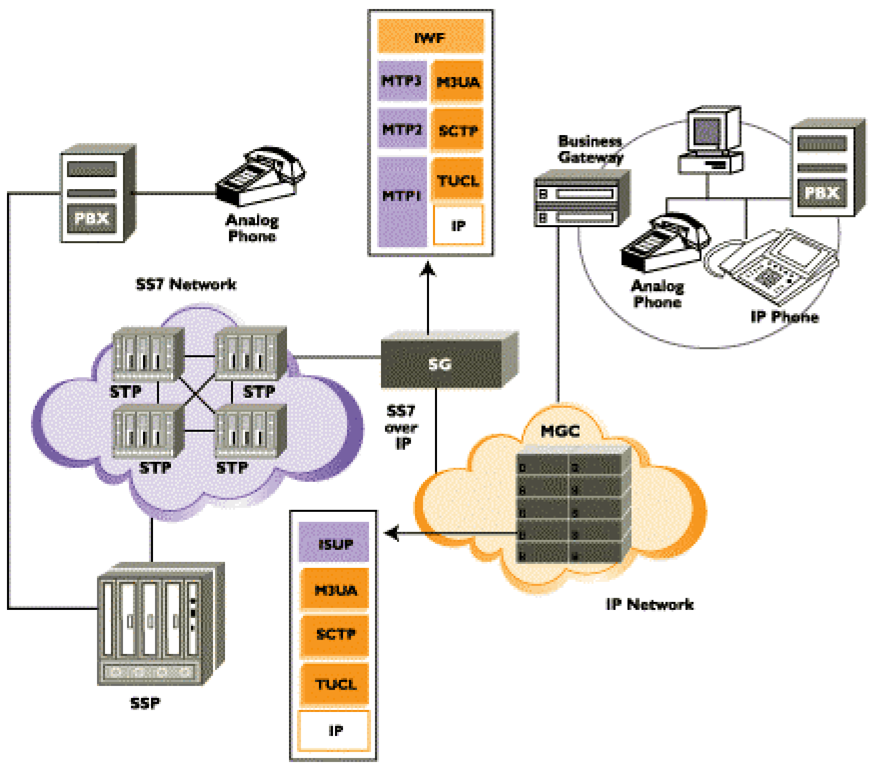
SIGTRAN STACK DIAGRAM



SIGTRAN STACK COMPONENTS:

Trillium's SIGTRAN family of protocols extends the value of a carrier's legacy telephone networks by carrying signaling traffic across SS7 and IP-based networks, and facilitates seamless and ubiquitous deployment of applications and services regardless of the underlying network infrastructure. In these coexisting and converging SS7/IP signaling networks, SIGTRAN plays a crucial role, and is being deployed in ever-growing number of network scenarios and integrated into network elements such as Signaling Gateways, Media Gateway Controllers, IP-resident databases, Third Generation Serving General Packet Radio Service Support Nodes, Radio Network Controllers, and Softswitches to accelerate development of converged voice and data solutions.

SIGTRAN NETWORK ARCHITECTURE



Trillium's SIGTRAN SOLUTIONS:

Trillium's SS7/SIGTRAN solution encompasses the SCTP, IUA, SUA and M3UA functionality defined by the IETF SIGTRAN Working Group.

Trillium's SCTP features:

- Persistent associations.

- Reliable data transport to have missing and duplicated datagrams.
- Elimination of head-of-line blocking.
- Immediate delivery of our-of-band data.
- Detection of session failure.
- User-controlled heartbeat generation.
- Congestion control algorithms.
- Bundling of multiple application PDUs to improve link use.
- Random tag and authentication cookie security mechanisms.

Trillium's IUA features:

- Transport of Q.921/Q.931 boundary primitives.
- Communication between protocol management modules on the SG and MGC.
- Supports management of active associations between the SGs and MGCs.

Trillium's M3UA features:

- Mapping of primitives received from the MTP Level 3 user layer to the corresponding SCTP primitives, associations, streams, and vice-versa.
- Same upper interface as MTP Level 3.
- Carries MTP Level 3 User Part Signaling, such as ISUP, SCCP, and TUP.
- Management of SCTP transport associations between the SG and MGC/IP databases.
- Nodal Interworking Function that provides seamless mapping between MTP Level 3 and M3UA.
- Active association control and failover.
- Seamless interworking of MTP Level 3 network management functions between SS7 and IP domains.

Trillium's SUA features:

- Carries SCCP User Part signaling such as ISUP, TCAP and RANAP.
- Allows seamless interworking between SCCP in both the SS7 and IP domains.
- Supports operation between SCCP peers in an all-IP domain.

➢ WIRELESS TECHNOLOGIES

Trillium Digital Systems' Wireless software source code solutions enable network equipment vendors to build a complete suite of wireless communication devices—from core network and radio-access network elements to mobile terminals for Second Generation (2G) digital wireless, General Packet Radio Service (GPRS), and the emerging Third Generation (3G) wireless networks.

The rapid and efficient deployment of new wireless data and Internet services has emerged as a critical priority for communications equipment manufacturers. Network components that enable wireless multimedia services are recognized as fundamental to the next-generation network infrastructure.

Wireless data services are expected to see the same explosive growth in demand that Internet services have seen in recent years. Audio, video, and data applications are converging to the point at which users of data and multimedia are demanding anytime, anywhere services for business and personal use. This need for ubiquitous wireless access is driving new business opportunities and innovation in the wireless access and infrastructure.

