EXHIBIT C

SIP: Session Initiation Protocol

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Abstract

   This document describes Session Initiation Protocol (SIP), an
   application-layer control (signaling) protocol for creating,
   modifying, and terminating sessions with one or more participants.
   These sessions include Internet telephone calls, multimedia
   distribution, and multimedia conferences.

   SIP invitations used to create sessions carry session descriptions
   that allow participants to agree on a set of compatible media types.
   SIP makes use of elements called proxy servers to help route requests
   to the user's current location, authenticate and authorize users for
   services, implement provider call-routing policies, and provide
   features to users. SIP also provides a registration function that
   allows users to upload their current locations for use by proxy
   servers. SIP runs on top of several different transport protocols.

1 Introduction

   There are many applications of the Internet that require the creation
   and management of a session, where a session is considered an
   exchange of data between an association of participants. The
   implementation of these applications is complicated by the practices
   of participants: users may move between endpoints, they may be
   addressable by multiple names, and they may communicate in several
   different media - sometimes simultaneously. Numerous protocols have
   been authored that carry various forms of real-time multimedia
   session data such as voice, video, or text messages. The Session
   Initiation Protocol (SIP) works in concert with these protocols by

enabling Internet endpoints (called user agents) to discover one
another and to agree on a characterization of a session they would
like to share. For locating prospective session participants, and
for other functions, SIP enables the creation of an infrastructure of
network hosts (called proxy servers) to which user agents can send
registrations, invitations to sessions, and other requests. SIP is
an agile, general-purpose tool for creating, modifying, and
terminating sessions that works independently of underlying transport
protocols and without dependency on the type of session that is being
established.

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 [27] - users can maintain a single
externally visible identifier regardless of their network location.

SIP supports five facets of establishing and terminating multimedia
communications:

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

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

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

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

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

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