ATL IP Telephone SIP Summary

Introduction

ATL has released a new range of IP Telephones – the Berkshire 5000, IP250, IP300S (basic business IP telephones) and IP400 (Multimedia over IP telephone, MOIP or videophone).

ATL's telephone range has been carefully designed to the SIP (Session Initiation Protocol) standards, and tested to be compatible with SIP based systems and equipment such as IP PBX and other SIP based phones.

This document is intended to clarify the SIP protocol, and ATL's telephones compatibility with both the SIP protocol and other products or systems using SIP as their standard for IP based voice and video communication.

1 SIP Protocol design

A goal for SIP was to provide a superset of the call processing functions and features present in the public switched telephone network (PSTN) permit familiar telephone-like operations are present: dialing a number, causing a phone to ring, hearing ringback tones or a busy signal. Implementation and terminology are different. SIP also implements many of the more advanced call processing features present in Signalling System 7 (SS7), though the two protocols themselves could hardly be more different. SS7 is a highly centralized protocol, characterized by highly complex central network architecture and dumb endpoints (traditional telephone handsets). SIP is a peer-to-peer protocol. As such it requires only a very simple (and thus highly scalable) core network with intelligence distributed to the network edge, embedded in endpoints (terminating devices built in either hardware or software). Many SIP features are implemented in the communicating endpoints as opposed to traditional SS7 features, which are implemented in the network. Although many other VoIP signaling protocols exist, SIP is characterized by its proponents as having roots in the IP community rather than the telecom industry. SIP has been standardized and governed primarily by the IETF while the H.323 VoIP protocol has been traditionally more associated with the ITU. However, the two organizations have endorsed both protocols in some fashion.

SIP works in concert with several other protocols and is only involved in the signaling portion of a communication session. SIP acts as a carrier for the Session Description Protocol (SDP), which describes the media content of the session, e.g. what IP ports to use, the codec being used etc. In typical use, SIP "sessions" are simply packet streams of the Real-time Transport Protocol (RTP). RTP is the carrier for the actual voice or video content itself.

The first proposed standard version (SIP 2.0) was defined in RFC 2543. The protocol was further clarified in RFC 3261, although many implementations are still using interim draft versions. Note that the version number remains 2.0.
SIP is similar to HTTP and shares some of its design principles: It is human readable and request-response structured. SIP proponents also claim it to be simpler than H.323. However, some would counter that while SIP originally had a goal of simplicity, in its current state it has become as complex as H.323. SIP shares many HTTP status codes, such as the familiar '404 not found'. SIP and H.323 are not limited to voice communication but can mediate any kind of communication session from voice to video or future, unrealized applications.

2 Overview of SIP Functionality

SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls. SIP can also invite participants to already existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility - users can maintain a single externally visible identifier regardless of their network location.

SIP supports five facets of establishing and terminating multimedia communications:

1. User location: determination of the end system to be used for communication;

2. User availability: determination of the willingness of the called party to engage in communications;

3. User capabilities: determination of the media and media parameters to be used;

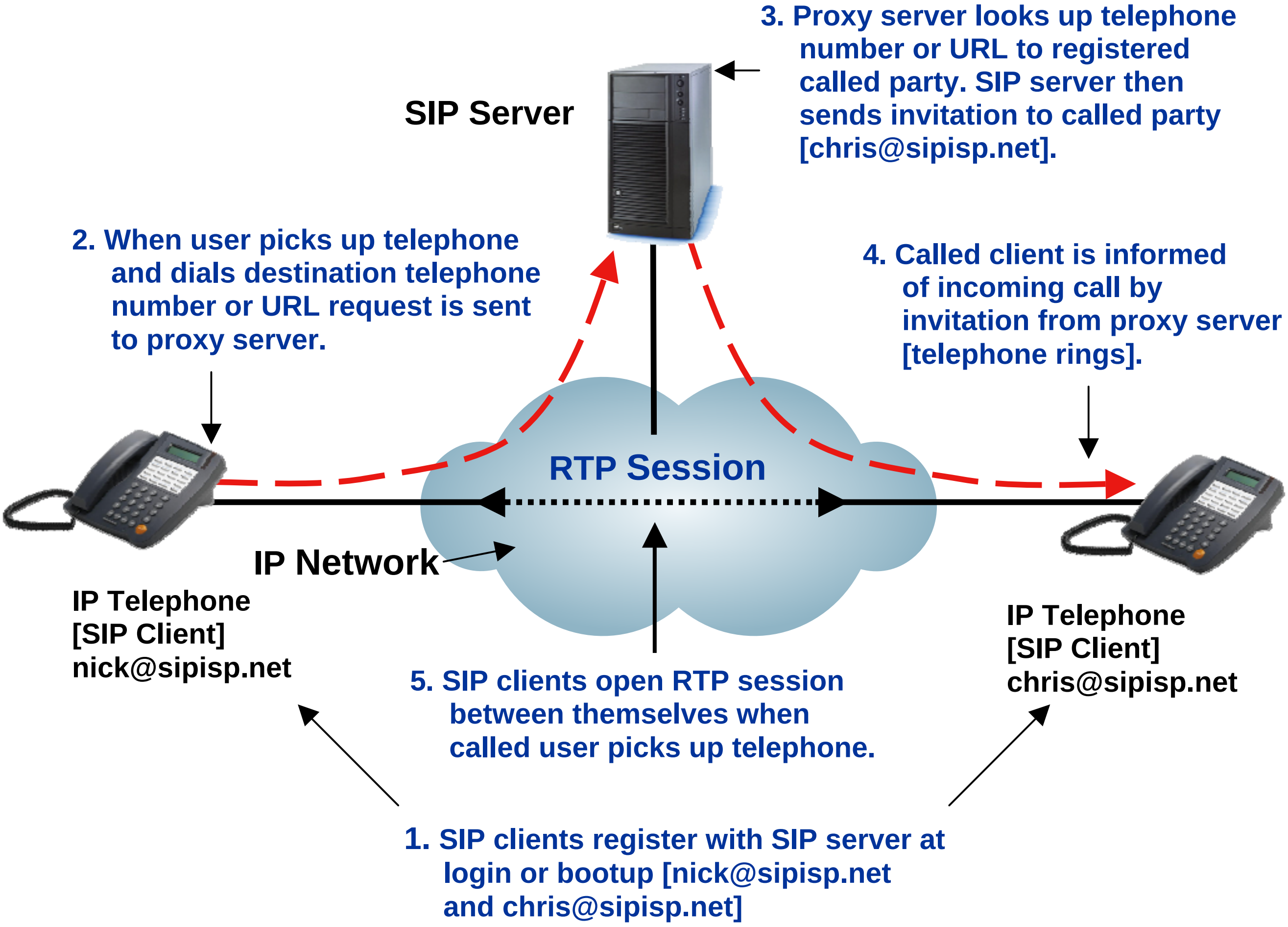
4. Session setup: "ringing", establishment of session parameters at both called and calling party;

5. Session management: including transfer and termination of sessions, modifying session parameters, and invoking services.

A graphical overview of a SIP call set-up is shown below.

SIP is not a vertically integrated communications system. SIP is rather a component that can be used with other IETF protocols to build a complete multimedia architecture. Typically, these architectures will include protocols such as the Real-time Transport Protocol (RTP) (RFC 1889 [28]) for transporting real-time data and providing QoS feedback, the Real-Time streaming protocol (RTSP) (RFC 2326 [29]) for controlling delivery of streaming media, the Media Gateway Control Protocol (MEGACO) (RFC 3015 [30]) for controlling gateways to the Public Switched Telephone Network (PSTN), and the

Session Description Protocol (SDP) (RFC 2327 [1]) for describing multimedia sessions. Therefore, SIP should be used in conjunction with other protocols in order to provide complete services to the users. However, the basic functionality and operation of SIP does not depend on any of these protocols.



SIP does not provide services. Rather, SIP provides primitives that can be used to implement different services. For example, SIP can locate a user and deliver an opaque object to his current location. If this primitive is used to deliver a session description written in SDP, for instance, the endpoints can agree on the parameters of a session. If the same primitive is used to deliver a photo of the caller as well as the session description, a "caller ID" service can be easily implemented. As this example shows, a single primitive is typically used to provide several different services.

SIP does not offer conference control services such as floor control or voting and does not prescribe how a conference is to be managed. SIP can be used to initiate a session that uses some other conference control protocol. Since SIP messages and the sessions they establish can pass through entirely different networks, SIP cannot, and does not, provide any kind of network resource reservation capabilities.

The nature of the services provided make security particularly important. To that end, SIP provides a suite of security services, which include denial-of-service prevention, authentication (both user to user and proxy to user), integrity protection, and encryption and privacy services.

SIP works with both IPv4 and IPv6.

3 ATL SIP Server

ATL has designed our products to be as close to “Plug and Play” as possible. In order for our range of IP phones and MOIP phones to be as user friendly as possible, they are preconfigured to work with ATL’s SIP server. This also allows ATL to ensure and maintain the quality of service associated with our phones when used with a public broadband connection (as opposed to used within a corporate LAN environment).

4 Summary

ATL SIP based products are intended for use with any SIP based system. ATL can qualify specific systems as required if not already covered by our compatibility testing. ATL will also ensure any reported issues are resolved as smoothly as possible.