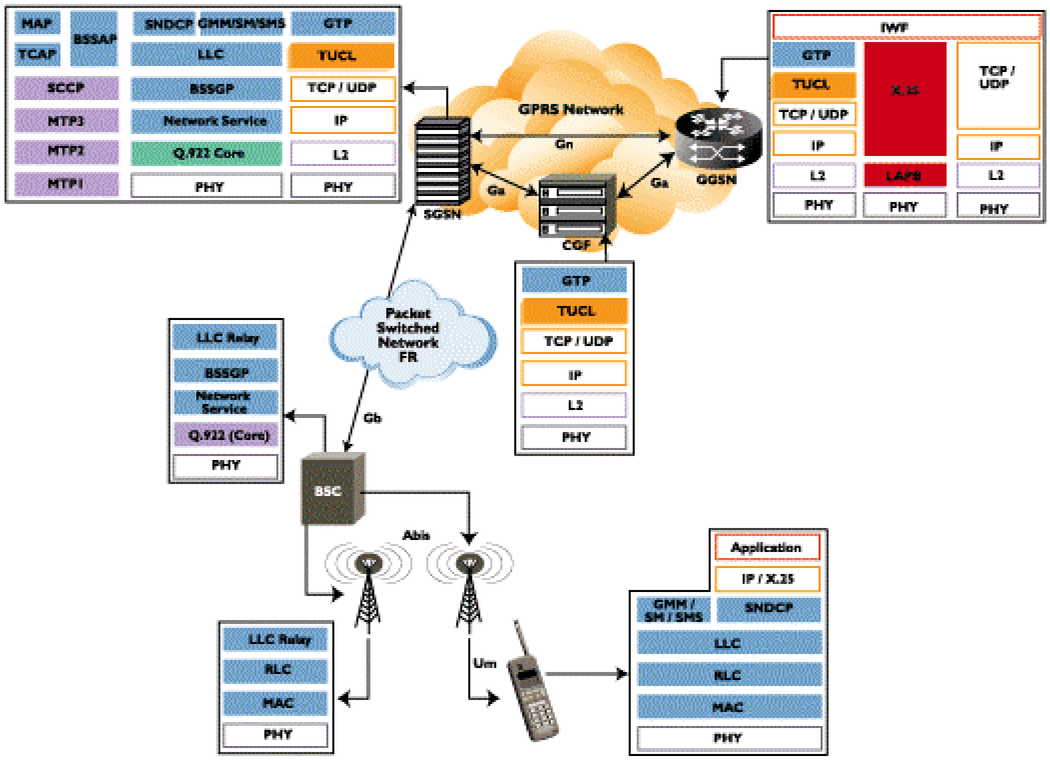
GPRS: General Packet Radio Service.

Trillium Digital Systems' General Packet Radio Service (GPRS) software source code solutions enable network equipment vendors to build a complete suite of communications device from GPRS Support Nodes to GPRS Mobile Terminal for the emerging high-speed, wireless packet data networks.

GPRS enables wireless service providers to offer high-speed packet data transmission over the Global System for Mobile Communications (GSM) and IS-136 networks. GSM and IS-136 are based on the Time Division Multiple Access (TMDA) technology.

The European Telecommunications Standards Institute (ETSI) has chosen GPRS as the data transfer mechanism of choice for GSM Phase 2+ and as a migration path to the Universal Mobile Telecommunications System (UMTS). The UMTS is ETSI's recommendation for IMT-2000, the 3G wireless initiative from ITU-T.

GPRS STACK DIAGRAM

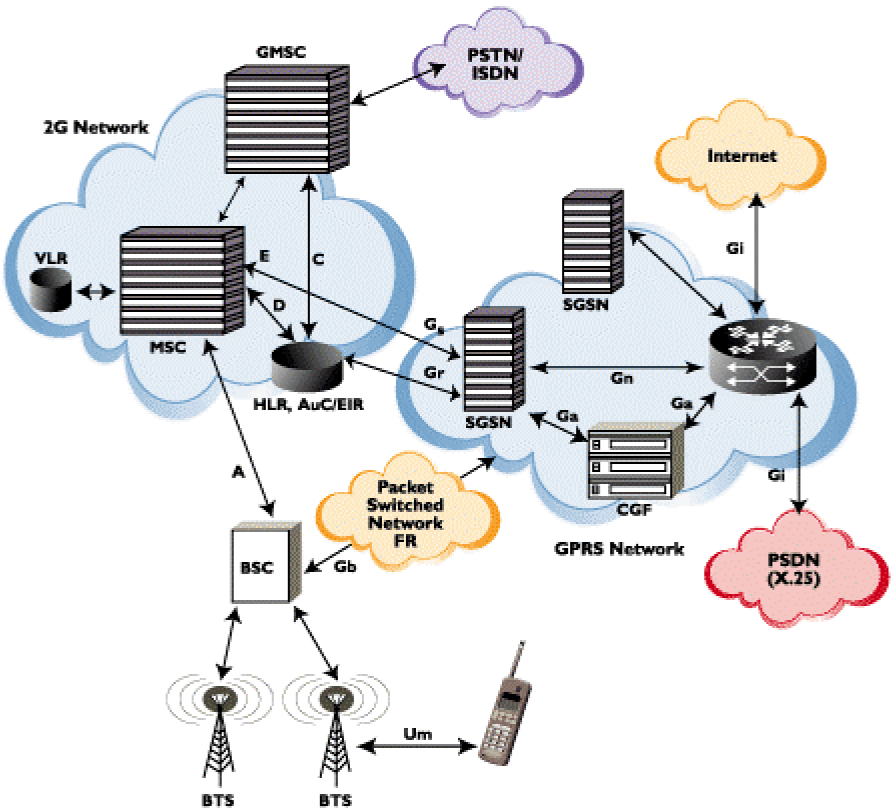


The following is a brief description of each protocol layer in the GPRS network infrastructure:

