

EXHIBIT Q

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CONFIDENTIAL - SUBJECT TO FRE 408

Peter J. McAndrews
McAndrews Held & Malloy Ltd.
500 West Madison Street
34th Floor
Chicago, Illinois 60661

Re: *ESN, LLC v. Cisco Systems, Inc., and Cisco-Linksys, LLC*

Dear Mr. McAndrews:

I write in response to your letter of December 20, 2007. We are disappointed that you continue to refuse to meet in person to discuss the merits of ESN's case and potential settlement. As a show of good faith, enclosed is a set of charts that compare Cisco patents US Pat. No. 6,584,108 (" '108 Patent") and US Pat. No. 6,275,574 (" '574 Patent") against claims 1, 9, and 13 of ESN patent US Pat. No. 7,283,519 (" '519 Patent). The '108 Patent and the '574 Patent arose out of the development efforts related to Cisco products. These charts are only a representative subset of prior art that invalidate the '519 Patent claims.

Cisco is ready to meet with ESN to review the attached charts and to discuss other prior art and Cisco products, which were publicly available before the critical date, that invalidate the '519 Patent claims. We look forward to engaging in a positive and productive settlement discussion with you before the end of this month.

Very truly yours,

Victoria F. Maroulis/SKS

Victoria F. Maroulis

Enclosure
51301/2348670.1

This document contains work product of Quinn Emanuel Urquhart Oliver & Hedges, LLP. Information is provided for discussion purpose only, subject to Fed. R. Evid. 408.

The following charts contain representative claim analyses. These Cisco patents are not the only prior art that invalidate ESN's patent-in-suit. There are numerous other prior art, including Cisco products that were publicly available before the critical date, that show that ESN's patent claims are invalid.

Preliminary comparison of claims 1 and 13 of US Pat. No. 7,283,519 to US Pat. No. 6,584,108, Method and Apparatus for Dynamic Allocation of Multiple Signal Processing Resources Among Multiple Channels in Voice Over Packet Data Network Systems, Chung et al., filed September 30, 1998, issued June 24, 2003.

Claim 1	
A network device comprising:	FIG. 1 is a system block diagram of a Multiservice Access Concentrator (MAC) 100 of an embodiment of the present invention for routing integrated data, voice, and video traffic. (6:10-13)
(1) a plurality of communication interfaces, including: (a) a telephone line interface, (b) a computer data interface, and (c) a broadband network interface.	The MAC uses a software-configurable wide area network (WAN) trunk to seamlessly integrate data, voice, and video into existing networks using common switch and network hardware ... (6:23-27) The interface pots of the MAC of an embodiment comprise a single *Ethernet port* 102, *two serial ports 104-106 that support speeds up to 2 Mbps*, and either *six analog voice ports* or a single digital voice access port (T1/E1) ... (6:45-49)
(2) a processor	With reference to FIG. 1, the interface ports of the MAC 100 are coupled to *a central processing unit* (CPU) 199... (7:46-47)
(3) a machine-readable storage medium which during use (a) stores a call processing application and (b) service profiles, and which (c) stores executable instructions to mediate communications between the plurality of communication interfaces, the instructions causing network device to:	FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The *internetwork software operating system that provides kernel services, network services and routing capability* comprises the voice processing subsystem 200... (7:24-28). FIG. 10 shows the data and voice flows when using a serial port as a network interface in an embodiment of the MAC 1000 of the present invention. *Data is received into a routing engine 1002 and passed to an I/O handler 1004. Voice channels are received from a private branch exchange, keyset, or telephone 1006. The received voice channels are routed to a voice call handler 1008, a voice compression engine 1010, and to an I/O handler 1004.*

	The I/O handler 1004 couples the data and voice to serial port 0 as the network interface. (14:23-32) The MAC, in providing voice services, *supports a wide range of call management configurations comprising configurations for dialing on-net extensions, off-net numbers, and local calls*, but the embodiment is not so limited. The MAC supports call set-up options comprising, but not limited to: on-net dialing; auto-dial, or private line automatic ringdown (PLAR); off-net dialing; and tandem switching. (15:63-16:3). The MAC of an embodiment supports on-net dialing when a call originator dials an extension by entering a phone number; the call is connected within the wide area network. *For on-net calls, a flexible call numbering plan allows dialing to any port on any system in the network by dialing a unique prefix that identifies the port or group of ports on the destination system.* (16:4-10). The MAC of an embodiment supports off-net dialing, wherein *when a caller dials "9", or another pre-programmed digit/digits, the MAC automatically connects the caller extension to a channel connected directly to a PSTN.* (16:23-26).
(4) detect network signaling events or trigger points in a telephone call;	The MAC supports analog voice streams using E&M, FXS, and FXO voice signal standards. (7:2-7) In operation, an analog voice port or digital voice port provides a voice signal to the Voice Signal Process module 202. *The Voice Signal Process module 202 translates the voice signal to a call setup message.* (7:29-32).
(5) invoke the call processing application in response to the detected network signaling events or trigger points, the call processing software operating according to the parameters defined in the service profile	The call setup procedure comprises calling the Voice Channel Manager 206 to allocate a DSP for connection to rhe receiving voice port via the PCM bus. *Furthemorer, the call setup procedure signals the Voice Signal Process module 202 to enable the allocated DSP to start the DTMF sampling for the dialing digits. Upon collection of enough digits by consulting the dial-mapper 208, a setup message is provided to the Tandem Switch Module 210. The Tandem Switch Module 210, using the Dial-peer and Dial-mapper 208, locates the permanent victual connection (PVC) number of the remote extension in order to provide a setup message.* (7:32-45)

