Introduction:

SIP is an application-layer control (signalling) protocol used to initiate, modify, and terminate interactive communication sessions, such as voice, video, chat, interactive games, and virtual reality. It is works independently on underlying transport protocol. In addition, it does not dependent on the type of sessions being established.

Basic Functions of SIP:

SIP provides and enables four basic functions including Name Translation and User Location, Feature Negotiation, Call Participant Management, and Call Feature Changes.

The main function of the SIP is to provide Name Translation and User Location that carry out mapping of descriptive information to location information. By employing Name Translation and User Location, SIP enables user mobility through a mechanism that allows requests to be proxied or redirected to the user's current location as the precondition of ensuring the call reaches the called party. Furthermore, Feature Negotiation Fuction allows participants to agress on a set of compatible media types since not all the participants are able to support the name level features, such as video etc.

SIP provides Call Participant Management fuction so that a participant has ability to bring other users onto a call or cancel connection to other users even during the course of the call. Similarly, users can also be placed on hold. Besides, call feature changes in SIP allows a user to change characteristics during a call.

Why SIP?

SIP has been developed purely as a mechanism establish e\sessions by initiating, terminating and modifying session without knowing the details of sessions, which means it is suitable to be employed in different architecutures and deployment scenario. In addition, SIP invitations are used to creat sessions and carry session descriptions that allow participants to agree on a set of compatible media types. In this way, SIP is not restricted to any particular media type , and can therefore exploit the extending range of multimedia technologies.

It is clear that SIP is not a session description protocol or a resource reservation protocol because it does not perform any conference control and also
has nothing to do with quality of service. Although SIP is not an integrated communication system protocol, it is a fundamental component that can be employed with other protocols to build complete multimedia architecture and implement various kinds of services. For example, SIP can work in a framework with other protocols that include SOAP, HTTP, XML, VXML, etc.

SIP uses an email-like addressing. Each user is clearly identified through a hierarchical URL (for example, sip: caizhenxiong@mcmaster.ca) so that it is just as easy to redirect someone to another phones as it is to redirect someone from a web page to another. Moreover, 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 are able to comprise images, audio, authorization tokens etc.

SIP is a request-response and text protocol that are closely related to two other Internet protocols, HTTP and SMTP; as a result, SIP works well alongside Internet applications. Since SIP was originally designed to be a modular component of a larger ip telephony solution and thus functions well with a large number of IP related protocols as well. By employing SIP, IP telephony becomes an important web application and merges easily into other Internet services. As a word, SIP is a simple toolkit that Internet and telephony service providers can use to build converged voice and multimedia services. For a field as beginning and fast changeing as IP telephony, where many technologies, problems and solutions are still under debate, flexibility is extremely important. SIP is exactly part of this flexibility is extremely important, as it works well with a wide range of protocols, each addressing a different aspect of the problem space. The advantage of SIP is its ability to work with many competing technologies and exploit to newer and better ones.

By providing its own reliability mechanism, SIP is independent of the packet layer, therefore it only needs an unreliable datagram service, such as the services over UDP. SIP provides hop-by-hop and end-to-end authentication, as well as end-to-end encryption by using s/MIME. In fact, the baseline SIP provides a series of security services, such as integrity protection, authentication, encryption, privacy services and "deniel-of-sevice prevention".

Conclusion:
By fulfilling the basic functions and re-using other protocols, SIP is made flexible and scalable. Software implemented by using the basic SIP protocol can be expanded with additional calabilities and is actively being exploited for many multimedia applications.

Bibliography
http://www.sipcenter.com/aboutsip/whatissip.html
http://www.zvon.org/tmRFC/RFC3261/Output/chapter2.html
http://rfc3261.x42.com
SIP: Session Initiation Protocol

Request received
\[\text{pass to TU}\]
\[V\]
\[+----------+\]
  \[|
  \[| Trying \[----------+\]
  \[|
  \[+--------+ 200–699 from TU \[send response\]
  \[|
  \[1xx from TU \[send response\]
  \[|
  \[|\]
  \[| Request V 1xx from TU\]
  \[send response+----------+send response\]
  \[|
  \[| Proceeding\]
  \[|<--------\]
  \[+----------+\]
  \[| Trnsprt Err\]
  \[| Inform TU\]
  \[| 200–699 from TU \[send response\]
  \[| Request V\]
  \[send response+----------+\]
  \[|
  \[| Completed <----------\]
  \[|<--------\]
  \[+----------+\]
  \[| Trnsprt Err\]
  \[| Inform TU\]
  \[| Timer J fires\]
  \[|<--------\]
  \[+----------+\]
  \[| Terminated\]

Figure: non-INVITE server transaction