- Sub-Network Dependent Convergence Protocol (SNDCP): protocol that maps a network-level protocol, such as IP or X.25, to the underlying logical link control. SNDCP also provided other functions such as compression, segmentation and multiplexing of network-layer messages to a single virtual connection.
- Logical Link Control (LLC): a data link layer protocol for GPRS that functions similarly to Link Access Protocol-D (LAPD). This layer assures the reliable transfer of user data across a wireless network.
- Base Station GPRS Protocol (BSSGP): processes routing and quality of service (QoS) information for the BSS. BSSGP uses the Frame Relay Q.922 core protocol as its transport mechanism.
- GPRS Tunnel Protocol (GTP): protocol that tunnels the protocol data units through the IP backbone by adding routing information. GTP operates on top of TCP/UDP over IP.
- GPRS Mobility Management (GMM/SM): protocol that operates in the signaling plane of GPRS, handles mobility issues such as roaming, authentication, selection of encryption algorithms and maintains PDP context.
- Network Service: protocol that manages the convergence sub-layer that operates between BSSGP and the Frame Relay Q.922 Core by mapping BSSGP's service requests to the appropriate Frame Relay services.

- BSSAP+: protocol that enables paging for voice connections from MSC via SGSN, thus optimizing paging for mobile subscribers. BSSAP+ is also responsible for location and routing updates as well as mobile station alerting.
- SCCP, MTP3, MTP2 are protocols used to support Mobile Application Part (MAP) and BSSAP+ in circuit-switched PLMNs.
- Mobile Application Part (MAP): supports signaling between SGSN/GGSN and HLR/AuC/EIR

GPRS NETWORK ARCHITECTURE



Trillium's GPRS SOLUTIONS

Trillium offers a comprehensive software solution that encompasses all the new components of the GPRS network. Trillium's GPRS solution can be interfaced with the existing Public-Switched Telephone Network (PSTN) infrastructures. And Trillium's GPRS software interworks with Trillium's MAP-GSM Phase 2+ software, so that existing GSM systems can be easily upgraded to support GPRS services.

GPRS Products and Protocols:

Gateway GPRS Support Node (GGSN)

- GTP, TUCL, X.25, TCP/UDP/IP, LAPB, MAP, TCAP, SCCP, MTPI, MTP2, MTP3, MPLS

Serving GPRS Support Node (SGSN)

- GTP, BSSGP, LLC, GMM/SM, SNDCP, Q.922, TUCL, NS, BSSAP+, SCCP, MAP, TCAP, MTPI, MTP2, MTP3

Base Station Controller (BSC)

- RLC/MAC, BSSGP, NS, Q.922

Mobile Station (MS)
- RLC/MAC, LLC, SNDCP, GMM/SM, MM

Home Location Register (HLR)
- MAP, SCCP, TCAP, MTP1, MTP2, MTP3, X.25, LAPB

Mobile Switching Center (MSC)
- SCCP, MTP1, MTP2, MTP3, MAP, BSSAP+

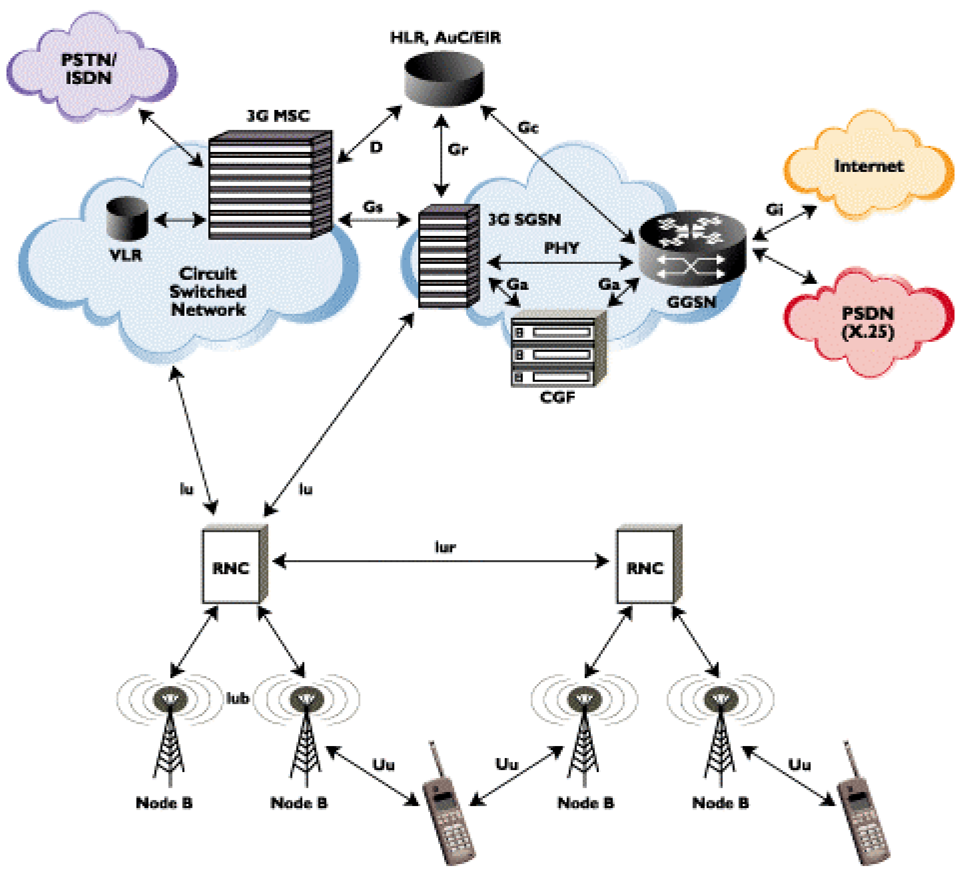
Charging Gateway Function (CGF)
- GTP, TUCL