Claim 13	
A method for establishing a voice-over-packet network architecture, the method comprising:	
(1) locating a system management platform in a shared packet network, the system management platform collecting call log data from a plurality of network devices; and	Management and configuration of the MAC of one embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (17:6-18).
(2) distributing the plurality of network devices that each include (a) a telephone line interface, (b) a computer data interface, (c) a broadband network interface terminating a link from the shared packet network, (d) a processor, and	See discussion under claim 1, (1) and (2).
(e) a machine-readable storage medium storing processor-executable instructions to control telephone calls, the instructions causing each network devise to route telephone calls in a peer-to-peer fashion over the shared packet network and to send call log data to the system management platform.	See discussion under claim 1, (3). See also FIG. 2 and FIG. 10. Furthermore, the routing table includes information elements comprising a preference-order and a last-use timestamp... The last-use timestamp, in tracking successfully completed calls and call attempts, is an important attribute in that it allows load balancing. Moreover, network peers may be associated with various administrative metrics that reflect various attributes of the network interface... (26:17-30).

Preliminary comparison of Claim 9 of US Pat. No. 7,283,519 to US Pat. No. 6,275,574, Dial Plan Mapper, Oran, filed December 2, 1998, issued August 14, 2001.

Claim 9	
A network device comprising:	
(1) a broadband network interface (2) a plurality of interfaces, including a telephone line interface and a computer data interface; (3) a processor; and	The invention relates to telephone systems and more particularly to a dial plan mapper used for routing telephone calls to different telephone networks. (1:4-5). The dial plan mapper 20 can be integrated into a variety of existing IP hosts such [as] *VoIP routers, Voice over Frame Relay (VoFR) routers*, and H.320/H.323 gateways. Other types of telephony systems connected to the IP network 22 may also use the dial plan mapper 20 and also come within the scope of this invention. (3:16-18).
(4) a machine-readable storage medium that stores processor-executable instructions to provide SIP agents, the instructions causing the network device to: (a) provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface, and (b) the instructions further causing the network device to implement a SIP proxy server that mediates all SIP communication over the broadband network interface involving non-SIP telephone.	FIG. 2A is a detailed diagram of one of the gateways 18 or 28 in the telephone system shown in FIG. 1. The gateways 18 and 28 include a session application 34 that responds to telephone calls. (3:24-27). Telephones 14 are coupled either directly or through a PSTN, PBX 30, etc. telephony interface 36 to the session application 34. The telephony interface 36 consists of whatever circuitry abstracts the signaling part of the telephony hardware attached to the gateway. The IP network 22 is coupled through an IP interface 38 to the session application 34. The session application 34 provides call translation between the telephone interface and the IP interface 38. (3:31-39). FIG. 4 shows a call coming through a local telephone 14 or a telephony interface such as an FXS, FXO, PBX trunk, etc. The session application 34 begins listening for a dial string when informed of a telephone off-hook condition by a telephony driver in telephony interface 36.... Each time a digit 40 arrives, it is appended to an accumulated dial string 42, and the dial plan mapper 20 is called and supplied with the accumulated digital string 42. This process repeats until the dial plan mapper 20 returns either a match or an indication that the dial string cannot possibly match an entry... The dial plan mapper queries the configuration database 32 for the longest match 44. The match pattern 44 ... is used as an index to identify associated configuration information 46. The session application 34 uses the returned session protocol, and session target from the configuration information 46 to initiate a session with a destination for the presented dial

	string 42, such as phone 15 on IP network 22. (4:43-65). The session application can also process Session [Initiation] Protocol (SIP)invitations, and if the invited user is reachable from one of the local telephony interfaces, the user is called and an attempt is made to connect him to the conference. (6:14-18). See FIG. 6 showing SIP as one of the available session protocol options. FIG. 6 is a table showing the different inputs and outputs of the dial plan mapper 20. (7:4-5) ... Session _ protocol indicates what call-establishment procedure to use for communicating with the call destination. (7:38-40).