➢ 3G WIRELESS

The third generation of mobile communications will greatly enhance the implementation of sophisticated wireless applications. Users will be able to utilize personal, location-based information and interactive services. Many corporations are restructuring their business processes to take full advantage of the opportunities provided by emerging wireless data services. Many advanced wireless services are already available, and the introduction of 3G wireless will add to their ubiquity and usefulness.

Trillium Digital Systems' Wireless software source code solutions enable network equipment vendors to build a complete suite of wireless communication device from core network elements to mobile terminals--for the emerging Third Generation (3G) wireless network.

3G NETWORK ARCHITECTURE



Trillium's 3G Solutions

Trillium's wireless protocol solutions are high-performance products that implement the complex signaling protocols required in the 2G/2G+/GPRS/3G wireless multimedia networks. The software solutions enable network equipment vendors to build a complete suite of wireless communication devices.

Trillium Protocols Enabling the Wireless World:

- 3G MAP
- Q.922/LAPF
- GSM MAP
- Q.933
- IS-41 MAP
- LLC
- SNDCP
- BSSGP
- Network Services
- LLC
- RLC/MAC
- SCTP
- Q.2140
- Q.2630
- M3UA
- Q.2160
- GTP
- SCCP
- TCAP
- MTP-3
- MTP-2
- RNSAP
- PDCP
- RANAP
- BSSAP+
- CAMEL (CAP)
- SUA
- GMM/SM

Trillium's services can be leveraged to provide complete Wireless solutions, including:

- Interworking Call Control
- Element manager application

Trillium's 3G wireless communications software will be compliant with 3G standards currently under development by the Third Generation Partnership Project (3GPP), a consortium of international organizations. We provide communications equipment and semiconductor manufacturers with complete software solutions for every element of the evolving 3G wireless network and future all-IP broadband wireless networks, ranging from mobile handsets to carrier-class equipment.

Trillium already offers SS7, ATM, and GPRS for GSM, all of which can be used to build 2G/2G+/3G wireless voice and data networks that interface with the existing PSTN infrastructure.

Trillium offers 3G portable software products including:

- SS7 over IP
- AAL2 Signaling.
- Broadband SS7.
- GPRS Tunneling Protocol (GTP).
- Radio Access Network Application Part (RANAP).
- Mobile Application Part (MAP).
- Camel Application Part (CAP).
- Voice over IP Protocols (H.323, SIP, GCP).
- Fault-Tolerant/High-Availability for increased network reliability.
- Distributed architecture to enable scalability and enhanced performance and to complement its existing wireless communications software portfolio.

Third generation (3G) wireless technology represents a shift from voice-centric to multimedia services: voice, data, video, and fax. Trillium's 3G wireless solutions enable increased capacity and higher data rates supporting equipment built on new modes of communication, business transactions, mobile two-way video and virtual private networks, and virtual home entertainment, among many other advancements.

The implementation of 3G wireless systems raises several critical issues, such as successful backward-compatibility to air interfaces as well as deployed infrastructures. The existence of legacy networks in most regions of the world makes compatibility and interworking between new 3G systems and legacy networks a crucial task that manufacturers of 3G networks must overcome to ensure the acceptance of this new technology by service providers and end-users.

3G Solutions have the following features:

- Fully portable, hardware-independent products enable integration into any processor and Operating System (OS).
- Simple, fully featured, and flexible interfaces.
- Mature product, integrated into a wide variety of solutions, ensures interoperability.
- Accurately implements standards and assures interoperability.
- High-performance, key building blocks, combined with expert Professional Services, and unparalleled technical support.
- Continuous monitoring and implementation of evolving standards.
- Small footprint and small dynamic memory enable optimized solutions.
- Consistent architecture enables the efficient use of resources and eases integration.
- Unparalleled service for standards updates, together with complementary solutions, enable customers to meet market demands.
- Consistent fault-tolerant and distributed architecture enables:
    - Developing scalable and reliable products.
    - Integrating both fault-tolerant and non-fault-tolerant components.

Telecommunications service providers and network operators are embracing the recently adopted, global 3G wireless standards to address rising customer demands and to provide new services.

The 3G wireless technology concept represents a shift from voice-centric services to multimedia-oriented services: voice, data, video, and fax.

With the increased capacity and higher data rates offered by 3G Wireless technologies (384kbps to 2mbps), new modes of communication, business transactions, information research, and entertainment are just a wireless stepping stone away.

The 3G network will initially use an ATM-based transport architecture migrating later to an all IP transport architecture.

Trillium's 3G portable software products are used in these network elements:

- Node B: RLC/MAC, NBAP, ATM (Q.2630, Q.2150.2, Q.SAAL, IME), AAL2/AAL5 drivers.
- 3G Gateway GPRS Support Node (GGSN): GTP, MAP, TCAP, SCCP, MTP3, MTP2.
- 3G Serving GPRS Support Node (SGSN): GTP, RANAP, MAP, GMM/SM, SS7 over IP (SCTP, M3UA, IUA, SUA), SS7 (SCCP, MTP3-B, TCAP), ATM (Q.2140, Q.SAAL, CIP, PLOA), CAP.
- Radio Network Controller (RNC): PDCP, RLC/MAC, RRC, RANAP, RNSAP, NBAP, GTP, SS7 over IP (SCTP, M3UA, IUA, SUA), ATM (Q.2630, Q.2150.2, Q.SAAL, Q.2140, CIP, PLOA), SS7 (MTP3-B, SCCP, TCAP).
- Mobile Station (MS): RLC/MAC, RRC, SNDCP, GMM/SM, PDCP.
- Home Location Register (HLR): SS7 (MAP, TCAP, SCCP, MTP3, MTP2).
- Mobile Switching Center (MSC): BSSAP+, RANAP, ATM (Q.SAAL, Q.2140), SS7 (SCCP, MTP3, MTP2, ISUP, TCAP, MAP), CAP).
- Charging Gateway Function (CGF): GTP.

- **BROADBAND TECHNOLOGIES:**

Increasing demand for ubiquitous, remote, and instant access to data, voice, and video content have fueled the growth of broadband technology, which provides high bandwidth and user-selectable Quality of Service (QoS).

Trillium Digital Systems' Broadband software source code solutions enable network equipment vendors to build a complete suite of communication devices—from core network elements to customer premises devices—in technologies such as ATM, Frame Relay, Broadband SS7, xDSL, cable modem, MPLS, and Third Generation (3G) wireless.

AAL 2

Trillium Digital Systems was the first company to offer *ATM Adaptation Layer 2* AAL2, an ATM portable signaling software product that handles blocking, unblocking, and reset, and call routing, based on the configured routing information; its flexible APIs provide all the information required to allow the user application to manage resources and CIDs, and to allow the user application-defined call routing during call establishment.

AAL2 STACK COMPONENTS:

ATM is essentially a packet-switched network with fixed-size packets known as cells. These cells are switched, based on a Virtual Path Identifier (VPI) and Virtual Channel Identifier (VCI). A connection in an ATM network is achieved by tying together a series of VPIs/VCIs at multiple interfaces.

A basic ATM network consists of a set of ATM switches and Network Interface Cards (NICs) connected by point-to-point ATM links. ATM is also used as a transport mechanism for various other technologies, such as Third Generation wireless (3G), MPLS, and Broadband SS7.

ATM technology provides high-bandwidth and user-selectable Quality of Service (QoS). User-selectable QoS means that a user application can specify the transmission delay, error rate, and bandwidth requirements on a per-connection basis.

ATM control plane protocols can be divided into six categories:

- The signaling protocol is used to establish the ATM SVC, consisting of the layer 2 data link management protocol and layer 3 signaling protocol.
- The Routing protocol is used to route an ATM signaling control message through the ATM network. The ATM Forum defines the PNNI for dynamic routing and the IISP for static routing.
- The Application protocol is used to map legacy Ethernet/Token-ring and IP applications on ATM. The ATM Forum defines LANE and MPOA, and the Internet Engineering Task Force (IETF) defines CIP and PPP over AAL5.
- The Interface Management protocol is used to manage ATM links and register ATM addresses.
- Drivers are for ATM chipsets to enable the ease of integration.

- The Interworking Call Control (ICC) product implements the call control functionality for ATM and enables interworking between ATM, SS7, ISDN, and other signaling protocols

Trillium's AAL2 Solutions
Some of the architectural features of Trillium’s ATM solutions are:

- Extensive debugging/trace functionality.
- Status and statistics information access, via the management interface.
- Error-logging functionality.
- Element control in the protocol software.

- Q.93B Signaling Layer
- Q.SAAL Signaling Layer 2
- IME
- PNNI Routing
- PLOA
- AAL Driver
- AAL2 Signaling
- Interworking Call Control

Trillium’s Q.93B Signaling Layer 3 portable software product supports:

- UNI 3.0/3.1/4.0, Q.2931/Q.2971, IISP, and PNNI 1.0 signaling procedures.
- Point-to-point and point-to-multipoint calls.
- Virtual channel allocation.
- Leaf-initiated join procedures.
- ATM anycast and group addresses.
- Proxy signaling, virtual UNI, and soft PVCs.
- Partial decoding of signaling messages.
- Fault-Tolerant/High-Availability (FT/HA), using the Q.93B Protocol-Specific Function (PSF).
- Signaling interworking using the Q.93B Protocol-Specific Interface Function (PSIF).

Trillium’s Q.SAAL Signaling Layer 2 portable software product supports:

- Complete Q.SAAL1/Q.SAAL2 and Q.2110/Q.2130 procedures.
- Error detection by selective retransmission.
- Data retrieval by the local user.
- Active or passive mode for the SAAL connection.
- Multiple SAAL connections over the same ATM link

Trillium’s IME portable software product supports:

- Complete ILMI 3.0/3.1/4.0 procedures.
- Proxy agent procedures.
- Auto-configuration procedures, including PVC.
- Multiple variable bindings in a single SNMP PDU.
- GetRequest, GetNextRequest, SetRequest, GetResponse, and Trap SNMPv1 PDUs.
- Easy addition of new MIBs.

- Periodic polling for keep-alive.
- Flexible use of the VCCs.
- Authentication and security features.

Trillium's PNNI Routing portable software product supports:

- PNNI version 1.0 procedures.
- Single- or multi-peer group models.
- Inside and outside links; version negotiation; Hello protocol-over-inside or –outside links; physical links or virtual paths; ATM anycast; SVC-based RCC; SVC-based RCC Hello protocol.
- Database synchronization upon startup using the neighbor peer-state machine.
- PTSE flooding, aging, and refreshing.
- PGL election procedure and capability.
- LGN horizontal-link Hello protocol, and link and nodal aggregation.
- Summarizing internal and external addresses.
- DTL processing and generating, crankback and rerouting of calls, and reachable addresses.
- Route catches (pre-computed and otherwise) to quickly look up popular destinations.
- Using the Dijkstra algorithm with configurable optimization criteria for route computing.
- Point-to-multipoint call topology tree for bandwidth-efficient call routing.

Trillium's PLOA portable software product supports:

- IETF's Classical IP over ATM (RFC 1577 and RFC 2225)—server and clients.
- Multi-protocol encapsulation over ATM adaptation layer 5 (RFC 1483).
- MPOA Client (MPC), MPOA Server (MPS), LAN Emulation Client 2.0, LAN Emulation Services 2.0, and PPP-over-AAL5.
- VC sharing among multiple applications using the LLC/SNAP encapsulation, to reduce VCC usage.
- AESA (formerly NSAP) and E.164 format addresses.
- UNI 3.0, UNI 3.1, UNI 4.0, and Q.2931 signaling.

Trillium's AAL Driver portable software supports:

- AAL5 implementation for MPC860 and MPC8260 processors.
- Message mode operations.
- Transmit and receive chaining.
- Error detection during reassembly.
- Maximum frame length of 1 to 65535 bytes.
- Receive address filtering.
- Programmable reassembly, time-out period selection for receive operations.
- Up to 65534 virtual channels.
- Cell scrambling for serial-mode SCCs.
- Flexible configuration; that is, all the options for address matching, such as using an internal lookup table, external address compression, or CAM.
- Loopback operation.

Trillium's AAL2 portable software product supports:

- AAL2 signaling procedures of Q.2630.1
- NNI-STC over MTP Level 3 (Q.2150.1)
- UNI-STC over SSCOP (Q.2150.2)
- Adding and deleting multiple AAL2 paths (ATM VCCs) between adjacent AAL2 nodes
- CID allocation
- Managing the AAL2 path
    - Blocking
    - Unblocking
    - Reset
- Call routing, based on the configured routing information
- Flexible APIs to provide all the information required to allow the user application to manage resources and CIDs, and to allow the user application-defined call routing during call establishment
- Provisioning multiple transport links between the adjacent AAL2 nodes, when the signaling transport is SAAL

Trillium's Interworking Call Control portable software supports:

- Switching Call Control for Q.93B, B-ISUP, ISUP, and ISDN
- Voice Trunking Over ATM (VTOA) as specified in the AF-VTOA-0089 and AF-VTOA-0113 phase 1 ATM Forum specifications.
- Interworking between
    - B-ISUP (ITU) and ISUP (ANSI)
    - B-ISUP (ITU) and ISUP (Q.767)
    - B-ISUP (ITU) and ISUP (ITU)
    - ISUP (ITU) and ISUP (ANSI)
    - B-ISUP (ITU) and Q.930/Q.931
    - Q.93B and B-ISUP
    - Q.93B and Q.930/Q.931
    - Q.930/Q.931 and ISUP
- Enbloc and overlap signaling
- Distributed call processing
- Resource allocation and verification
- Flexible resource allocation algorithms
- Dynamic and static binding of resources
- Dynamic and static routing
- Connection Admission Control
- B-ISUP and ISUP maintenance procedures
- Call prioritization while routing
- Generating Call Detail Records
- Generating call flow traces
- Maintaining procedures for B-ISUP and ISUP
- Flow control at interface

- **ATM; Asynchronous Transfer Mode:**

ATM is essentially a packet-switched network with fixed-size packets known as cells. These cells are switched, based on a Virtual Path Identifier (VPI) and Virtual Channel Identifier (VCI). A connection in an ATM network is achieved by tying together a series of VPIs/VCIs at multiple interfaces.

A basic ATM network consists of a set of ATM switches and Network Interface Cards (NICs) connected by point-to-point ATM links. ATM is also used as a transport mechanism for various other technologies, such as Third Generation wireless (3G), MPLS, and Broadband SS7.

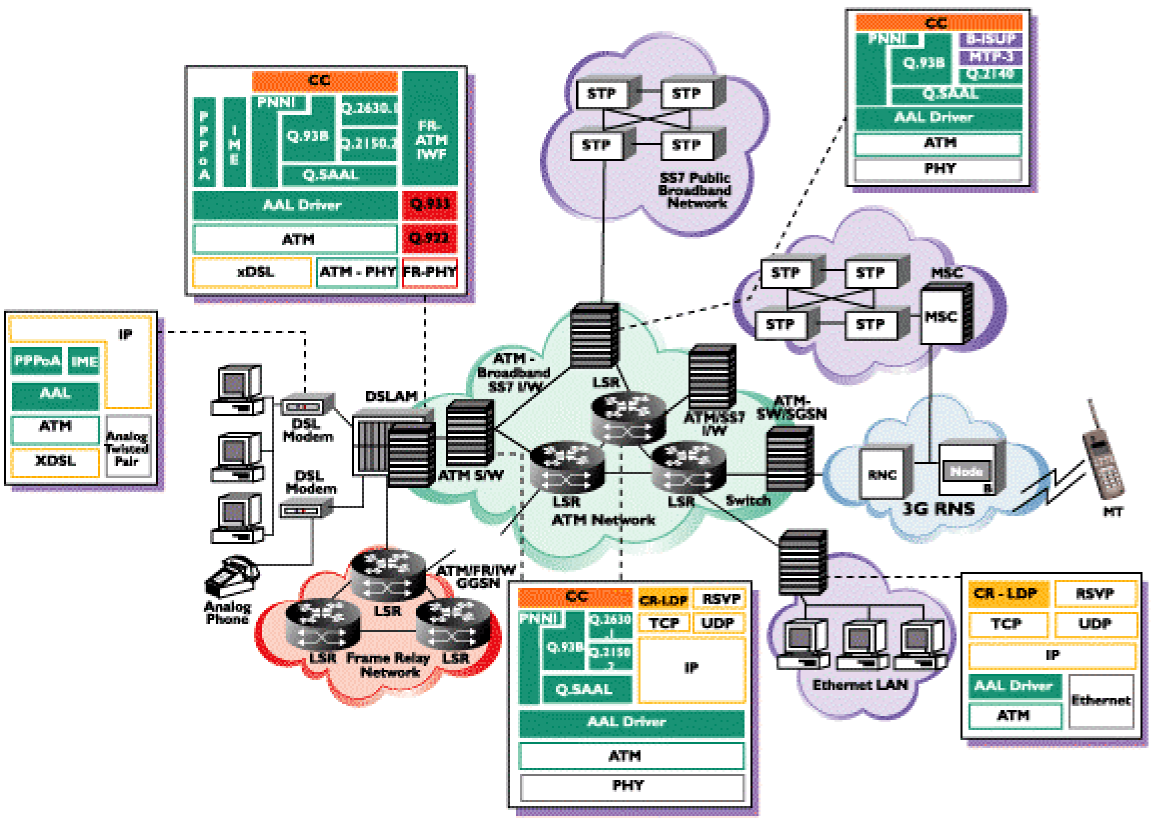
ATM technology provides high-bandwidth and user-selectable Quality of Service (QoS). User-selectable QoS means that a user application can specify the transmission delay, error rate, and bandwidth requirements on a per-connection basis.

Trillium Digital Systems' ATM software code solutions enable network equipment vendors to build a complete suite of communication devices, from a core network element to a customer premises device to efficient transport multimedia traffic over an ATM transport.

ATM control plane protocols can be divided into six categories:

- The Signaling protocol is used to establish the ATM SVC, consisting of the layer 2 data link management protocol and layer 3 signaling protocol.
- The Routing protocol is used to route an ATM signaling control message through the ATM network. The ATM Forum defines the PNNI for dynamic routing and the IISP for static routing.
- The Application protocol is used to map legacy Ethernet/Token-ring and IP applications on ATM. The ATM Forum defines LANE and MPOA, nd the Internet Engineering task Force (IETF) defines CIP and PPP over AAL5.
- The Interface Management protocol is used to manage ATM links and register ATM addresses.
- Drivers are for ATM chipsets to enable the ease of integration.
- The Interworking Call Control (ICC) product implements the call control functionality for ATM and enables interworking between ATM, SS7, ISDN, and other signaling protocols.

ATM NETWORK ARCHITECTURE



Trillium's ATM Solutions
Some of the architectural features of Trillium's ATM solutions are:

- Extensive debugging/trace functionality
- Status and statistics information access, via the management interface
- Error-logging functionality
- Element control in the protocol software

- Q.93B Signaling Layer
- Q.SAAL Signaling Layer
- IME
- PNNI Routing
- PLOA

- AAL Driver
- AAL2
- Interworking Call Control

➢ MPLS; Multi-Protocol Label Switching

MPLS is an emerging standard rapidly gaining acceptance by both vendors and ISPs. MPLS addresses the "Four S problem" of achieving:

- Speed
- Security
- Scalability
- Service guarantees

MPLS initially grew as a way of improving the forwarding speed of routers; however, in today's network, MPLS offers new capabilities for large scale, carrier-grade IP networks.

MPLS is an accelerated data transmission technology that uses a method of label lookup. A short, fixed-length label is appended/swapped to the packets that traverse through an MPLS domain. Once that packet reaches the destination edge of the network, the label is stripped away. The MPLS protocol derives its performance by simply switching labels on packets within the network.

MPLS is an efficient way of integrating IP and ATM networks because it allows high-volume IP traffic to traverse core ATM infrastructures. Traffic Engineering and Virtual Private Networks (VPNs) are other applications that can harness the MPLS technology.

MPLS network devices are called Label Switched Routers (LSRs), and are classified as Label Edge Routers (LERs), operating at the edge of the network, and Label Core Routers (LCRs), operating in the core of the network. LCRs may commonly be called LSRs.

The path that a data packet uses to reach its destination is referred to as the Label-Switched Path (LSP) and the distribution of labels within the MPLS domain is carried out by the signaling protocols such as LDP/CR-LDP RSVP-TE.

Trillium Digital Systems' Multi-Protocol Label Switching (MPLS) software source code solutions enable end-to-end, high performance implementation of the MPLS standard across the backbone networks. MPLS plays an important role in the routing, switching, and forwarding of packets through the next generation network to meet the service demands of the network users.

MPLS STACK DIAGRAM:

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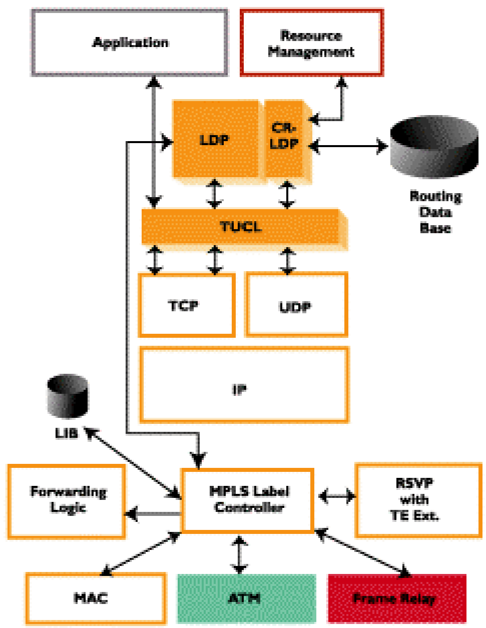
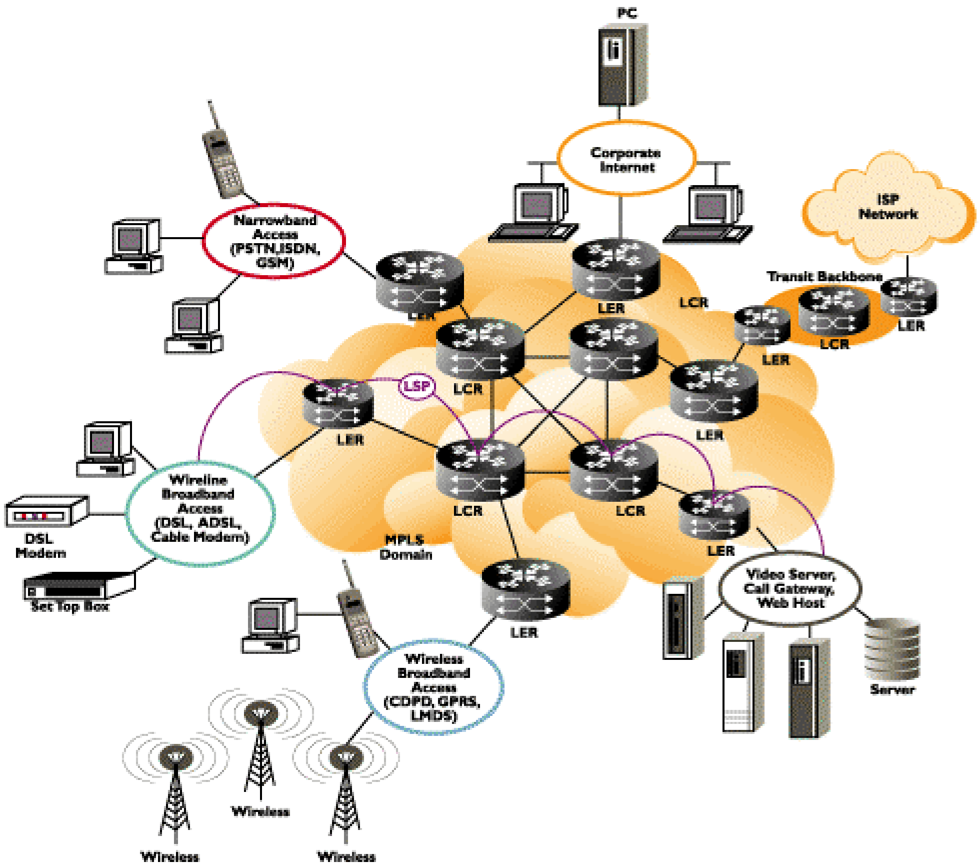


MPLS STACK COMPONENTS:

The core MPLS components can be broken down into the following parts:

- Network layer (IP) routing protocols
- Edge of network layer forwarding
- Core network label-based switching
- Label schematics and granularity
- Signaling protocol for label distribution
- Traffic engineering
- Compatibility with various Layer-2 forwarding paradigms (ATM, frame relay, PPP)

MPLS NETWORK ARCHIECTURE



Trillium's MPLS SOLITIONS:

Trillium's MPLS portable software product conforms to the following IETF drafts:

- MPLS-LDP
- MPLS-TRFENG
- MPLS-MIB
- MPLS-ATM
- MPLS-CR-LDP
- MPLS-SHIM

The control plane functionality of Trillium's MPLS stack includes:

- Label management
- Conservative or liberal label retention
- Request of Control-driven label bindings
- Basic and extended discovery mechanisms
- Downstream on Demand and Unsolicited label distribution support
- CR-LDP
- Label stacking
- Loop detection

- Load balancing
- Stream merging
- Stream aggregation

The TCP/UDP Convergence Layer (TUCL) portable software product is a generic protocol software layer that can be used as a transport layer with the MPLS stack. It provides transparency to the underlying TCP/UDP/IP stack interfaces and enables portable implementation of TCP/UDP/IP applications.

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RFC’s:

- RFC 2705
- RFC 2805
- RFC 2897
- RFC 3054
- RFC 3050
- RFC 3015